



## **ABOUT THE VCE-18 DYNAMIC RANGE ENHANCER**

Le VCE-118 est un outil dynamique unique basé sur les percées qui ont eu lieu lors de l'avènement de la technologie VCA au début des années soixante-dix. Sa réputation d'arme secrète réside dans sa mise en œuvre spéciale du ratio, qui permet à la fois la compression et l'expansion, combinée à son mode linéaire spécial comme carburant puissant pour la dynamique de flexion en tant qu'effet créatif.

Naturellement, le VCE-118 offre la compression percutante ultime connue des premiers produits phares du compresseur VCA comme le célèbre 160. Mais il ne s'arrête pas là, puisque sa gamme dynamique élargie surprend avec des effets intrigants rappelant les gates et les transitoires shapers. Le

mode linéaire peut aller encore plus loin avec des escapades sonores comme des rythmes fortement pompants ou une ambiance silencieuse.

## THE CONTROLS

### Power (Of/On)

Engage ou contourne complètement l'ensemble de la chaîne de traitement.

### Threshold Level (0.-46dB)

En tournant le bouton de seuil dans le sens des aiguilles d'une montre, vous obtenez des niveaux de seuil inférieurs. Surveillez les voyants LED ci-dessous/au-dessus pour surveiller l'emplacement de votre signal par rapport au niveau de seuil actuel (voir ci-dessous).

### Ratio Knob ( $\infty$ :1...1:2)

Le bouton de rapport tournant dans le sens inverse des aiguilles d'une montre permet d'obtenir des vitesses de compression de 1 :1 à  $\infty$  :1. En tournant le bouton dans le sens des aiguilles d'une montre, la plage dynamique est étendue à partir de rapports compris entre 1 :1 et 1 :2.

## Mode (Above Threshold/Linear)

Utilisez le bouton de mode pour régler le mode de fonctionnement. En mode « Au-dessus du seuil », le traitement de la dynamique n'est appliqué qu'aux niveaux supérieurs au niveau de seuil actuel. Cela permet des applications générales de compression et de limitation des pics. En mode « Linéaire », l'ensemble de la plage dynamique est traité en fonction du réglage du ratio. Par exemple, -80...+20 dB sera compressé à -40...+10 dB si votre ratio est de 2 :1 avec un niveau seuil de 0 dB. Cela donne toutes sortes d'effets intéressants et extrêmes tels que des tambours de pompage, une ambiance morte, etc. qui peuvent être réglés à votre goût via la vis de mixage (voir ci-dessous). Dans les deux modes, le niveau seuil s'applique comme le point au-dessus ou autour duquel la compression ou l'expansion a lieu.

*AVERTISSEMENT : Une compression ou une dilatation excessive vers le haut peut générer des dépassements de signal et des pics de volume inattendus. Par exemple, des sections presque silencieuses ou très silencieuses de votre*

*signal peuvent être tirées vers le haut lors de l'utilisation de taux de compression élevés en mode linéaire. Utilisez la vis HR (voir ci-dessous) ou réduisez le rapport et abaissez le seuil pour éviter cela.*

## Headrom "HR" (-6...+2dB)

Utilisez la vis de la marge de manœuvre pour régler le niveau au-dessus duquel les pics de signal seront écrêtés. C'est idéal pour apprivoiser les dépassements transitoires autrement causés par une forte expansion vers le haut ou une compression vers le bas en mode linéaire.

### **Mix (0...100%)**

Utilisez la vis de mixage pour régler le mélange croisé souhaité entre le signal sec et le signal traité.

### **Trim (-20...20dB)**

Le trim est un gain d'appoint linéaire appliqué avant le nœud de mixage.

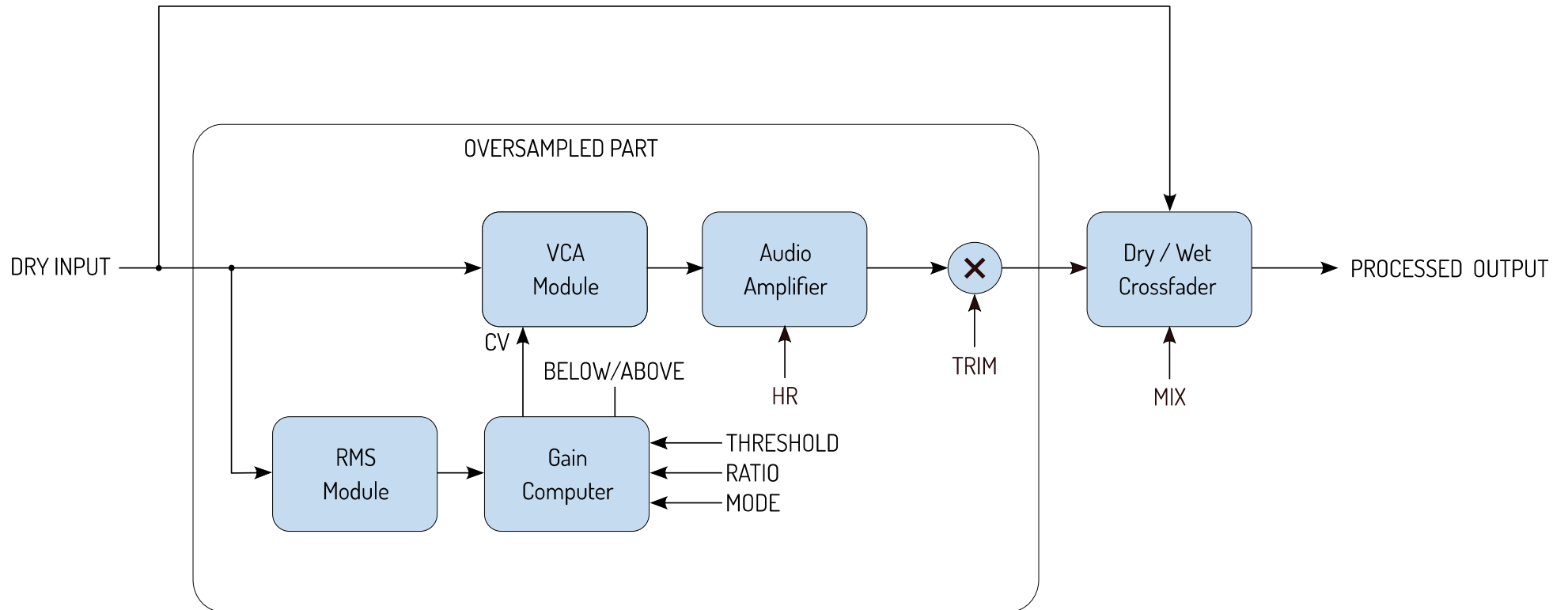
## **METERING**

### **Below / Above LEDs**

Les deux LED au-dessus du bouton de seuil indiquent si votre signal est actuellement en dessous (jaune) ou au-dessus (rouge) du niveau de seuil.



# SYSTEM BLOCK DIAGRAM



## KEYBOARD SHORTCUTS

### VST hosts

Ctrl/CMD+Click: Reset to default

Alt+Drag: Switch to circular knob mode

Shift+Drag/Mouse wheel: Fine control

### AU / AAX hosts

Alt+Click: Reset to default

Shift+Drag/Mouse wheel: Fine control

### All hosts

Double Click: Reset to default / jump to value

## SYSTEM REQUIREMENTS

- Intel compatible or Apple Silicon CPU
- 2GB RAM
- Mac OS 10.12 or newer (Mac users)
- Windows 7 or newer (PC users)
- A display resolution of 1280×1024 pixels or more
- A VST2, VST3, AAX or AU compatible 64-bit host (Mac users)
- A VST2, VST3 or AAX compatible 32-bit or 64-bit host (PC users)

## FAQS AND TECHNICAL SUPPORT

An overview of Frequently Asked Questions can be found on our website under the following link

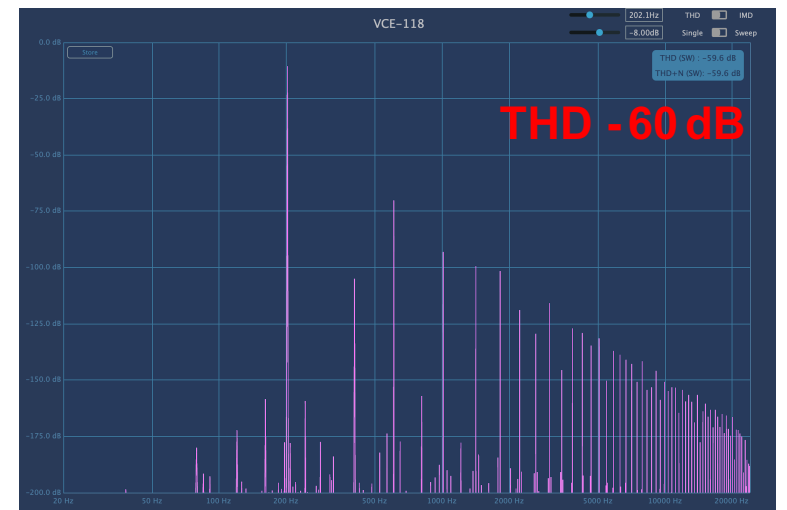
<https://fuseaudiolabs.com/faq.html>

Our tech support can be reached via the following link

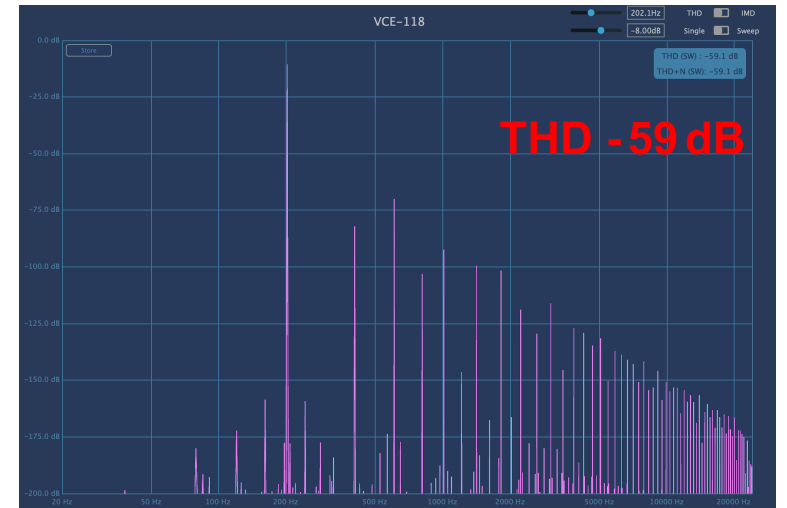
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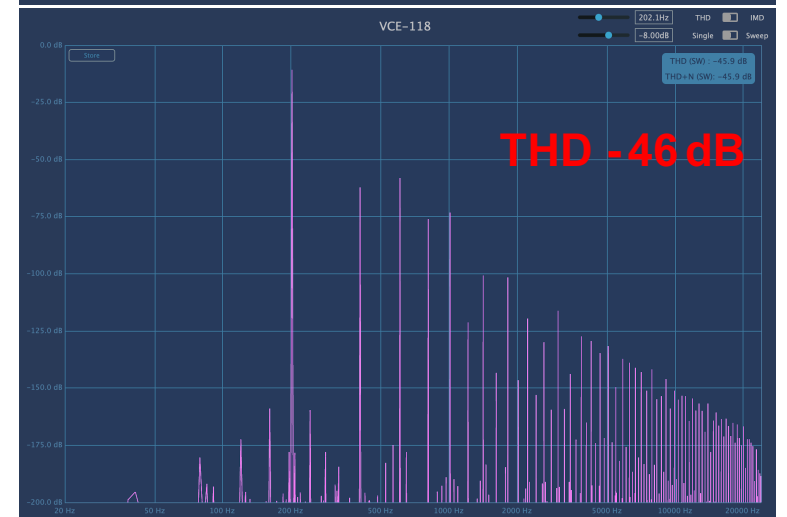
HR +22 dB



