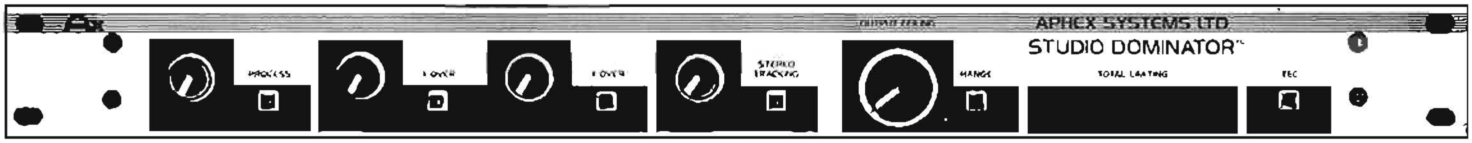
STUDIO DOMINATOR™

MODEL 700



OPERATING GUIDE
&
SERVICE MANUAL

APHEX SYSTEMS LTD. 13340 Saticoy Street, North Hollywood, CA 91605 (818) 765-2212

THE STUDIO DOMINATOR

TABLE OF CONTENTS

1.0 INTRODUCTION

The Studio Dominator from Aphex Systems, Ltd. is a stereo multiband peak limiter designed to fit a wide range of audio applications. Through the use of multiband techniques along with new proprietary circuits , the audibility of limiting action has been greatly reduced, especially when compared to conventional limiters. This means that greater limiting depth is possible, resulting in higher loudness with maintained audio quality. At virtually any limiting depth, the Studio Dominator is free of "hole punching", "dullness", and most other sound deterioration normally associated with limiters. As a peak overshoot protection limiter, the Studio Dominator is undetectable in line while it absolutely prevents peak levels from exceeding a user settable output level. In addition, the desired limiting effects of greater audio density, increased "punch", etc., are readily available with the Studio Dominator.

The Studio Dominator is easy to adjust and use. It is recommended, however, that the user read this operating manual in order to fully understand the capabilities of the device and how to best set up for operation.

The Studio Dominator will find applications in all parts of the audio industry. Some examples are:

* Sound Contracting- protection of amplifiers and speakers from overload; increased loudness; maximized use of available power.

* Recording - preventing sudden peak overload of mixer or recorder; tightening tracks; special effects, etc.

* Digital Sampling - Obtaining good full scale samples free from peak overload, i.e., no more missed samples.

* Digital recording - insuring clean recording by stopping A/D clipping of peaks and overshoots.

* Satellite Uplink - Modulation control to prevent splattering on high frequency audio, reduced distortion, better signal-to-noise ratio.

* Broadcasting - AM and FM Modulation control for greater loudness; cleaner sound; use in production for greater consistency of tapes, punchier voiceovers.

* Location Film Shoots - anti-crash for dialog and sound effects recording.

* Post Production - Soundtrack peak control; managing difficult dialog; controlling transient sound effects.

* Optical Recording - preventing "valve clash" with higher average level for low distortion and better signal-to-noise performance.

* Analog Disk Mastering - peak control for higher allowable average cutting levels; less limiter degradation to the program; brighter, punchier sound.

* C/D Mastering - peak and density control for more accurate digitizing, cleaner sound requiring less error correction on playback; no limiter induced sound degradation.

2.0 FUNCTIONAL DESCRIPTION

Traditionally, limiters have been designed specifically for a particular application, or with a multitude of interacting controls that must be adjusted to adapt the unit to a particular purpose. Because the Studio Dominator circuitry is "intelligent", only a few basic adjustments are necessary to get proper results in any application. The user simply sets the OUTPUT CEILING to the level where peaks must absolutely stop (such as 100% modulation or just under system overload), and sets the DRIVE control to obtain the desired amount of limiting. Some additional controls are provided, but they are mainly for those who wish to obtain effects from the limiter.

2.1 Multiband vs. Wideband processing

Much of the Studio Dominator's high performance is due to a newly created multiband processing technique that overcomes the problems previously encountered with both wide- and multi-band limiting.

One of the main problems of wideband limiting is "spectral gain intermodulation" or "hole punching", the audible effect of the more powerful low frequencies, such as a bass drum, causing the whole program, including the high frequencies, to drop momentarily in level. Wideband limiters also tend to sound dull for this reason.

Multiband limiting attempts to solve this problem by splitting the audio into two or more bands, and processing them separately. But more bands means more parameters to control and the summing of the separate bands to deal with. For a three band limiter, the number of controls could easily run to twelve or more per channel!

The Studio Dominator uses program dependent, "intelligent" circuits to eliminate all but the most necessary controls. This dynamic control also tends to make the limiting less audible by providing the right amount of processing at the right time.

2.2 ALT (Automatic Limit Threshold)

Though free of "hole punching" multiband limiters generally do not keep the output peak amplitude at a consistent level. This is due to the varying amount of band energies being added together together with different amounts of limiting in each band. Such a characteristic is usually unacceptable (at least in broadcast applications), so a final clipper is added to chop down the final level to a reliable amplitude at the expense of added distortion. Another solution is to add a wideband limiter at the output, but that only re-introduces the problems of wideband limiting and negates the advantages of multiband. A new solution had to be found.

The answer is a new circuit concept termed ALT (Automatic Limit Threshold, Pat. pending). Incorporated in the Studio Dominator, ALT accurately manages output summing levels in a way that produces no distortion and maintains the full advantages of multiband limiting.

In the ALT system, the limiter peak output is constantly monitored and adjusted to a reference level by controlling the limiter thresholds. Below-threshold gain is not changed, so low level signals are not modified by the ALT process. Since ALT has a finite reaction time, a soft "catch clipper" is used following the final summing to stop any brief overshoot transients. This does not produce any audible distortion because the ALT prevents all but the briefest overshoots, too short to hear.

By these methods, the summed signal peak ceiling is guaranteed to remain at a calibrated value. The maximum output level can now be made accurately adjustable by a single front panel control. This is a highly desirable feature not commonly found on limiters.

Due to intelligent interplay between the ALT control, the attack timing of the individual bands and the catch clipper, the "come on" of the limiter is extremely smooth sounding if it can be heard at all.

2.3 EQ, Release Time and TEC

To allow for variations in operating conditions and tailoring of the sound, several features are available on the Studio Dominator.

The band crossover frequencies are selectable, with the drive levels into the low and high bands adjustable by $\pm$ 6dB. This allows static and dynamic equalization effects to be obtained. As the bands get deeper and deeper into limiting, the EQ effect will diminish as all bands approach equal amplitude.

The release time allows the user to obtain more or less density (loudness) under heavy limiting. A quicker release time gives greater loudness by maintaining a small peak-to-average ratio. The release time may also be used for maximum transparency at different average amounts of limiting.

All limiters, by their nature, remove transients that create the live feel in audio. A unique circuit called TEC (Transient Enhancement Circuit, Pat. pending) restores some of this transient "feel". Output level is not changed, but a psychoacoustic effect is accomplished through slight envelope modification of the components below threshold. TEC may be switched in or out as desired.

2.4 The Gain Control Circuits

Extremely low distortion limiting is accomplished by the use of type 1537A VCAs in each band coupled with individual precision comparator/pulse charge gain control circuits. The attack time is sufficiently fast to prevent audible overshoot clipping (with the help of ALT) but is slow enough to prevent "hole punching" by transients such as record pops, etc. The limiting thresholds for all bands are equal and are DC programmed by the ALT bus. Release time is likewise equal and DC controlled.

2.5 Tracking Control

TRACKING locks the gain control signals together for each band. This provides the most stable stereo image. However, greatest average loudness is acheived when the two channels are operated independently.

2.6 Output Ceiling Control

The calibrated OUTPUT CEILING control is the only control on the Studio Dominator that is not a DC control. It is a precision attenuator that is adjustable in 1dB steps from -2dBU to +21dBU. In-between levels can be accomplished by an internal trimpot if desired. This setting represents the actual <u>peak</u> voltage of the output signal, not RMS voltage. This limit will not by exceeded under ANY conditions of limiting. The RANGE control selects HI or LOW range of the OUTPUT CEILING knob.

2.7 Factory Options

To adapt the Studio Dominator to the widest variety of applications, several factory options are available.

a) Pre/De-emphasis - Allows insertion of a precise preemphasis curve prior to limiting. Deemphasis is jumper selectable to restore flat response if desired. 75 uSec is standard, others available.

b) Lowpass Filter - In addition to the above, an overshoot corrected lowpass filter of exceptional quality may be installed to restrict bandwidth for broadcast or digital recording. 15kHz is standard.

c) Matrix/Dematrix - For AM stereo, this option allows L+R and L-R limiting with selectable de-matrixing.

These options are available as factory installed only, and are identified by separate model numbers. Refer to Section 7.0 for further information.

3.0 A DISCUSSION OF GAIN REDUCTION

There are several misunderstandings about limiting and compression among audio professionals. Compression and limiting are sometimes referred to interchangeably, for example.

