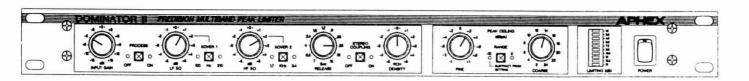
DOMINATOR II

PRECISION MULTIBAND PEAK LIMITER



OPERATING GUIDE



WARNING

TO REDUCE THE RISKS OF FIRE OR ELECTRIC SHOCK, DO NOT EXPOSE THIS APPLIANCE TO RAIN OR MOISTURE

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1.0 INTRODUCTION

The new world standard

After its introduction in 1985, the Studio Dominator became the world standard for peak limiting because of its transparency and effectiveness. However, unwilling to rest on its already established reputation, Aphex Systems, Ltd. has continued to develop new techniques. The result is the Dominator II.

As a direct descendant of the original, the **Dominator II** has the same primary functions and applications. Its many performance improvements lie in the areas of dynamic range, distortion, flexibility, and ease of setup.

Although those familiar with the Studio Dominator already know much of the information in this manual, Aphex recommends that these people as well as those new to Aphex Dominators read this manual.

Absolutely zero overshoot

The Dominator II is a stereo multiband peak limiter with absolutely zero overshoot. Once the **PEAK CEILING** is set, there is no higher amplitude in the output. Awareness of headroom limitations and the price of exceeding those limitations, important for any audio application, is critical for most, especially digital. The Dominator II allows users to work confidently, creatively, and quickly by freeing them from the fear of "crashing".

Wide range of applications

Achieving this **brick wall** result while retaining complete fidelity is the art and science of the Dominator II. Because of its exceptional function and performance, it has applications in a great many audio fields, including:

- BROADCASTING-PRODUCTION, AND TRANSMISSION FOR AM, FM, AND TV
- RECORDING
- SAMPLING
- MIXING
- MASTERING-CD, FILM, AND ANALOG DISK
- SOUND REINFORCEMENT
- SATELLITE AND STL UPLINK
- LOCATION RECORDING
- VIDEO POST-PRODUCTION
- TAPE DUPLICATION
- TELECONFERENCING

1.1 DYNAMIC RANGE CONTROL

Essential to a full understanding of the operation and application of the Dominator II is a knowledge of the definitions and applications for each type of gain control: leveling, compression, limiting, and clipping.

1.1.1. Leveling

Constant output level

Leveling has high compression ratios with slow attack and release times. Because of its very slow time constants, leveling has no effect on short term changes in average levels or on transient peaks. Therefore, it is used to maintain a constant output level without affecting the short term dynamics by adjusting for long term changes in the input signal. Typically, the threshold is set low so that low level signals will be brought up.

1.1.2 Compression

Creative effects

Compression has low ratios with faster attack and release times. A compressor forces a wide dynamic range into a smaller range. The size of the resulting dynamic range depends on the level of the threshold, the height of the ratio used, and the speed of the attack and release. The higher the ratio and the faster the time constants, the greater the effect on the short term dynamics (the actual quality and fidelity of the sound); these effects are often used creatively. Normally, since one of the desired results of compression is to bring up low level signals, the threshold is set low.

1.1.3 Limiting

High levels kept down Limiting has high compression ratios (usually defined as greater than 8:1), fast attack and, depending on the particular application and desired sound, slow or fast release times. Since it is normally used to keep high levels down, the threshold is set high. Inasmuch as the ratios are high, as the input is driven further into limiting, the dynamic range of the output becomes tighter, an effect that may or may not be desirable. If the attack times are set to control only the average level, and allow for peaks above the threshold to pass, the limiter is considered a **program limiter**. When the speed of the attack times is increased to control peaks, the limiter is considered a **peak limiter**.

1.1.4 Clipping

An absolute brick wall

Regardless of a limiter's attack time, there is a finite amount of time before the detector circuits cause the gain circuits to reduce the output below threshold. Also, extremely fast gain modulation introduces unwanted audible effects. Therefore, clipping with its infinite ratios and instantaneous attack and release, functions as an absolute **brick wall**. To control the peaks, clipping literally shaves off the peaks of the wave above threshold without changing the amplitude of the remainder of the wave. Done properly, clipping is inaudible and, under certain circumstances, actually enhances the audio, Done improperly, however, clipping produces very audible, obnoxious effects.

2.0 FUNCTIONAL DESCRIPTION

Drawbacks of traditional peak limiters overcome

Traditionally, peak limiters have been designed with fast attack times: the faster the attack time, the lower the amount of overshoot above threshold. The drawback to these fast attack times, however, is that the limiter is triggered off each transient, even extremely short ones. The sonic result is hole punching and overall lower density (peak to average ratio). On the other hand, while slower attack times result in better sonic performance, the overshoots cause distortion, or the overall level must be reduced to accommodate the overshoots. The Dominator II overcomes both drawbacks by combining limiters with fairly slow attack times and clipping, in an interactive, self-adjusting manner.

Adjustable threshold

The Dominator II has been designed as a unity gain device with an adjustable threshold. Since there is zero overshoot over threshold, users have only to set the **PEAK CEILING** to the level at which peaks must stop (for example: 100% modulation). In other words, set it and forget it!

2.1 MULTIBAND VERSUS WIDEBAND PROCESSING

A significant problem with wideband processing is "spectral gain intermodulation" which occurs when one part of the spectrum controls the level of another part. A typical situation is a vocalist being 'sucked down' whenever the kick drum hits.

Since most energy is contained in the lower frequencies, they tend to control the level of the entire spectrum. When lower frequencies are above the limit threshold, higher frequencies are attenuated, causing the output to be dull.

Multiband processing for user flexibility Multiband processing solves these problems by splitting the audio into two or more frequency bands, and processing each band separately. However, more bands often result in many more parameters to control, including a method of summing the bands together again. While this creates user flexibility, it also requires different settings for almost every different source.

Fewer controls

Since the Dominator II uses **program dependent**, **intelligent circuits** that reduce the number of controls, users have the flexibility to shape the sound while quickly and easily achieving consistent, effective limiting.

2.2 ALT (AUTOMATIC LIMIT THRESHOLD)

[Reference: Fig. 2.1 ALT BLOCK DIAGRAM, Page 2.3]. A multiband processor splits the audio into separate bands, limits each band individually, and then sums the bands together again. Even though each band's peak output is predictable, summing the bands together produces an unpredictable peak output.

One conventional approach to making the summed output predictable is to use a wideband limiter after the summing. Unfortunately, this introduces all the drawbacks of wideband limiting identified above.

Another approach is to use a clipper on the summed output, but this causes too much clipping distortion if the summed output is too high. To avoid this distortion, the limiters' thresholds are set very far below the clipper threshold. The drawback to this approach is a loss of loudness and, because of the lower thresholds, a much greater amount of processing.

Predictable
peak output and
maximum
loudness with no
audible distortion

Instead of either of these approaches, the Dominator II uses a patented method to produce a predictable peak output while maintaining maximum loudness without audible distortion: the Automatic Limit Threshold (ALT). With this method, the outputs of the three bands are summed and sent to the ALT detector circuit. If the sum exceeds a reference value, the ALT reduces the thresholds of the individual limiters. When the summed output falls below the reference value, the limit thresholds return to their original setting.

Self-adjusting finite attack time

The ALT circuit has a self-adjusting finite attack time. The amount of time it takes to lower the thresholds of the limiters is the length of time the limiters' overshoot may be in the clipper. The reference value of the ALT in relation to the clipper determines the depth of clipping. The DENSITY control sets both parameters. When it is set higher, the ALT reference gets closer to clipping and the attack time is slower, producing more clipping. When DENSITY is set lower, the opposite occurs. The 0 RCH position for the DENSITY control emulates the standard parameters of the original Studio Dominator, Model 700; this is recommended for general use.