HR +3 dB

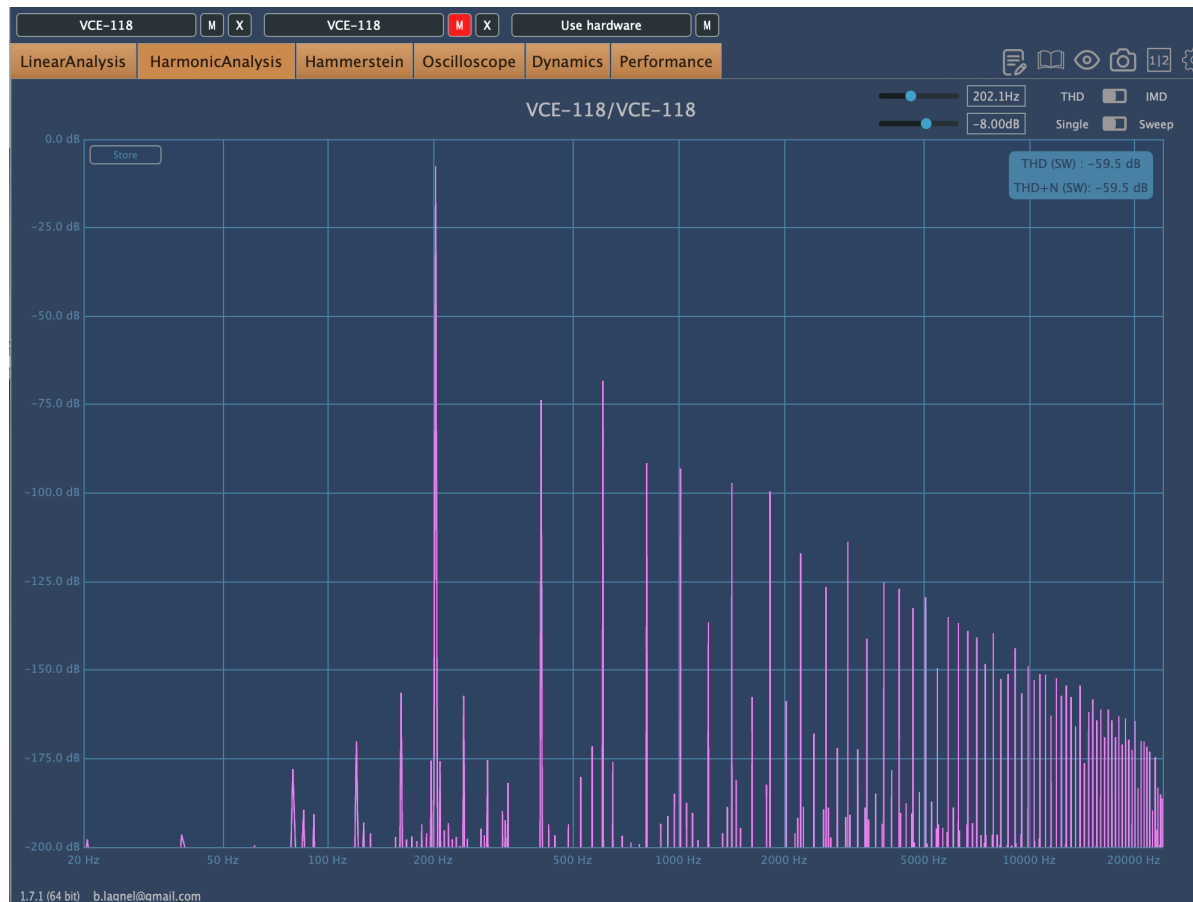


HR -6 dB



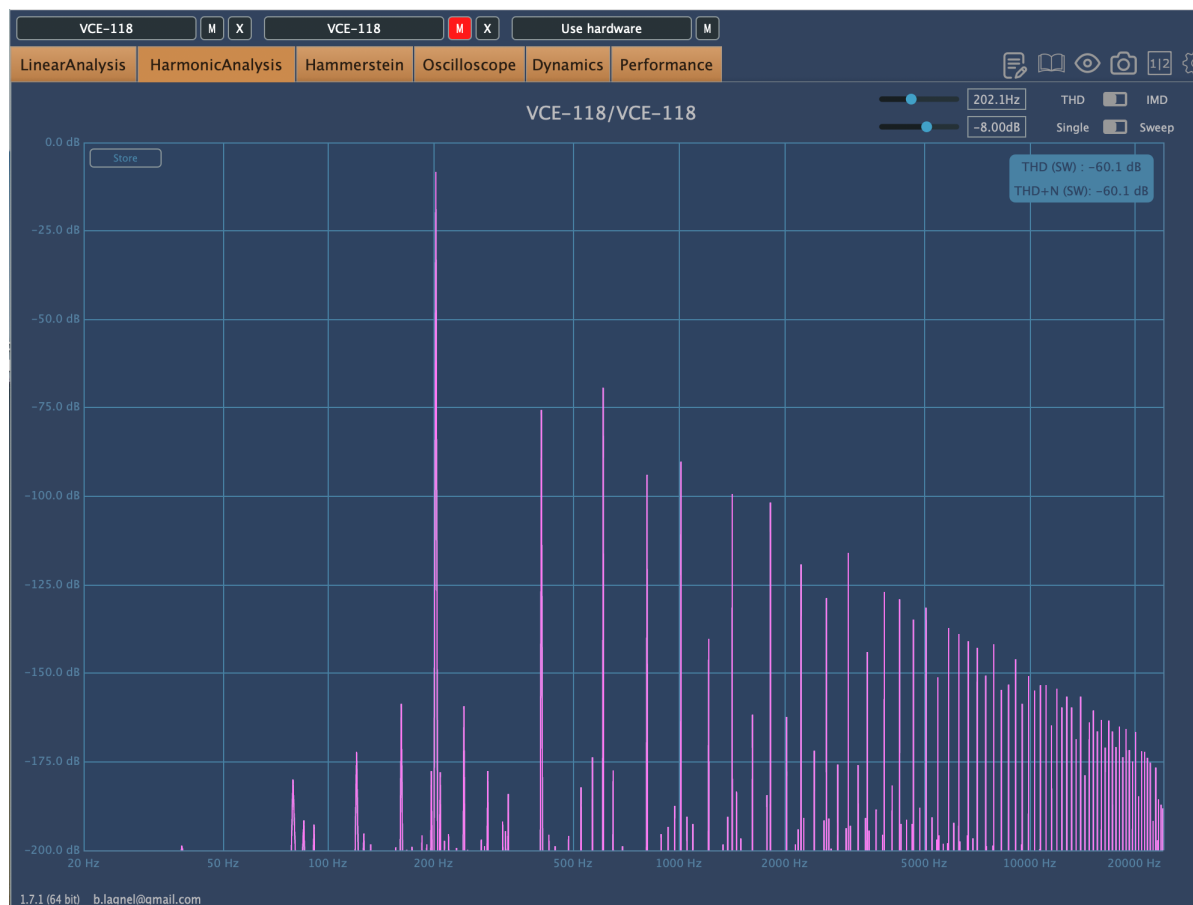


HR +3 dB





HR +3 dB





**dbx**  
**118**

BELOW      ABOVE

●      ●  
THRESHOLD

ABOVE  
THRESHOLD



LINEAR



2V      10MV

LEVEL

**DYNAMIC RANGE  
ENHANCER**

COMPRESS

1.2      1.0      1.2  
1.4      2.0      1.4  
3.0      5.0      1.6  
∞      2.0      1.8

EXPAND

**INDICATEURS LED INFÉRIEURS/AU-DESSUS**

**(Indicateurs de seuil de diodes**

**électroluminescentes) Lorsque BELOW (la LED orange) s'allume, le signal d'entrée du 118 est inférieur au niveau défini par la commande THRESHOLD. Lorsque ABOVE (la LED rouge) s'allume, le signal vers le 118 est au-dessus du niveau défini par le THRESHOLD CONTROL.**

**COMPRESS/EXPAND (Contrôle)** Ajuste la quantité de compression ou d'expansion créée par le 118.

Définition de la commande COMPRESS/EXPAND sur une valeur dans la plage EXPAND (dans le sens des aiguilles d'une montre à partir du réglage « 1.0 ») fait que le 118 agit comme un EXPAND. Si vous réglez la commande COMPRESS/EXPAND sur une valeur de la plage COMPRESS (dans le sens inverse des aiguilles d'une montre par rapport au réglage « 1.0 »), le 118 agit comme un compresseur. Lorsque la commande est réglée sur 1.0 (position 12 heures), le 118 agit comme un amplificateur de ligne conventionnel, n'ayant aucun effet sur le signal.

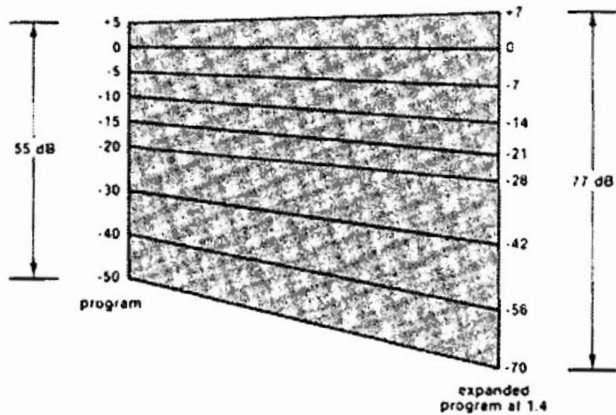


**AU-DESSUS DU SEUIL/LINÉAIRE (commutateur)** En position LINÉAIRE (interrupteur « in »), le 118 fonctionne sur toute la plage de niveaux de signal d'entrée.

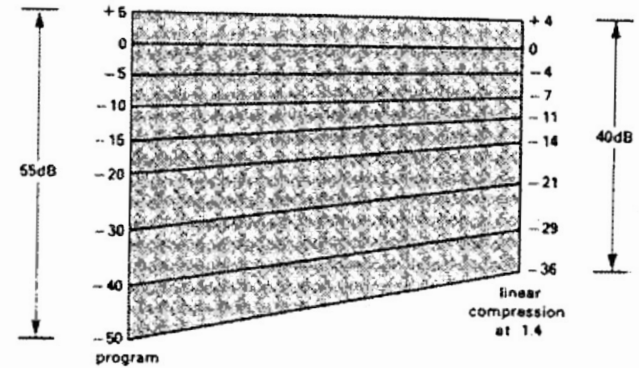
• **AU-DESSUS DU SEUIL** (interrupteur « out »), le 118 n'agit que sur les signaux au-dessus du niveau réglé par la commande THRESHOLD.

**THRESHOLD (Control)** Définit le niveau de « décision » : en mode ABOVE THRESHOLD, le THRESHOLD est le niveau au-dessus duquel le 118 commence la compression ou l'expansion ; en mode LINEAR avec la commande COMPRESS/EXPAND réglée sur un réglage EXPAND, le THRESHOLD est le niveau au-dessus duquel le 118 augmente les niveaux de signal, et en dessous duquel il diminue les niveaux de signal ; en mode LINEAR avec la commande COMPRESS/EXPAND réglée sur un réglage COMPRESS, le THRESHOLD est le niveau au-dessus duquel le 118 diminue les niveaux de signal, et en dessous duquel il augmente les niveaux de signal. (Reportez-vous à la figure 3, A et C.)

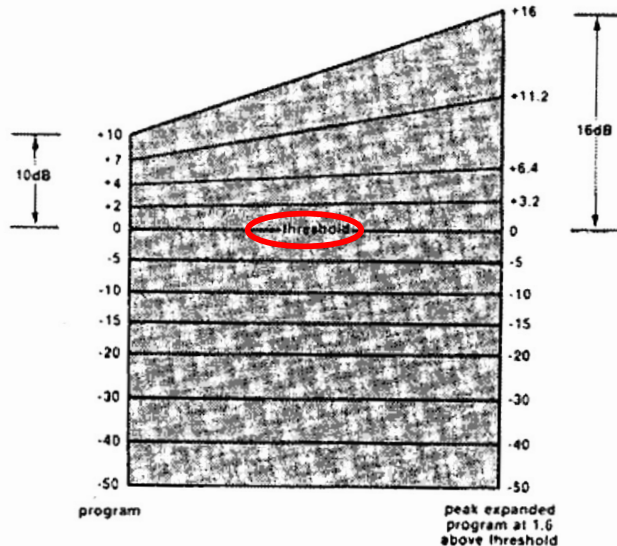




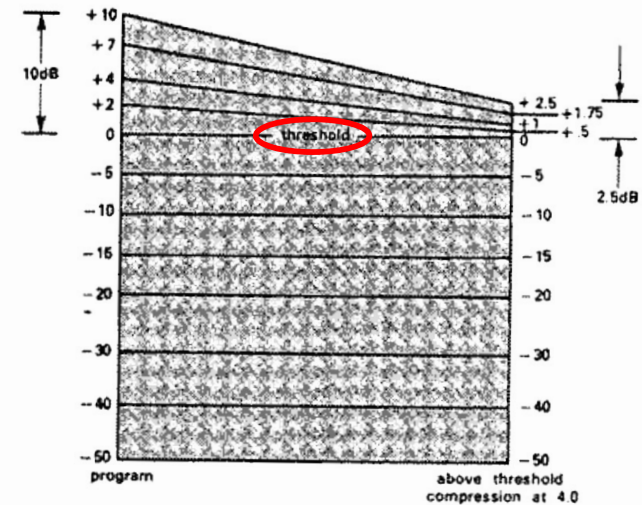
A — Linear expansion operates over the entire dynamic spectrum of music irrespective of frequency (20-20,000Hz), making loud passages louder and quiet passages quieter, reducing audible noise. Expansion ratios are adjustable. For example: 1.0 = 0% increase in dynamic range; 1.2 = 20%; 1.5 = 50%; 2.0 = 100%.



C — Linear compression reduces the entire dynamic range of the music irrespective of frequency (20-20,000Hz).



B — Above threshold expansion is used to make loud passages louder above a user adjustable threshold level when no low level dynamic alternation is desired.



D — Above threshold compression has no effect on low level signals. When signal level reaches the adjustable threshold, the dynamic content of the music is decreased but only above that threshold. Higher compression ratios may be used in above threshold compression than in linear compression; however, all ratios are available.

These Graphs illustrate the Actions of the 118 in the **LINEAR** and **ABOVE THRESHOLD** Modes.



# dbx Model 118

dynamic range enhancer

## INSTRUCTION MANUAL



## INDEX

BRIEF OPERATING INSTRUCTIONS	2
INTRODUCTION	5
Dynamic Range and Why it is Important	5
dbx Expansion	6
DETAILED OPERATING INSTRUCTIONS/GENERAL NOTES	6
CONNECTIONS	7
OPERATION	15
When to Use Expansion	15
How to Use the 118 as an Expander (Linear Expansion)	15
When to Use Compression	16
How to Use the 118 as a Compressor (Linear Compression)	16
When to Use Peak Unlimiting	17
How to Use the 118 as a Peak Unlimiter/Peak Limiter	17
Mono Operation	17
Quad Operation	17
Using the 118 with a Tape Noise Reduction System	18
When to Use dbx Tape Noise Reduction Plus Expansion	18
How to Record and Playback with Expansion	18
When to Record and Playback dbx-Encoded Tapes with Compression	18
How to Peak Limit After Decoding a dbx-Encoded Program	19
COMPRESSORS, LIMITERS AND EXPANDERS, A SUMMARY	19
Compressor	19
Peak Limiter	19
Expander	19
Peak Unlimiter	21
How the 118's Compression/Expansion Circuits Work	21
SIMPLE PRECAUTIONS WHEN USING ANY PROGRAM EXPANDER	22
SPECIFICATIONS	23
dbx PRODUCT WARRANTY	24
FACTORY SERVICE	24
SCHEMATIC	25
GLOSSARY	I

**WARNING: TO PREVENT FIRE OR SHOCK HAZARD,  
DO NOT EXPOSE THIS APPLIANCE TO RAIN OR MOISTURE.**



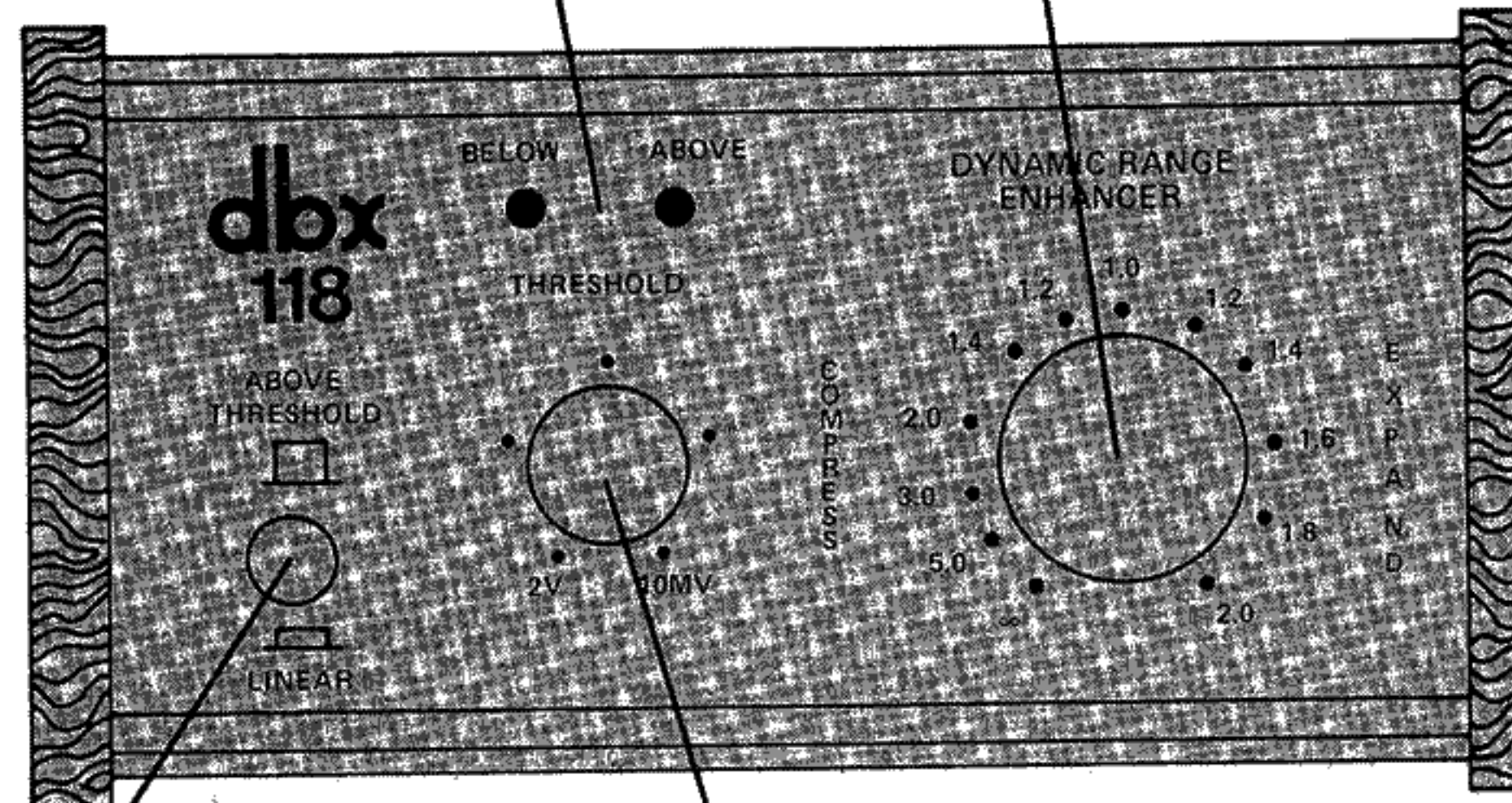
## BRIEF OPERATING INSTRUCTIONS

2

Fig. 1 — 118 Front Panel

**BELOW/ABOVE LED INDICATORS (Light Emitting Diode Threshold Indicators)** When BELOW (the amber LED) glows, the input signal to the 118 is below the level set by the THRESHOLD control. When ABOVE (the red LED) glows, the signal to the 118 is above the level set by the THRESHOLD CONTROL.

**COMPRESS/EXPAND (Control)** Adjusts the amount of compression or expansion created by the 118. Setting the COMPRESS/EXPAND control to a value in the EXPAND range (clockwise from the "1.0" setting) causes the 118 to act as an expander. Setting the COMPRESS/EXPAND control to a value in the COMPRESS range (counterclockwise from the "1.0" setting) causes the 118 to act as a compressor. When the control is set at 1.0 (12 o'clock position), the 118 acts as a conventional line amplifier, having no effect on the signal.



**ABOVE THRESHOLD/LINEAR (Switch)** When in the LINEAR position (switch "in"), the 118 functions over the entire range of input signal levels. When in the ABOVE THRESHOLD position (switch "out"), the 118 acts only on signals above the level set by the THRESHOLD control. (Refer to Figure 3, B & D.)

**THRESHOLD (Control)** Sets the "decision" level: in the ABOVE THRESHOLD mode, the THRESHOLD is the level above which the 118 begins compression or expansion; in the LINEAR mode with the COMPRESS/EXPAND control set to an EXPAND setting, the THRESHOLD is the level above which the 118 increases signal levels, and below which it decreases signal levels; in the LINEAR mode with the COMPRESS/EXPAND control set to a COMPRESS setting, the THRESHOLD is the level above which the 118 decreases signal levels, and below which it increases signal levels. (Refer to Figure 3, A & C.)