To clearly understand the operation of the Studio Dominator, it is good to have a standard definition of terms, and have a clear understanding of the differences between leveling, compression, limiting, and clipping. Therefore, this section will discuss the terminology, offer what we think are the most accurate definitions, and describe the basic principles.

3.1 Leveling

"Leveling", or AGC, is control of long-term average levels. The ratio is fairly high (>10:1), with slow attack and release times, and a low threshold. Because of the slow operation, leveling has little or no effect on short-term changes in average level or on transients.

3.2 Compression and Limiting

"Compression" is a popular contraction of the general term "dynamic range compression", where "dynamic range" refers to the difference between the minimum signal level and the maximum signal level from an audio source over a period of time. Limiting may be defined as a subset of compression because the dynamic range is reduced through limiting. However, because there are significant differences between what are called "limiters" and "compressors", it is best not to equate compression and limiting in application oriented thinking.

Compressors are used principally where it is desired to force an audio signal to maintain a more constant power level. This may be to improve the signal-to-noise ratio in a system, to correct excessively varying volume levels from a program source, or for effects like gathering greater loudness. One way to think of this is that compression tries to keep levels from falling too low. Limiting, in contrast, is used where it is required that a signal be stopped from exceeding some specific level. These applications include prevention of overmodulation, speaker protection, overcutting protection, system overload protection, transient peak elimination, etc. For certain applications it is necessary to combine a compressor and a limiter to get the desired results. This might be where we are required to maintain a high average signal level constantly near to but not exceeding the system overload point. Broadcasting is an example of this application.

The parameters which mainly differentiate between limiters and compressors are: compression ratio, operating threshold, and timing. The circuits to create both types of processors are essentially similar, using a variable gain cell and a detector circuit to control the cell. Only the above parameters need be changed to realize either a limiter or compressor from such a circuit. Changes in any of the three parameters will result in significant sonic differences. Therefore, there is not only a science to designing a good compressor or limiter, but a good measure of art as well. It is important for the user to appreciate these factors because he plays an artistic role in the way he uses and adjusts the equipment.

3.3 Compression Ratio

"Compression ratio" refers to the ratio of a decibel change of input level to the corresponding decibel change of output level of a device. Thus, if a 2dB change of input caused a 1dB change of output, the compression ratio would be 2:1. Perfect linear circuits produce a 1:1 compression ratio. Compressors and limiters use compression ratios ranging from about 1.1:1 to about 30:1. This ratio is sometimes referred to as the compression "slope". Variablé slope compressors are available which begin compressing with a small ratio and gain ever increasing ratio as the input signal rises in level. Most compressors and limiters generate a more or less constant slope, however. In general, compressors operate with low ratios of 1.1:1 to 3:1; levelers and limiters with greater ratios, from 7:1 to 20:1.

3.4 Operating Threshold

The operating threshold, either refered to as the "compression threshold" or "limiting threshold" depending on the processor, is the input referred signal level where dynamic range modification begins to take place. All limiters and most compressors operate as linear amplifiers with a relative slope of 1:1 below this threshold. The most notable exception to this rule is a compressor associated with some "companding" noise reduction systems where the compression slope begins close to the noise floor of the compressor circuit. That is to say that constant ratio compression takes place over nearly the entire dynamic range of the system. In all other compressors, the compression threshold is a user variable setting which determines the "depth" of compression which will be used. The lower the threshold setting, the greater amount of compression is obtained. For limiters, the limiting threshold is also a user adjustment.

The Studio Dominator, rather than making the threshold user variable, operates with a self controlled threshold. A variable input gain control is then provided to adjust the "drive" to the gain controlling and detector circuits, allowing the Output level control to be precisely calibrated. This method offers the advantage of optimized noise and distortion design in the circuitry, calibrated output level, and simplified operation.

3.5 Attack and Release Times

Timing parameters are extremely critical to the operation of limiters and compressors. The two timing factors of greatest importance are attack time and release time (sometimes called recovery time).

The attack time is the time the processor requires to bring the input signal under 90% control after the input level exceeds the operating threshold. Limiters usually incorporate fast attack times to prevent sudden signal increases from escaping amplitude control. Compressors usually incorporate slower attack times to prevent washing out of transient sounds. Because of this, compressors will generally produce overshoots at the output well above the average signal level. Often, the attack time is not adjustable by the user, and thus becomes part of the characteristic sound of the particular model of compressor.

Release time is the time required by the processor to restore itself to 90% of full gain when the input signal drops below threshold. This time can vary from a fraction of a second to many seconds depending on the particular device. Very often this parameter is made user adjustable both for limiters and compressors even if the attack time is internally fixed. Longer release time results in less loudness because the gain does not restore low levels quickly. Faster release time results in correspondingly greater loudness, but this sometimes comes at the cost of greater distortion (especially to low frequencies) and more audibility of the gain control effects.

3.6 Limiters and Limiting

As previously discussed, in order to be effective, a limiter typically has a threshold above 0VU, a high compression ratio(7:1 or greater), and fast attack and release times. Compressors usually operate at much lower ratios from, say, 1.1:1 to 3:1 in order to maintain a pleasing sound quality because gain reduction is taking place over a larger dynamic range. Compressors operating with a high compression ratio are mostly used when a sound effect is intended. Recording engineers may use a heavily driven limiter as a compressor when they are looking for more "sustain" or a different "texture", for example. A limiter driven to heavy gain reduction effectively becomes a compressor.

Let us now depart from further elaborations about compressors, and focus on two types of limiters: "peak limiters", and "program limiters". At this point it is important to solidify the definition of "peak level". This term is taken to mean two different things in the audio industry. Sometimes it is used to refer to the highest (or peak) VU level occuring in an audio program. Other times it is used to refer to the actual peak amplitude of the audio wave, which would be measured on a peak program meter (ppm) or an oscilloscope display, but not on a VU meter. Let us define "peak level" as the true peak waveform amplitude, and "peak program" as the highest VU level.

3.6.1 Program Limiters

With these definitions in mind, a "program limiter" can be described as a limiter having all the previously defined characteristics of a limiter, and whose detection circuits result in limiting the "peak program" level (as measured on a VU meter) to a specified level, usually zero VU, without regard for the peak level.

3.6.2 Peak Limiters

A "peak limiter" is used where the absolute audio waveform amplitude cannot be allowed to exceed a given value. Most audio systems produce particularly objectionable distortion when driven to overload, so peak limiters are often used to maintain safe peak levels below any system clipping to guarantee a distortion free system. Limiters in this application are sometimes referred to as "protection limiters". Another application of peak limiters is "modulation limiting" in a broadcast situation where a specified peak level is defined as 100 percent modulation which must not be exceeded.

Peak limiters are usually characterized by fast attack and release times, a very high compression ratio and a high threshold such that no gain reduction takes place for signals whose peak level is under the overload point of the system. Limiters with very fast release tend to generate distortion under continuous gain reduction, but for protection limiting where limiting is brief, the fast action tends to be practically unnoticeable, maintaining a clean sounding audio system.

3.7 Clippers

Most peak limiters incorporate a final clipper to catch any amplitude overshoot that may escape the variable gain element due to the short but finite attack time of the detector circuit. A properly designed clipper has a virtually instantaneous action to catch the briefest of transients. This "catch clipper" usually has little work to do, and because it clips only brief transients is inaudible in the circuit. Clipping is for protection, and performs no audible gain reduction.

At this point, the reader should be well aware of the differences between limiters and compressors, and how they might be used.

4.0 APPLICATIONS

Some common applications of limiters are:

* Obtaining more loudness
* Obtaining more "punch"
* Preventing transmitter overmodulation
* Preventing tape overload
* Preventing system peak overload
* Preventing overcutting
* Preventing satellite uplink overmodulation
* Audio P.A. system protection
* Special Effects

Conventional limiters are more suited to certain of these uses than to others. The Studio Dominator, because of its intelligent circuitry and optional configurations, is equally well suited to all limiter uses. This manual will address several general categories of limiter application and show typical setups which can serve as the basis for your particular operation.

4.1 Broadcasting

FM broadcast processing requires special consideration because of the pre-emphasis curve and bandwidth restrictions involved. An option card (and a few other simple modifications) can be installed in the Studio Dominator to produce accurate operation for FM transmission. With this option, two operating conditions may be chosen. First, pre-emphasis can be selected which causes the Studio Dominator to output a properly pre-emphasized signal ready for direct feed into a non pre-emphasized stereo generator. This is the preferred approach in most cases. Second, accurate de-emphasis may be selected after limiting to restore flat system response for feed to a pre-emphasized stereo generator.

NOTE: If the stereo generator used does not include the necessary left and right channel 15kHz overshoot compensated lowpass filters, then the lowpass filter option should be specified in the Studio Dominator. Many stereo generators include lowpass filters, but they are not overshoot compensated, which results in loss of modulation capability. In this case, the filter should be bypassed or removed and the Studio Dominator filter used.