Global stereo balance and imaging It is important to know that because there is only one ALT circuit controlling both channels equally, this method provides global stereo balance and imaging by ensuring that both channels always limit at the same threshold. Since this does cause an interaction if the Dominator II is used as two independent channels, Aphex does not recommend such a practice.

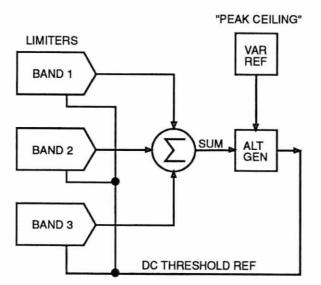


Fig. 2-1. ALT Block Diagram

2.3 EQUALIZATION

As indicated above, wideband limiting often causes dullness. To prevent this, a common practice is to follow the limiter with an equalizer set to boost the high frequencies. The drawback is that the equalizer adds directly to the overshoot.

Three-band processing with selectable crossover frequencies

Another practice is to attenuate the lower frequencies fed to the detector circuits. While this diminishes spectral gain intermodulation, the drawback is that when there is substantial energy in those lower frequencies, it creates unacceptable amounts of overshoot. The Dominator II eliminates these problems by using three-band processing with selectable crossover frequencies between low and mid-band and between mid and high-band, with adjustable input levels to the low and high-band. Also, the crossover filters are designed so that when the EQ controls are set flat and the input is below threshold, the Dominator II will pass perfect square waves. This means that the Dominator II equals a well designed wideband limiter in below threshold audio performance, and vastly outperforms any wideband device in performing its function above threshold.

Flexibility to shape the sound and maintain absolute peak ceiling The selection of the crossover frequencies and the input level to each band helps determine the **frequency response** of the output. When the input is below the limit threshold, the EQ will give a response equal to the change in input in each band. Since the compression ratio of the limiter is essentially infinite, a change in the input level above threshold will not cause any change in the output level. The EQ provides user flexbility to shape the sound and still maintain an absolute peak ceiling.

Adjustable output density





-12 +12 -15 dB +15 INPUT GAIN

No clicks, pops, or unintended signals

Attack time preset and equal for limiters

Stable stereo image without "smearing"

2.4 RELEASE TIME

The release time allows users to adjust the density of the output. With faster release times, the output is consistently maintained as close as possible to maximum. Extremely fast release times result in a very tight dynamic range, and also may cause low frequency distortion. Slower release times reduce the distortion and provide a rounder, softer sound. Very slow release times tend to make the output much lower in average amplitude, especially on transient inputs.

2.5 GAIN CONTROL CIRCUITS

An important design goal for all Aphex products has been to make a signal processor able to **do nothing** before it performs its function. That is, it must be able to pass audio as if it were nothing more than a straight wire--no noise, distortion, or color.

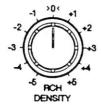
Essential to achieving that goal is the use of the Aphex VCA 1001 high performance integrated gain control element. Its measured specifications are outstanding as a signal amplifier, but even more importantly, its dynamic performance surpasses all other devices. Others produce distortion or amplitude anomalies with complex program audio which do not appear with steady tone test measurements. Of extreme importance is that other devices also create anomalies when gain is changed rapidly. The best known anomaly is DC shift or control feedthrough whose sonic results include clicks, pops, or unintended signal fed to the control circuits. The Aphex VCA 1001 is completely free from these anomalies, even in the most demanding situations such as a peak limiter.

Each band has its own limiter and detector. The attack time is preset and equal for all limiters. The release time is adjustable and also equal for all limiters. The threshold is adjustable from the front panel Peak Ceiling control and is also adjusted by the ALT.

2.6 TRACKING CONTROL

Tracking forces the limiting of each band in each channel to follow the same band in the other channel. The channel with the greatest amount of limiting will cause the same band in the other channel to have an equal amount of limiting. This control maintains a stable stereo image. Without the tracking control, increased stereo loudness can be achieved but with potentially inaccurate or "smeared" stereo imaging, depending on the amount of limiting. For light limiting, non-tracking may be preferred in many cases.

Greater waveform power



Adjustable depth and duration of clipping

2.7 DENSITY

The DENSITY control is one of the most important improvements incorporated into the Dominator II. As indicated in section 2.2 ALT, the DENSITY control adjusts the parameters of the ALT circuit. In effect, this allows users to determine the RELATIVE CREST HEIGHT (RCH) of the audio output signal. With a higher RCH setting, the below-threshold peak levels increase, producing greater power in the waveform as well as greater relative loudness.

By setting the ALT reference level and attack time, this control essentially adjusts the average **depth** of clipping, and the relative **duration** of clipping allowed. High clipping depth and duration create greater loudness, but may generate high distortion. Low clipping depth and duration reduce clip distortion, transferring more work to the limiters. Loudness and 'punch' are traded for low distortion.

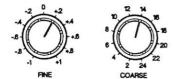
For those interested in the technical aspects, when the DENSITY control is in the 12 o'clock position, the limiter thresholds are 3dB below the clipper threshold. Fully clockwise, the limiter thresholds are at the same level. Fully counterclockwise, the limiter thresholds are 6dB below the clipper.

2.8 PEAK CEILING CONTROL

Many audio applications require that the **peak output** be trimmed to be as high as possible. To fulfill that requirement, the Dominator II was designed with two controls for the output ceiling. The **COARSE** control switches the ceiling, in 2dB steps, from +2 to +24dBu (peak). The **FINE** control adjusts the ceiling +1 to -1dB from the **COARSE** setting, in 0.2dB steps.

There is a third control, **RANGE**, which adjusts the internal gain structure by adding 10dB at the input, and subtracting 10dB at the output. These three controls give users the ability to trim the peak output to within 0.2dB over a 34dB range.

The PEAK CEILING controls set the threshold of limiting (the maximum peak output). If the input level remains below threshold, adjusting the PEAK CEILING controls does not affect output levels.



Peak output trimmed to within 0.2dB over 34dB range

2.9 INPUT AND OUTPUT CIRCUITS, BYPASS RELAY

PROCESS

OFF ON

No interference from transmitters

Bypass mode prevents line loading disturbance

Balanced and unbalanced output lines driven with no difficulty

Fail-safe characteristic for critical systems For maximum audio performance, the Dominator II has **servo-balanced transformerless** audio I/O circuits. With these, perfect interfacing with any system, balanced or unbalanced, high or low impedance, is extremely simple.

The input stage and AC power input receptacle include radio frequency filtering to reject interference from transmitters and allow the Dominator II to be used in typical broadcast racks.

The **servo-balanced** input stage has a selectable 600 ohm termination resistor for systems which need to be loaded. When **Bypass** mode is selected, the resistor is lifted to prevent line loading disturbance. For systems that don't require a load resistance, users simply do not select the termination; unterminated, the input impedance is 19.5k Ohm.

The servo-balanced output stage properly drives any load of 600 ohms or greater. The output impedance of 65 ohms can drive long capacitive lines effectively, terminated or unterminated. The especially unique characteristic of the servo-balanced output stage is its ability to drive balanced and unbalanced output lines without difficulty. For unbalanced output, the unused output pin (2 or 3) is GROUNDED to pin 1. This causes the output stage to shift all drive automatically to the hot pin only, and adjust gain to provide full output level (no 6dB loss as with other circuits).