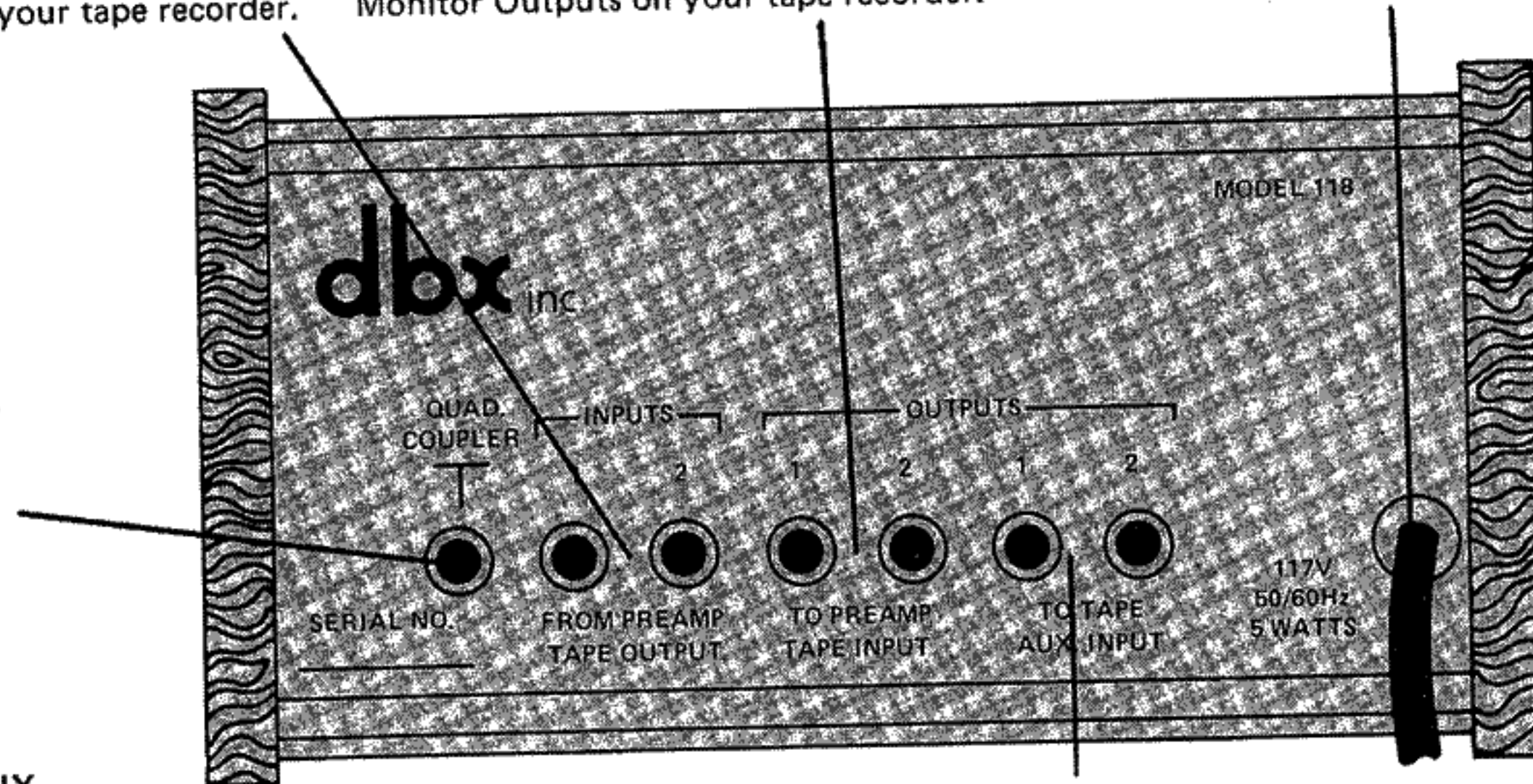
Fig. 2 — dbx 118 Rear Panel

**INPUTS** Connect the cables from the Pre-amp Tape Outputs (Tape Recorder Outputs) on your receiver, preamplifier, or integrated amplifier/preamplifier to these jacks. If you have a tape recorder, these cables were originally connected to the Line or Aux inputs on your tape recorder.

**OUTPUTS** Connect cables from the 118's TO PREAMP TAPE INPUT jacks to the Preamp Tape Inputs (Tape Monitor Inputs) on your receiver, preamplifier, or integrated amplifier/preamplifier. If you have a tape recorder, these cables originally came from the Aux, Line or Monitor Outputs on your tape recorder.

**AC POWER CABLE** Connect the 118's Power Cable to any 117 volt, 50 or 60Hz outlet. The 118 requires only 5 watts (maximum) AC power, which means that you may leave it connected to the AC line continuously. Alternately, plug the 118 into a switched outlet on your receiver or integrated amp/preamp. The 118 has no AC power (On-Off) switch.

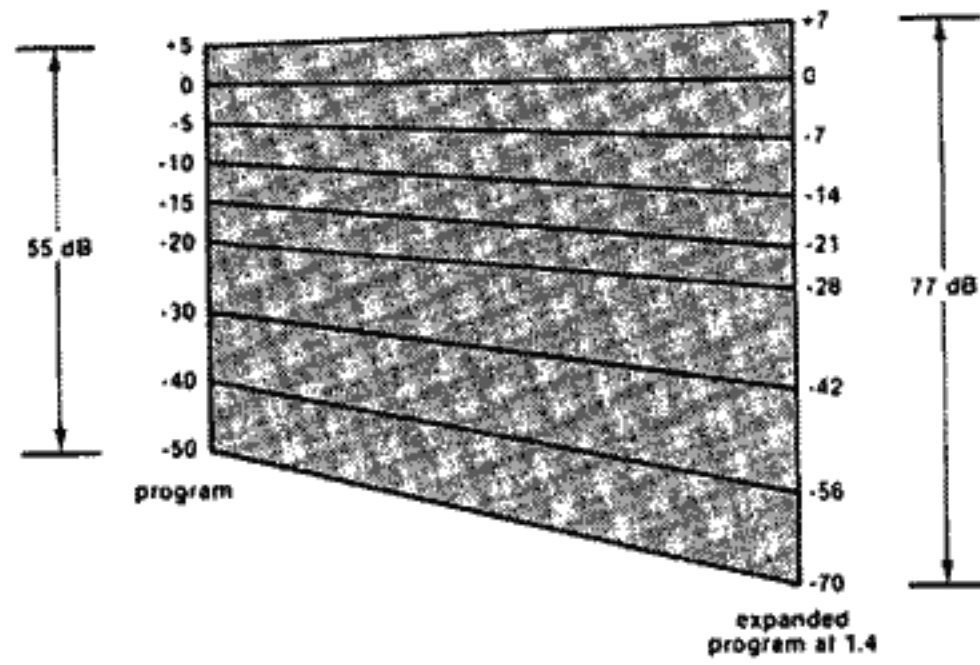
**QUAD COUPLER** If you have two dbx 118's, and wish to expand or compress a four channel program, connect a cable between the QUAD COUPLER jacks on each unit, connect the INPUTS and OUTPUTS of one unit to your system's Front channels, connect the INPUTS and OUTPUTS of the other unit to your system's Rear channels, and operate the two units normally. (Matrix four channel programs may be dbx processed in their 2-channel encoded form using one 118.)



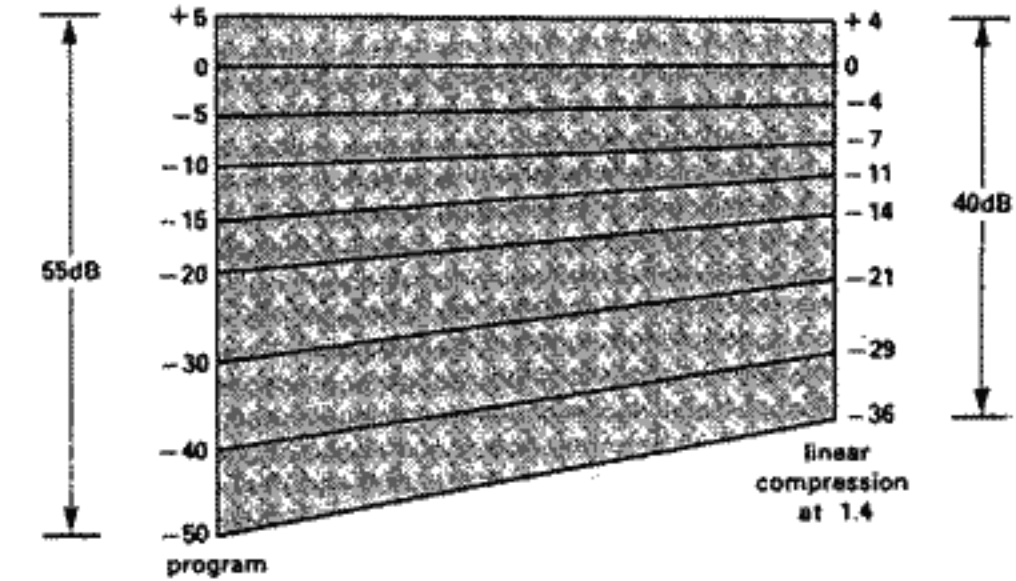
**CAUTION:** If you connect the TO TAPE AUX INPUT jacks to the inputs of your tape recorder, and connect the outputs of your tape recorder to the Aux Inputs of your receiver, preamplifier or integrated amplifier/preamplifier, potentially damaging oscillation is possible. This could occur in some systems when you attempt to record and monitor at the same time. To protect your sound system, do not switch the Input Selector Switch on your receiver, preamplifier or integrated amplifier/preamplifier to the Aux position when you are recording.

If you have a tape recorder, connect cables from the 118 TO TAPE AUX INPUT jacks to the Aux or Line Inputs on your tape recorder. Connect the Aux, Line or Monitor Outputs of your tape recorder to the Aux Inputs of your receiver, preamplifier or integrated amplifier/preamplifier. Alternately, if your equipment has two Tape Monitor Loops, connect the Aux, Line or Monitor Outputs of your tape recorder to the Tape Monitor Inputs (Play Inputs) of the second Tape Monitor Loop.

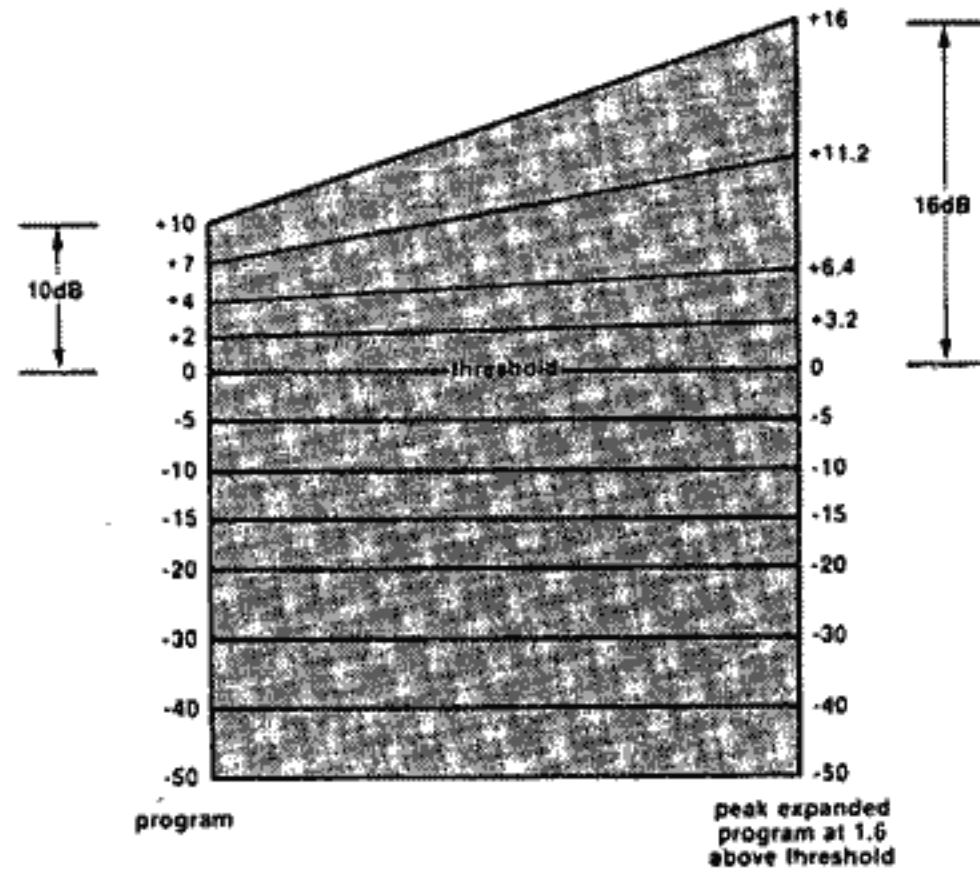




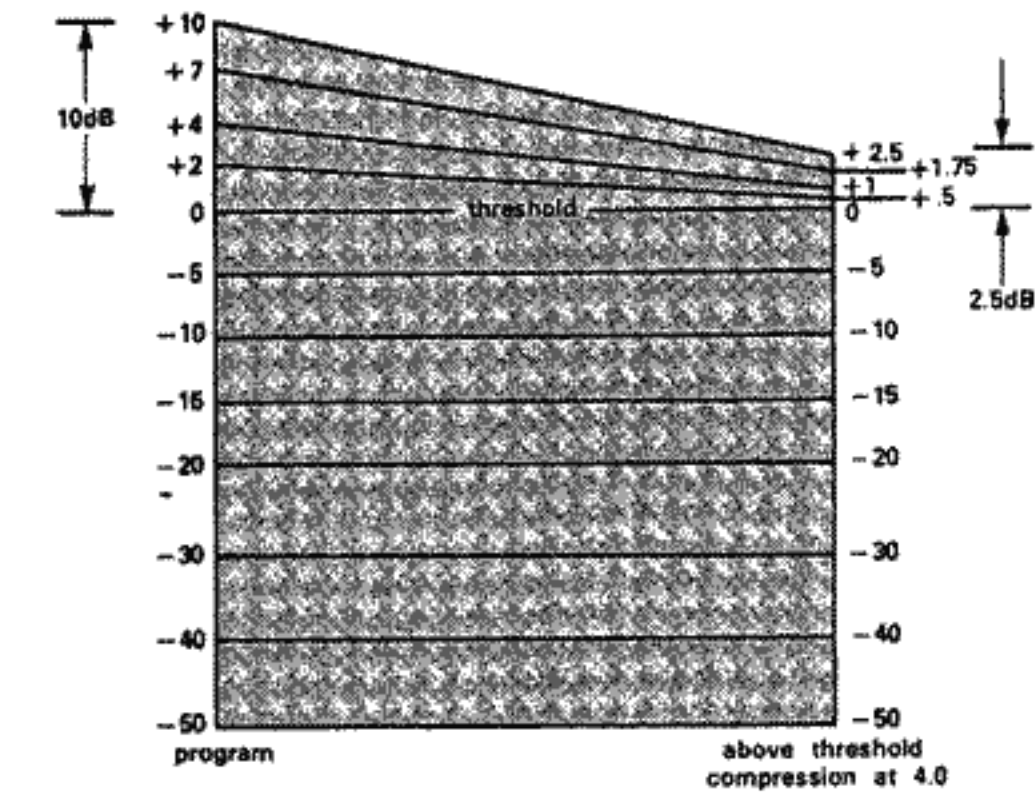
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C — Linear compression reduces the entire dynamic range of the music irrespective of frequency (20-20,000Hz).



B — Above threshold expansion is used to make loud passages louder above a user adjustable threshold level when no low level dynamic alternation is desired.



D — Above threshold compression has no effect on low level signals. When signal level reaches the adjustable threshold, the dynamic content of the music is decreased but only above that threshold. Higher compression ratios may be used in above threshold compression than in linear compression; however, all ratios are available.

Fig. 3 — These Graphs Illustrate the Actions of the 118 in the LINEAR and ABOVE THRESHOLD Modes.

## INTRODUCTION

A live musical performance is an exciting experience, the kind of experience that record manufacturers and radio stations *attempt* to capture for reproduction in home listening. The most significant shortcoming of tapes, records and radio broadcasts is the lack of dynamic range, a shortcoming which is due to the dynamic limitations in the recording and transmission media. Yet even when using the best speakers, amplifiers, tuners, turntables, and tape machine, much of the excitement of the original, live performance is lost due to the dynamic range limitations of source material.

