dbx 160X

PROFESSIONAL SINGLE-CHANNEL COMPRESSOR/LIMITER

INSTRUCTION MANUAL

1.0 INTRODUCTION

1.1 PRODUCT DESCRIPTION

The dbx Model 160X is a professional single channel compressor/limiter that features an exclusive *combination of the dbx Over Easy and "hard knee"* compression characteristics. dbx Over Easy compression permits extremely smooth, almost inaudible transitions into compression due to the gradual change of compression ratio around the threshold. The 160X offers the user the choice of using the Over Easy curve or a mathematically precise "hard" threshold—at any compression ratio selected. The Over Easy curve, coupled with dbx's true-RMS level detector, wide-range Blackmer voltage-controlled amplifier (VCA) and feed-forward circuitry, makes it *possible to achieve large amounts of compression without adverse audible side effects.* These features also allow the 160X to actually be set for *INFINITY+™ compression.* This is a special negative compression effect whereby the program dynamics are inverted above the set threshold (i.e., the higher the input level, the lower the output level).

The detectors of *two Model 160X's can be coupled so that stereo programs are compressed with stable imaging,* by means of a simple patch cable between the two units. *The stereo coupling can be disengaged instantly* by a front panel pushbutton.

Program-dependent attack and release times assure natural sound without the need for continuous manual adjustments. *Threshold and compression ratios are adjustable over a wide range,* so virtually any line level signal can be processed effectively. *Output gain is also adjustable* over a very wide range (±20 dB). These features make the Model 160X compatible with the operating levels of virtually all professional sound and creative audio equipment.

For metering, the 160X provides a true RMS dual wide-range LED array which simultaneously shows *the amount of gain reduction* up to 40 dB* *and the input or output level* from −40 dB to +20 dB, depending on the setting of a front panel switch.

The 160X also features a true *"hard-wired by-pass"* switch, separate detector inputs, Tip/Ring/Sleeve phone jacks as well as barrier strip connectors, balanced active inputs and +24 dBm single-ended output drive capability which can be field-modified to provide transformer or active balanced operation.

The dbx Model 160X is well suited to a broad range of applications including tape recording, disc mastering, radio and TV production and broadcast, live concert sound reinforcement, mobile and theatrical production.

IMPORTANT FOR UK USERS

The wires in this mains lead are colored in accordance with the following code:

Blue: Neutral
Brown: Live

As the colors of the wires in the mains lead of this apparatus may not correspond with the colored markings identifying the terminals in your plug, proceed as follows:

The wire that is colored blue must be connected to the terminal that is marked with the letter N or colored black;

The wire that is colored brown must be connected to the terminal that is marked with the letter L or colored red.

Ensure that all terminals are securely tightened and that there are no loose strands of wire.

WARNING

This unit must be protected by a 3-amp fuse, preferably using a fused plug. Also, do not remove the cover without first disconnecting the unit from the mains supply.

**The 160X is capable of greater than 60 dB of compression.*

1.2 160X FRONT PANEL

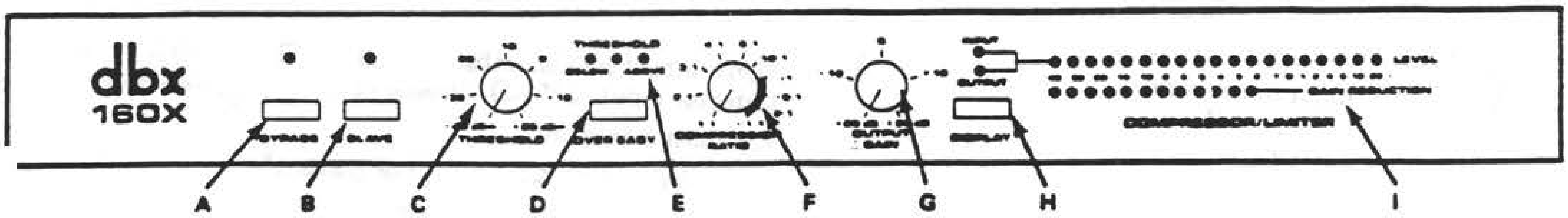

Fig. 1-1 – Model 160X Front Panel

A. BYPASS SWITCH AND INDICATOR

Depressing the BYPASS button creates a "hard-wire bypass" of the 160X's circuitry by connecting the input directly to the output; the LED above the switch turns ON in Bypass mode.