A high quality audio **relay** provides a true hard-wired bypass function. **Bypass** connects the output connector directly to the input connector and lifts the input termination resistor, if it was selected. The limiter input stage is still connected to the input connector; thus, the limiter remains active for instant insertion in line. The Bypass function can be initiated in one of three ways: powering down or power failure; front panel **Process Off/On** switch; or remote control (rear panel jack). This configuration provides a fail-safe characteristic for critical systems where the audio signal must be bypassed through the Dominator II if the AC line or internal power supply fails.

2.10 MODEL 723 PRE AND DE-EMPHASIS

Reduced noise for broadcast and transmission links

DE-EMPHASIS
(MODEL 723 ONLY)

Pre-emphasis is an equalization curve expressed as a time value based on the product of a resistor and capacitor. The higher the value, the greater the equalization. This has been used as a noise reduction technique for broadcast and transmission links.

Primarily, there are two world standards: **50** and **75** microseconds. Starting flat at approximately 1KHz, 50 microsecond pre-emphasis increases almost 12dB at 15KHz; 75 microsecond pre-emphasis increases over 17dB at 15KHz.

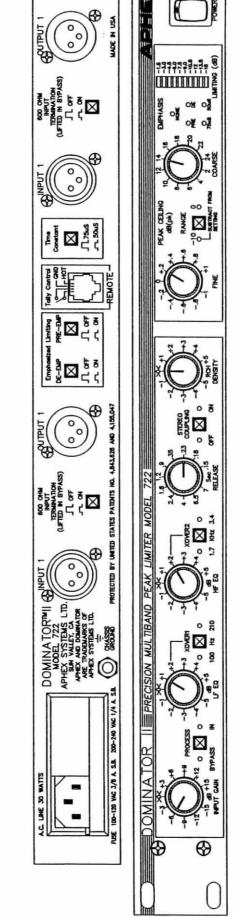
The Dominator II Model 723 has pre-emphasis (either 50 or 75 microseconds) added after the input circuit and before the limiters. It also has a complementary de-emphasis circuit (which may be switched out of circuit) after the final limiter and before the output stage.

When the de-emphasis circuit is in circuit, the audio output of the Model 723 is flat if the input is below threshold. As the input increases above threshold, the output takes the shape of the de-emphasis curve.

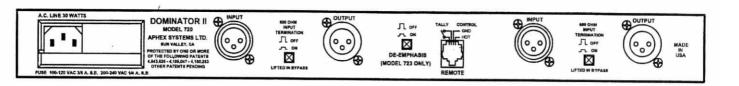
redeficient for the woder 722 Dominator with the and De-Emphasis

The Model 722 replaced the Model 723. The functions and circuitry are the same except that the pre-emphasis in the Model 722 can be switched off while it was 'on' all the time in the Model 723.

The advantage of the Model 722 is that the pre-emphasis and de-emphasis can be switched on and off and the curve can be selected (50 or 75μ seconds) by rear panel switches. The other advantage is that the status of the pre and de-emphasis is displayed on the front panel.



3.0 INSTALLATION



3.1 UNPACKING

Your Dominator II was carefully packed at the factory in a container designed to protect the unit during shipment. Nevertheless, Aphex recommends making a careful examination of both the shipping carton and the unit for any sign of physical damage.

If damage is evident, do not discard the container or packing material! Contact the carrier immediately to file a claim for damages since the consignee (you) must make all shipping claims.



3.2 INPUT SENSITIVITY

Input sensitivity is selected by a front panel RANGE switch to allow the Dominator II to be installed before input requirements are determined. See Section 2.8.

3.3 MOUNTING

The Dominator II occupies one standard rack unit of space (1-3/4 inches), 19 inches wide and 8 1/2 inches deep. To mount the unit properly:

- •Allow at least 3 inches of space in the rear for connectors.
- •Make sum them is at least 1/2 inch of clearance above and below the unit for cooling.
 - •Mount the unit with the cusbioned rack screws proyided.

Installation

3.4 CONNECTORS

The audio connections are made via standard 3-pin XLR connectors. Pin 1 is **ground**. Since the inputs and outputs are servo-balanced, either pin 2 or pin 3 may be **high**. If the Dominator II will be driving an unbalanced load, Aphex recommends that the grounded side be attached to ground at the output connector. Also, it is important to ensure polarity from input to output.

The remote control connector, shown below in Figure 3-1, is a standard RJII telephone plug. (Note: The Dominator II will work without any connection in that jack.)

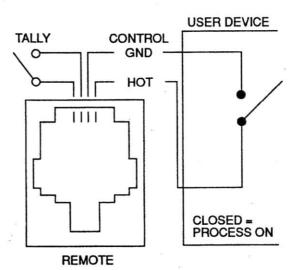


Fig. 3-1. Pin-out diagram for the remote

3.5 IMPEDANCES

The input impedance, 19.5k Ohms unterminated, may be driven easily by any output source. There is a 600 Ohm selector switch on the rear panel for each channel of systems which require loading. When the Bypass mode is selected, this loading is lifted.

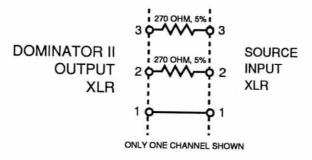
The output impedance of 65 Ohms will properly drive any load of 600 Ohms or greater at the calibrated peak output level. It is not recommended to convert the output to 600 Ohms for any reason; however if this is absolutely required, the circuit below is recommended.



NOTE DO NOT USE ANY SORT OF MATCHING TRANSFORMER TO ESTABLISH A 600 OHM OUTPUT IMPEDANCE.

The 270 Ohm resistors may be installed in the XLR output cable connectors, in the patch bay, or other interface. It is not easy to add them inside the Dominator. Remember that the output level becomes very dependent on loading when the build out resistors are added, and a 50% (6dB) loss of level will be experienced for a 600 Ohm load.

Fig. 3-2. Circuit for creating a 600 Ohm source impedence at output of Dominator II



3.6 AC LINE CONNECTOR

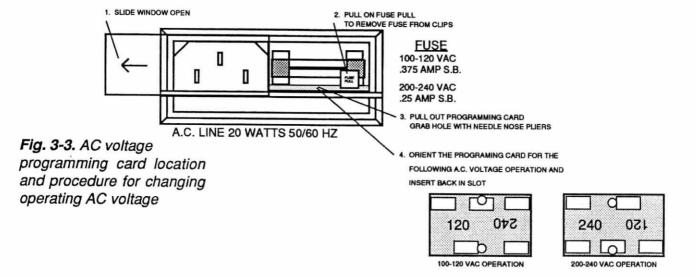
AC mains power is supplied to the unit via an integrated receptable/fuse holder on the rear panel.

3.7 MAINS VOLTAGE SELECTION

The Dominator II may be set for any standard mains voltage, as follows:

- Remove the power cord from the chassis receptacle.
- · Slide the plastic cover to the left to uncover the fuse compartment.
- · Pull the fuse lever to remove the fuse
- Pull out the small circuit board and re-insert with the appropriate voltage (100, 120, 220, or 240) readable through the fuse holder window.
- Re-insert the fuse and AC line connector.

Figure 3-3 below illustrates this procedure:



3.8 FUSE SELECTION

100 120 volts 375 amps slo-blow

220-240 volts .25 amps slo-blow

3.9 DE-EMPHASIS BYPASS: MODEL 723 ONLY

Dominator II Model 723 has a pre-emphasis circuit at the input, and a complementary de-emphasis circuit at the output. The de-emphasis circuit may be switched in or out of the circuit via a switch on the rear panel. When the switch is IN, the de-emphasis is in circuit.