### **Dynamic Range and Why it is Important**

Dynamic range can be defined as the difference in sound level between the loudest and the quietest passages in music, measured in decibels, or dB. Acoustical sound level is measured in decibels of sound pressure level, or dB SPL. The dB expresses the smallest change in sound level that the average ear can detect. 0dB SPL is the threshold of human hearing, the faintest sound most people can hear. The loudest sound the average person can tolerate without physical pain is approximately 120dB SPL (some people can tolerate 130dB, others leave the room when the sound level reaches 110dB). Thus, human hearing has a maximum dynamic range of about 120dB.

Noise is something with which we are all too familiar. It is the lower restricting factor of the dynamic range of most recorded material. Typical sources of noise are phonograph disc surfaces, turntable rumble, tape hiss and FM broadcast hiss. "Rumble" and "scratch" filters can remove some of this noise, but unfortunately, they also remove parts of the program. A simple but effective way to remove noise, or to reduce it significantly without subtracting anything from the music, has long been sought but, until recently, has been difficult to achieve.

When the original program's quietest sounds are too low

in level, they are buried in the background noise of the tape, phonograph disc or FM broadcast. When the original program's loudest sounds are too high in level, they can cause distortion. Since the dynamic range of a tape machine, disc, or radio broadcast is not wide enough to accommodate the full dynamic range of a live performance (up to 100dB), the program is often "compressed." This means that its dynamic range is reduced by making the loud passages quieter, and/or the quiet passages louder. Long ago, engineers realized that if there were some way to reverse the compression process used in recording, thereby restoring the dynamic range of the program, the quality of recorded music would be significantly improved. Until recently, the increase of dynamics and the reduction of noise was not successfully achieved in any consumer product.

dbx has developed techniques that make a dramatic improvement in the dynamic range of any recorded musical material. The patented dbx Professional and dbx II Tape Noise Reduction Systems, as packaged in the dbx 150 series and 120 series, are designed to preserve dynamics and keep noise out of original tape recordings. dbx II processing has also been used to produce phonograph discs with full dynamic range and no audible surface noise.

Unlike these dbx tape noise reduction systems, which operate by encoding a program before recording and decoding it upon playback, the dbx linear decibel compression/expansion system, as packaged in the Model 118, operates on any program material. With the 118, you can enhance the dynamic range and reduce the noise of any record, pre-recorded tape, or FM broadcast, adding to your listening enjoyment. When using a 118 in combination with a dbx tape noise reduction system, such as our Model 122, 124, 152, 154, 157, or others, you can actually make recordings from a conventional phonograph disc, FM broadcast or another tape recorder that have lower noise



and wider dynamic range than the original source.

#### **dbx Expansion**

dbx tape noise reduction systems do not reduce noise that is present in the original program, but dbx expanders do. Much of the noise in an FM broadcast, or on a phonograph disc is only audible during quiet passages of the music. By expanding the program dynamics, the loud levels are made louder, and the quiet levels, *including the noise*, are made quieter. Simultaneously, much of the dynamic range and, therefore the excitement, of the original performance is restored. In addition, these same expansion circuits can be used to compress a program, and there are several situations where compression is desirable. For example, by compressing a tape which is being recorded for use in an automobile tape player, the quietest levels will be high enough to be above the road noise, yet the loudest levels do not create ear splitting volume levels. Using compression, you do get increased loudness without the usual increase in distortion.

By restoring dynamic range and lowering noise, the dbx 118 can make an old, tired record collection sound almost like new again, and can make an FM broadcast sound as good as your best conventional phonograph discs. The increase in dynamic range depends on the dynamics of the original program, and on the setting of the COMPRESS/EXPAND control. For example, a 60dB program, when subject to 1.5 expansion, becomes a 90dB program. (Refer to Figure 3A, Page 4.) For this reason, it is possible to exceed the dynamic range capabilities of a tape recorder if a program is expanded before being tape recorded. To "capture" the full dynamics of an expanded program on tape, a dbx tape noise reduction system may be utilized. This manual describes how you can fully enjoy these benefits, making the dbx 118 an even more valuable addition to your sound system and to your listening enjoyment.

## DETAILED OPERATING INSTRUCTIONS GENERAL NOTES

#### **Power Mains**

The dbx 118 is wired for 117V AC, 50 or 60Hz operation. (240V AC operation is available outside the continental U.S.A., and is so indicated on a rear panel label.)

#### **Mounting**

The 118 is supplied in a handsome cabinet that can be placed almost anywhere. Avoid mounting the 118 above any source of heat or strong electro-magnetic fields, such as those generated by power amplifiers or vacuum tube equipment.

#### **Level and Impedance**

The input levels, output levels and impedances of the 118 are designed for most hi-fi and semi-pro equipment. The 118's input impedance is 50K-ohms which means it can be driven from high or low impedance sources. The 118's input will accept a wide range of voltages. The output will drive up to 7 volts into 5000-ohms or greater. This means that you can confidently connect the 118 to almost any type of equipment found in a normal hi-fi or semi-pro system. See the Specifications section on Page 23 for more detail.

#### **Grounding**

The 118 is not grounded to the power mains through its power cord. It obtains a ground via the shields of the interconnecting audio cables from the preamplifier (or receiver) and/or from the tape machine. No special ground wires need be connected between the 118 chassis and any other equipment. If you hear hum, try reversing the AC plug, and if that does not help, check to make sure all audio cables are properly installed, and that there are no broken conductors in any of the cables. Also try cleaning the connectors on your cables since hum and radio station pickup are sometimes caused by microscopic dirt particles between

## CONNECTIONS

the mating surfaces of the connectors.

### Cable Length

Some installations will tolerate longer cables, but whenever possible it is a good idea to keep signal cable runs under 10 feet. This will minimize hum and noise pickup, and also help prevent high frequency losses caused by cable capacitance.

**NOTE:** Do not connect the 118's AC POWER CABLE to the AC mains until all signal connections have been completed. In addition, it is a good idea to turn off the power to all of your system components when making signal connections to avoid transients (pops or oscillations).

Connect your 118 according to one of the following methods:

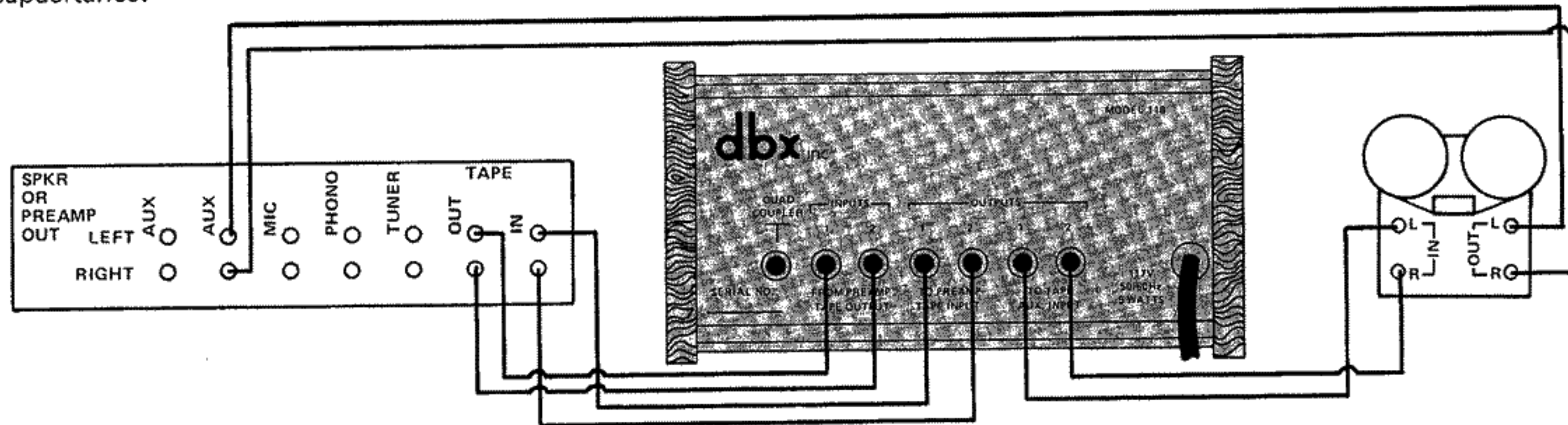


Fig. 4 – Installation of the 118 in a Receiver, Preamplifier, or Integrated Amplifier/Preamplifier having One Tape Monitor Loop.

To listen to your phonograph, tuner, or tape machine, select the corresponding Phono, Tuner or Aux input on the preamp. With the Tape Monitor Switch OFF, you will hear the unprocessed audio. Switching ON the preamp's Tape Monitor substitutes dbx-processed audio for unprocessed audio at the preamp, receiver, or main amp output. However, dbx-processed audio is always fed to the tape machine's record input (unless the COMPRESS/EXPAND control is set at 1.0). Since the tape machine's line output is connected to the amplifier's auxiliary input, off-the-tape monitoring during recording can only be done if the tape machine has a suitable headphone jack.

**CAUTION:** (Regarding the Setups in Figures 4, 5, 7, 9 and 10) When the tape machine's outputs are connected to your preamp's Aux Inputs, the tape machine's Line Outputs are connected to its own Line Inputs (via the preamp's Record Output jacks). Thus, when the preamp's Source Selector Switch is in the Aux position, a feedback oscillation can occur. To prevent such oscillation, do not set your preamp's Source Selector Switch to the Aux position when recording. As an added precaution, to avoid speaker or amplifier damage in the event of feedback, turn down the amplifier's volume control when first playing a tape until you have determined the installation is correct and there is no feedback.



The Tape Monitor Switch should normally be ON so that the speakers are fed with the correct equalization. Switching OFF the Tape Monitor removes both the 118 and the equalizer. Comparisons of compressed, expanded, or non-dbx processed audio may be made by rotating the 118's COMPRESS/EXPAND control (a setting of 1.0 is equivalent to no dbx processing).

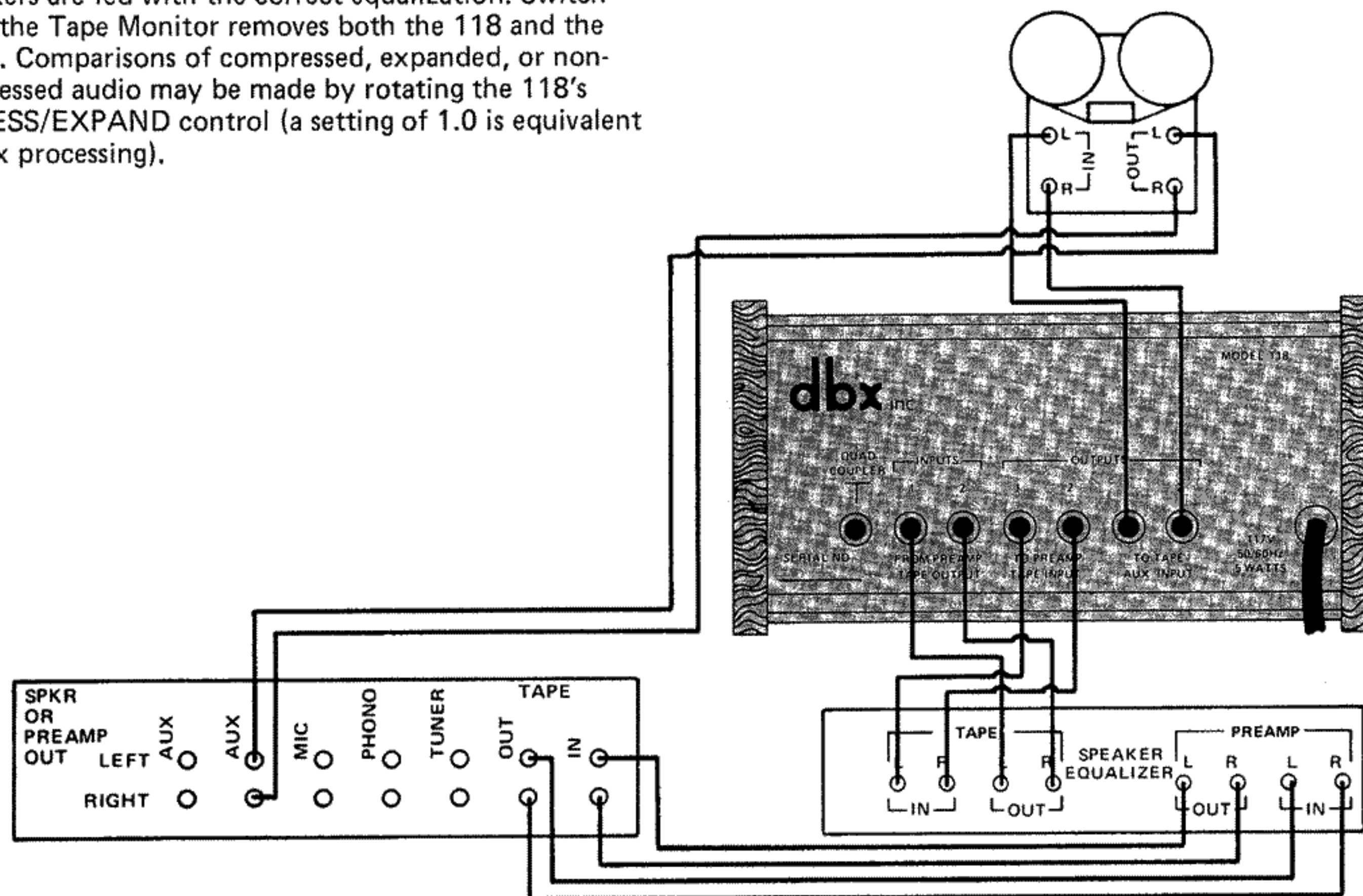


Fig. 5 — Installation of a dbx 118 in conjunction with a Speaker Equalizer, such as one of the Bose or Electro Voice Interface series, or any Graphic Equalizer having its own Tape Monitor Loop, such as those manufactured by S.A.E., Soundcraftsmen, Altec, and others.

If an external equalizer is connected, it should be placed after the dbx unit, as illustrated. The 118 cannot be switched out of the circuit, so to avoid signal processing, set the 118's COMPRESS/EXPAND control at 1.0 (12 o'clock).