B. SLAVE SWITCH AND INDICATOR

Depressing the SLAVE button on one 160X of a stereo pair determines that the *other* 160X will be the Master (the controlling unit). The LED above the SLAVE button turns ON when the 160X is in SLAVE mode, indicating that the front panel functions (with the exception of BYPASS and DISPLAY select switches) are deactivated and under control of the Master 160X. When neither 160X is in SLAVE mode, each will operate normally as a mono compressor/limiter.

C. THRESHOLD CONTROL

Adjusting this knob sets the threshold of compression from -40 dBm (7.8 mV) to 20 dBm (7.8 V). In hard-knee compression mode, the threshold is defined as that point above which the output level no longer changes on a 1:1 basis with changes in the input level. In Over Easy mode, the threshold of compression is defined as the approximate middle of the Over Easy threshold region, as shown in Figure 3-1.

D. OVER EASY SWITCH

Depressing this button changes the threshold mode to dbx's Over Easy characteristic, and allows the middle (amber) THRESHOLD LED to turn ON when the signal is in the Over Easy region. When this button is out, the 160X operates as a hard-knee compressor/limiter.

E. THRESHOLD INDICATORS

These three LEDs indicate the relationship of the input signal level to the threshold of compression. The green "BELOW" LED is ON when the signal is below threshold and the red "ABOVE" LED is ON when the signal is above threshold. When the 160X is switched to Over Easy mode, the amber LED is ON when the signal is in the Over Easy region (Refer to Figure 3-1).

F. COMPRESSION RATIO CONTROL

Rotating this control in a clockwise direction increases the maximum amount of compression from 1:1 (no compression) up to infinity:1 (no increase in RMS output regardless of input level increases above threshold); further clockwise rotation increases compression into the INFINITY+ region, up to a maximum of -1:1 (i.e., a 1 dB *decrease* in input level causes a 1 dB *increase* in output level). In the INFINITY+ region, the 160X inverts the program dynamics for special effects.

G. OUTPUT GAIN CONTROL

Adjusting this control varies the amount of fixed gain (up to ±20 dB) in the 160X's output amplifier stage. The OUTPUT GAIN control *does not* interact with the threshold of compression.

H. DISPLAY FUNCTION SWITCH AND INDICATORS

Depressing this button causes the upper LED array to display the INPUT level to the 160X. With the button out, the OUTPUT signal level is displayed. A pair of LEDs immediately above the DISPLAY switch indicates the selected display status.

I. LEVEL AND GAIN REDUCTION DISPLAYS

The upper row of 19 LEDs displays either the INPUT or OUTPUT level, as selected with the DISPLAY switch. This display is factory set so that 0 dB is equivalent to +4 dBm input or output level (1.23 V rms), but may be reset with the rear panel METER CALIBRATION trimmer. (See Section 3.5 for recalibration instructions.)

The lower row of 12 LEDs displays up to 40 dB of GAIN REDUCTION being caused by the 160X. (Fixed gain changes commanded by the OUTPUT GAIN control are not displayed by the gain change LEDs but are reflected in the output level display.)