Setting up the limiter for FM operation is basically similar to the general set-up procedure. A program signal is fed into the Studio Dominator and drive is adjusted to produce at least 8dB of limitng. Release time should be full fast initially. Raise the peak ceiling control to approach 100 percent modulation on program peaks. The output control operates in 1dB steps so it will probably be necessary to use the stereo generator input trimmers to achieve the final exact 100 percent modulation setting. If there are no such trimmers on your stereo generator, you may access the output trimmers inside the Studio Dominator for a fine adjustment of output level.

Once the modulation limits are set by the above technique, the DRIVE, RELEASE TIME, and other controls may be adjusted to obtain the desired on-air sound. Use of the TEC may be particularly beneficial since a greater sense of transient dynamics can be obtained. Additionally, by not turning on the tracking mode, wider stereo imaging and higher average loudness can be obtained. The loss of center image stability is not as severe with the Studio Dominator as expected because the ALT circuit intelligently tracks the limiting thresholds of left and right channels.

4.2 Recording and Mastering

a) Tracking

If the tape saturates, if the electronics run out of headroom, or if the digital recorder reaches peak input, - you can no longer "fix it in the mix." By setting the Studio Dominator output ceiling just below the system's peak level, these problems will never again occur.

Recording with boosted high end can be safely handled by the Studio Dominator, preventing tape saturation while eliminating the need for another equalizer in the signal path.

Using the Studio Dominator, the output of the mic preamp or input fader may be increased to make a "hotter" recording. This will effectively increase the signal-to-noise performance so that noise reduction may no longer be necessary.

b) Mixing

Very often it is desired that a lead track ride out front of the rest of the mix. By using the Studio Dominator on that track, maximum loudness can be achieved and maintained, allowing it to stay out front without having to drop the level of the rest of the mix.

Since the ratio of the Studio Dominator is infinite, any track driven into consistent deep limiting will not move, regardless of input level. This is especially useful for bass.

c) Mastering

The ideal for mastering is an invisible transfer from tape to lacquer at the maximum level possible. When using a conventional limiter to prevent overcutting, the bass or kickdrum modulates the high end, causing "holes" and loss of transient "feel". If the lead track is very transient, the whole mix may "duck" during peaks. Due to its multiband technology and intelligent control circuits, the Studio Dominator will not cause these effects. At the same time, a higher level disc may be safely cut with more predictable groove spacing.

4.3 Live Sound

As with recording electronics, P.A. equipment has headroom limitations. Although power amps amd speakers may be more forgiving than digital audio to short term peaks, various types of audible, objectionable distortion may be generated. Furthermore, the amps and speakers themselves may be driven to failure. The Studio Dominator not only prevents system overload while maintaining audio quality, but actually improves overall system performance. It does this by enabling all parts of the sound system to safely run at their maximum operating levels, continuously. A great increase in effeciency may be realized now that headroom is no longer such a consideration.

Stage monitors always demand maximum controlled loudness. The Studio Dominator's limiting, crossover and EQ controls allow precise tailoring of the sound for maximum level before feedback.

4.4 Other Applications

Once the characteristics of the Studio Dominator are understood, other applications become obvious, such as -

* Film and Video post-production- The limited dynamic range and unpleasant sound of overmodulation in these media make the Studio Dominator an essential tool in any post house.

* Tape Duplication- Master tapes can be recorded at a higher level, making the dupes "hotter" with lower noise and transient distortion.

* Satellite Transmission - Signal-to-noise ratio can be improved, and pre-emphasis induced high frequency modulation distortion avoided. Overshoots from companding can be reducede for greater channel modulation and lower distortion.

* Digital Sampling for synthesizers - Very transient acoustic sounds can be properly sampled without worrying about sampler drive levels.

4.5 Use with the Aphex Compellor (Compressor/Leveler)

As previously described, compressors bring low level signals up, and limiters hold high level or peak signals down. Using both devices in series gives you both control of average levels (compression) and peak control (limiting).

Since the Compellor is the most transparent compressor/leveler and the Studio Dominator the most effective, flexible and transparent peak processor, the combination of the two devices is the ultimate in dynamics control.

4.6 Frequency Response Tailoring

As program audio reaches the onset of limiting, low frequencies tend to be limited first, due to their greater energy. The end result is a slight perception of "brightening" the sound, enhancing impact and transient feel of percussive sounds as well as aiding intelligibility in most applications.

In the event that absolute flatness of "dynamic" frequency response is desired, a pre- and de-emphasis card may be added to tailor the limiting curve as needed. See section 7.0 for more information on factory options.

5.0 INSTALLATION

5.1 Unpacking

Your Studio Dominator was carefully packed at the factory, and the container was designed to protect the unit from rough handling. Nevertheless, we recommend careful examination of the shipping carton and its contents for any sign of physical damage which could have occured in transit. If damage is evident, do not destroy the container or packing material. Immediately notify the carrier of a possible claim for damage. Shipping claims must be made by the consignee.

5.2 Input Sensitivity Set

Before installation the Studio Dominator must be set for the correct input sensitivity corresponding to the desired operating level (OVU). It may be set to either -10 or +4 dBm. The switch for each channel is located inside the unit just behind the input jacks on the main PC board. Simply slide the switch to the desired position. Towards the power supply is +4dBm, away is -10dBm. The +4dBm position is suitable for operating levels from 0 to +10dBm.

5.3 Mounting

The Studio Dominator occupies one standard rack unit of space (1 3/4") 19" wide, with a depth of 8 1/2". Allow at least an additional 3" of depth for connectors. Be sure that there is at least 1/2" of air space around the unit for cooling. Mount with the cushioned rack screws provided.

5.4 Connectors

The audio inputs and outputs are made with standard XLR type 3-pin connectors on the rear panel. Equal phase is maintained from input to output, so the user may elect to use either pin 2 or pin 3 as high to match his balanced system.

5.5 I/O Considerations

5.5.1 Impedances

The input is of high (160 kOhms) impedance and may be easily driven by any output source. In the case of certain situations that need to see a 600 Ohm load, a 600 ohm resistor may be tied across pins 2 and 3 on the input plug.

The output is of an active, transformerless type with an output impedance of 20 Ohms between pins 2 and 3. When driving certain transformer coupled loads, It may be necessary to install 287 Ohm 1% build out resistors in series with pins 2 and 3 to create a 600 Ohm source impedance.

5.5.2 Balanced vs. Unbalanced Use

Although of balanced configuration, the inputs may be driven single ended (unbalanced) on pin 2 or 3, tying the other pin to pin 1 ground. It is recommended to use pin 3 to maintain phase matching with the outputs, which use pin 3 as high in the unbalanced mode.

A rear panel switch located near each output connector converts the outputs from balanced to unbalanced. In the unbalanced mode, Pin 2 is grounded and Pin 3 is driven high with a 6 dB increase in level. This equalizes the signal level in balanced and unbalanced modes, and maintains the output ceiling calibration for both modes.

5.6 AC Line Connector

A.C. mains power is supplied to the unit via an integral receptacle/fuse holder on the rear panel which meets all of the international safety certification requirements. Check to be sure that the unit is configured to match your A.C. mains voltage by inspecting the voltage programming tag located with the fuse holder. No power switch is provided on the unit since power can be removed by unplugging the power cord at the chassis for maintenance.

5.6.1 Voltage Selection

As shipped, the Studio Dominator can be adapted to any standard AC mains voltage simply by reprogramming the voltage programming card in the fuse holder. This is done by the following procedure:

1. Remove the power cord from the chassis receptacle.

2. Slide the clear plastic cover to the left to uncover the fuse compartment.

3. Remove the fuse by prying out the "fuse pull" lever.

4. Pull out the small printed circuit programming tag. The tag has four voltages printed on it which are 100,

220 on one side, and 120, 240 on the other side. Orient the tag so the required voltage is readable on the top left side of the tag, and reinsert the tag in the fuse holder. You should now be able to read your correct mains voltage through the window of the fuse holder.

5.6.2 Fuse Selection

Be sure to use the correctly rated fuse for your mains voltage. These fuses are:

100-120V, 0.25A Slo-Blo
220-240V, 0.125A Slo-Blo

6.0 FRONT PANEL CONTROLS

On the front panel, from left to right you will notice five control knobs and six pushbutton switches. Of these, the "drive" and "output ceiling" controls are the most important.

6.1 Drive

The drive control adjusts the amount of limiting which will take place for a given input level. It is similar to a threshold control except that no change in peak output level occurs as the drive is adjusted for more limiting. An ordinary threshold control would reduce the output level as more limiting is obtained, and thus require an output gain make-up control to restore the desired peak output level. In this case, it would be impossible to calibrate an output gain make-up control to any specific output level, since that would depend on the threshold setting. The Studio Dominator therefore is much simpler to use than the usual limiters with threshold controls.