4.0 FRONT PANEL CONTROLS

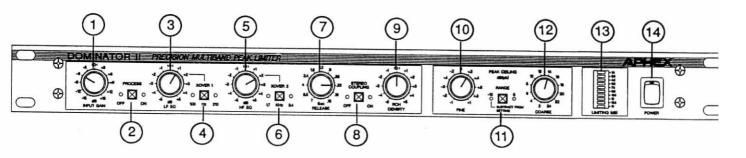


Fig. 4-1 Front Panel Controls

- 1. INPUT: Adjusts input level ±15dB. Center detent is unity gain.
- PROCESS IN/OUT: Switches the audio through the unit or the bypass relays.
- 3. LF EQ: Adjusts input level to the low frequency band ±5dB. Center detent is "flat" response below limiting.
- 4. LF XOVER: Switches the crossover point between low and mid-bands from 100Hz to 210Hz.
- 5. HF EQ: Adjusts input level to the high frequency band ±5dB. Center detent is "flat" response below limiting.
- HF XOVER: Switches the crossover point between mid and high bands from 1.7KHz to 3.4KHz.
- 7. RELEASE TIME: Adjusts release time of the limiters between 150msec (fully clockwise) and 7 sec (fully counterclockwise)
- 8. STEREO COUPLING: Forces the limiting in each band to be equal to the limiting in the same band of the other channel, if the band in the other channel has a greater amount of limiting.
- DENSITY: Adjusts the RELATIVE CREST HEIGHT (RCH) of the output. The higher the RCH (clockwise), the louder the output. The lower the RCH, the lower the average level in the output.
- 10. FINE: Adjusts the PEAK CEILING BY ± 1 dB in 0.2dB steps.
- 11. RANGE: Switched to the -10 position, it adds a 10dB boost in the input and a 10dB cut in the output. Therefore, the PEAK CEILING is 10dB lower than the settings shown on the COARSE and FINE controls.

- 12. COARSE: Switches the PEAK CEILING, in 2dB steps, from +2 to +24dBu(pk).
- 13. METER: Displays the limiting, from gain reduction and clipping, in the channel with the greatest amount of limiting.
- **14. POWER:** Switches AC mains power on and off. The OFF position engages bypass relays.

5.0 BASIC SETUP

This procedure makes the Dominator II a unity gain device with nominal operating parameters and peak output at the proper level. Refer to Section 2.0 FUNCTIONAL DESCRIPTION and Section 6.0 APPLICATIONS for information on using EQ, RELEASE TIME, and DENSITY controls as well as other setup recommendations.

- 1. Set the RANGE switch as needed. Switch to -10 if the input is low level (-10dBV, -7.8dBm), or to 0 if the input is high level (0 to +8dBm).
- 2. Set INPUT AND EQ controls to 0 detent position, LF XOVER at 100Hz and HF XOVER at 1.7KHz, TRACKING in OFF POSITION.
- 3. Set RELEASE TIME AND DENSITY controls fully clockwise.
- 4. Set PEAK CEILINGcontrol to a level below the estimated maximum peak input level to the following device, e.g., recorder, transmitter, etc.
- 5. Feed tone or program at normal level; adjust INPUT control to generate at least 6 to 8dB of limiting.
- 6. Raise PEAK CEILING control while observing peak indicators on the input of the following device, e.g., modulation monitors, peak meters, overload indicators, etc. If no peak indicators are available, listen to the audio for distortion.



NOTE

Be sure that there is limiting as the PEAK CEILING control is increased; adjust INPUT control as necessary.

 Once the PEAK CEILING has been established, return INPUT, DENSITY, and RELEASE TIME to center detent. Switch TRACKING to ON.

6.0 APPLICATIONS

6.1 RECORDING

Good recording during tracking

Both analog and digital recording media suffer from excess or insufficient record level. If the level is too low, an analog track will be noisy; a digital track will lack resolution and sound grainy. If the level is too hot, an analog track will sound compressed or pinched, while a digital track will sound harsh and badly distorted. Since these problems cannot be properly "fixed in the mix" it is necessary to take measures that ensure good recording during tracking.

Riding faders during recording is not always the best approach to getting correct record levels. The recordist may not always know the performance, and artists seldom play at exactly the same levels as in rehearsals. Especially when multiple tracks are being recorded in a hectic situation, artists and engineers cannot be expected to keep all levels optimized. All too often, the results are some bad tracks.

Consistent hot tracks without overload When the Dominator II is set up according to Section 5.0 Basic Setup and used during recording, a hot track can be made consistently and the recording engineer will be freed from worries about overload. A digital track can then utilize the maximum resolution available without ever crashing. An analog track will sound clean and more quiet. This applies to mixdown as well as to mulitrack masters. A hot two track master can be generated using the full dynamic range of the medium without overload distortion.

NOTE

Finding the absolute peak input crashpoint of a particular recorder may be difficult because of the absence of any peak indicators. In that event, program should be recorded at various peak input levels and monitored for distortion. Once the peak input level has been established, it should be noted for later use in setting up the Dominator II.

Even if peak indicators are provided, they may not indicate the true crashpoint. It may be wise to verify the actual maximum peak input as if there were no peak indicators.



6.1.1 Digital Recording and Sampling

While overload characteristics of analog tape may be a desirable effect for certain types of program, overload characteristics of digital should be avoided under all circumstances.

Maximum benefit of digital medium The obvious and most often used method to avoid this effect is to run the input level sufficiently low to prevent any peaks from causing overload. The difficulty with this method is that digital loses one bit of resolution for every 6dB drop in level. And, as resolution decreases, the negative sonic characteristics of digital become more prominent (grunge, inaccurate high frequencies, and loss of 'air'). No amount of processing will increase the resolution---in fact, the reverse! Digital signal processing, including digital to digital conversions, actually further decreases resolution! The Dominator II, used simply as a protection device, will guarantee maximum benefit of the digital medium.

6.1.2 Stereo Recording

Stable imaging for binaural recording

Stable imaging is often a high priority for stereo recording, and critical for binaural recording. The Dominator II should be set up according to Section 5.0 Basic Setup. Also, it is important to make sure that TRACKING is switched to ON.

6.2 MIXING

When assembling a mix, it is important to keep the highest peak levels of the elements within a fairly close range. This will allow for a hotter master later on, without the necessity for a large amount of limiting at the mastering stage.

Adjustable for different elements

Using the Dominator II on each individual element will also give users the flexibility to adjust the unit differently for different elements. This is an appropriate place to try various settings of crossovers, EQ levels, DENSITY and RELEASE times.

Multitrack mixing often requires layering of the various tracks to achieve depth and allow the more important tracks to maintain prominence. The level range (window) for each track may be maintained by riding faders or using some form of gain reduction.

Program limiter function for tight "windows"

Effective for voice-overs



If it is desirable to have a tight "window" for the average level, the Dominator II may be used as a PROGRAM LIMITER. In this case, average levels must be driven further into limiting. That is accomplished by turning up the INPUT control, by driving the input to the Dominator II to a higher level, or by lowering the PEAK CEILING. The amount of limiting on the display should be at least 6 to 8dB.

One application of this type of effect is for voice-overs in which the voice must "ride" over the music or effects bed at a consistent level. The tightness of the window is determined by the speed of the release time. The faster the release time, the tighter the window. The slower the release time, the more open the window.

NOTE

Beware of radical settings. If the release time is too fast, there may be an increase in audible distortion caused by the limiters following the low frequency waveforms. Even if there is no increase in distortion, the tightness may become unnatural.