1.4 SPECIFICATIONS

Input Impedance	Signal input: 50 kΩ, unbalanced; 100 kΩ, balanced. Detector input: 230 kΩ, unbalanced; 460 kΩ, balanced.
Input Level	+24 dBm maximum.
Output Impedance	22 Ω, designed to drive 600 Ω or greater.*
Output Level	+24 dBm into 600 Ω or greater.
Threshold Range	Variable from –40 to +20 dBm (7.8 mV to 7.8 V RMS).
Compression Ratio	Over Easy: Program-dependent, affected by THRESHOLD, COMPRESSION RATIO settings (COMPRESSION RATIO control determines maximum compression ratio), continuously variable from 1:1 to ∞:1 to –1:1. Hard-knee: COMPRESSION RATIO setting defines exact compression ratio, continuously variable from 1:1 to ∞:1 to –1:1.
Maximum Compression	>60 dB
Threshold Characteristic	Over Easy or hard-knee (switch selectable).
Attack Time (1)	Program-dependent: 15 ms for 10 dB increase in input level (above threshold), 5 ms for 20 dB, 3 ms for 30 dB.
Release Time	Program-dependent; varies automatically from 0-500 ms, affected by settings of front panel controls.
Output Gain	Variable from –20 to +20 dB.
Slew Rate	>10 V/us
Dynamic Range (2)	>113 dB
Equivalent Input Noise (unweighted)	<–89 dBm, 20 Hz – 20 kHz.
Frequency Response	+0, –1dB, 20 Hz – 20 kHz.
Distortion Below Threshold (3)	2nd harmonic 0.07%; 3rd harmonic 0.07%.
Distortion Above Threshold (4)	2nd harmonic 0.07%; 3rd harmonic 0.02%.
Metering	19 LED INPUT or OUTPUT display from –40 to +20 dB, 12 LED GAIN REDUCTION display from –1 to –40 dB.
Meter Zero Set	–15 dBm to +10 dBm.
Indicators	BELOW/threshold/ABOVE (green, yellow, red), INPUT (red), OUTPUT (red), SLAVE (yellow), BYPASS (red).
Controls and Switches	THRESHOLD, COMPRESSION RATIO, OUTPUT GAIN, DISPLAY function switch, meter zero adjust, BYPASS switch, SLAVE switch, OVER EASY switch.
Connectors	Input/output: TRS phone jacks and barrier terminal. Detector: barrier terminal Strapping: TRS phone jack
Dimensions	1-3/4" H x 19" W x 9-1/4" D (4.4 cm x 48.3 cm x 18.4 cm).
Weight	6.5 lbs (3.0 kg)
Power Requirements	115/220 VAC ±10%, 50-60 Hz, 8 W
Accessories	AB-1 active balanced output card.

(1) Measured in the infinite compression region of the threshold curve, time required to reduce signal by 63% of level increase (above threshold).

(2) Defined as the difference between the maximum signal level and the "A" weighted noise floor as measured at the output.

(3) Measured at 1 kHz, 0 dBm input and output.

(4) Figures are typical at infinite compression, 1 kHz, 0 dBm input and output – 2nd harmonic is relatively unaffected by compression ratio, time constants and frequency, while 3rd harmonic decreases with slower time constants, higher frequencies and lower compression ratios.

**Transformer for isolated floating output available directly from:*
Jensen Transformers, 10735 Burbank Blvd., North Hollywood, CA 91601. (See Section 2.3.4.)

3.0 OPERATION

3.1 OUTPUT CONTROL

This control sets the amount of fixed gain in the 160X's output stage over a range of ±20 dB. Fixed gain added or subtracted by the OUTPUT control from signal passing through the 160X is not affected by the setting of the THRESHOLD control. Gain changes brought about when the input signal exceeds the THRESHOLD reference setting are in *addition* to those caused by the OUTPUT control.

3.2 THRESHOLD CONTROL

In hard-knee mode this control sets a reference level above which input signals will be processed by the 160X's gain change circuitry in the manner defined by the setting of the RATIO control. Input signals which fall below this level will pass through the 160X unprocessed (except for fixed gain changes directed by the OUTPUT control).

In Over Easy mode, signals begin to gradually activate the 160X's gain change circuitry as they *approach* the THRESHOLD reference level and they do not get fully processed in the manner defined by the RATIO control until they have passed somewhat above the THRESHOLD reference level. In Over Easy mode there is no distinct point at which processing begins, and the THRESHOLD setting corresponds to a point on the input/output transfer curve midway between the onset of processing and that point at which the transfer curve corresponds to the setting of the RATIO control (Figure 3-1).