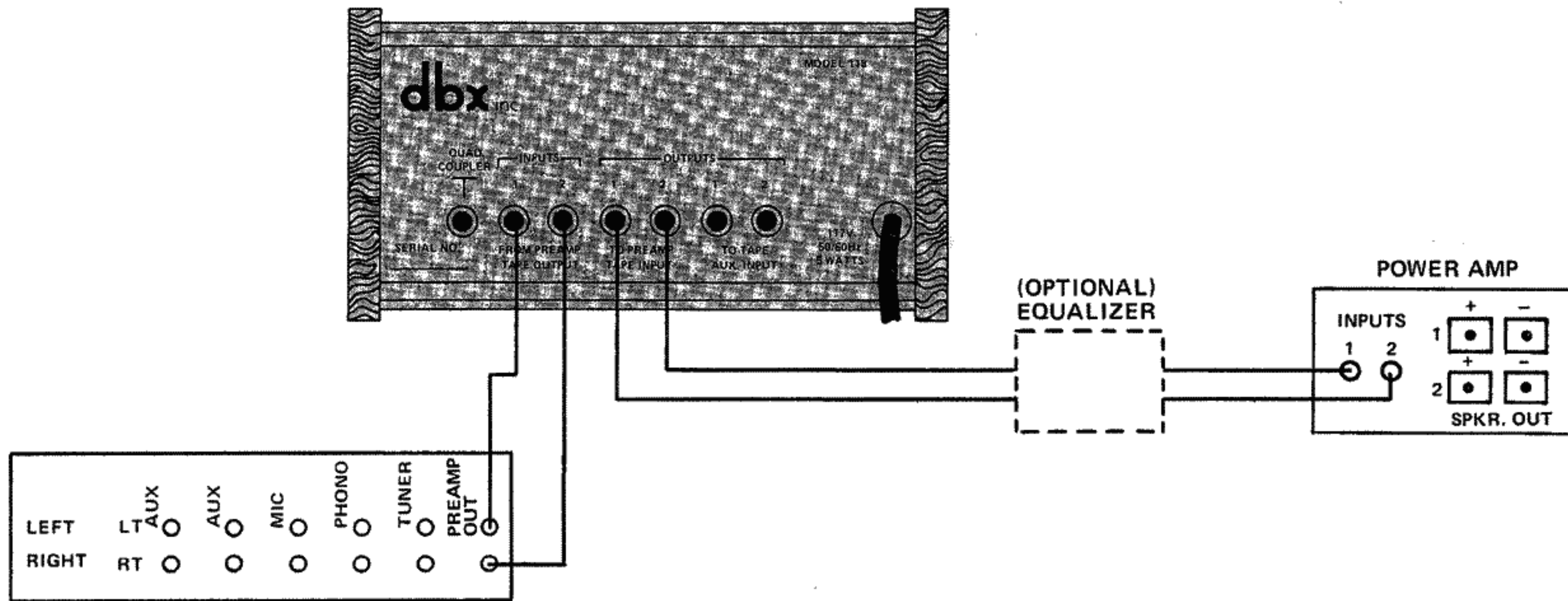
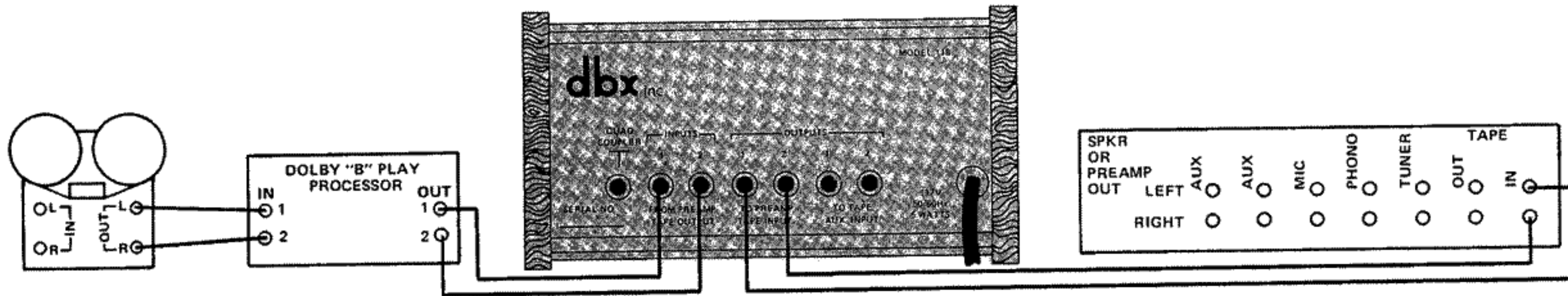
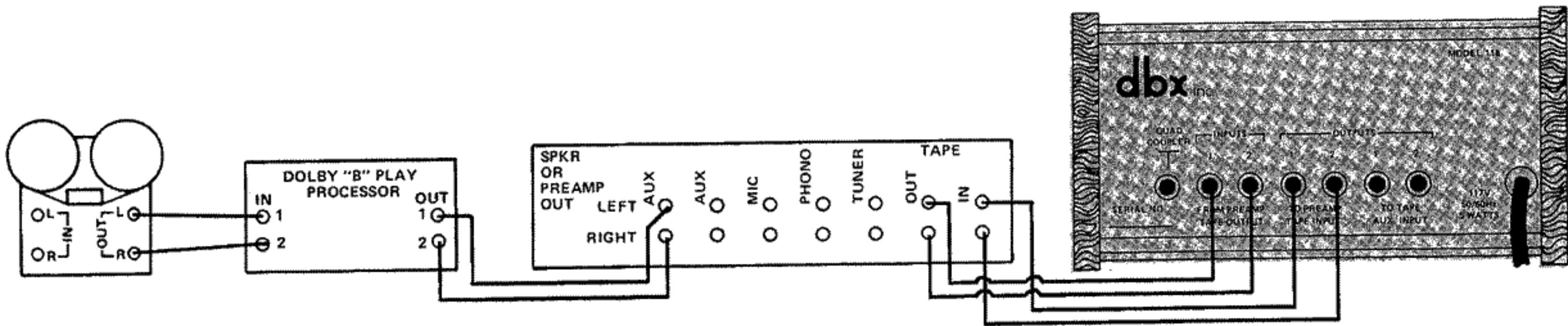


Fig. 6 – Installation of a dbx 118 in a Sound System where the Preamp has no Tape Monitor Loop, and a separate Power Amplifier and Preamplifier are utilized. This same setup applies where an Integrated Amp/Preamp or Receiver has Preamp Out/Main Amp in Jacks.

A dbx 118 may be used to expand audio after Dolby 'B' decoding, thereby achieving additional noise reduction, and increase the dynamic range.



"A" shows the model 118 connected in the Play side of the tape monitor for playback only.



"B" shows the model 118 connected to the amplifier tape monitor loop. The Dobby 'B' playback is connected to the amplifier aux inputs. This will allow all functions on your amplifier to be expanded simply by changing your amplifier selector switch.

Fig. 7 — Installation of a 118 in conjunction with a Dolby 'B' Noise Reduction Decoding. for additional Noise Reduction after Dolby

To tape record a phonograph disc, FM broadcast, or any source plugged into the preamp's Aux Input, select the appropriate source with the preamp's Source Selector Switch. To make compressed recordings, set the Tape Monitor Switch for "Dub A to B" so that the program will first pass through the 118, and then to the tape machine; with the tape machine set for off tape monitoring, the compressed and recorded program will be heard while

recording. To expand conventional recordings upon playback, set the Tape Monitor Switch for "Dub B to A," and the expanded program will be heard. To hear an unexpanded tape, set the preamp Tape Monitor to "Play B" or leave the switch in "Dub B to A" but set the 118 COMPRESS/EXPAND control at 1.0. To bypass dbx processing completely, set the Tape Monitor switch to "Source."

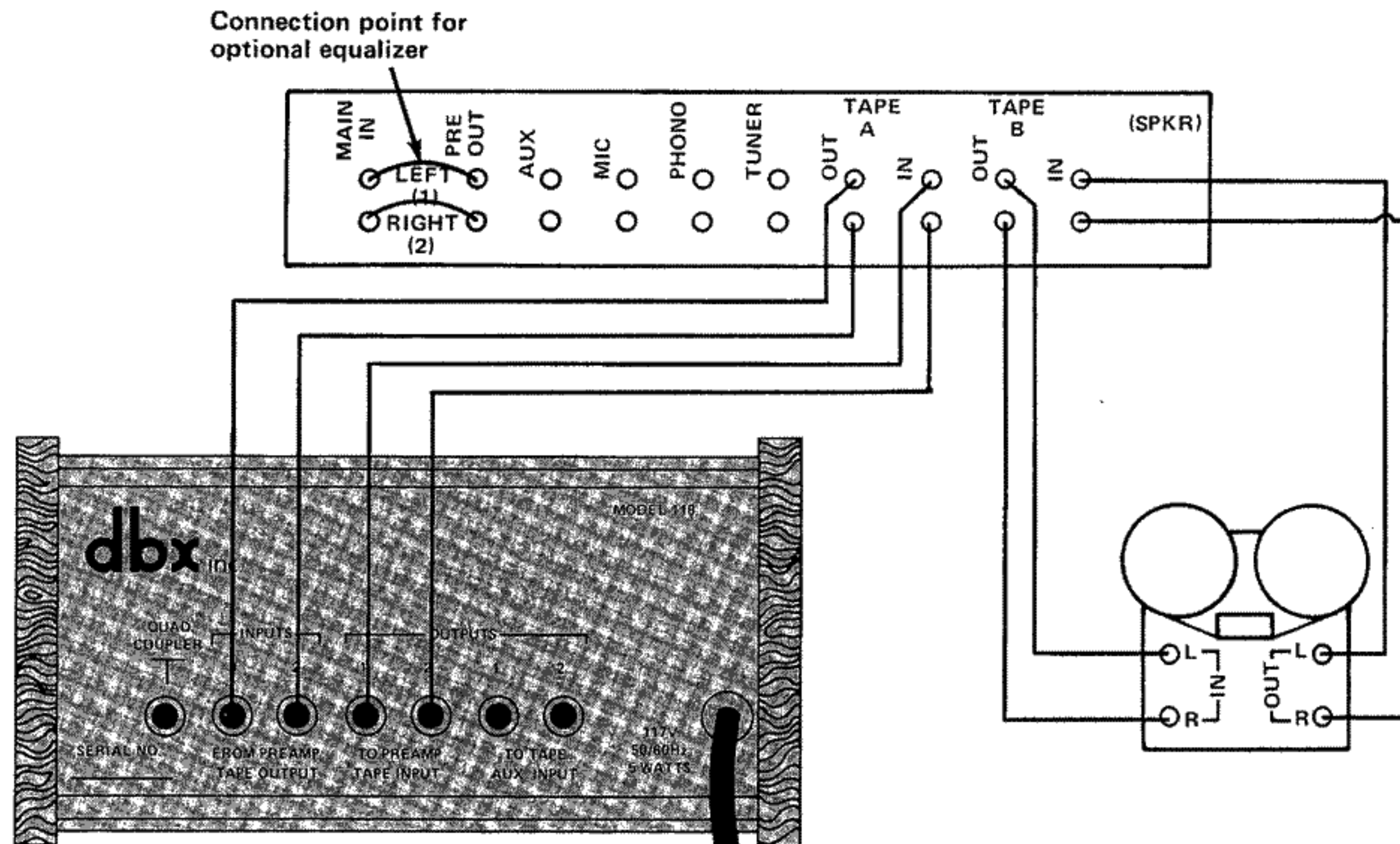


Fig. 8 — Installation of a 118 in conjunction with a Preamp, Integrated Amp/Preamp, or Receiver having Two Tape Monitor Loops, a Setup that permits Off-the-Tape Monitoring via the Main Sound System (not via Headphones, as in Figure 4).



The best results are achieved by dbx processing the quad-encoded stereo program, and there is the additional advantage in that only one 118 is required (2 channels instead of 4). The stereo Record Output of the preamp goes through the 118, where it is expanded or compressed, and returns to the quad decoder for decoding of the four channels and returns as a four channel program. A matrix encoded disc or FM broadcast can be played with dbx expansion to reduce noise prior to matrix decoding. When making matrix-encoded tape recordings for playback in automobiles, or for background music, use dbx compression prior to recording. (When making tape copies of matrix encoded programs, it is a good idea to play the tape on the same machine used to make the recording. This minimizes phase errors which can cause the decoder to incorrectly position the sound within the four channel perspective.)

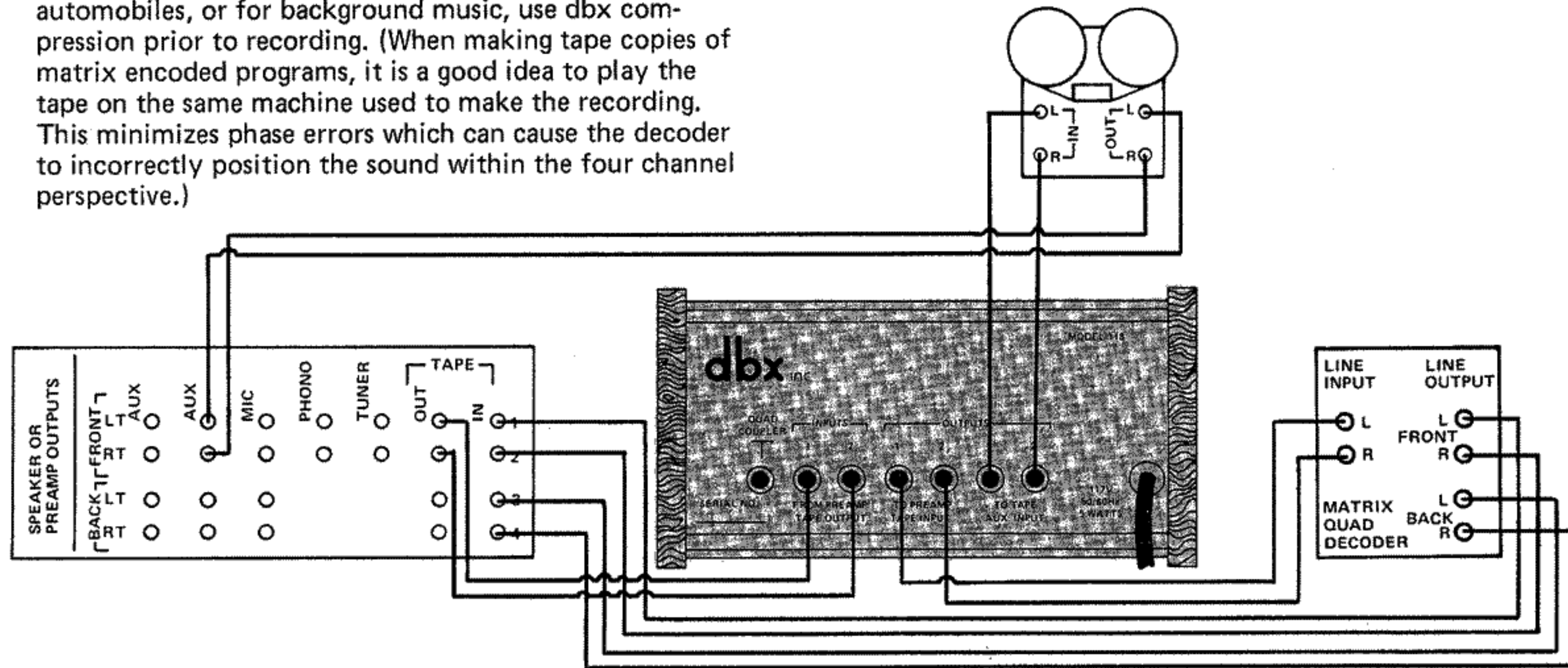


Fig. 9 – Installation of the 118 with a Matrix-derived Four Channel (Quad) Sound System.

Connect a cable between the QUAD COUPLER jacks of the two 118's. Front panel control settings of the two 118's should be identical.

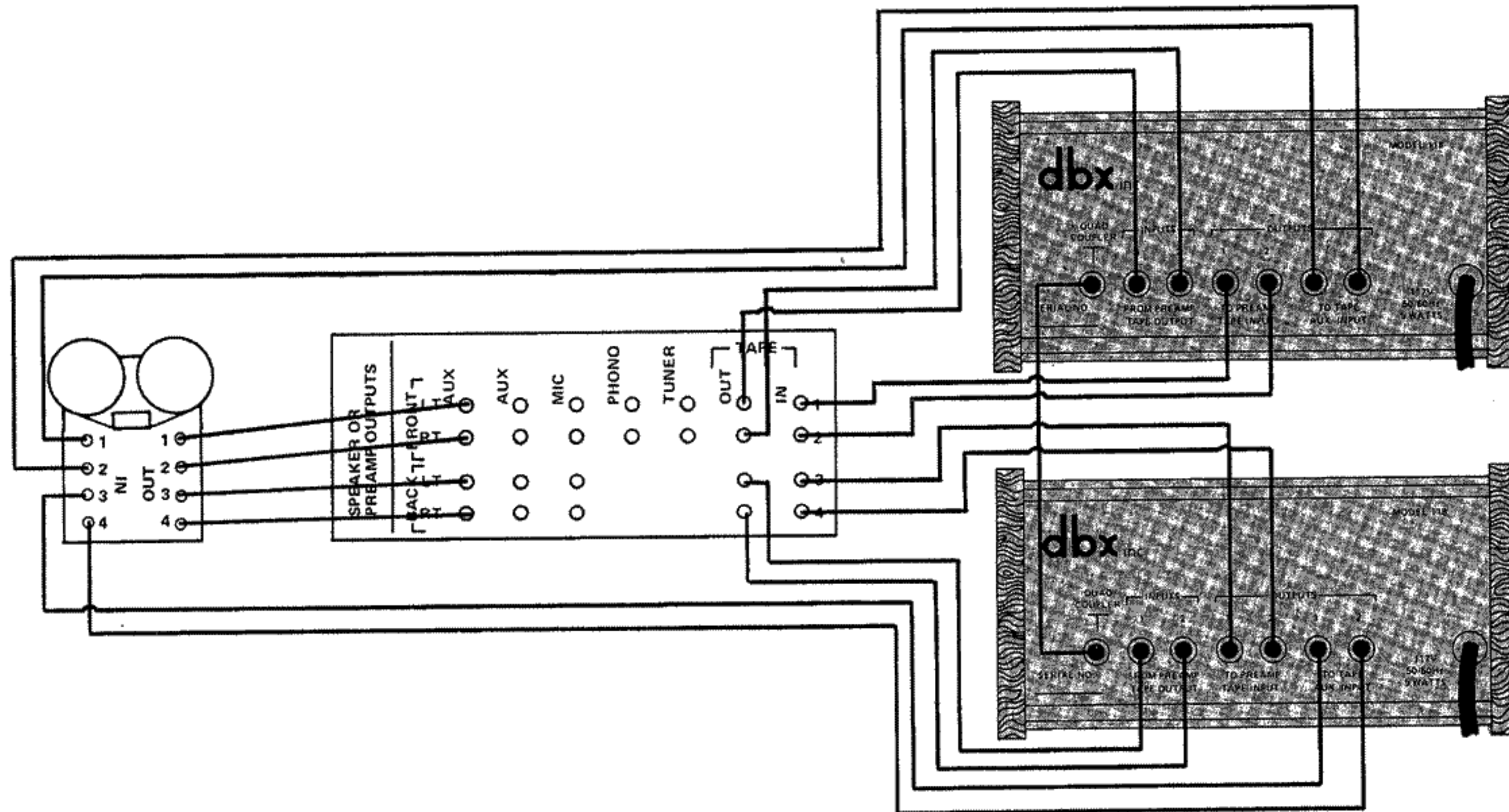


Fig. 10 – Discrete Quad System utilizing Two Model 118's.