6.2 Output Ceiling

The output ceiling control sets the absolute maximum peak output voltage from the limiter. Expression of the ceiling level is in units of peak dBu. For those not familiar with peak dBu, this is merely the peak amplitude of an audio wave relative to 1.1 volts. The 1.1 value is derived from the peak value of a sine wave producing 1 milliwatt into a 600 ohm load. Thus, dBu is the voltage equivalent to dBm without regard for load impedance, and peak dBu is the voltage equivalent to peak dBm without regard to load impedance. Since the Studio Dominator output impedance is very low, it acts as a voltage source and will thus produce the same calibrated output voltage into any normal load impedance. Thus, dBu is a convenient unit of output level calibration since it directly correlates to dBm into a 600 ohm line and is more-or-less a standard of measurement in audio systems which mix 600 ohm lines with high impedance lines. The output range of the Studio Dominator is -2 to +21 dBu in 1 dB steps.

The range switch sets the range of the output ceiling control to either -2 to +9 or +10 to +21 dBu.

To determine the output level from the Studio Dominator in peak volts (Ep) when dBu is specified, use the following equation:

$$Ep = \text{Antilog} \frac{dBu}{20} \times 1.1 \text{ volts}$$

example: output level +18 dBu;

$$E_p = \text{Antilog} \frac{18}{20} \times 1.1 = 8.74 \text{ Vpeak}$$

To determine the proper output ceiling setting in dBu if the peak output voltage (Ep) is specified, use the following equation:

$$\text{dBu} = 20 \times \text{Log} \frac{E_p}{1.1}$$

example: required output ceiling = 9 V peak;

$$\text{dBu} = 20 \times \text{Log} \frac{9}{1.1} = 18.3 \text{ dBu}$$

Note that the RMS output of the limiter will not necessarily equal .707 times the peak output level. This is because most audio waveforms have a much higher peak to average amplitude ratio than pure sine waves.

6.3 Secondary Controls

The secondary controls on the Studio Dominator and their functions are as follows:

6.3.1 LF EQ-
Adjusts the relative drive to the low band limiter.

6.3.2 LF X-OVER-
Selects 80 or 160 Hertz as the low-to-mid crossover frequency.

6.3.3 HF EQ-
Adjusts the relative drive to the high band limiter.

6.3.4 HF X-OVER -
Selects 1700 or 4500 Hertz as the mid-to-high crossover frequency.

6.3.5 PROCESS ON-OFF -
Activates the limiter circuits when "on" and deactivates limiting when "off". Equalization, drive, TEC, and output ceiling controls remain active in both positions.

6.3.6 TRACKING -
Causes left and right channel limiters to track gain reduction band-for-band when "on".

6.3.7 RELEASE TIME -
Adjusts the limiter gain reduction release time.

6.3.8 TEC-
Transient enhancement circuit which psycho-acoustically increases the transient "feel" of the audio when "on".

Most of the above controls are sufficiently self explanatory. A few words about equalization and TEC are in order, however.

6.4 The EQ Section

When the EQ controls are at 12 o'clock, all the bands are driven equally from the crossover filters. This results in flat response through the limiter. A 6dB boost or cut can be obtained for drive to the low and high band limiters using the EQ controls. For audio signals below threshold, the equalization will follow the the EQ settings. However, when the signals rise into limiting, any EQ boost will be reduced or eliminated as each band approaches maximum limiting.

6.5 TEC

The TEC circuit restores articulation to the audio by manipulating the below-threshold audio envelope without changing the average power content or peak amplitude of the audio wave. It can be used to bring out more punch in an audio source, or to improve the dimension and imaging qualities of the audio. A noticeable increase in the perceived loudness of the audio is an additional benefit of the TEC process. Depending on audio source type and quality, the effect of TEC may be subtle or quite noticeable.

7.0 FACTORY OPTIONS

Several options are available to configure the Studio Dominator for special applications. To insure proper operation of the limiter, these options are offered factory-installed only. Separate model numbers are assigned to the different option versions to avoid confusion. Studio Dominators already sold can later be changed to another version. Consult Aphex Systems for further information.

Some audio transmission and recording media require an audio bandwidth limiting means to prevent cross-channel splash or aliasing distortions from occuring. Such systems typically use time or frequency multiplexing to convey two or more audio channels over a single carrier. FM stereo broadcasting, satellite uplinking, and digital recording are a few good examples of systems requiring audio band limiting.

Generally, both recording and transmission media also incorporate some form of audio pre-emphasis which is mainly used to improve signal to noise ratio. FM broadcasting and many satellite uplinks, for example, use 75 microsecond high frequency pre-emphasis. Digital recorders usually employ pre-emphasis ahead of the A/D converter.

In virtually all of these kinds of audio transmission or recording media, it is important to fully use the available dynamic range of the channel in order to maximize the signal-to-noise ratio and minimize small-signal distortions. It is also desirable to utilize the maximum possible audio bandwidth within the physical constraints of the channel to obtain the maximum audio fidelity. Meeting these two objectives requires the use of specialized audio processing technology including pre-emphasis recognition and lowpass filtering.

Another kind of limiting is required for AM Stereo and other "matrixed" two-channel audio systems where processing the sum and difference signals through separate channels is required.

7.1 Model 702 - With Matrix/De-matrix network

This card converts the discrete left and right channel input audio to sum and difference information which is then sent to the two limiter channels. After limiting, selectable de-matrixing is available to output either discrete left and right audio or peak limited L+R and L-R audio.

The results from matrixed processing are not totally obvious but are interesting. The first notable effect is that the recovered left and right channels will not necessarily be well peak limited. For this reason, FM stereo broadcasting cannot fully utilize this type of limiting without serious overmodulation difficulties. (Due to the modulation vector summation of all signal components in the FM stereo multiplex signal, it is necessary to limit the discrete left and right channels to properly control modulation.)

Another result of matrixed limiting is that the stereo separation may become greatly enhanced if the limiters are not operated in "Tracking" mode. This is because the difference component, which is usually small compared to the sum component, can achieve higher relative signal level through separate limiting. The effect is usually desirable for AM stereo transmission, because increasing the relative difference energy level does not reduce the monaural amplitude modulation and improves the stereo signal-to-noise ratio. In this case, the AM radio listener who switches his receiver to stereo mode or tunes from a mono station to a stereo station with a stereo tuner hears the stereo signal become louder, with the center audio approximately as loud as the previous mono audio but with significant additional left and right audio level. The "hole-in-the-middle" effect produced by this type of operation is usually not noticed since the center is not perceived to diminish as much as the sides are perceived to increase.

It should be noted that if the Studio Dominator is placed in "Tracking" mode, the sum and difference information will always remain in normal proportions, and therefore the de-matrixed stereo will not obtain enhanced separation. The sum signal, being usually the greater in level, will thus control the gain reduction of both the sum and difference channels. In the unlikely case that the difference channel should contain the greater signal, the sum channel will track the difference, and thereby remain in normal proportion. Operating in this mode will not usually diminish the sum amplitude since it is normally always in control of the limiter gain. Thus, monaural AM will not be sacrificed. The AM stereo signal may not achieve the greatest possible loudness, but the stereo sound will resemble the original separation. The Studio Dominator provides simple front panel selection of the "Tracking" mode.

Some applications other than AM Stereo which make use of matrixed limiting are satellite stereo transmissions, and tape recording.

The chief advantage of matrixed recording is that azimuth errors upon playback show up as stereo high frequency separation reductions rather than monaural high frequency loss. Some radio stations record music tape cartridges in the matrixed format often with noise reduction to improve the reproduced consistency and quality. Using the matrixed Studio Dominator in the recording system can allow higher average recorded levels without distortion or compressed sound. This can eliminate the need for noise reduction and the attendant loss of detail and quality of the sound.

Stereo program distribution via satellite is often matrixed to allow simple one-channel recovery of the monaural component for radio and television stations not capable of stereo transmission. Simple de-matrixing is then performed at two-channel down-links to recover the stereo. One common problem which occurs with satellite audio distribution is sibilance or high frequency distortion due to accidental carrier overdeviation. This happens because inefficient or nonexistent modulation limiting at the uplink transmitter. In some cases, program originators have had to reduce the audio levels laid on tape, reduce the send levels, or incorporate a de-esser to compensate for this distortion. The matrixed Studio Dominator can eliminate these problems by maintaining undistorted well modulated stereo satellite carriers with highly transparent audio fidelity.

The preferred point to insert the Studio Dominator is, of course, at the satellite uplink transmitter. However, a program producer having no technical control over the contracted uplink can be reasonably assured of cleaner satellite transmission if the matrixed Studio Dominator is used at the studio for protection limiting ahead of the outgoing feed to the satellite uplink station.

7.2 Model 703 - With Pre- and De-Emphasis

Limiting according to an equalization curve such as 75 microsecond pre-emphasis can be accomplished by incorporating this card. Internal input and output patch points are accessed by the card, and the pre-emphasis is inserted ahead of limiting. Selectable de-emphasis is provided after limiting to restore flat response below threshold if required.