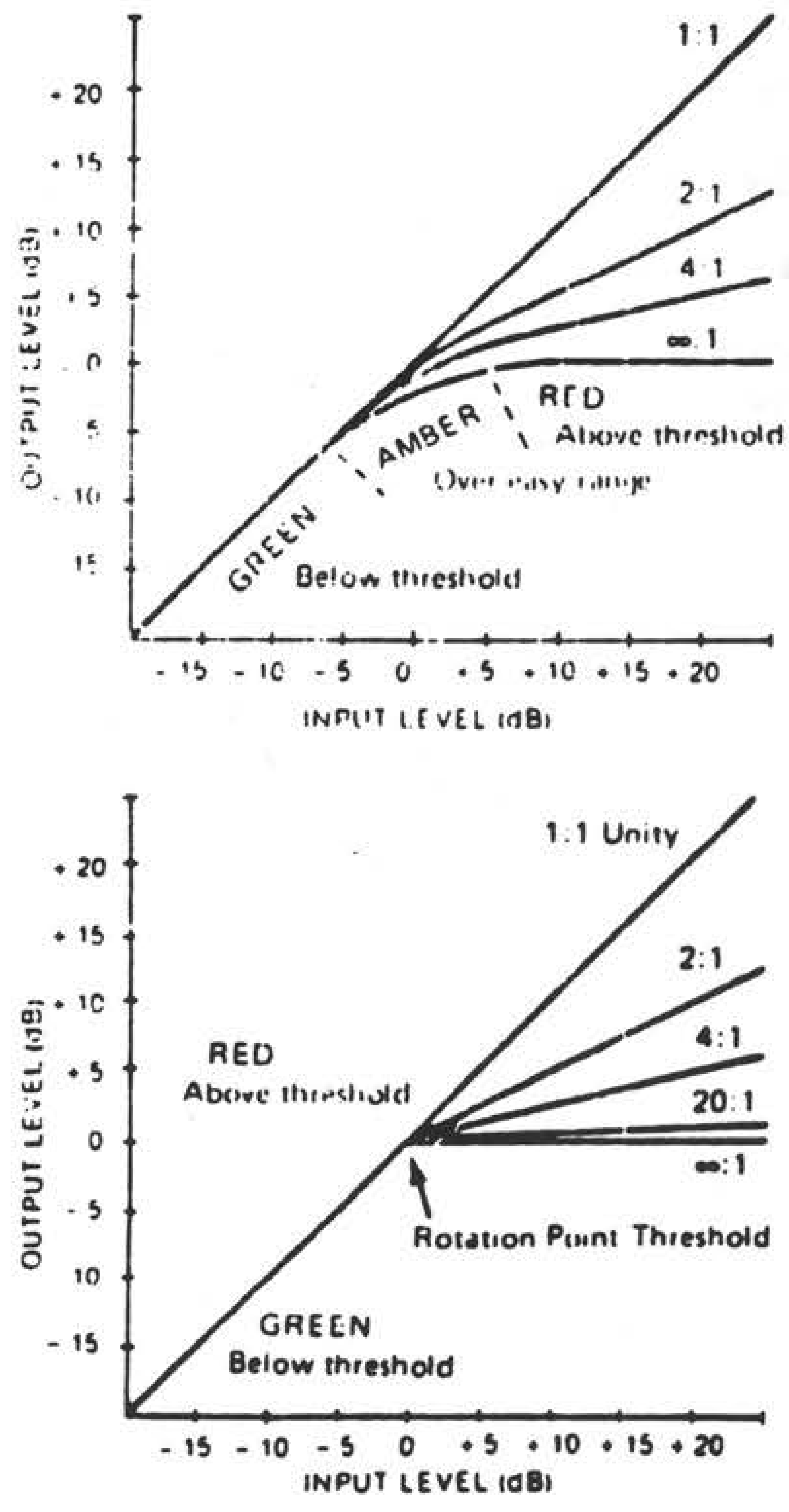

Fig. 3-1 – Over Easy vs. Hard-Knee Compression

NOTE: The THRESHOLD setting relates to the signal level seen by the DETECTOR input. In normal operation, the program signal input is connected directly to the detector input. If this is not the case, the signal actually present at the detector input will determine whether and how the 160X processes the signal coming into its SIGNAL INPUT.

3.3 RATIO CONTROL

When an input signal is above the THRESHOLD reference level, the setting of this control determines the number of dB by which the input signal must change in level to produce a 1 dB increase in the signal level at the output of the 160X. A setting of 2:1 indicates an input:output ratio wherein a 2 dB increase in input signal (above threshold) will produce a 1 dB increase in output signal. A setting of ∞:1 indicates that an infinite increase in input level would be required to raise the output level by 1 dB.

The 160X's RATIO control covers the entire range from 1:1 to ∞:1 and, in addition, goes to Infinity+ (negative) ratios. At a setting of -1:1, the above threshold input signal must *decrease* by 1 dB in level to *increase* the signal at the output of the 160X by 1 dB.

The control curve of the RATIO potentiometer has been designed to provide total operator control, with scale expansion at the subtle lower ratios for easy, repeatable settings.