The dbx 118, used as an expander, restores dynamics of recorded or broadcast programs. In this capacity, as a single-ended device, the 118 does achieve a limited amount of noise reduction. However, dbx makes tape noise reduction systems that contain circuitry capable of sup-

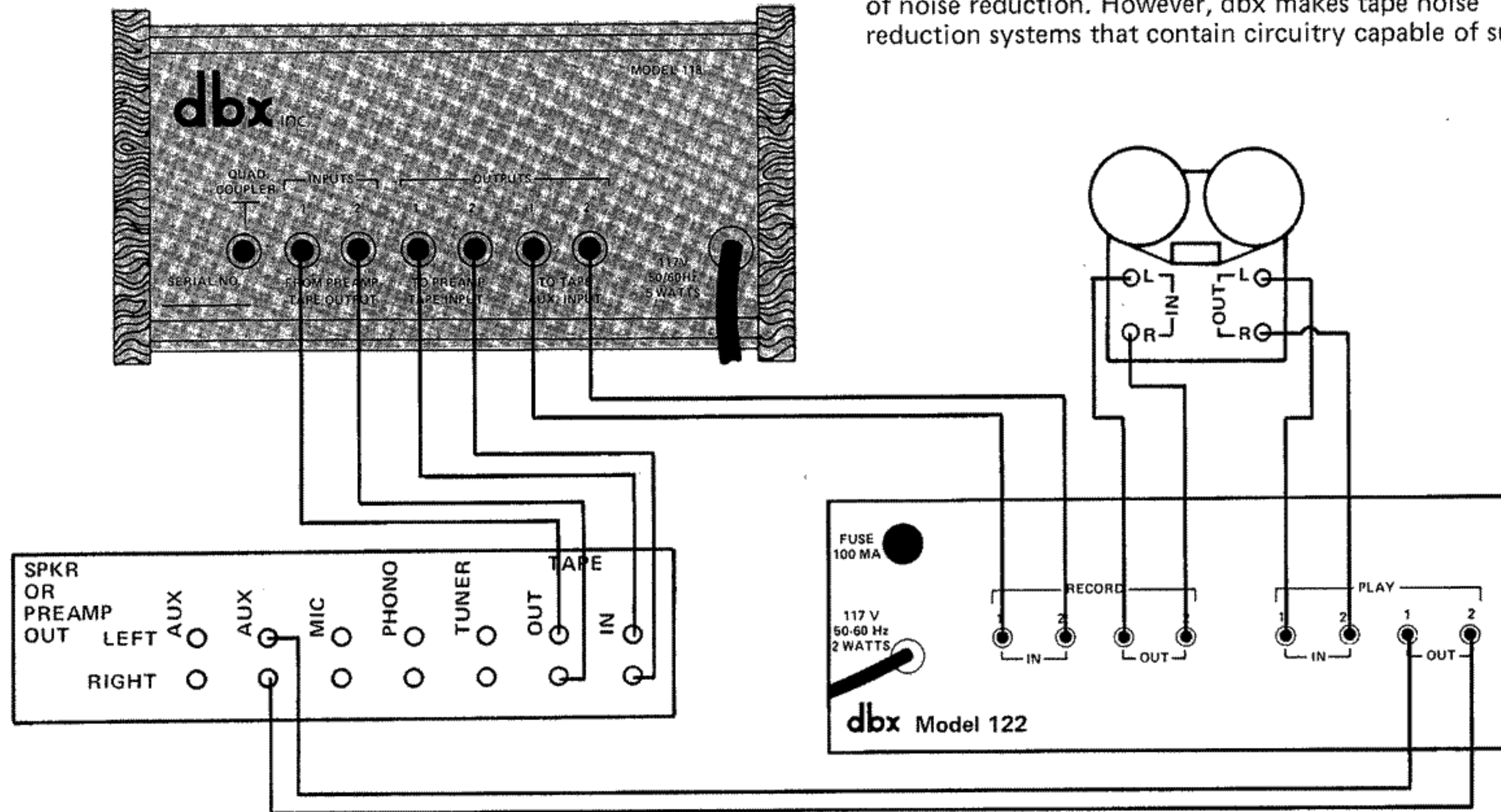


Fig. 11 – Using the dbx 118 in conjunction with a dbx Tape Noise Reduction System (the Model 122 is illustrated, but any dbx professional or dbx II unit is suitable). This setup is ideal for Expanding the Dynamics of a conventional Phonograph Disc or an FM Broadcast and then making a Low-Noise, Wide Dynamic Range Recording of that Program.



## OPERATION

15

pressing up to 30dB of tape noise. The thought occurred to many dbx users, and to dbx engineers, that it would be desirable to restore the "lost" dynamics and reduce noise on a conventional phonograph disc or broadcast, and then capture that program on tape preserving the increased dynamic range while adding no audible noise in the tape recording process. This is exactly what the installation in Figure 11 is designed to achieve. (The dbx Model 128 is a product which essentially duplicates these functions by including the equivalent of a Model 118 and a Model 122 in a single unit.)

When recording, set the 118 to the desired amount of expansion (which can be monitored by turning the preamp Tape Monitor ON). The expanded signal will then be encoded by the dbx 122 (or other dbx tape noise reduction system) so that it will fit within the dynamic range capability of your tape recorder. Upon playback, the 122 will decode the tape so that it can be heard without further 118 processing. Thus, when playing the tape, switch the preamp to Aux and leave the Tape Monitor OFF. Alternately, use the 122 to make an encoded recording that has not been pre-expanded, and expand the tape upon playback, after dbx decoding, by switching ON the Tape Monitor.

At first glance one may ask why this combination of expansion and noise reduction can't be done with a Dolby 'B' tape noise reduction system. The reason is that the Dolby system will only reduce noise 7 to 10dB; it will not allow wide dynamic range music to be recorded on tape. The dbx 122 (or other dbx tape noise reduction system) is capable of far more noise reduction (compression and expansion) than the Dolby 'B' system, and it easily accommodates any expanded program from the 118.

### When to Use Expansion

The 118's expansion circuitry can actually reduce the noise present in an original source, while it restores lost dynamic range. For example, you can expand a compressed FM broadcast and increase its dynamic range to the point where it is comparable to your best conventional phonograph discs and, at the same time, significantly reduce the FM hiss. Or you can expand a conventional disc and dramatically reduce surface noise and turntable rumble at the same time you increase its dynamics to restore sound to near that of a live performance.

### How to Use the 118 as an Expander (Linear Expansion)

1. With your system's Master Volume control turned down low, switch the 118's ABOVE THRESHOLD/LINEAR switch to the LINEAR position.
2. Set the COMPRESS/EXPAND control to the desired expansion ratio (on the right hand side of the control scale). An expansion ratio of 1.2 creates a 20% increase in dynamic range, 1.4 creates a 40% increase, 1.6 a 60% increase, and so forth . . . up to 2.0, which creates a 100% increase in dynamic range.

The amount of expansion is continuously adjustable from 1.0 (a 1:1 ratio for no effect) to 2.0 (a 1:2 ratio for maximum effect), for two reasons. First, different listeners desire varied degrees of expansion to suit their musical tastes. Second, various program sources have used different amounts of compression when recorded, and will therefore require different amounts of dbx expansion to achieve satisfactory results when played back. Most users prefer to set the COMPRESS/EXPAND control between 1.2 and 1.5 (an expansion ratio of 1:1.2 to 1:1.5) depending on the recorded material. Too much expansion often results in the audible variation of background noise that is usually



described as "breathing" or "pumping." One way to adjust the control is to increase the amount of expansion until it sounds extreme, then back off a little.

3. Slowly increase your system's Master Volume. Set the THRESHOLD control to a point where the BELOW THRESHOLD indicator (amber) is illuminated during quiet passages, and the ABOVE THRESHOLD indicator (red) is illuminated during loud passages. The setting is not critical. If the control is set high (counterclockwise), the program volume may decrease somewhat; if the control is set low (clockwise), the program volume may increase somewhat. The numbers around the THRESHOLD control refer to the actual input voltage (the "threshold") at which the 118 decides whether to expand upward (increase the volume) or expand downward (decrease the volume and the noise). When the BELOW THRESHOLD LED is illuminated, the 118 is decreasing the program level, and when the ABOVE THRESHOLD LED is illuminated, the 118 is increasing the program level.

#### **When to Use Compression**

Use compression for any situation where the average level must be relatively constant, for instance, when playing or recording background music. At the low listening levels of background music, quieter passages must be kept loud enough to be heard, yet loud passages must not be too loud. Similarly, if you are recording music for playback in your automobile or any other high noise area, compression can keep the quiet sounds above the road noise, while it holds the loud sounds below distortion-causing levels. Compression is also useful for a live recording of several people with different voice levels. In this case, the compressor acts like an automatic volume control and saves you the task of "gain riding."

#### **How to Use the 118 as a Compressor (Linear Compression)**

1. With your system's Master Volume set for low volume levels, switch the ABOVE THRESHOLD/LINEAR switch to the LINEAR position.

2. Set the COMPRESS/EXPAND control to the desired compression ratio on the left hand side of the control (see Figure 12C and Page 6). Note that the control is continuously adjustable from 1.0 (a 1:1 ratio for no effect) to infinity (an infinity to one ratio for extreme compression). As with the expansion side of the control, a good way to adjust the control would be to turn it toward the infinity setting until the effect seems extreme, then back off a little to adjust for the desired sound.

3. Slowly turn your system's Master Volume control back up to a desirable listening level. Set the THRESHOLD control to a point where the BELOW THRESHOLD indicator (amber) illuminates during quiet passages, and the ABOVE THRESHOLD indicator (red) illuminates during loud passages. The numbers around the THRESHOLD control refer to the actual input voltage, the threshold at which the 118 decides whether to compress downward (decrease the volume) or compress upward (increase the volume). Refer to the glossary for a further definition of "threshold."

If the THRESHOLD control is set high (counterclockwise), the program volume may decrease somewhat; if the control is set low (clockwise), the program volume may increase somewhat. The amount of compression that takes place is not affected by the setting of the THRESHOLD control, and adjustment is non-critical. Now adjust the COMPRESS/EXPAND control for the desired amount of compression.

118 increases gain only above the threshold point and does not act on levels below the threshold point.

**Peak Unlimiter**

A peak unlimiter is an expander with a high expansion ratio such as 1:1.4 to 1:1.6 (settings of 1.4 to 1.6 on the COMPRESS/EXPAND control) which is used to counteract the effects of peak limiting described earlier. Although the 118 works best with moderate expansion ratios while in the LINEAR mode, *HIGH EXPANSION RATIOS WOULD NORMALLY BE USED TO PROVIDE PEAK UNLIMITING IN THE ABOVE THRESHOLD MODE.*

**How the 118's Compression/Expansion Circuits Work**

An expander or compressor is made by combining two devices. One device changes the level of the audio (by changing its own gain), and the other senses the audio level to control the gain change device. The 118 uses VCA's (voltage controlled amplifiers) as the level changing devices. The 118's VCA gain is linearly proportional, in dB, to the controlling voltage from the level sensing device. The 118's level sensing devices respond to the RMS (root-mean-square) value of the sum of the stereo input signals, converting that RMS value to a DC (direct current) control voltage, which changes in proportion to the dB change of input signal. Depending on the setting of the COMPRESS/EXPAND control (i.e. compress or expand) the polarity of the control voltage is reversed (to plus or minus), and the fraction of a volt per dB of level change is varied. The control voltage from the level sensing device is fed to the VCA where it modifies the audio level appearing at the 118's outputs.

RMS level sensing is used, rather than peak or simple average sensing, for several reasons. First, the human ear and brain perceive loudness in a way that more closely follows the RMS value than any other value. Thus, when the

118 expands or compresses, it does so on the same basis the human ear uses to determine loudness. In contrast, a peak sensing system would respond to short term transients that do not contribute to loudness and can lead to audible distortion. RMS level sensing results in very natural changes in level. dbx uses their own, patented voltage controlled amplifiers and RMS detectors that work over a wider dynamic range and have lower noise than any other such devices available.

A particularly interesting feature of linear decibel compression and expansion is that it works evenly over the entire dynamic range at all times, regardless of level. This means that the gain changes are very gradual, spread evenly over the dynamic spectrum. Competitive systems which concentrate gain changes only at the louder or quieter ends of the dynamic range must, necessarily, cram a lot of gain change into a small area. Obviously natural sounding audio quality cannot benefit from such processing. Furthermore, the use of a threshold type system necessitates tricky compression/expansion adjustment that would require repeated attention with almost every new program. The linearity of the dbx system means such matching is unnecessary.

The peak unlimiting capability (or peak limiting capability) of the 118 is a special purpose feature. When the 118 is in the ABOVE THRESHOLD mode, it operates by "ignoring" all signals below the preset THRESHOLD level (set by the THRESHOLD control). Above that level, the compression or expansion works just the same as when the 118 is in its LINEAR mode.

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## SIMPLE PRECAUTIONS WHEN USING ANY PROGRAM EXPANDER

22

The 118 (or any expander) places greater demands on your power amplifier and speakers. Whether or not a given amplifier is of adequate power rating is not always easy to determine; it depends on the sensitivity of the speakers, the distortion characteristics of the amplifier and the levels at which music is being played.

Set for 1.5 expansion (1:1.5), the 118 will expand a good 60dB classical recording to about 90dB of dynamic range. Full realization of the benefit of this dynamic range when playing at loud levels requires both a hefty power amplifier and speakers that can take the high power. If you have such equipment, the results will be breathtaking. Fortunately, such components are not mandatory for enjoyment of the 118 at moderate listening levels.

The most important point is this; if the speakers and

amplifier cannot handle wide dynamic range, and if the expander "tries" to drive them to a wide dynamic range, excessive clipping distortion (overdrive) may occur. To avoid this unpleasant effect, use good speakers and a reasonably large amplifier. If distortion still occurs, it will probably be noticed only with programs that have a good dynamic range to begin with, and which do not need large amounts of expansion. In such cases, a reduction in the expansion ratio setting or lowering of the listening level will avoid distortion. A good expander is a powerful tool, and, as with any tool, it can be used to excess. Used properly, the expander can turn an old record collection into a treasure of new listening enjoyment, and it can turn a boring selection of compressed and limited FM broadcasts into an exciting new source of listening pleasure.



## SPECIFICATIONS

<b>COMPRESSION</b>	Continuously adjustable from 1.0 (1:1) to infinity (infinity:1).
<b>EXPANSION</b>	Continuously adjustable from 1.0 (1:1) to 2.0 (1:2).
<b>INPUT NOISE</b>	90dB below 1 volt
<b>INPUT LEVEL</b>	30 volts RMS, maximum
<b>INPUT IMPEDANCE</b>	50k-ohms
<b>MAXIMUM OUTPUT LEVEL</b>	7 volts RMS at 1kHz into a 5k-ohm load
<b>LEVEL ADJUSTMENT</b>	Adjustable for unity gain for signal levels of 10 millivolts to 2 volts.
<b>FREQUENCY RESPONSE</b>	±0.5dB 20Hz to 20kHz
<b>ATTACK TIME</b>	12 milliseconds typical; faster for rapidly rising signal level.
<b>RELEASE RATE*</b>	Follows signal envelope decrement up to a rate of 180dB per second. Attack and release times vary with the rise and decay of a musical signal. Time responses are optimized to simulate the time response of the human ear.
<b>TOTAL HARMONIC DISTORTION</b>	0.1% typical, 20Hz to 20kHz, at COMPRESS/EXPAND setting of 1.0.
<b>POWER REQUIREMENTS</b>	117 volts AC; 50 or 60Hz. (240 volt operation available as an option.)
<b>POWER CONSUMPTION</b>	5 watts
<b>DIMENSIONS</b>	7¼" wide (184.2 mm); 10¼" deep (260.4 mm); 3¾" high (95.25 mm).
<b>WEIGHT</b>	5 lbs. (2.3 kg)

*\*Attack and release times automatically vary with rate of change of program level (attack and release times "track" the signal envelope).*

Specifications subject to change with notice or obligation.

## dbx PRODUCT WARRANTY

24

All dbx products are covered by a Limited Warranty. Consult your warranty card or local dealer for details.