This option is shipped with 75 microsecond pre-emphasis as standard. A different equalization curve can be obtained by changing a capacitor or resistor. 50 microsecond pre-emphasis for European broadcasting can be shipped upon request.

Because the audio dynamics become radically different when FM broadcast pre-emphasis is incorporated, the limiter must be modified in several ways to most efficiently accommodate the signal. The following changes are made at the factory when Model 702 is ordered.

1. The ALT generator has an additional timing resistor included.
2. The high frequency limiters are given a slower attack time.
3. The high frequency limiters are given a faster release time.
4. The final summing limiter is given slightly more clipping.

7.3 Model 704 -With Pre/De-Emphasis and Lowpass Filter

This card contains the circuits of Model 702 described above (including the required limiter modifications) with the addition of an overshoot corrected 15KHz lowpass filter for FM broadcasting or other bandwith restricted applications.

The pre-emphasized Studio Dominator with lowpass filter limits spectral energy according to the amplitude constraints of the pre-emphasis curve to allow maximum dynamic range utilization of the pre-emphasized audio channel. Further, a "brick-wall" filter sweeps away all signficant energy which may be present outside the 15KHz upper frequency limit. Because of the multiband limiting and automatic limiting threshold features, sonic effects of the limiter are greatly diminished. A very transparent "unprocessed" sound can be obtained while gaining significant signal-to-noise improvements and complete protection from common effects such as sibilance breakup, main-to-sub crosstalk, etc.

The Studio Dominator's lowpass filter is a high quality ninth-order eliptical active filter type which is flat in frequency response within 0.2dB to 15KHz but is down at least 90dB at the 19KHz pilot frequency. A modified state-variable topology is used to attain a sonically excellent filter having very high stability with low noise and distortion. Out-of-band rejection is 60dB or greater above 19KHz. Phase matching is excellent between left and right channels throughout the passband and into the reject band.

The lowpass filter is inserted after all limiting. This enables the Studio Dominator to produce very well band limited extremely accurate peak modulation having high dynamic stereo separation (low main to sub crossstalk) in composite FM.

NOTE: The audio output from a Studio Dominator equipped with pre-emphasis and lowpass filters is suitable for direct connection to an FM stereo generator.

8.0 <u>OPERATING INSTRUCTIONS/SET-UP</u>

Setting up the Studio Dominator for operation is extremely simple. Once the input sensitivity has been set correctly (see sec. 5.2), the general set-up procedure is as follows:

8.1 Beginning

Start with these settings:

a. All knobs at 12 o'clock position.
b. PROCESS ON.
c. LF X-OVER at 80 Hz.
d. HF X-OVER at 1700 Hz.
e. TRACKING OFF.
f. OUTPUT CEILING anywhere.
g. RANGE on low or hi.
h. TEC OFF.

8.2 Drive

Set the drive level by inputting a typical program source to the limiter and observing the "total limiting" display. Increase or decrease the drive to obtain a nominal 6 dB indication on this display.

8.3 Output Set

Set the output level to the desired value using the Range switch and Output Ceiling selector. If the exact output level required is not known, the setting can be done experimentally with program audio.

NOTE 1: A tone burst or program audio must be used to set the peak ceiling output level because a steady state tone through the limiter will settle to 3 dB (peak) under the ceiling level. This is an important characteristic of the Studio Dominator. More can be learned about this in the theory section of the manual.

NOTE 2: At this point, the Studio Dominator is set to maintain a precise peak ceiling. Unlike most other limiters, the Studio Dominator output level will not need any correction even if equalization, drive, or any other control is adjusted <u>any</u> amount! In many installations, the output level will <u>never</u> need to be readjusted once the correct level is set.

8.4 Drive Set

Drive is now readjusted to accomplish the desired limiting action. If only "protection" limiting is desired, then the drive should be reduced so that only the highest peaks which would ordinarily "crash" the downstream equipment cause limiting. This condition causes the limiter to be totally transparent within the safe dynamic range of the audio system. When the excessive peaks occur, instant undetectable limiting takes place. Occasional limiting action assures adequate output level for optimal signal-to-noise performance.

If increases in loudness or density are desired, greater amounts of total limiting may be used. As greater average limiting is used, the audible effects of limiting will become more pronounced. This is unavoidable because of dynamic range compression. However, the Studio Dominator can be driven to greater than 20dB of total limiting without inducing more distortion. The unwanted effects of pumping, hole punching, dullness, and other typical limiter side effects are much less audible with the Studio Dominator than for other limiters. This makes it very feasible to use heavy limiting to achieve high density where previously other limiters would produce too many annoying side effects.

8.5 Release Time

Adjust the release time setting for the desired limiter characteristics. The desirable release time is usually a function relative to the amount or depth of limiting. For a large amount of total limiting, fast release time will give greater loudness and more density while a slow release time will give less density and less average loudness but more natural sound.

For shallow limiting where only infrequent audio peaks are driven into limiting (such as in "protection" limiting applications), the fastest release setting will provide nearly undetectable limiter action.

The release setting may be readjusted at any time without interacting with other controls, and can be readjusted to compliment any combinations of drive and equalization which may be created by the user.

NOTE: After the output level, drive, and release time are chosen, adjusting the Studio Dominator is essentially complete. Equalization and use of TEC are optional but the user should experiment with them to determine if the effects are desirable for the particular application.

8.6 TEC

Experiment with the TEC by switching it "on" and "off". Allow about a half-second for it to come fully on. The effect of TEC disappears instantly when it is switched off. You should notice a subtle increase in articulation of the sound. This takes the form of greater consonant detail in voices, and greater "edge" or "punch" to precussive and transient sounds in music. You may also detect an increase in the loudness of "inner" sounds of music such as low level voices and harmonies.

8.7 Equalization

Finally, you can experiment with the equalization controls. Remember that the EQ is still active when the process is switched "off", so there is no "in/out" comparison available for EQ. Because the EQ controls adjust the input level to the low and high band limiters, the effect of any equalization "boost" may be nulled by the gain reduction of the limiter bands depending on the amount of limiting. The heavier the limiting, the less EQ boost you can achieve. Therefore, it is often best to set up a desired EQ without limiting (process "off") and then activate limiting and adjust drive. Equalization is provided with the Studio Dominator mainly for effects, and there are no hard rules for its use. Anything that works for you is allowed.

9.0 TECHNICAL DESCRIPTION

Since the Studio Dominator is basically a symmetrical stereo system, many circuits are identical in the left and right channels. Therefore, component designations used in the descriptions of these circuits will be taken from the left channel only for simplicity.

9.1 INPUT CIRCUIT

A unique precision instrumentation topology is incorporated which employs a common mode signal cancelling servo. Due to common mode cancelling by the servo at the input (prior to signal amplification), common mode signal distortion mechanisms which exist in conventional balanced input circuits are eliminated.

The instrumentation amplifier operates on the same principles as the well known classical circuit, except where the common mode servo is concerned. Therefore, only the servo will be described.

The outputs of buffers U101A and U101B are normally equal and out of phase for differential mode input signals. R108 and R109 sum these signals into inverting gain stage U102B. Any output signal from U102B represents amplified common mode information out of phase with the common mode signal at the buffer inputs. The signal from U102B is summed passively with the input signals through R102 and R103 to cancel the common mode signals at the input buffers. The operation of this circuit may alternately be described as a servo stage which monitors the difference between zero volts and the detected common mode signal, then outputs a correction signal to null the common mode signal at the input to the instrumentation amplifier.

9.2 CROSSOVER

Three frequency bands are generated by the crossover circuit composed of U508, U509, U510, and associated components. The high and low band generators are essentially passive first order highpass and lowpass filters buffered by output voltage followers U510, and U508, respectively. The mid band generator, U509, is a differencing amplifier which subtracts the low and high band information from the total input signal. In this manner, three phase-matched frequency bands are produced which can be directly added to perfectly reproduce the input signal.

Tuning the crossover frequencies is accomplished by FET switching of timing components. Low crossover tuning is accomplished by switching C112 in parallel with C111 while high crossover tuning is accomplished by switching R123 across R122. The JFET switches are controlled by d.c. signals from the front panel selector switches.

9.3 VCA LIMITERS

Each crossover filter outputs its signal to an individual VCA limiter stage. There are, then, three VCA limiter stages per channel. Since all VCA limiters are identical, only the left channel low band limiter will be described in full detail.

The heart of the VCA limiter is U303, a type 1537A voltage controlled attenuator device. U301A, B and U302A, B compose the input and output support circuits for U303. Input audio is present on TP301 while output audio is present on TP302. Pin 8 of U303 is the gain control port and is fed from three sources through a passive mixer/attenuator using R318 as the shunt element. Zero volts on this terminal allows maximum gain of the VCA circuit (10dB). Negative voltage on pin 8 causes signal attenuation.