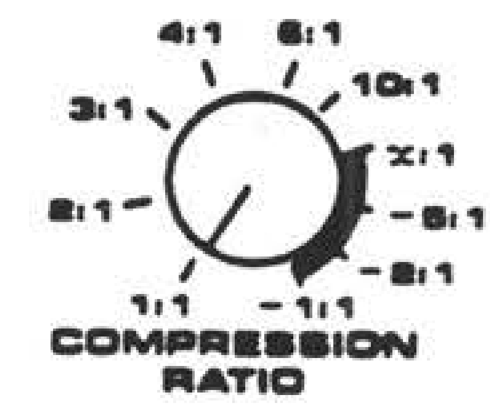

Fig. 3-2B – Ratio Control

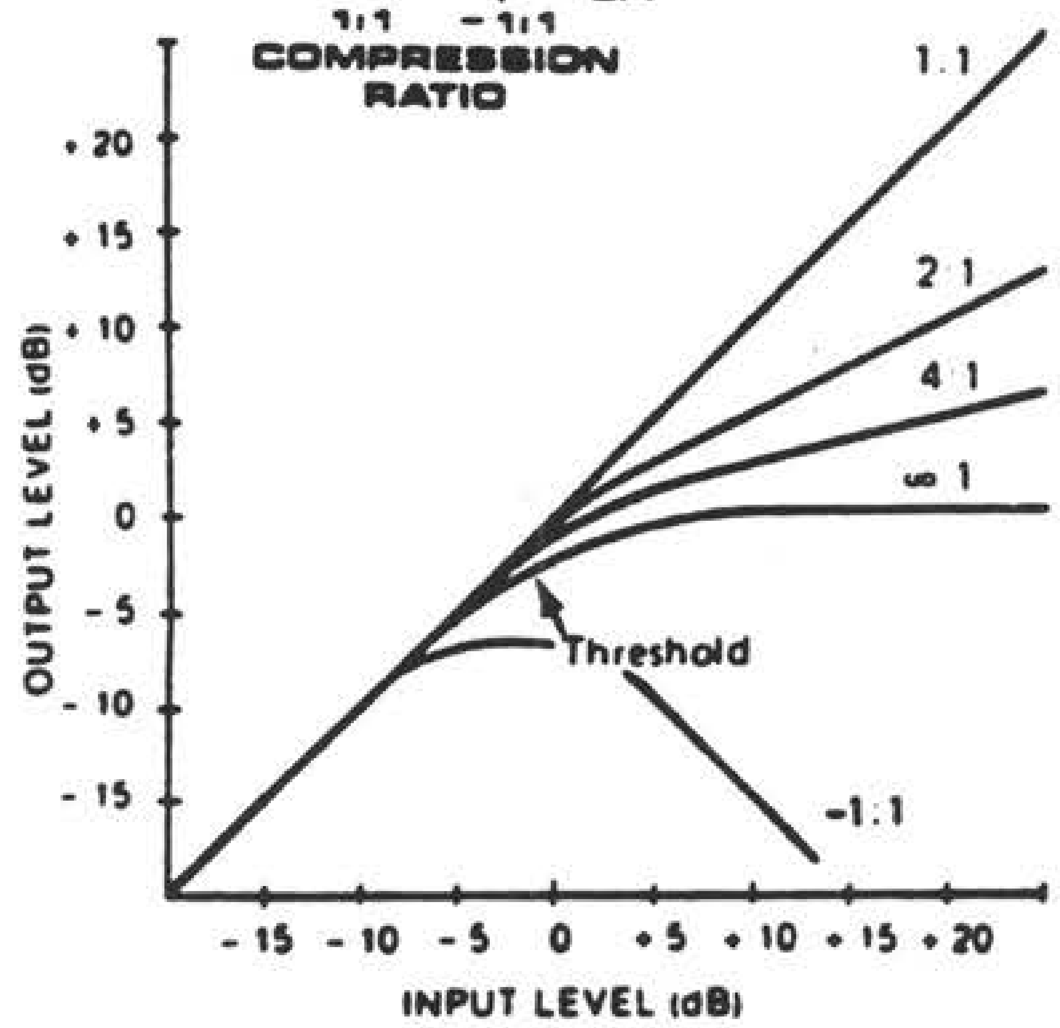

Fig. 3-2A – Typical Transfer Curves

Fig. 3-2 – Ratio

3.4 STEREO STRAPPING

Two channels of program material do not necessarily constitute a stereo program. A stereo program is one where the two channels are recorded and/or mixed to create the illusion of a single, unified "panorama" of sound. The stability of the psychoacoustic "image" of each sound source within the stereo program depends upon its ability to maintain a specific phase and amplitude relationship from left to right channel.