## FACTORY SERVICE

The dbx Customer Service Department is prepared to give additional assistance in the use of this product. All questions regarding interfacing dbx equipment with your system, service information or information on special applications will be answered. You may call during normal business hours — Telephone: 617-964-3210 or write to:

**dbx, Inc.**  
**71 Chapel Street**  
**Newton, MA 02195**  
**Attn: Customer Service Department**

Should it become necessary to have your equipment factory serviced;

1. Please repack the unit including a note describing the problem along with the day, month and year of purchase.

2. Send the unit freight prepaid to:

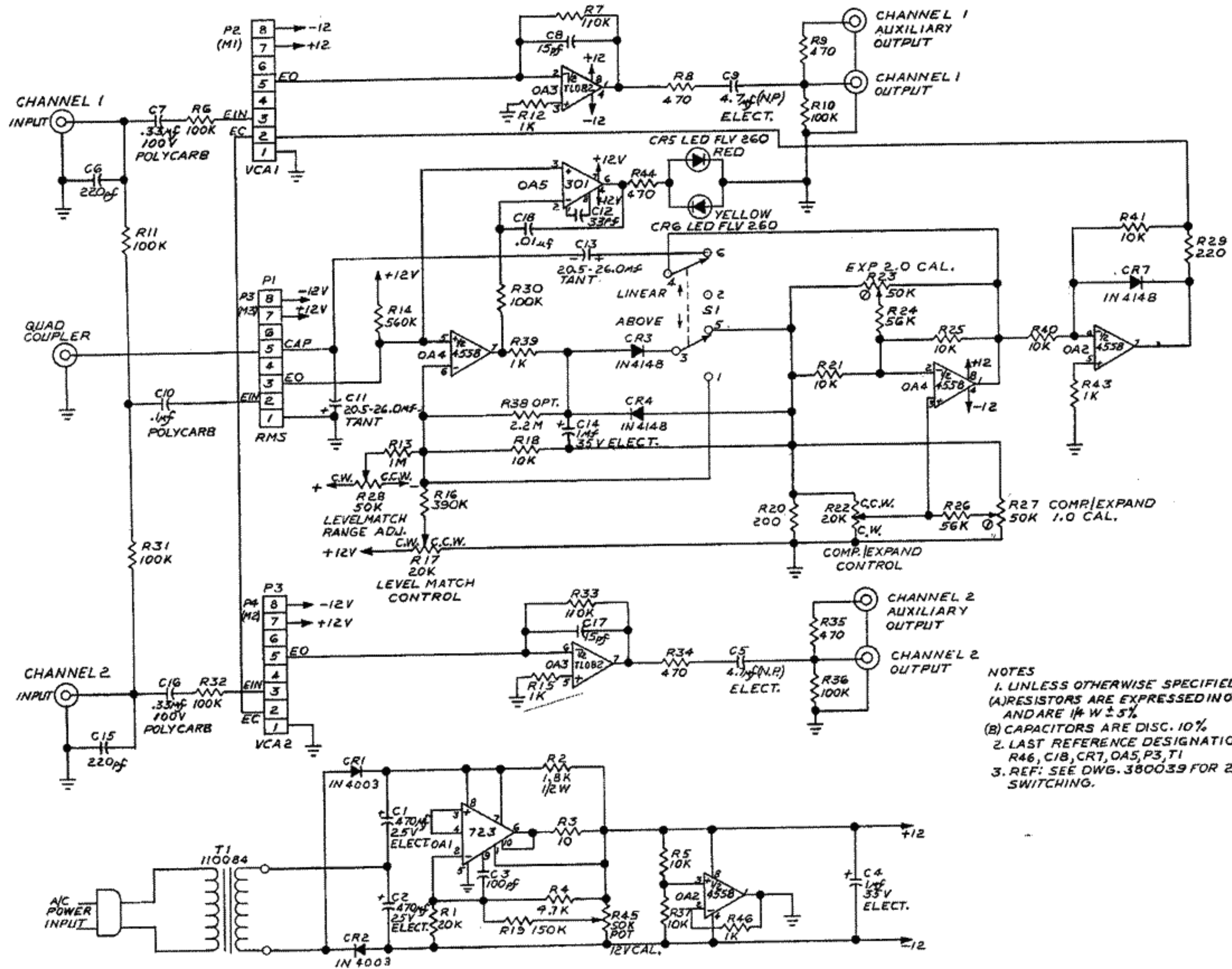
**dbx, Inc.**  
**224 Calvary Street**  
**Waltham, MA 02154**  
**Attn: Repair Department**

3. We recommend that you insure the package and send it via United Parcel Service wherever possible.

4. Please direct all inquiries to the dbx Customer Service Department.

Outside the United States — contact your nearest dbx dealer for the name of an authorized repair center.

# SCHEMATIC



- NOTES
1. UNLESS OTHERWISE SPECIFIED (A) RESISTORS ARE EXPRESSED IN OHMS AND ARE 1% W ± 5% (B) CAPACITORS ARE DISC. 10%
  2. LAST REFERENCE DESIGNATION R46, C18, CR7, OA5, P3, T1
  3. REF: SEE DWG. 380039 FOR 240V SWITCHING.

## GLOSSARY

### Asperity Noise

This is a swishing type of background noise that occurs with tape recordings in the presence of strong low frequency signals, especially when there are no high frequency signals to mask the hiss. Asperity noise is caused by minute imperfections in the surface of the tape, including variations in the magnetic particle size in the tape's oxide coating. The imperfections increase or decrease the strength of the magnetic field passing the play head in a random manner, resulting in audible noise. Asperity noise may be present even when no program is recorded. When a program is recorded, asperity noise becomes superimposed on the signal, creating modulated asperity noise, or "modulation noise." Using high-quality tape with a calendered surface helps reduce asperity and modulation noise (calendered tape is pressed smooth by high-pressure rollers).

### Attack Time

Attack time may mean different things, depending on the context. In music, the time it takes for a note to reach its full volume is the attack time of the note. Percussive instruments have short attack times (reach maximum volume quickly) and wind instruments have long attack times (reach maximum volume more gradually).

When a compressor (or expander) changes the level of an incoming signal, the circuitry actually requires a finite amount of time to complete that change. This time is known as the attack time. More precisely, the attack time is the interval (usually measured in milliseconds or microseconds) during which the compressing or expanding amplifier changes its gain from the initial value to 90% of the final value.

### Aux Input (Aux Level)

Aux inputs, an abbreviation for auxiliary inputs, are low sensitivity jacks provided on most hi-fi and semi-professional equipment. Aux inputs (also known as "aux level" or "line level" inputs) have "flat" frequency response and are intended to be used with preamplified signals. Aux-level (line-level) signals are medium-level, higher than microphone levels, but not enough power to drive a speaker. The advantage to these levels is that they are less susceptible to hum and noise than are microphone levels. Typical items which might be connected to aux inputs are tape machine "play" outputs, tuner outputs, and dbx "play" outputs. Mic-level or phono-level signals are considerably lower in level than aux inputs (approx. -60 to -40dBV), so they will not produce adequate volume when connected to an aux input. Moreover, phono cartridge outputs require RIAA equalization which is not provided by aux inputs.

### Bandwidth

Bandwidth refers to the "space" between two specific frequencies which are upper and lower limits; alternately, bandwidth refers to the absolute value of the range of frequencies between those limits. Thus, a filter which passes frequencies from 1,000Hz to 10,000Hz may be said to have a bandwidth of 1kHz-10kHz, or it may be said to have a 9kHz bandwidth (10kHz minus 1kHz equals 9kHz).

Bandwidth is not necessarily the same as frequency response. Bandwidth may be measured at low levels, and frequency response at higher levels. Moreover, bandwidth may refer only to certain portions of the circuitry within a piece of equipment, whereas frequency response may refer to the overall performance of the equipment. Thus, while the overall input-to-output frequency response of dbx type II equipment is 20Hz to 20kHz, the bandwidth of the RMS detection circuitry within that equipment is 30Hz to 10kHz.

### Bass

The low audio frequency range below approximately 500Hz. For the purpose of discussion or analysis, the bass range may be further divided into upper bass (250 to 500Hz), mid bass (100-200Hz), low bass (50-100Hz), and ultra-low bass (20-50Hz).

### Bass Boost

An accentuation of the lower audio frequencies (bass frequencies), whereby they are made louder than other frequencies.

### Bi-amplified

Descriptive of a sound system which utilizes a low level cross-over network to divide the full-spectrum audio signal into low and high frequency ranges. These ranges are then fed to separate power amplifiers, which in turn feed low frequency speakers (woofers) and high frequency speakers (tweeters).

### Bias

Bias, as the term is used in tape recording, is a very high frequency signal (usually over 100kHz) that is mixed with the program being recorded in order to achieve linear magnetization of the tape. If only the audio program were applied to the recording head, a very distorted recording would result because lower-energy portions of the program would not be able to overcome the initial magnetization threshold of the tape (known as hysteresis).

The frequency of the bias signal is not critical, so long as the record and erase bias are synchronized. However, the bias:



energy level has a direct effect on the recorded level, background noise, and the distortion. It is sometimes necessary to reset the bias level for optimum performance with different types of recording tape, and professional tape machines are equipped with continuously variable bias controls; many consumer tape machines are now equipped with bias selector switches.

### **Clipping**

Clipping is a very distorted sound. It occurs when the output capabilities of an amplifier are exceeded, and the amp can no longer produce any more voltage, regardless of how much additional gain or how much more input signal is present. Clipping is relatively easy to see on an oscilloscope, and it is sometimes audible as an increase in harmonic distortion. In severe cases of clipping (hard clipping), sine-waves begin to resemble square waves, and the sound quality is very poor. Often, the maximum output level of an amplifier is defined as that level where clipping begins to occur. There is a phenomenon known as input clipping, and this may occur where the input signal is so high in level that it exceeds the level-handling ability of the transformer and/or of the input amplifier. Clipping also occurs when tape is saturated by excessive record levels.

So-called "soft clipping" is usually the result of transformer saturation, and it may be somewhat less objectionable than the "hard clipping" that occurs when output voltage limits are reached. Aside from degrading the sound quality, clipping can damage loudspeakers. Output clipping may be avoided by reducing the level of the input signal, reducing the gain of the amplifier, or using a larger amplifier. Input clipping may be avoided by reducing the level of the incoming signal, and then increasing the gain of the amplifier.

### **Clipping Level**

This is the signal level at which clipping just begins to occur. Clipping level is not always easy to define. It may be a matter of visually judging the waveform on an oscilloscope as the level is increased; alternately, clipping level may be defined as the level at which harmonic distortion reaches a given value. Tape clipping, or saturation, is defined as the 3% harmonic distortion level.

### **Compression**

Compression is a process whereby the dynamic range of program material is reduced. In other words, the difference between the lowest and highest audio levels is "squeezed" into a smaller dynamic range. A compressed signal has higher average level, and therefore may have more apparent loudness than an uncompressed signal, even though the peaks are no higher in level. Compression is

achieved with a compressor, a special type of amplifier that decreases its gain as the level of the input signal increases. The amount of compression is expressed as a ratio of the input dynamic range to the output dynamic range; thus, a compressor that takes a program input with 100dB of dynamic range and yields an output program of 50dB dynamic range may be said to have a 2:1 compression ratio.

### **Compressor**

A compressor is an amplifier that decreases its gain as the level of the input signal increases to reduce the dynamic range of the program (see "compression"). A compressor may operate over the entire range of input levels, or it may operate only on signals above and/or below a given level (the threshold level).

### **Crossover Frequency**

In loudspeaker systems and multi-amplifier audio systems, the transition frequency (actually a frequency range) between bass and midrange or midrange and treble speakers or amplifiers.

### **Crossover Network**

A circuit which divides the audio spectrum into two or more frequency bands for distribution to different speakers (high level crossover) or different amplifiers which then feed different speakers (low level crossover).

High level crossovers are usually built into the speaker cabinet, and are passive (they require no power supply). Low level crossovers are used in bi-amplified or tri-amplified sound systems. They are usually self-contained, and come before the power amplifiers. Low level crossovers may be passive or active; active low level crossovers are known as "electronic crossovers."

### **Damping Factor**

The ratio of loudspeaker impedance to the amplifier's output source impedance. Damping describes the amplifier's ability to prevent unwanted, residual speaker movement. The higher the numerical value, the better the damping.

### **DB (Decibel) also, dBv dBV dB SPL dBm dB**

One dB is the smallest change in loudness the average human ear can detect. 0dB SPL is the threshold of human hearing whereas the threshold of pain is between 120 and 130dB SPL. The term dB is an abbreviation for decibel, or 1/10 of a Bel. The decibel is a ratio, not an absolute number, and is used to express the difference between two power, voltage or sound pressure levels. (dB is 10 times the logarithm of a power ratio or 20 times the logarithm of a voltage



or sound pressure ratio.) If the number of "dB's" are referenced to a given level, then the value of the dB number becomes specific.

**dBv** expresses a voltage ratio. 0dBv is usually referenced to 0.775V rms. Thus, 0dBv=0.775V, +6dBv=1.55V (twice 0dBv), +20dBv=7.75V (ten times 0dBv), etc.

**dBV** expresses a voltage ratio and is similar to dBv, but 0dBV is usually referenced to 1V rms. Thus, 0dBV is 2.2dB higher than 0dBv.

**dB SPL** expresses a Sound Pressure Level ratio. dB SPL is a measure of acoustic pressure (loudness), not acoustic power, which would be measured in acoustic watts. 0dB SPL is equal to 0.0002 dynes/square centimeter (the threshold of human hearing at 1kHz). As with dBV, an increase of 6dB SPL is twice the sound pressure, and an increase of 20dB SPL is an increase of 10 times the sound pressure.

**dBm** expresses a power ratio. 0dBm is 1 milliwatt (.001 watts), or 0.775V rms delivered to a 600-ohm load. +3 dBm=2 milliwatts, or 1.55V into 600 ohms (twice 0dBm), +10dBm=10 milliwatts, or 7.75V into 600 ohms (ten times 0dBm), etc. dBV and dBm are numerically equal when dealing with 600-ohm circuits. However, when the impedance is other than 600 ohms, the value of dBV remains the same if the voltage is the same, whereas the value of dBm decreases with increasing impedance.

**dB** alone, without any suffix, doesn't mean anything unless it is associated with a reference. It may express the difference between two levels. Thus, the difference between 10dBV and 15dBV, the difference between 0dBm and 5dBm, and the difference between 90dB SPL and 95dB SPL are all differences of 5dB.

#### **Decay Time**

Decay time may mean different things, depending on the context. A compressor's decay time is also known as its release time or recovery time. After a compressor (or expander) changes its gain to accommodate an incoming signal, and the signal is then removed, the decay time is the amount of time required for the circuitry to return to "normal." More precisely, the decay time is the interval (usually measured in microseconds) during which the compressing or expanding amplifier returns to 90% of the normal gain. Very fast decay times can cause "pumping" or

"breathing" effects, whereas very slow decay times may cause moderate-level program which follows high-level program or program peaks to be too low in level.

#### **Decoder**

When a circuit restores an original program from a specially treated version of that program, the circuit may be said to decode the program. The equipment or circuit which performs this function is known as a decoder. Decoders must be used only with programs which have been encoded by complementary encoding circuitry. Typical decoders include: FM tuners that use multiplex decoders to extract left and right stereo signals from left-plus-right and left-minus-right signals, matrix quadraphonic decoders that extract four channels of program from the stereo program on encoded recordings, and dbx decoders that retrieve wide-dynamic range programs from the compressed programs on dbx-encoded recordings.

#### **De-emphasis & Pre-emphasis**

De-emphasis and pre-emphasis are related processes that are usually done to avoid audio noise in some storage or transmission medium. Pre-emphasis is a boost at specific higher frequencies, the encoding part of an encoding/decoding system. De-emphasis is an attenuation at the same frequencies, a reciprocal decoding that counteracts the pre-emphasis. In dbx noise reduction, de-emphasis is performed by the decoder (the play circuitry). The de-emphasis attenuates high frequencies, thereby reducing tape modulation noise and restoring the original frequency response of the program before it was dbx encoded. There are other types of pre-emphasis and de-emphasis. For example, in FM tuners, de-emphasis is used to compensate for special equalization (known as 75-microsecond pre-emphasis) applied at the station's transmitter.

#### **Dynamic Range**

The dynamic range of a program is the range of signal levels from the lowest to the highest level. In equipment, the dynamic range is the "space," in dB, between the residual noise level and the maximum undistorted signal level. A program with wide dynamic range has a large variation from the softest to the loudest passages, and will tend to be more lifelike than programs with narrow dynamic range.

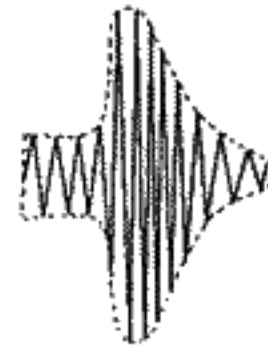
#### **Encoder**

When a circuit processes an original program to create a specially treated version of that program, the circuit may be said

to encode the program. The equipment or circuit which performs this function is known as an encoder. Encoded programs must be decoded only with complementary decoding circuitry. Typical encoded programs include: FM multiplex broadcasts, matrix quadraphonic recordings, and dbx encoded recordings.