One source of gain control voltage comes via the "Drive Bus". R320 couples this control voltage to the VCA. The drive bus sets the basic operating gain of all the VCA's by this means.

The low and high band limiters receive an "Equalization" control voltage from the front panel "EQ" controls. The mid band limiter receives no such signal. The result is that the basic operating gain of the low and high band limiters can be adjusted for "boost" and "cut" relative to the mid band.

The final source of gain control is the limiter sidechain composed of U304 and associated components. A control voltage is generated by this circuit which attenuates the VCA when peaks are detected above the limiter threshold in the following manner:

A sample of VCA output audio is coupled to the (-) input of comparator U304A. A reference voltage is coupled to the (+) comparator input. The source of this reference voltage is the ALT bus which supplies a base voltage of approximately 2.0VDC. This value changes to a lower value when the Automatic Limit Threshold circuit is operating as will be later described.

The output of U304A remains positive, clamped to about +0.6 volts by D301, when the sampled VCA audio is low in amplitude. When the positive audio peak exceeds the ALT voltage, the U304A output swings to negative 13 volts. The negative pulse from U304A passes through blocking diode D302 to charge C304 via R323 which determines the charging rate. Buffer U304B couples this negative voltage to the VCA gain control port via R321, thus attenuating the audio signal until the output amplitude falls just below the ALT value. An extremely high compression ratio limiter is thus produced.

Restoration of the gain control voltage to zero is accomplished by the discharge path provided by Q301. The rate of discharge is determined by the "Release Time" control which programs Q301 as a current sink.

9.4 SUMMING FINAL LIMITER

The three VCA limiters output their signals to the summing final limiter composed of U511 with associated components. The totally processed audio signal comes out of this circuit and is sent to the "Peak Ceiling" circuit for final output amplitude programming.

The summing final limiter uses a pair of light emitting diodes as soft clippers to catch transient peaks and limit them to approximately 1.9 volts amplitude. This defines the maximum peak ceiling of the output signal at this point. ALT action (later described) maintains consistent peak levels just below the clipping point and anticipates the amount of clipping to maintain inaudible peak protection.

A JFET switch comprising Q103, R134, and D106, removes the soft clipper LED's from the summing amplifier in "process off" mode to eliminate clipping.

9.5 PROCESS ("PROC") SAMPLE RECTIFIER

Outputs from the three VCA limiters are summed into a full wave rectifier composed of U103A/B and associated components. D104 serves to cause the output of U103B to contain a slight d.c. offset equal to one diode drop. This biases the "or" summing diode D105 to the verge of conduction under zero signal conditions to improve the dynamic range of the proc sample bus. The proc sample signal is utilized by both the ALT and TEC circuits.

9.6 OUTPUT CEILING CIRCUIT

Audio from the summing final limiter is coupled to the wiper of S506, the "Output Ceiling" selector. Resistors R601 through R612 are selected by S506 to set the stage gain in 1 decibel increments.

Feedback gain is switched by CMOS switch U507 which is controlled by the "Range" switch. A change of 12 decibels is provided by this gain switch to appropriately rescale the range of the output ceiling circuit.

VR6I1 provides a $\pm$ 2dB gain trim for final output calibration. Output from this stage is a balanced differential signal which is subsequently coupled to the line amplifier stage.

9.7 OUTPUT LINE AMPLIFIER

Twin inverting driver stages are used to drive the output jack with low source impedance and high output current capability. Switch S102 selects the output configuration to balanced or unbalanced mode. In balanced mode, the gain of each driver is unity and pins 2 and 3 of the output connector are driven in phase with the input connector signal. In the unbalanced mode, S102 shorts pin 2 to ground, disconnects the pin 2 driver stage, and doubles the gain of the pin 3 driver stage to restore correct output voltage swing.

U104A/B operates as a high speed current boosted amplifier capable of driving low impedance loads. Direct coupling is used to eliminate any audio baseline shifts which can disturb the accuracy of the peak level.

9.8 AUTOMATIC LIMIT THRESHOLD (ALT) CIRCUIT (Pat.Pending)

The proc sample signal is sampled and compared to a reference voltage by comparator U503. When peaks on the proc sample bus exceed the reference value, then U503 sinks current into pin 7 from the TEC module which contains the remainder of the ALT circuit. Through a proprietary process (pat. pending) the ALT signal is thus produced which is buffered by U501 and thus coupled to the ALT bus.

9.9 TRANSIENT ENHANCEMENT CIRCUIT (TEC) (Pat.Pending)

U502A acts to compare the instantaneous value of the proc sample signal to a historical average. Pulses thusly generated are coupled to the TEC module which contains computing elements to create a control signal. U501 buffers the TEC control signal and couples the signal to the Drive Bus driver circuit.

9.10 METERING CIRCUIT

U502B is a summing stage which receives input from the display bus. The display bus receives "ored" signals representing the AGC voltage from each VCA limiter of both channels. The "oring" is possible because of the blocking diodes in each VCA limiter. R535 provides slight forward bias on the "oring" diodes to linearize the function. Current through R534 cancels this bias offset at the display driver output. D507 and D506 cause U502 to act as a positive polarity amplifier to eliminate any slight negative output voltage as may otherwise occur due to temperature drifts.

U506 is a conventional 10-segment L.E.D. driver to provide the "Total Limiting" panel display.

10.0 TEST AND ALIGNMENT PROCEDURES

The Studio Dominator is an interactive device, with many of the circuits depending on and working with others. Therefore, the following test procedure should be used in its entirety only, as a spot test of a single function may not provide valid information.

10.1 MAIN BOARD

10.1.1 Power-Up

a. Apply power and immediately check for overheating of the output transistors. Also look for other possible signs of power rail overloads.
b. Check all LED's and mechanical operation of the pushbuttons.

10.1.2 Beginning Settings

a. DRIVE pot full CCW.
b. RELEASE TIME full CW.
c. Equalizers CENTER.
d. All pushbuttons GREEN.
e. TEC off.
f. OUTPUT CEILING at +6.
g. INPUT LEVEL SELECT (internal) set at -10

10.1.3. Input Stage Test

a. Feed -10dBm 1kHz into both channels.
b. Measure the level at H504 jumpers 1-2 for left channel and 10-11 for right channel. The level should be 0dBm ± 1/2 dB. Left and right should be equal within ± 0.2dB.
c. Switch the input level switches to +4 position. The level at the jumpers should drop 13dB and still match within ± 0.2dB.

10.1.4 VCA Balancing

With NO AUDIO INPUT, balance all six sets of limiter trims. Perform the following procedure for each limiter band:

a. Check test points 301(401) to insure no gross DC offset exists. Less than 50 millivolts is OK.
b. Monitor test point 302(402).
c. Set DRIVE full CCW.
d. Measure the DC value on TP302(402).
e. Set DRIVE full CW.
f. Adjust VR301(401) to obtain the same voltage on TP302(402) as in step d.
g. Quickly repeat steps c thru f to obtain minimum shift as the DRIVE pot is turned through the full sweep.
h. Now adjust VR302(402) to obtain zero volts on TP302(402). The setting of the DRIVE pot should not matter.
i. Run the DRIVE pot through a full sweep and verify minimum shift from zero volts is obtained.

When all six limiters are shift and offset nulled, proceed to the next step.

10.1.5 Gain Verification

a. Feed the generator into the LEFT CHANNEL at -10dBm, 1 KHz. Check that input sensitivity is set for -10 on both channels.
b. Set PROCESS to ON.
c. Raise DRIVE to full CW.
d. Check that two to four total limiting LED's are on.
e. Repeat the procedure for RIGHT channel input.

10.1.6 Limiter Test

a. Raise the generator output to 0dBm, RIGHT channel feed only.
b. Observe that all the limiting LED's are on.
c. With the RELEASE TIME pot full CW, reduce the DRIVE control to verify nominal releasing of gain reduction.
d. Measure the output level with 6 to 10 dB of limiting and the OUTPUT CEILING at +6. It should be very close to 0dBm. Trim the output trimmer for exactly 0dBm output.
e. Repeat the procedure for a LEFT CHANNEL feed.

10.1.7 ALT Test

a. Feed left channel 0dBm at 1Khz and obtain about 6dB of limiting.
b. Set output level to -3dBm using the OUTPUT CEILING selector.
c. Change input frequency to 100Hz.
d. Output level should rise about 2.8dB.
e. Repeat the procedure for the right channel.

10.1.8 Output Stage Test

a. Feed signals into both inputs at 1KHz. Set PROCESS to "off".
b. Measure the output level from the left channel with a balanced floating voltmeter.
   The DRIVE control can be adjusted to get an easy reading.
c. Alternate the BAL/UNBAL switch and verify the output level remains constant.
d. Repeat the procedure for the right channel.