If two independent compressors are used to process the stereo program, a loud sound occurring in one channel will cause a gain reduction only in that channel. This gain reduction would cause the perceived image of any sound spread between the two channels to move toward the side which had not been compressed, because the spread signal would be momentarily softer in the compressed channel. This can be avoided by linking the two compressors in such a way that both channels receive the same amount of compression — an amount equal to the maximum "needed" by either channel at that moment. On the 160X, this is accomplished by means of the STRAPPING jacks; a cable between these jacks permits the RMS detectors of both units to "talk" to one another — but only when one of the units' SLAVE buttons is depressed. Each compressor then senses the incoming signal level, which is compared to the threshold set on the master (the unit whose SLAVE button is *not* depressed). If the compression is required by *either* channel, both channels are subject to compression — the same amount of it — so the stereo image remains stable.

When compressing a stereo program with a pair of 160Xs, only the master unit controls need be adjusted.

3.5 METER CALIBRATION

The INPUT/OUTPUT LEVEL DISPLAY in the 160X is factory-calibrated to indicate "0" when the signal level is +4 dBm (1.23 V) at either the input or output of the 160X, depending on the DISPLAY function switch position. (The METER CALIBRATION control does not affect the GAIN CHANGE LEDs.)

To recalibrate the LEVEL DISPLAY, depress the DISPLAY button to meter the INPUT LEVEL, and feed a 1 kHz signal at the selected nominal operating level (the level desired for a "0 dB" meter indication) to the 160X's SIGNAL INPUT. Then adjust the rear panel METER CALIBRATION control until the meter indicates "0 dB."

4.0 TYPICAL APPLICATIONS

4.1 SMOOTHING VARIATIONS IN VOCAL LEVEL

When the distance between a vocalist and a microphone changes, variations in signal level occur. Start with the 160X adjusted for low compression (around 2:1) and adjust the THRESHOLD control for optimum results, then increase RATIO if necessary.

4.2 SMOOTHING VARIATIONS IN INSTRUMENT LEVEL

To achieve a smoother electric-bass sound, compress the instrument's output with a ratio of about 4:1. Compression lessens the loudness variations among the strings and increases the sustain. Other instruments, such as horns, vary in loudness depending on the note being played, and benefit similarly.

NOTE: When compressing a stereo program with a pair of 160Xs, the factors affecting a compression curve, and the actual compression and threshold settings, are like those previously covered with reference to single channels of program material. However, it will generally be found that large amounts of compression are more audible in a mixed stereo program than they might be on the separate tracks that were mixed to create the program.

4.3 RAISING A SIGNAL OUT OF A MIX

Since reducing dynamic range increases the *average* signal level and meter readings by a small amount, a single track can be raised out of a mix by boosting its level slightly and applying compression. It is also possible to separate certain vocals or instruments from a mono program already mixed by following the procedure in Section 4.6.1.

4.4 PREVENTING TAPE SATURATION

With programs of widely varying levels, compression can prevent recording levels from saturating tape tracks (see also Section 4.7.1).

4.5 SPEAKER PROTECTION

Compressors are frequently used to prevent excessive program levels from damaging drivers in a sound-reinforcement system. Limiting also benefits intelligibility by allowing low-level input signals to be reproduced through the system at higher volume. In a musical performance, this provides additional intimacy as the vocalist's whispers are heard clearly at each seat in the house. The Over Easy curve utilized by the 160X permits a very high amount of compression (10:1 or greater) to be used in many situations. Vocalists and musicians don't get the sense of being "choked back", but high average levels can be maintained without the speaker damage which would normally occur due to excessive heat buildup in the drivers.

In circumstances where the 160X is expected to cause no change in gain unless an emergency arises (wildly excessive levels) some operators set the RATIO to ∞:1, the THRESHOLD to the highest permissible level, and operate the unit in hard-knee mode. (In hard-knee mode, the 160X will perform in the manner of its predecessor, the 160.) As a general rule, the compressors should be *as close to the amplifiers as possible* in the signal chain. If the 160X is placed before the EQ, for example, a potentially damaging boost in EQ won't be seen by the unit and the speakers may be damaged. (See Section 4.6.2.) For maximum sound-pressure levels, large sound-reinforcement systems frequently use a separate compressor on each output of the electronic crossover(s). For a stereo sound-reinforcement system, stereo strapping cables should be run between the 160Xs in each band (low-low, mid-mid, etc).