6.3 MASTERING:CD, VINYL, FILM, TAPE DUPLICATION

As in recording, the main goal in mastering is to achieve a clean, hot transfer without overload distortion and without coloration. Many applications, however, require that the master be as loud as possible.

Loudness is defined in several different ways. Here, loudness is defined as how high the average level (VU or RMS) is in relation to the peak level in terms of amplitude and duration. The higher the average level and the longer it stays at that high level, the louder it will seem to be.

Loudness tuning procedure provides maximum loudness

If the goal is to achieve a maximally loud master, the Dominator II should be set up initially according to Section 3.0, followed by a loudness tuning procedure, as follows:

Make sure that there is 2 to 6dB of limiting by turning up the INPUT
control or increasing the input level to the Dominator II. The first 6dB of
limiting increases the loudness. A greater amount of limiting will turn the
Dominator II into a PROGRAM LIMITER, and begin affecting the
apparent dynamics.

Turn up the DENSITY control. This will increase the output level into the clipper which will increase average level. Because the amount of clipping will be increased, it is necessary to stop when any distortion becomes audible.

Higher average level for longer with faster RELEASE time 3. Speed up the RELEASE time. This will allow the average level to stay at a higher level for a longer time. If only 2 or 3dB of limiting is used, then the RELEASE time may be used at full fast. If greater amounts of limiting are used, slower release times must be used to avoid distortion.

6.4 SOUND REINFORCEMENT

Brick wall peak limiting may not be as apparently critical for sound reinforcement because of the forgiving nature of some amplifiers and speakers to short term "spikes". The problem is that a conventional limiter must be set well below the peak input of the amplifier to reduce the height and duration of the over-threshold spike. That will then introduce all the problems of conventional limiters explained in Section 1.0 Introduction.

Sound reinforcement without overload

One of the most important advantages of using the Dominator II for sound reinforcement is that all the available power may be applied without fear of overload. If the Dominator II gives an increase in average level of only 3dB (it usually can achieve greater loudness), it effectively has doubled the power amplification. In effect, the Dominator II turns a 10,000 watt system into a 20,000 watt system that is completely overload protected!

6.4.1 Processor Speaker Systems

Problems of conventional limiters avoided

To avoid some of the problems of conventional limiters, some crossover manufacturers use limiters in each band. Typically, these limiters have fast attack which causes an entire band to "duck" when a spike goes over threshold in that band. Other manufacturers provide sliding crossover frequencies and limiters to allow maximum level in each band, an approach that causes different equalization characteristics as the program changes its peak to average characteristics. Using the Dominator II in front of these systems ensures against these negative effects by establishing a maximum peak level into the downstream processors.

6.4.2 Reverberant Rooms

Dullness and loss of intelligibility eliminated

Greater protection, equalization, loudness, and layering A common problem sound designers face is the necessity to limit lower frequencies to below the level of excitation of the reverberant field. The problem with conventional limiters is that the entire spectrum will be limited to the same point which causes dullness and loss of intelligibility. The multiband design of the Dominator II eliminates this problem. The LF crossover should be set at 210Hz. Other settings should follow Section 3.0.

6.4.3 Input Limiting and Mixing

As in recording tracks and multitrack mixing, the Dominator II may provide protection, equalization, greater loudness, and layering (program limiting). See Section 6.1 and 6.2 for suggested settings.

6.5 BROADCASTING

6.5.1 Production

A major goal in preparing audio for broadcast is to have the same sound off air as in the studio. When production staff lacks understanding of broadcast requirements, this goal is often unachievable.

One problem is that a studio environment allows for a much wider dynamic range than a typical living room. While broadcasters recognize this problem and employ compressors to reduce the dynamic range, the compressors make decisions that should have been made in the studio. The result is a changed sound.

Another problem is that production staff are concerned only with average levels. The result is that various elements will have much higher peak values than other elements which causes the broadcast limiters to clamp down harder on those peaks, which results in "hole punching".

Still another problem is caused by pre-emphasis. As explained in Section 2.10, 75 microsecond pre-emphasis boosts 15KHz over 17dB relative to 1KHz. Many producers want their material to be brighter and brighter, especially as they get older and older. The additional high frequency content in the program combined with the pre-emphasis greatly overworks the broadcast limiters. The results are most apparent on sibilance which becomes very "spitty" and smeared while transient material such as applause becomes "crackly".

The problem of too wide dynamic range may be handled by fader moves or use of an intelligent compressor such as the Compellor. If layering of the elements is critical, the Dominator II may be used as described in Section 6.2.

Off air sound close to studio sound

The problem of varying peak outputs can be handled simply and effectively by setting the Dominator II according to Section 5.0 so that the PEAK CEILING is set at the maximum input to the recorder. This allows the mixers to continue working with VU meters without worrying about peaks. It also allows an overall hotter mix which sails through the broadcast limiters relatively untouched.

The third problem, too much high frequency, may be handled by simply reducing the high frequency content in the program. However, because that solution is often unacceptable, there is a need to control the high frequency content automatically. The Dominator II, Model 723, has additional circuitry which has the appropriate pre-emphasis in the input, and complementary deemphasis in the output. This results in flat response under limiting, and a peak output shaped to the pre-emphasis above limiting.

Use of the Model 723 in the studio causes less loudness and less brightness because of the greater amount of high frequencies that the Dominator II must control. The off air sound, however, will be very close to the studio sound.

6.5.2 AM, FM, TV, Cable Transmission

Greater loudness with original sound quality

The object of broadcasting should be the transmission of the program material without any changes. Reality, however, is much different. The necessity to handle different program levels automatically, as well as the much more sonically destructive requirement to be loud do indeed cause changes to the audio. The Dominator II was designed to achieve greater loudness while retaining the original sound quality.

Better, more predictable AM processing

AM broadcasting is bandwidth limited. AM processors are quite aggressive because many of the audible artifacts are out of band. The Dominator II in front of these processors will give them less to do and in this way help them work both better and more predictably. The Dominator II should be set to achieve loudness according to Section 6.3. The DENSITY control may be used more aggressively since the artifacts will be out of band.

FM broadcasting is more sensitive to processing because of its wider bandwidth. Most commercial stations demand loudness which requires greater amounts of processing. Typically, there are slow gain riders followed by pre-emphasized limiters, followed by a stereo generator, followed by a composite clipper. It isn't surprising that listeners cannot listen for very long periods of time!

Less work and greater predictability for final limiter

If there is a pre-emphasisized limiter in the broadcast chain, the Model 720 should be used directly in front of it. The Dominator II should be set up for maximum loudness with tolerable distortion as described in Section 6.3. This will allow the final limiter to work less, and more predictably. The Dominator II is particularly useful in between a multiband compressor and the final limiter, since the multiband compressor typically generates a tremendous number of peak overshoots.

The Model 723 may be used as a final limiter. For this application, the stereo generator must contain the required 15KHz lowpass filters. Aphex highly recommends bypassing the pre-emphasis circuit in the stereo generator, allowing the Model 723 to provide all pre-emphasis. In this case, the Model 723 has its de-emphasis turned OFF. However, if the stereo generator pre-emphasis cannot be defeated, the Model 723 must then have its de-emphasis turned ON.

Greater fidelity, punch, and dynamic feeling for FM stations The Dominator II will give an FM station greater fidelity, punch, and the feeling of dynamics while maintaining competitive loudness. Since TV audio in most of the world is FM, the problems for TV audio are the same as for FM radio. The additional TV audio problems stem from a much wider dynamic range and the fact that most of the program is dialog. Because people are more sensitive to distortion on dialog than on music, processing must be of higher quality, and must not be aggressive.