10.1.9 Equalizer Test

a. Feed a square wave signal at 1KHz into both channels. Switch PROCESS "off".
b. Look at waveforms on H504 at the jumper locations 3-4 for left, and 8-9 for right channel.
   By adjusting the equalizer controls, a good square wave should be obtainable (a slight difference between equalizer settings for both channels can be expected)
   EQ controls should be very near center (12 O'Clock).

10.2 POWER/METERING BOARD

10.2.1 Power-Up

a. Apply power and check power supply regulator outputs for nominal voltages.
b. Check all red and green LED's and mechanical operation of the pushbuttons.

10.2.2 Beginning Settings

a. DRIVE pot full CCW.
b. RELEASE TIME full CW.
c. Equalizers CENTER (12 O'Clock)
d. All pushbuttons GREEN.
e. TEC off.
f. OUTPUT at +6.

NOTE: ALL TESTS ARE CONDUCTED WITH AN INPUT TO THE LEFT CHANNEL ONLY EXCEPT STEP 10.2.6.

10.2.3 ALT Test

a. Set the generator for 1KHz at 0dBm but turn off the generator output. Switch PROCESS to ON.
b. Measure the ALT voltage on pin 7 of U501. It should be about 1.8 to 2.0 volts DC.
c. Switch on the generator and raise the DRIVE from minimum while watching the ALT voltage. The ALT voltage should drop to about 1.4 volts before full drive is reached.

10.2.4 Metering Test

a. Set the generator to -10dBm at 1KHz. Adjust the drive control until the ALT voltage just starts to drop.
b. The first limiting LED should just be coming on.
c. Switch the generator output to 0dBm. All 10 limiting LED's should go on.

d. Now reduce the generator output in 1dB steps and observe the LED's go off one at a time.

10.2.5 TEC Test

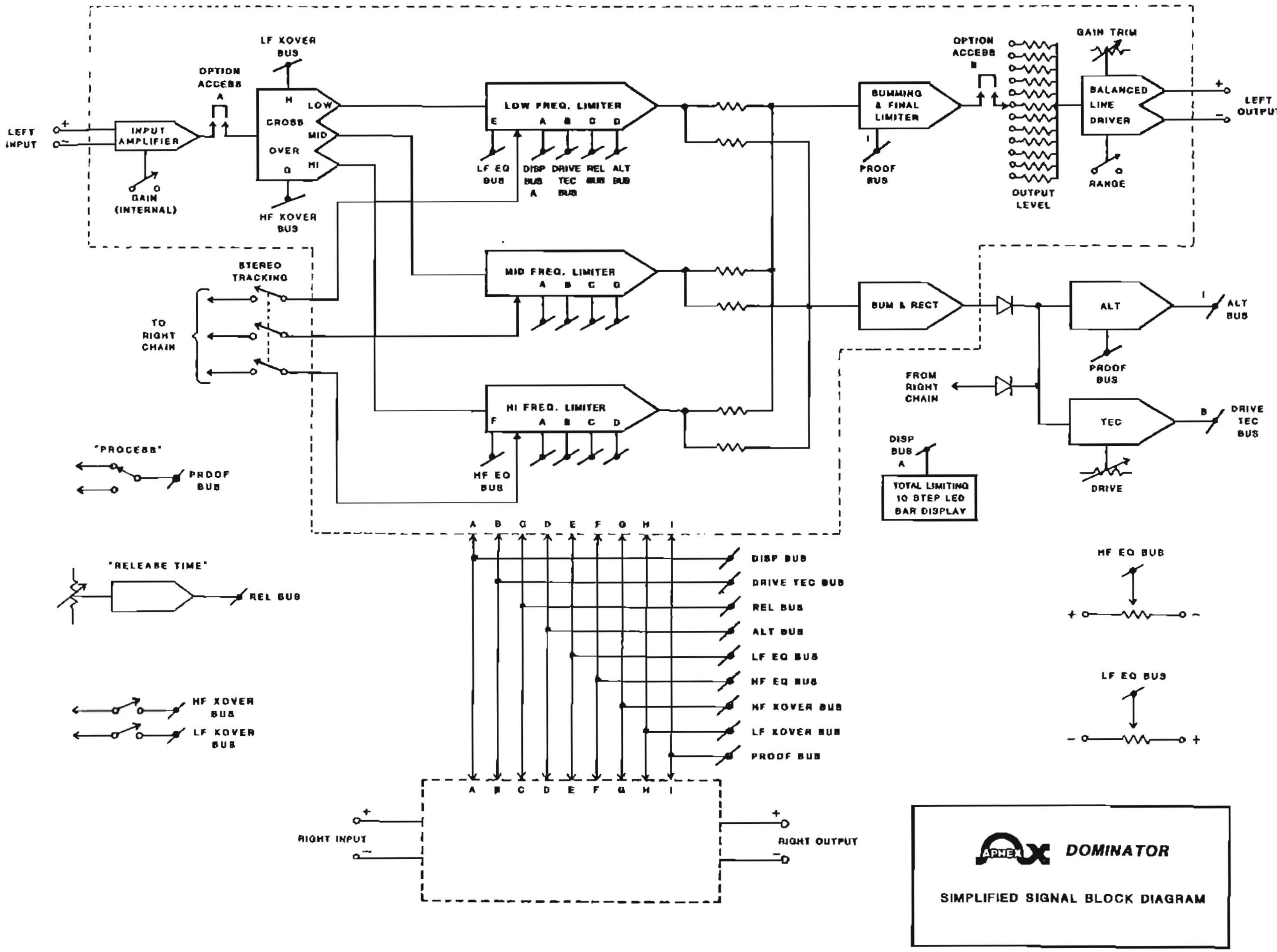
a. Set generator to 0dBm at 5KHz.
b. Set PROCESS to OFF.
c. Raise DRIVE to obtain about 0dBm output.
d. Turn TEC "on".
e. Observe the audio output on a triggered scope

f. Turn the generator off and on several times and observe the triggered waveform. There should be a high amplitude beginning portion followed by a ramped decreasing portion followed by a long steady lower amplitude portion.

10.2.6 Output Stage Test

a. Turn TEC "off", and PROCESS "on".
b. Raise DRIVE until at least two limit LED's come on. Measure the output level and adjust VR801 for exactly 0dBm out. (The OUTPUT CEILING selector should be at +6dBu.)
c. Repeat the procedure for a RIGHT channel input.