### Envelope

In music, the envelope of a note describes the change in average signal level from initial attack, to peak level, to decay time, to sustain, to release time. In other words, the envelope describes the level of the note as a function of time. Envelope does not refer to frequency.



The outline is the envelope,  
the signal is within the envelope.

In fact, any audio signal may be said to have an envelope. While all audio frequencies rise and fall in instantaneous level from 40 to 40,000 times per second, an envelope may take many milliseconds, seconds or even minutes to rise and fall. In dbx processing, the envelope is what "cues" the rms level detection circuitry to compress and expand the signal; the peak or average level of individual cycles of a note would be useless for level detection because the gain would change much too rapidly for audibly pleasing sound reproduction.

### EQ (Equalization)

EQ or equalization, is an intentional change in the frequency response of a circuit. EQ may be used for boosting (increasing) or cutting (decreasing) the relative level of a portion of the audible spectrum. Some EQ is used for achieving sound to suit personal listening tastes, while other types of EQ are specifically designed to correct for non-linearities in the system; these corrective EQ "curves" include tape (NAB or CCIR) equalization, and phonograph (RIAA) equalization. In a sense, the pre-emphasis and de-emphasis used in dbx processing are special forms of equalization.

There are two common types of Equalization curves (characteristics): PEAKING and SHELIVING. Shelving EQ is used in most Hi-Fi bass and treble tone controls. Peaking EQ is used in Hi-Fi midrange tone controls, in graphic equalizers, and many types of professional sound mixing equipment.

EQ is performed by an equalizer, which may be a specially built piece of equipment, or it may be no more than the tone control section of an amplifier. Graphic equalizers have many controls, each affecting one octave, one-half octave, or one-third octave of

the audio spectrum. (An octave is the interval between a given tone and its repetition eight tones above or below on the musical scale; a note which is an octave higher than another note is twice the frequency of the first note.)

### Expander

An expander is an amplifier that increases its gain as the level of the input signal increases, a characteristic that "stretches" the dynamic range of the program (see "expansion"). An expander may operate over the entire range of input levels, or it may operate only on signals above and/or below a given level (the threshold level).

### Expansion

Expansion is a process whereby the dynamic range of program material is increased. In other words, the difference between the lowest and highest audio levels is "stretched" into a wider dynamic range. Expansion is sometimes used to restore dynamic range that has been lost through compression or limiting done in the original recording or broadcast; expansion is an integral part of compander-type noise reduction systems, including dbx. Expansion is achieved with an expander, a special type of amplifier that increases its gain as the level of the input signal increases. The amount of expansion is expressed as a ratio of the input dynamic range to the output dynamic range; thus, an expander that takes a program input with 50dB of dynamic range and yields an output program of 100dB dynamic range may be said to have a 1:2 compression ratio.

### Fundamental

A musical note is usually comprised of a basic frequency, plus one or more whole-number multiples of that frequency. The basic frequency is known as the fundamental, and the multiples are known as harmonics or overtones. A pure tone would consist of only the fundamental.

### Ground Compensated Output

This is a sophisticated output circuit that senses the potential difference between the ground of the dbx unit and the shield ground of unbalanced inputs to which the dbx unit is connected. Ideally, the dbx unit and the input of the following device should be at the same level (potential). However, where grounding is not "right" (where so-called "ground loops" exist), this circuit calculates the ground error and adds a correction signal to the high side of the output, thereby cancelling much of the hum, buzz and noise that might otherwise have been introduced by ground loops.



### Harmonic Distortion

Harmonic distortion consists of signal components appearing at the output of an amplifier or other circuit that were not present in the input signal, and that are whole-number multiples (harmonics) of the input signal. For example, an amplifier given a pure sine-wave input at 100Hz may produce 200Hz, 300Hz, 400Hz, 500Hz, 600Hz and even 700Hz energy, plus 100Hz, at its output (these being the 2nd, 3rd, 4th, 5th, 6th and 7th order harmonics). Usually, only the first few harmonics are significant, and even-order harmonics (i.e. 2nd and 4th) are less objectionable than odd-order harmonics (i.e. 3rd and 5th); higher harmonics may be negligible in comparison to the fundamental (100Hz) output. Therefore, rather than specifying the level of each harmonic component, this distortion is usually expressed as T.H.D. or Total Harmonic Distortion. While T.H.D. is the total power of all harmonics generated by the circuitry, expressed as a percentage of the total output power, the "mixture" of different harmonics may vary in different equipment with the same T.H.D. rating.

### Harmonics

Overtones which are integral multiples of the fundamental.

### Headroom

Headroom refers to the "space," usually expressed in dB, between the nominal operating signal level and the maximum signal level. The input headroom of a circuit that is meant to accept nominal -10dB levels, but can accept up to +18dB without overdrive or excessive distortion, is 28dB (from -10 to +18 equals 28dB). Similarly, the output headroom of a circuit that is meant to supply nominal +4dBm drive levels, but that can produce +24dBm before clipping is 20dB. A circuit that lacks adequate headroom is more likely to distort by clipping transient peaks, since these peaks can be 10 to 20dB above nominal operating signal levels.

### I.M. (Intermodulation Distortion)

Intermodulation distortion consists of signal components appearing at the output of an amplifier or other circuit that were not present in the input signal, that are not harmonically related to the input, and that are the result of interaction between two or more input frequencies. I.M. distortion, like harmonic distortion, is usually rated as a percentage of the total output power of the device. While some types of harmonic distortion are musical, and not particularly objectionable, most I.M. distortion is unpleasant to the ear.

### Impulse Response

Related to the rise time of a circuit, the impulse response is a measurement of the ability of a circuit to respond to sharp sounds, such as percussion instruments or plucked strings. A circuit with good impulse response would tend to have good transient response.

### Level Match

The dbx noise reduction system is unlike competitive systems in that there is no one threshold at which compression or expansion begins. Instead, compression occurs linearly, with respect to decibels, over the full dynamic range of the program. By necessity, there is an arbitrary signal level which passes through the encoder and decoder without being changed in level. This level is known as the level match point (transition point). Some dbx equipment provides for user adjustment of the level match point, for monitoring purposes only. Although this is not necessary for proper encode/decode performance, by setting the level match point to be approximately equal to the nominal (average) signal level, there will be no increase or decrease in level as you switch from monitoring "live" program to monitoring dbx-processed program.

### Limiter

A limiter is a type of compressor, one with a 10:1 or greater compression ratio. A limiter with a high compression ratio (120:1) can be set so that no amount of increase in the input signal will be able to raise the output level beyond a preset value. The difference between limiting and compression is that compression gently "shrinks" dynamic range, whereas limiting is a way to place a fixed "ceiling" on maximum level, without changing the dynamic range of program below that "ceiling," or threshold.

### Line Level (Line Input)

Line level refers to a preamplified audio signal, in contrast to mic level, which describes a lower-level audio signal. The actual signal levels vary. Generally, mic level is nominally -50dBm (with typical dynamic range of -64dBm to +10dBm). Line level signals vary, depending on the audio system. Hi-Fi line levels are nominally -15dBV, whereas professional line levels are nominally +4dBm or +8dBm (with typical dynamics ranging from -50dBm to +24dBm).

Line inputs are simply inputs that have sensitivities intended for line level (preamplified) signals. Often, the nominal impedance of a line level input will be different than the nominal impedance of a mic level input.

**Modulation Noise**

Modulation noise is a swishing type of background hiss that occurs with tape recordings in the presence of strong low frequency signals. The noise depends on the level of the recorded signal; the higher the recorded signal level, the higher the modulation noise. Modulation noise has typically been "masked," hidden by the dominant signal and/or by the background hiss of the tape. However, when the background hiss is removed, as with dbx processing, modulation noise could become audible. This would happen primarily with strong, low-frequency signals, but in fact it is minimized by dbx's pre-emphasis and de-emphasis.

**Octave**

In music or audio, an interval between two frequencies having a ratio of 2:1.

**Overshoot**

When a compressor or expander changes its gain in response to a fast increase or decrease in level, the maximum gain change should be directly proportional to the actual signal level. However, in some compressors the level detection and gain changing circuitry develop a kind of "inertia," over-reacting to changes in level, increasing or decreasing the gain more than the fixed ratio asked for. This over-reaction is known as overshoot, and it can cause audibly non-linear compression (distortion). dbx circuits have minimal overshoot, so they provide highly linear compression and expansion.

**Peak Level**

An audio signal continuously varies in level (strength, or maximum voltage) over any period of time, but at any instant, the level may be higher or lower than the average. The maximum instantaneous value reached by a signal is its peak level (see RMS level).

**Phase Shift**

"Time shift" is another way to describe phase shift. Some circuitry, such as record electronics and heads, will delay some frequencies of an audio program with respect to other portions of the same program. In other words, phase shift increases or decreases the delay time as the frequency increases. On an absolute basis, phase shift cannot be heard, but when two signals are compared to one another, one having a phase shift relative to the other, the effects can be very noticeable, and not very desirable. Excessive phase shift can give a tunnel-like quality to the sound. Phase shift also can degrade the performance of compander type noise

reduction systems which depend on peak or average level detection circuitry.

**Power Amplifier**

A unit that takes a medium-level signal (e.g., from a pre-amplifier) and amplifies it so it can drive a loudspeaker. Power amplifiers can operate into very low impedance loads (4-16 ohms), whereas preamplifiers operate only into low impedance (600 ohms) or high impedance (5,000 ohms or higher) loads. Also known as a main amplifier, the power amplifier may be built into an integrated amplifier or a receiver.

**Preamplifier**

A device which takes a small signal (e.g., from a microphone, record player), or a medium-level signal (e.g., from a tuner or tape recorder), and amplifies it or routes it so it can drive a power amplifier. Most preamplifiers incorporate tone and volume controls. A preamp may be a separate component, or part of an integrated amplifier or of a receiver.

**Pre-Emphasis** (See "de-emphasis")

**Receiver**

A single unit that combines tuner, preamp and power amplifier sections.

**Release Time or Release Rate** (See "decay time" and "attack time")

**Rise Time (Attack Time)**

This is the ability of a circuit to follow (or "track") a sudden increase in signal level. The shorter the rise time, the better the frequency response. Rise time is usually specified as the interval (in microseconds) required to respond to the leading edge of a square-wave input.

**RMS Level**

RMS level (Root Mean Square) is a measurement obtained by mathematically squaring all the instantaneous voltages along the waveform, adding the squared values together, and taking the square root of that number. For simple sine waves, the RMS value is approximately 0.707 times the peak value, but for complex audio signals, RMS value is more difficult to calculate. RMS level is similar to average level, although not identical (Average level is a slower measurement).



**Sub Harmonic**

A sub-multiple of the fundamental frequency. For example, a wave the frequency of which is half the fundamental frequency of another wave is called the second sub harmonic of that wave.

**Sub Woofer**

A loudspeaker made specifically to reproduce the lowest of audio frequencies, usually between 20Hz and 100Hz.

**Synthesizer**

An ELECTRONIC MUSIC SYNTHESIZER is an audio processor that has a built-in sound generator (oscillator), and that alters the envelope of the sound with voltage controlled circuitry. Synthesizers can produce familiar sounds and serve as musical instruments, or they can create many unique sounds and effects of their own.

A SUB HARMONIC SYNTHESIZER is a device which is not used to create music, but to enhance an existing audio program. In the case of the dbx Model 100, the unit creates a new signal that corresponds to the volume of the input signal, but is at 1/2 the frequency of the input signal.

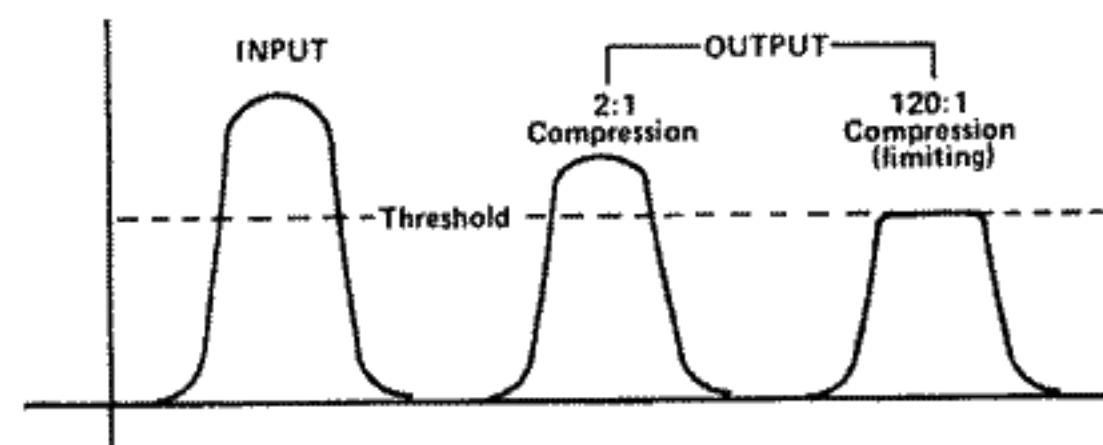
**Tape Saturation**

There is a maximum amount of energy that can be recorded on any given type of magnetic tape. When a recorder "tries" to record more energy, the signals become distorted, but are not recorded at any higher levels. This phenomenon is called tape saturation because the magnetic oxide particles of the tape are literally saturated with energy and cannot accept any more magnetization.

**T.H.D. (Total Harmonic Distortion) (See "Harmonic Distortion")****Threshold**

Threshold is the level at which a compressor or limiter ceases to have linear gain, and begins to perform its gain-changing function (i.e., where the output level no longer rises and falls in direct proportion to the input level). In most systems, the threshold is a point above which the level changes, although there are compressors that raise signal levels below a threshold point. Some compander-type noise reduction systems, such as Dolby<sup>®</sup>\* have upper and lower threshold between which the gain changes; these systems require careful level calibration for proper encode/decode performance. dbx noise reduction systems have no threshold at which compression or expansion factors change, so level calibration is not critical.

\*'Dolby' is a trademark of Dolby<sup>®</sup> Laboratories, Inc.

**Tracking Accuracy**

Tracking refers to the ability of one circuit to "follow" the changes of another circuit. When two volume controls are adjusted in exactly the same way, the corresponding "sameness" of the output levels can be expressed as the tracking accuracy of the controls.

The level detection circuitry in a dbx encoder senses the signal level, changes the gain, and creates an encoded signal. The corresponding "sameness" of the original signal and the encoded/decoded signal can be expressed as the tracking accuracy of the noise reduction system. (dbx systems are non-critical for the operator, and are built to close tolerances, so that tracking accuracy is excellent, even if the encoder and decoder are in different pieces of dbx equipment.)

**Transition Level (See Level Match)**

When a circuit has uniform compression or expansion throughout its full dynamic range, there must be some level which passes through the unit without being raised or lowered (where gain is unity). This unity gain level is the transition level or transition point. The transition point is a "window" 1dB wide, in a dbx encoder (compressor), all signals above the transition point are decreased in level, and all signals below the point are increased in level. Conversely, in a dbx decoder (expander), all signals above the transition point are increased in level, and all signals below the point are decreased in level. The transition level is similar to a "threshold," except it does not refer to a point at which compression or expansion factors change.



**Triamplified**

Similar to biamplified. A sound system where a passive crossover network creates three frequency ranges, and feeds three power amplifiers: one for bass, one for mid, and one for high frequencies. The amplifiers are connected directly to the woofers, midrange drivers and tweeters without a passive, high-level crossover network.

**Tuner**

A unit which receives radio broadcasts and converts them into audio frequency signals. May be part of a receiver.

**VCA (Voltage Controlled Amplifier)**

Traditionally, amplifiers have been designed to increase signal levels (to provide gain). If an amplifier were required to decrease the level (to attenuate), it could become unstable, and might even oscillate. The gain (amount of amplification) in these traditional amplifiers would be adjusted by one of three methods (1) attenuating the audio signal fed to the input of the amplifier, (2) attenuating the audio output of the amplifier, or (3) changing the negative feedback (feeding more or less signal from the output back to the input, but in reversed polarity).

The VCA is a special type of amplifier that can be used to increase or decrease levels over a wide dynamic range. Instead of using signal attenuation or negative feedback, the gain (or loss) is adjusted by means of an external dc control voltage. dbx has a unique, patented VCA design that has extremely low noise and very wide dynamic range; the dbx VCA is the heart of dbx noise reduction equipment.

**Woofers**

A loudspeaker which reproduces only low frequencies.