4.6 INFINITY+™ COMPRESSION

It has been noted that with a full-scale Infinity+ setting of -1.1, a 1 dB *decrease* of input level above THRESHOLD causes a 1 dB *increase* of output level. It can be seen how this differs from the full counter-clockwise setting of 1:1, where the transfer function is linear (e.g., a 1 dB *increase* of input level causes a 1 dB *decrease* of output level).

A total dynamic reversal of the entire program being processed will not occur unless the THRESHOLD is set very low (e.g., -40 dBm) and all the program material falls above the threshold level. More often, some of the program input will be below threshold, and will therefore have normal dynamics — that is, the output level will change linearly with changes of input level. As the input level rises above threshold, however, the output level will begin to decrease. The audible results will differ with different types of program material and different settings. With instruments having fast attack times, such as a picked guitar, a harpsichord, or a drum, a low threshold will all but eliminate the attack transient. Interesting effects can be obtained by setting a moderate threshold with instruments with a slower attack, such as wind instruments or bowed strings; the level will build normally up to the threshold, then begin going lower, the output will become very quiet as the instrument is generating its loudest sound, perhaps quieter than when the note first began.

4.7 USING THE DETECTOR INPUT

4.7.1 Detector EQ For Signal Enchancement

It is possible to separate certain vocals and instruments from a mix by frequency-weighted compression. With an equalizer inserted ahead of the detector input (but not in the audio path), the equalization settings do not shift the timbre or frequency response of the audio signal. They merely alter the threshold response of the compressor on a "frequency-weighted" basis (see Figure 4-1). With this arrangement, raising certain frequencies on the equalizer causes them to be suppressed in the audio signal. A relatively high threshold setting can allow normal sounds to be unaffected while solo and very loud sounds are held back. (Of course, when compression does occur, the level of the entire program is affected.) Depending on the threshold setting, lower-level fundamentals or harmonics will not cause compression, and the program is not subject to the phase shift normally caused by program equalization.

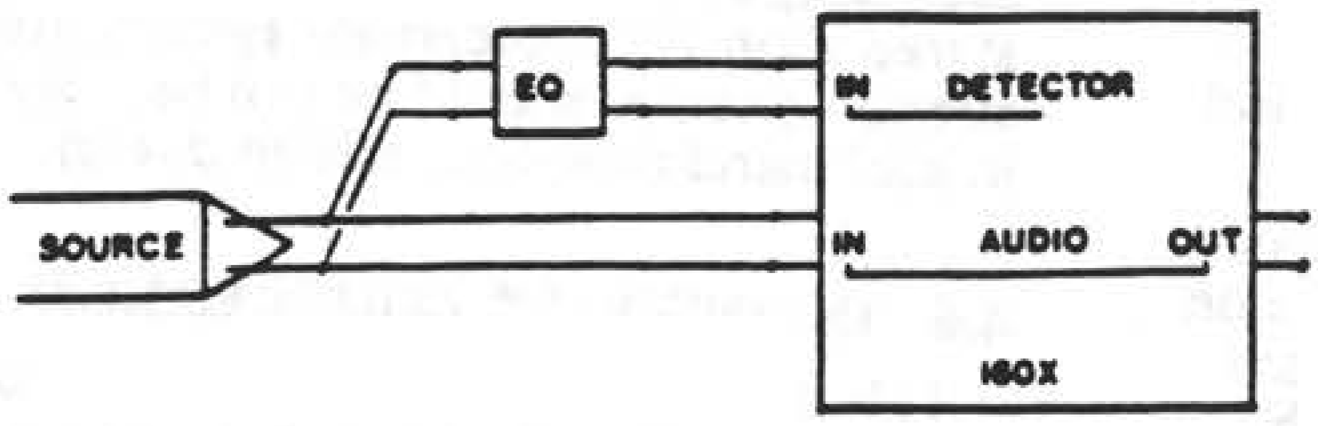

Fig. 4-1 – EQ In Detector Path

The converse of the above EQ technique may also be used; dipping the equalizer bands causes any sound with dominant energy in the affected register to pull the level up because the 160X will detect less need for compression.

