dbx Model 100 "Boom Box"

sub harmonic synthesizer

INSTRUCTION MANUAL

INDEX

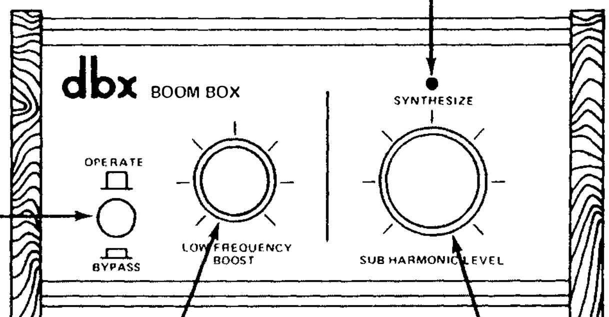
WARNING: TO PREVENT FIRE OR SHOCK HAZARD, DO NOT EXPOSE THIS APPLIANCE TO RAIN OR MOISTURE.

CONTROLS & CONNECTORS

Fig. 1 – Front Panel

SYNTHESIZE INDICATOR
This Light Emitting Diode (L E.D.) flashes whenever the Boom Box input contains a signal that is capable of causing sub harmonic synthesis to occur–it is ON whenever the Boom Box *potentially can create* new, ultra-low bass frequencies. The amount of that synthesized bass *actually added* to the program depends on the setting of the SUB HARMONIC LEVEL control. Normally, the L.E.D. will flash during loud bass notes, drum beats, etc., and it will flicker on quieter bass notes.

OPERATE/BYPASS SWITCH
Pushing this button IN places the unit in BYPASS mode, so the Boom Box has no effect. With the button OUT, the Boom Box is in OPERATE mode so it can enhance bass response by synthesizing and boosting low frequencies. This switch can be used to "A-B" compare the sound with and without Boom Box processing.

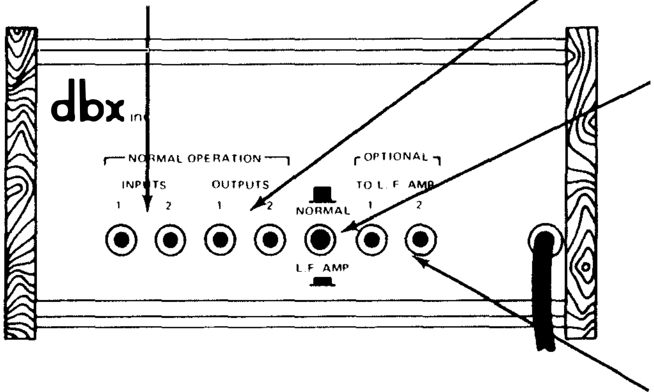


LOW FREQUENCY BOOST CONTROL
This control is a type of equalizer that can be used to "smooth" out the low frequency response of your sound system by filling in the "gap" between the moderately low frequency bass in the original program and the Boom Box's synthesized ultra-low bass. The LOW FREQUENCY BOOST control increases the level of low frequencies something like a BASS tone control, although its major effect is at lower frequencies than the typical bass tone control. With LOW FREQUENCY BOOST set fully counterclockwise, there is no effect; rotating the control clockwise increases the low bass. This control is independent of the SUB HARMONIC LEVEL control.

SUB HARMONIC LEVEL CONTROL
This control sets the amount of synthesized low frequencies (sub harmonics) added by the Boom Box. With the control set fully counter-clockwise, there is no effect; clockwise rotation adds synthesized extremely low bass to the program. The actual effect of this control also depends on the amount of bass present in the program.

Fig. 2 — Rear Panel

INPUT CONNECTORS
Connect the cables from your preamplifier, integrated amplifier or receiver's Tape Outputs (Record Out) to these connectors. (Alternately, connect the cables from your system's Preamp Out connectors, as shown in Figure 4A.)



OUTPUTS
Use these outputs for NORMAL operation: i.e., where the Boom Box is used with a single stereo amplifier. Cables from these outputs should be connected to your preamp, integrated amplifier or receiver's Tape Input (Play In) connectors. (Alternately, connect the cables to your system's Main Amp Input connectors, as shown in Figure 4A.)

When a second speaker system and power amplifier are used to handle the lowest frequencies, cables from the NORMAL OPERATION OUTPUT connectors are usually wired to the Main Amp Input connectors of the integrated amp or receiver.*

NORMAL/L.F. AMP SWITCH
For normal operation this pushbutton is OUT so the Boom Box adds low frequencies to the original input program, and the complete, enhanced program is fed to the NORMAL OPERATION OUTPUT connectors.

The NORMAL/L.F. AMP pushbutton is pressed IN only when a separate power amplifier and speaker system are used to reproduce Low Frequencies.* This causes the synthesized low frequencies, and certain other frequencies that have been boosted by the Boom Box's LOW FREQUENCY BOOST control to be fed to the OPTIONAL TO L.F. AMP connectors. The full-frequency original input program (but no boosted or synthesized lows) feeds the NORMAL OPERATION OUTPUT connectors.