11.0 BLOCK DIAGRAM



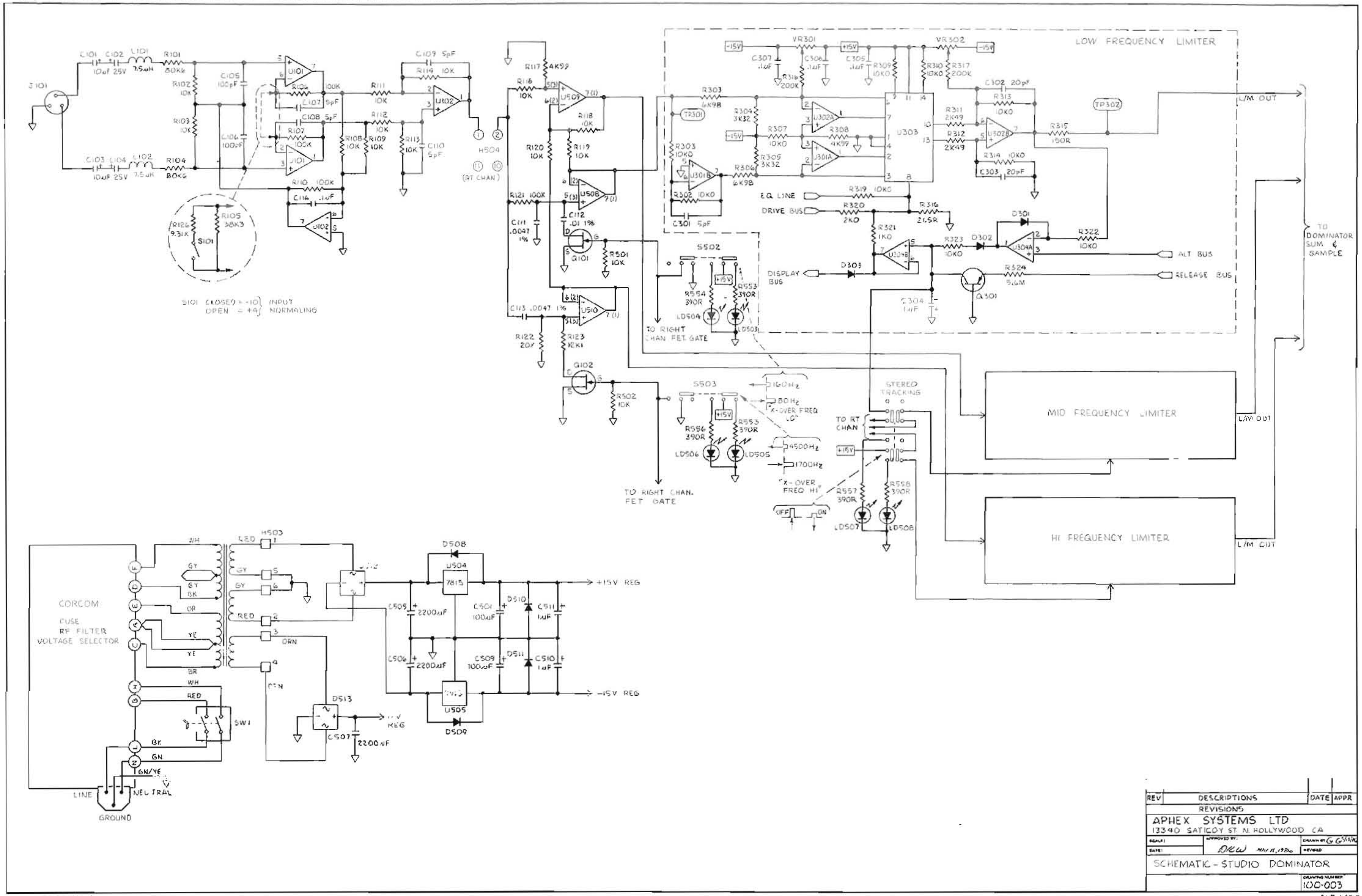
SIMPLIFIED SIGNAL BLOCK DIAGRAM

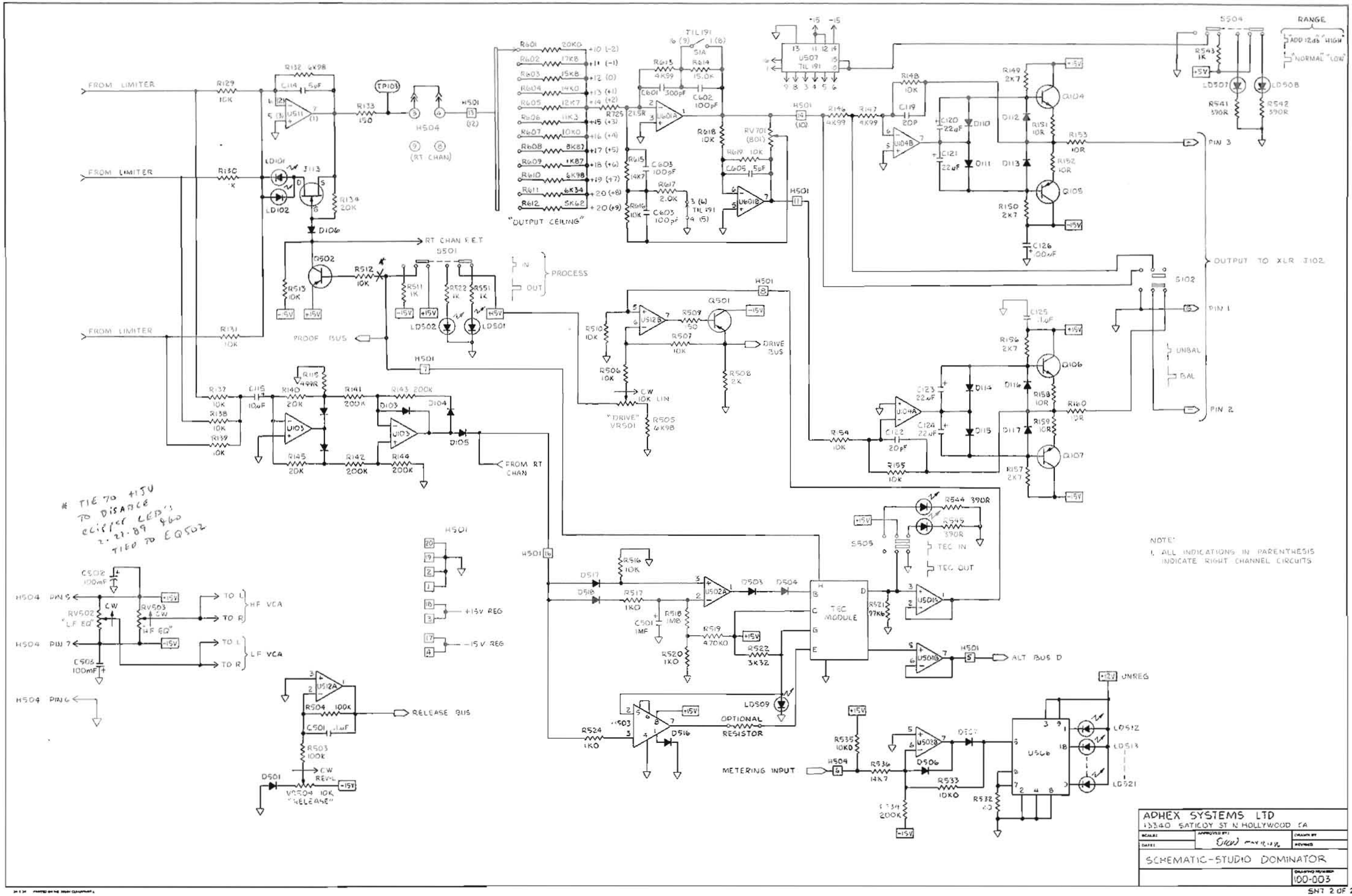
12.0 SCHEMATICS AND PARTS LISTS

12.1 Input / Bandsplit / Limiters and Power Supply

12.2 Summing / Peak Ceiling / Output Tec and Alt

12.1 INPUT / BANDSPLIT / LIMITERS AND POWER SUPPLY




FROM LIMITER
FROM LIMITER
FROM LIMITER
"OUTPUT CEILING"
RT CHAN F.E.T.
PROOF BUS
DRIVE BUS
"DRIVE" VR501
PROCESS
IN
OUT
FROM RT CHAN
RANGE
"ADD 12dB" HIGH
"NORMAL" LOW
PIN 3
PIN 1
PIN 2
OUTPUT TO XLR J102
UNBAL
BAL
TEC IN
TEC OUT
TEC MODULE
ALT BUS D
RELEASE BUS
"RELEASE"
METERING INPUT
OPTIONAL RESISTOR
HF VCA
LF VCA
"LF EQ"
"HF EQ"
+15V REG
-15V REG
12V UNREG
NOTE:
1. ALL INDICATIONS IN PARENTHESIS INDICATE RIGHT CHANNEL CIRCUITS
* TIE TO +15V TO DISABLE CLIPPER LED'S 2-21-89 #60 TIED TO EQ502
APHEX SYSTEMS LTD
13340 SATICOY ST N HOLLYWOOD CA
SCHEMATIC-STUDIO DOMINATOR
100-003
SHT 2 OF 2

13.0 EXPLODED VIEW AND MECHANICAL PARTS LISTS

14.0 SPECIFICATIONS

FREQUENCY RESPONSE

Below Threshold: 20Hz to 50kHz ± 0.2dB

TOTAL HARMONIC DISTORTION (THD)

Below Threshold: typically 0.01% or less 20Hz to 20kHz

Above Threshold (15dB limiting), typically:

20Hz - 0.1%
1kHz - 0.04%
15kHz- 0.17%

NOISE (Unweighted, 20Hz to 30kHz): 80dB below peak ceiling level

INPUT:

Type- RF Filtered true instrumentation differential balanced

Impedance- 160kOhms

Maximum Input Level- 20dB above normalized setting of -10 or +4

CMRR- Greater than 40dB

OUTPUT:

Type- Electronically balanced transformerless. Switchable to operate balanced or single-ended at full output.

Impedance- 20 Ohms

Maximum Output Level- Selectable in 1dB steps from -3 to +20dBu

CROSSOVER:

Type- Phase coherent, 6dB/Octave

Bands- 3

Crossover Frequencies- User selectable - 80 or 160Hz (lo-mid) and 1700 or 4500Hz (mid-hi)

	Bandlimiters	ALT	Final Limiters
ATTACK TIME:	5mSec	Program Dep.	Instantaneous
RELEASE TIME:	User Set 0.1-100S	"	"

GAIN TRACKING: ± 0.1dB (TRACKING ON)

COMPRESSION RATIO:

Infinite above threshold with zero overshoot

POWER REQUIREMENTS:

100, 120, 220 or 240 VAC ± 10% @ 20 Watts

SIZE:

1 3/4"H X 19"W X 9"D (excluding controls)

WEIGHT:

Net: 8 lbs.

Shipping: 10 lbs.

15.0 WARRANTY AND SERVICE INFORMATION

Aphex Systems, Ltd. warrants parts and labor for the Studio Dominator for a period of one year from the date of purchase. If it becomes neccessary to return a unit for repair, repack it in the original carton and packing material, if possible. If a warranty repair, enclose a copy of proof of purchase.

Send to:

APHEX SYSTEMS, LTD.
13340 SATICOY ST.
NORTH HOLLYWOOD, CA. 91605

16.0 APPLICATION NOTES