During the recording of cymbals and tom-toms, a compressor with an equalizer in the detector path can help prevent tape saturation. The equalizer can be adjusted for boost with a peak of about 5 kHz, causing the cymbal to be compressed on a very loud crash, stopping tape saturation at high frequencies where there is less headroom. However, gentle tapping of a drumstick or brushing of the cymbal will not be held back. Assuming the tom-tom is a lower-frequency instrument and can be better tolerated by the tape, it has less need for compression. The equalization in the detector circuit means that the compressor is not triggered as readily by a loud tom-tom beat as by an equally loud cymbal crash.

In the absence of a more sophisticated de-esser, small amounts of high-frequency boost in the detector path can frequently help in the processing of vocals which have been very brightly equalized or which suffer from a prominent "ess" sound.

4.7.2 Detector EQ For Speaker Protection

If a single compressor is to be used with a multi-way speaker system (i.e.,before the crossover, after EQ), the system operator is faced with the problem of keeping the entire system level down below the point of destruction of the most sensitive component. If, for example, mid-range drivers are frequently damaged, the whole system must be operated at a lower sound pressure level or additional mid-range drivers must be added. By inserting an equalizer in the detector path of the 160X, it can be made *more sensitive* to frequencies in the range handled by the sensitive drivers. The system can then be run at higher average levels and will only be dropped back when damaging signals are present.

4.7.3 Detector EQ For Broadcast

By inserting a pre-emphasis filter network in the detector path of a 160X processing pre-emphasized audio, higher average signal levels can be run within the headroom limitations of the broadcast chain.

4.7.4 Anticipatory Compression

By feeding the program directly to the 160X's detector input and sending the audio signal through a delay line before the audio input, the unit can "anticipate" the need for a gain change; see Figure 4-2. With some experimentation, the effect can be that of "zero" attack time at a given frequency. Additional signal delays beyond this "zero" time will then cause the compressor to finish reducing the gain before the leading edge of the loud passage even enters the signal input. This will suppress the program material preceeding the loud passage. The 160X will then begin to recover from compression (release) before the loud passage has dropped back down toward the set threshold. This will cause the output level to surge higher as the note or passage should be decaying. This special effect obtained with time delay sounds similar to the inversion of dynamic envelopes produced during reverse playback of a tape recording. When coupled with Infinity+ compression, highly unusual effects can be achieved.

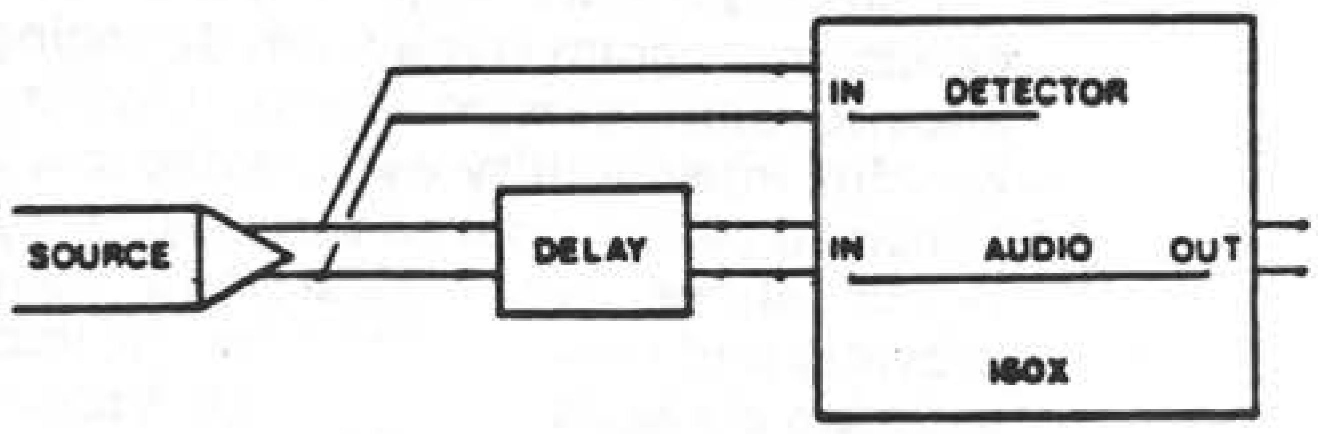

Fig. 4-2 – Delay In Signal Path