The Dominator II Model 723 with the de-emphasis circuit engaged should be used in front of a TV stereo generator because the generator has its own pre-emphasis.

If a stereo generator has its own processing, the Model 720 should be used; in this case, all processing in the generator should be bypassed except for the pre-emphasis limiter. Both Models should be set up according to Section 5.0 in order to retain a maximum of fidelity.

Cable TV presents essentially the same circumstances as broadcast TV when the cable operator must de-modulate signals from various sources and modulate again for distribution. Also, a cable operator often must add local spots to a source, spots that may have very different levels than the source.

Channel-to-channel level consistency with highest quality for cable TV

Aphex recommends using both the Compellor and the Dominator II (typically Model 723) on each channel to provide channel-to-channel level consistency while maintaining the highest quality. For this use, the Dominator II should be set up according to Section 5.0.

6.6 SATELLITE UPLINKS AND STLs

Satellite uplinks and STLs both suffer from limited dynamic range, and most use pre-emphasis to reduce high frequency noise. To maximize signal-to-noise, and protect from overload, many people use conventional limiters.

As indicated in Section 1.0 Introduction, these limiters have severe sonic drawbacks. In addition to sonic degradation, they do not provide protection from peak overloading because of pre-emphasis. Further, even if these limiters are brick wall they must be set 17dB below maximum peak input to the uplink. This then causes the high noise floor to become even more apparent, especially when downstream processors bring up the low level signals.

Maximum signalto-noise ratio with overload protection The Dominator II Model 723 is the perfect solution. It maximizes the signal-to-noise ratio and at the same time provides overload protection. It is particularly effective for mobile recording trucks which transmit back to the station or to a satellite. For this type of application, the Dominator II should be set up according to Section 5.0.

6.7 TAPE DUPLICATION

Maximum level with signal quality for tape duplication Tape duplication for audio and video cassettes requires a "hot" master for optimal performance. High speed duplication makes that requirement even more critical. The Dominator II will maximize the level on the master while maintaining signal quality. If the playback will be bandwidth limited, the Dominator II may be set for maximum loudness.

Because some audio tapes and duplication systems have much lower high frequency head room, Aphex recommends using a Dominator II Model 723 with a pre-emphasis curve similar to the peak response characteristics of the tape/system.

6.8 TELECONFERENCING

Speaker, amplifier, transmitter, and echo cancellor overload eliminated It is important, for both playback and transmission, to set and maintain peak levels. This is done by finding the maximum peak levels for the various stages, and then setting the Dominator II according to Section 5.0 in this manual. Doing this will eliminate such problems as speaker and amplifier overload, transmitter overload, and echo cancellor overload.

If there is a requirement for low levels to be brought up, the Compellor should be used in front of the Dominator II.

6.9 USING THE DOMINATOR II WITH OPTIMOD 8100A

Maximum benefit in broadcast chain achieved with Optimod As part of a broadcast chain including the Optimod, the Dominator II will help maintain loudness while providing greater clarity, openness, detail, and apparent dynamic range. The procedures that follow will assist the user in installing and setting the various pieces of equipment to achieve maximum benefit.

6.9.1 Equipment Configuration

- a. Feed the output of the studio to a COMPELLOR.
- b.Feed the output of the COMPELLOR into an AURAL EXCITER TYPE III.
- c. Feed the output of the AURAL EXCITER into the DOMINATOR II MODEL 720.
- d. Feed the output of the DOMINATOR II into the OPTIMOD.

6.9.2 Modifying the Optimod

- a. Replace cards 3 and 4 in the Optimod with Aphex 510 Cards. The Compellor and the Dominator II replace the functions of these cards with higher quality audio paths and more musical gain control. The Aphex 510 Cards consist of a high quality input stage and peak metering circuit. Aphex recommends removing card 5 also, since it no longer serves any purpose. All three cards should be stored carefully without disturbing the calibrations, for use if the Optimod ever needs to be restored.
- b HF LIMITING INDICATOR (OPTIONAL): The stock Optimod HF Limiting Indicator does not show any activity until there is approximately 8dB of limiting. An additional resistor for each channel on card 6 will provide more accurate indication.
 - Locate two 100K Ohm resistors on card 6, designated R628 and R662.
 They are just behind the HF Limiting control; refer to Figure J-8 in the OPTIMOD MANUAL.
 - Convert these two resistances to 26.5K Ohm by a 36K Ohm resistor across each one.

6.9.3 Setting Up the Equipment

a. Compellor

- Feed an 0 reference tone into the Compellor, and read the INPUT METER. If it does not indicate 0, adjust input and output settings as described in the manual.
- Set PROCESS BALANCE at 11 to 12 o'clock
- Adjust DRIVE so that 8 to 10dB is indicated on the GAIN REDUCTION METER.
- Adjust OUTPUT so that the OUTPUT METER reads 0.
- Set SILENCE GATE at 12 o'clock.

b. Aural Exciter Type III

- Using MODE A, set NR THRESHOLD so that the LED indicator shows 0 on low level signals.
- Set TUNE, PEAKING, NULL FILL, HARMONICS, and TIMBRE at 12 o'clock.
- With AX in, start mixing in the effect with the MIX control until the
 effect is audible.
- Aphex recommends reading the AURAL EXCITER TYPE III manual for guidance on adjusting the individual controls.

c. Dominator II

- Follow procedure in Section 3.0 of this manual for the initial setup.
- Adjust the EQ, INPUT, or PEAK CEILING controls so that there is approximately 3 or 4dB of limiting.
- Adjust the RELEASE and DENSITY controls for loudness without audible distortion.

d. Optimod

- Using a mono signal or test tone, adjust the INPUT LEVEL controls so
 that both left and right channel input levels show exactly 0VU on the
 meter. Switch the meter indication to L-R and trim to lowest value.
- Set HF LIMITING full clockwise (hard).
- Set CLIPPING to 0 to start.

7.0 TECHNICAL DESCRIPTION

The **Dominator II** is a stereo processor with two identical channels. The circuit description refers to the left channel, but by substituting the equivalent reference designators all information applies also the right channel.

7.1 I/O FUNCTIONAL DESCRIPTION

7.1.1 Servo Balanced Input Stage

[Reference: Fig. 68172-1B.SCH] The input stage consists of U101A&B. RN101, a precision resistor network, forms a bridge around U101B to receive the input signal. VR102 serves as a fine bridge balance trim to allow peaking the common mode rejection. U101A serves as a current to voltage converter to produce the output voltage.

When the input stage is not overloaded, there is no signal voltage at any opamp input node. This assures maximum input stage linearity and maximum common mode breakdown voltage. U101B establishes a servo loop to maintain this condition. U101B, pin 5, is grounded to the input jack ground, pin1. This point, the only point where the chassis is connected to the power ground system, becomes the zero volt signal reference to reduce the possibility of ground loop hum. Signal input on XLR, pin 2, feeds the RN101 bridge, causing U101B to produce an output signal on pin 7. Signal input on XLR, pin 3, feeds forward directly to U101A, causing an output signal at pin 1. Because RN101 bridges the current summing node of U101A, the output signal from U101B injects a complementary current into the output summing node. In this way, the input stage preserves perfect symmetry of input sensitivity.

If the input signal is unbalanced, pin 1 **or** pin 2 may be hot with no effect on gain. To prevent possible noise pickup, however, it is desirable to ground the undriven pin although it may be left open without negative effect.

The variable feedback resistance of VR501A establishes input gain. The variable trimmer VR101 compensates for resistance and linearity tolerances of the potentiometer. This allows precise gain matching of the two channels, and the establishment of precision unity gain throughput.

The input lines are passed through an RFI (radio frequency interference) filter consisting of R101, R102, L101 through L104, and C101 through C104. The filter, a modified butterworth fourth order lowpass filter, has a cutoff frequency of approximately 380 KHz. Virtually no phase shift is introduced within the audio range.

C105 through C108 block any offsets that arrive at the input. Sonic perfection is enhanced by the 'composite' capacitor approach--using a high grade mylar capacitor to bridge a high quality non polar electrolytic. This eliminates any problems of dielectric absorption or dissipation in the electrolytic.

7.1.2 Servo Balanced Output Stage

[Reference: Fig. 68172-5B.SCH] The output stage consists of U103A&B and U104A&B. U103A&B form a cross-coupled differential amplifier with the peculiar ability to sense if one of the outputs is shorted to ground. With no outputs shorted, the cross-coupling establishes a 50 percent gain reduction of both polarity amplifiers. If one output is shorted, the lack of cross-coupling to the other stage provides a doubling of gain in the unshorted stage. Doubling causes the active output to double in level, which provides correct gain and level to equal the balanced mode. Also, the doubling of cross-coupled feedback to the shorted stage produces a precise differential gain null, and removes the output current dumped to ground from the shorted stage.

Either output may be shorted; the circuit will behave identically in a symmetrical manner. VR105 functions to null the common mode output gain.

U104A&B function as a DC servo to eliminate any DC offsets at the outputs of U103A&B. To eliminate any squarewave tilt, the low frequency corner of this servo is less than 0.1Hz. Output coupling capacitors are thereby eliminated, which improves the sonic clarity of the **Dominator II**.

7.1.3 Relay Bypass

[Reference: Fig. 68172-1B.SCH] K101 is a 3-pole relay with two functions. First, it provides a direct path from the input XLR to the output XLR when deenergized. Second, it opens the connection of the selectable input terminating resistor when de-energized. This scheme allows a transparent bypass with no loading, while allowing a loaded or unloaded insertion in the line.

7.2 LEVEL TRANSLATOR

[Reference: Fig. 68172-2B.SCH] The circuit optimizes limiter signal to noise ratio (SNR) for low or high line levels. The RANGE switch dictates the state of this circuit. In the O position, the SNR is optimized for HIGH line level such as O, +4, or +8dBm. In the -10 position, SNR is optimized for -10dBV operating level.

The circuit inserts a switched gain stage (U509A) after the input stage but before limiting, and inserts a switched gain stage (U510A) after the limiter summing stage. These gain stages are orchestrated to add 10dB of gain ahead of limiting, and equally cut gain after limiting in the -10 RANGE position.

Several incidental circuits on the schematic deserve mention:

- U511A serves as a phase inverter to obtain the necessary differential drive to the servo balanced output stage.
- Q506 is a constant current source for the LED indicator string which includes all the switch position indicators except PROCESS IN/OUT.
- Q504 is a logic inverter for the RANGE function.
- Q503 temperature compensates the RELEASE control function.
- SW504 couples the left and right limiter control voltages.

7.3 BAND SPLITTER

[Reference: Fig. 68172-3B.SCH] R108 receives signal from the level translator circuit. U501A, U502A, and U503A comprise a modified state variable crossover. SW502 and SW503 switch low and high crossover frequencies, respectively. VR502A and VR503A, the LF EQ AND HF EQ controls respectively, provide ±5dB gain adjustment for low and high band outputs. The mid-band output is maintained at unity gain. U504A and U505A serve as inverting buffers to re-establish correct phase and load the EQ controls for correct taper shaping.

VR103 and VR104 trim flat response for center detent EQ positions precisely. The three band outputs are fed to three limiters, described in the following paragraphs.

7.4 BAND LIMITERS

[Reference: Fig. 68172-4B.SCH] The circuit shown is one of three identical limiters in each channel. The heart of the limiter is a voltage controlled amplifier composed of U303, the VCA, and support stages U301A and U301B. VR301 and VR303 are the coarse and fine control feedthrough null trims, respectively. VR302 is the DC offset trim. U302B is a phase flipper circuit which establishes the polarity of peak detection. (The phase flipper is controlled by a phase bus originating on the Aux. control card.) Q301 serves as a switch to short out the signal at pin 3 of U302B. If Q301 is open, U302B acts as a non-inverting unity gain stage. If Q301 is closed, U302B acts as an inverting unity gain stage.

A detector composed of U304A&B controls limiter gain reduction. U304A compares the peak amplitude of the audio signal from U302B to a DC reference voltage known as ALT. If the audio peak is greater than the ALT reference, the U304A output swings negative; otherwise, it remains positive.

When the comparator output is negative, C304 charges to a negative voltage through R317 and D302. The rate of charge (attack time) is regulated by the value of R317. U302A, a high impedance voltage follower, drives the VCA control point. As the voltage on C304 goes negative, the VCA begins attenuating. Within 5 milliseconds, the VCA gain is attenuated enough to bring down the peak output level below the ALT reference level, at which time the comparator output snaps positive again. The voltage, developed more slowly on C304, discharges through a constant current source consisting of Q302 and R319. The rate of discharge (release time) is determined by the programmed current through the current source which, in turn, is programmed by a voltage from the RELEASE TIME control.

When there is no limiting, the Q302 current source tends to draw C304 to a positive voltage. But the clamp circuit of U304B prevents the voltage on C304 from going more positive than zero volts by absorbing all the current source output under that condition.

7.5 BAND SUMMING

[Reference: Fig. 68172-3B.SCH] The three limiter outputs arrive at U506A, an inverting three input summing amplifier. The summed output at pin 1 feeds through header H503 to the clipper circuit located on the clipper Aux. board. The summed signal also feeds through header H502 to the ALT rectifier circuits on PCB 68-177 "AUX CONTROL."

7.6 PROGRAMMABLE CLIPPER BOARD

[Reference: Fig. 68-179.sch.] The summed limiter signal arrives through header H503. Ula is a buffer stage with unity gain. Input resistors R1 and R2 form a divider, clamped by the clipper circuit made up of transistor array QA1. The clipped signal returns to H503 and is passed on to the Level Translator circuit previously described in Section 7.2 of this manual.

The clipper circuit is a dual differential op-amp configuration that uses a simple high speed topology in a bipolar current clamping mode. Clipping occurs when the audio signal at the junction of R1 and R2 exceeds the bias voltage established by the clip ref. signal on H502, pin 5, a method that produces perfectly square clipping over a wide range of amplitudes. The clip ref. voltage, derived directly from the output ceiling control, follows a specific relationship to the limiter thresholds. In this way, clipping is coordinated correctly with the ALT system.

U2A&B serve as clip ref. buffers.

7.7 PRE/DE-EMPHASIS CIRCUIT

[Reference: Fig. 68180-1B.SCH and 68180-2B.SCH] This board adds pre/de-emphasis circuits to the programmable clipper circuit described in 7.6 above. U3B forms a precision pre-emphasis circuit imposed on the path from the input stage to the level translator stage on the main board.

U3A forms a precision de-emphasis circuit imposed between the limiter summing stage and the level translator stage of the main board. De-emphasis is selectable or defeatable by SW1.

7.8 AUX. CONTROL BOARD

[Reference: Fig. 68-177-1.sch and 68-177-2.sch.]

7.8.1 Peak Ceiling Reference Generator

The peak ceiling reference generator is composed of the circuit involving U1D. The peak ceiling switch selects one of twelve resistor pairs which set the DC output level on pin 14. These twelve steps represent the coarse limit threshold settings. Fine adjustment is made by a potentiometer, located on the main board between pins 3 and 4 of H101, that provides an offset gain of ±1dB. The peak ceiling ref. feeds the peak clipper through H101 as well as the ALT generator.

7.8.2 ALT Generator

The circuits of U1C and U2 form the ALT generator. U2 is a comparator which receives two signals: the peak ceiling reference voltage at the (+) input, and the full wave summed limiter output signal at the (-) input. When the peak signal is below the reference level, the output voltage rises to +15VDC. When the peak signal exceeds the reference signal, the output voltage is zero. A pulse train with constant rise time but variable pulse width is thus established out of the comparator.

C6 and R31 form a differentiator which converts the said pulse train into a constant pulse width, variable interval pulse train that causes Q1 to switch on and off accordingly. D6, keeping the differentiator centered, provides a symmetrical current path to offset the loading effect of the Q1 base current.

The charge on C5, the final ALT reference voltage buffered by U1C, is developed according to the description below.

Q1 is a chopper that modulates the discharge path of C5 by means of the pulse train developed by C6 and R31. Charging of C5 occurs between discharge pulses. The charge path reflects back through R32 and one section of the Density control. If U6D pin 13 is low, the Density control sets the initial ALT voltage as 50 to 100 percent of the Peak Ceiling reference. If U6D pin is hi, the Density control has no effect, and the initial ALT voltage is equal to the Peak Ceiling reference.

7.8.3 Test Tone Detector Description

This circuit is used to continuously detect if a test tone is being fed into the Dominator. The purpose is to automatically disable the Density control so a tone will always be allowed to reach the calibrated limiting amplitude, and then instantly revert to normal Density control during program audio. This circuit operates on the assumption that only a test tone and not any form of program audio will have a perfectly stable and constant amplitude for over three seconds.

The workings of the tone detector are not at all obvious. The head end of this circuit is comprised of the circuit of U1B. This is a unipolar amplifier having a nonlinear taper circuit created by LEDS LD1, and LD2 to extend the effective dynamic range of detection. Q2 and Q3 split the output signal as two rectifier outputs. Q2 drives the track and hold circuit of C11, R78, U6A, and U7A. Q3 drives a "motion detector" made up of U3A, U3B, and Q4. When the signal from the meter rectifier has no amplitude change, the output of U3B is stable. At even the slightest disturbance of amplitude, however, FU3B outputs a pulse resetting the zero charge of C10. If there have been no amplitude disturbances for about 250 milliseconds, C10 attains a charge of 10 volts. At this point, U6A, an open-collector comparator, generates a ground closure at its output. This allows C11 to drain out its tracking charge. The whole reason for this elaborate system is to allow the peak tracking voltage on C11 to be agile and follow program level changes, but to be totally ripple free for low frequency test tones.

The circuit of U7B, U6B, C, and D form a second "motion detector" which analyzes the character of the track and hold voltage of C11. The circuit of U7B is a lag-pump motion amplifier which works by greatly amplifying the difference between the original track-hold voltage and an integrated version of the same signal. The output of U7B is a voltage of about 7.5 volts which swings about if any change in amplitude of the track-hold voltage of C11 is present.

U6B and C act as a window detector which discharge the charge on C13 if motion is produced by U7B. When the charge of C13 is discharged, it forces the output transistor of comparator U6D to conduct, or switch on. This grounds the line to the low end of the Density control, allowing the Density control to generate the normal potentiometric output. When the voltage of C13 has not been discharged by a motion detect for about 3 seconds, the voltage rises beyond 7.5 volts and reverses the state of the U6D output. This defeats the potentiometer action of the Density control, as required for test tone limiting. The ALT timing capacitor C5 is also paralleled by under this condition switched capacitor C19 to greatly slow the response of the ALT circuit for tone testing.

7.8.4 Phase Detector

U5B receives audio signals from the two input stages, and acts as a summing stage. D1 and D2 provide threshold bias for D3 and D4, which form a peak differencer with C3, C4, R45, and R46. U5A acts as a comparator with hysteresis. The output of U5A operates Q6 which, in turn, provides a zero or -15VDC output to the Phase bus used by the band limiters. In this way, the peak asymmetry causes the Phase bus to switch levels according to the polarity with the greater peak amplitude.

7.8.5 Limiting Meter Driver

The input audio of the two channels is first full wave rectified, and then ored together. This is accomplished in an obvious manner by the circuits of U5C&D and U4C&D. U4B, a logarithmic amplifier, receives the ored signal. U4A, another logarithmic amplifier, receives the Peak Ceiling reference voltage. Since the two signals are of opposite polarity, adding their voltages is the same as subtracting their absolute values. This is performed by U3C whose output represents the exact amount of peak limiting, if any, in the **Dominator II** at any given instant. If the output is positive, there is no limiting. If the output is negative, limiting takes place in a decibel linear relationship to the magnitude of the voltage.

U3D and U1A form a peak holding circuit to capture the negative output of U3C, and drive the meter display circuit. The peak acquisition is virtually instantaneous, while the fallback time is variable by means of the current source consisting of Q1&2 of QA1. The release time bus controls the current source. Thus, the meter display indicates the release time of the limiters, useful for getting an idea of the Release Time setting.

7.9 METER DISPLAY

[Reference: Fig. 68-176.SCH] U1 drive 10 LEDs in a series string technique to conserve power supply current. VR1 trims the full scale sensitivity.

The Peak Ceiling switch, SWl, is co-located on this board.

9.0 SPECIFICATIONS

AUDIO SPECS: RANGE SETTING	OdB	-1OdB	
NOMINAL GAIN:	OdB ±15dB	Same	
OUTPUT NOISE:	-81dBu	-89dBu	
THD	<.005%	Same	
SMPTE IMD:	<.006%	Same	
DIM.:	<.006%	Same	
FREQ. RESP.:	±0.2dB 2Hz-75KHz	Same	
MAX INPUT (MIL):	+27dBu	+23dBu	
MAX OUTPUT (MOL):	+22dBu (RMS)*	+12dBu (RMS)*	
CROSSTALK:	>70dB up to 20KHz	Same	
DYNAMIC RANGE:	104dB	102dB	
CONTROLS	ADJ. RANGE		
INPUT GAIN	±15dB		
LFEQ	±5dB		
LF CROSSOVER	100Hz/210Hz		
HF EQ	±5dB		
HF CROSSOVER	1.7KHz/3.4KHz		
RELEASE TIME	150mSec to 7Sec		
DENSITY	-5 to +5 RCH		
OUTPUT CEILING	-9 to +25dB (PK)**		
I/O			
INPUT CIRCUITS	Servo Balanced Transf	ormerless	
OUTPUT CIRCUITS	Servo Balanced Transformerless		
INPUT CONNECTORS	3-Pin XLR Female		
OUTPUT CONNECTORS	3-Pin XLR Male		
INPUT IMPEDANCE	19.5K OHMS Unterminated;		
		nel Selectable Terminator	
	(Terminator Lifts in By		
OUTPUT IMPEDANCE	65 OHMS		
INPUT CMRR	Better than 60dB 20Hz to 10KHz		
INPUT RF REJECTION	Better than 40dB at 800 KHz, Better than 60dB		
	Above 2MHz		
MISCELLANEOUS		-	
POWER	100/120/220/240VAC 50/60Hz 30 Watts		
POWER FUSE	100/120VAC = .375A (SLO)		
	220/240VAC= .25A (S		
WEIGHT	5.6 Pounds		
SIZE	19"X 1.75" X 9.5"		

^{*}MOL is limited by the peak ceiling setting. The output stage is capable of +25 dBu into 600 OHMS.

^{**}dB (PK) = peak value of sinewave