TO L.F. AMP OUTPUTS
These connectors are not normally used. They are provided for "optional" operation when the Boom Box is used with two sets of power amplifiers and speaker systems, one set for only the boosted and/or synthesized low frequencies and one set for the low, mid and high frequencies in the original program.* Cables from the TO L.F. AMP outputs should be connected to the Input of the amplifier which drives the low frequency speaker system, usually the separate power

(continued on next page)

**See footnote on next page.*

amplifier. Both TO L.F. AMP outputs are paralleled and therefore carry the same signal information. This is done because at low bass frequencies the sound is non-directional and therefore does not require stereo separation.

NOTE: When two amplifiers are used, *usually* the more powerful amp is used to drive the low frequency speaker system because low frequency speakers *tend* to be less efficient then higher frequency speakers. The separate power amplifier driving the low frequency speakers need not be of the same quality as the power amplifier which drives the full range speakers. For instance, higher harmonic distortion may not be audible since the low frequency speakers will tend to "ignore" the higher frequency distortion components. In this way, the power amplifier section of a good quality but relatively small integrated amp or receiver (approximately 10 to 50 watts/channel) will be perfectly adequate for driving the full range frequency speakers, while a relatively inexpensive power amplifier (approximately 50 to 100 watts/channel) can be used to drive the low frequency speakers.

For the "ultimate" in bass response, a second amplifier and a special low frequency speaker system can be used to reproduce only the boosted and/or synthesized low frequencies, although such a setup* **is not necessary *for enjoyment of the Boom Box. When a separate L.F. amplifier is used, ideally the Boom Box should be placed between the Preamp Output and the Main Amp Input. This enables the preamplifier (or receiver) VOLUME control to simultaneously increase and decrease the volume level fed to both power amplifiers. If you placed the Boom Box in a Tape Monitor Loop (between the Record Out and Play In connectors) while using a second power amplifier for the low frequencies, you would have to make two perfectly synchronized VOLUME adjustments, one for each amplifier, because the preamp VOLUME would only affect the level in one power amplifier.* *(For details, see page 10.)*

BASS-LIKE YOU'VE NEVER HEARD IT BEFORE!

Have you ever wished you could feel the sound at home the same way you do at a live rock concert or a recording session? Now you can use your own Hi-Fi system to enjoy the "sock" of a live concert—sense the deep, chest-pounding beat of a drum, add amazing richness to low strings. All this is made possible by dbx's unique, patent-applied-for technology which recreates bass that was not recorded or broadcasted. The dbx Model 100, called the "Boom Box," adds new dimensions in bass response to virtually any modern Hi-Fi sound system.

Your Hi-Fi amplifier may be specified to have "flat" response from about 20Hz to 20,000Hz, but the chances are that you've never before heard anything coming out of it much below 40Hz to 50Hz for several reasons:

1. Most Hi-Fi speakers roll off in this region (they "poop out").

2. Little sound is ever recorded or broadcast below 50Hz.

Frequencies below 50Hz are considered extremely low bass, and are often sacrificed intentionally in order to get more time on a record, to get less signal distortion in tape recording and radio equipment, and to maximize speaker efficiency so that smaller amplifiers can be used. The Boom Box works by sensing the mid-bass frequencies that remain in the program, and by using them as a guide to recreate (synthesize) corresponding amounts of ultra-low bass. The Boom Box can also boost the bass existing in the original program in a way that dramatically improves the bottom end response of most speaker systems.

Technically, the Boom Box is known as a signal enhancement processor, and it performs two types of enhancement. It *synthesizes* bass at low frequencies, often well below anything in the recorded or broadcast program. In fact, the synthesized bass is at one half the frequency of the input, hence the bass is literally "twice as deep" (or low). This is done by the Boom Box's SUB HARMONIC SYNTHESIZER circuitry.* The synthesizer circuitry is set up to work only on bass frequencies, so it has no effect on the midrange or treble sound. The Boom Box's second enhancement function resembles the function of an equalizer or bass tone control in that it *boosts* the level of all bass frequencies that are part of the original program.

The big difference is that the Boom Box's LOW FREQUENCY BOOST enables a lot more bass to be obtained without making the music sound "muddy" or ill-defined. (The boost is accomplished with a special type of "peaking curve," rather than the typical "shelving curve" of a tone control.) The L.F. Boost also serves to fill in the "gap" created between the synthesized ultra-low bass and the lower limits of the previously existing bass. You simply can't do what the Boom Box does using any conventional Hi-Fi tone control—or even using a sophisticated equalizer. So hook up your Boom Box, set it up as described in the following pages, and get ready to really *experience* your favorite music for the first time.

The term SYNTHESIZER, as it relates to the Boom Box, bears no relationship to electronic music synthesizers such as those made by "Moog," "ARP," and others. The term "SUB HARMONIC" derives from the fact that almost all musical notes have* **harmonics, *overtones that are twice the frequency of the fundamental note, three times the fundamental, etc. Since the Boom Box creates frequencies at one half that of the fundamental note, these frequencies are known as sub harmonics.*

WIRING THE BOOM BOX INTO A TYPICAL HI-FI SYSTEM

The Boom Box is usually connected between the Record Out & Play In connectors (the Tape Monitor Loop) of your preamp, receiver or integrated preamp/ amplifier.* When a tape recorder is also connected in the Tape Monitor Loop, the Boom Box should be connected between the tape recorder's Play Output and the preamplifier's Play Input. Other connection schemes are also possible, as discussed in the section of this manual titled "USING THE BOOM BOX IN MORE COMPLEX SOUND SYSTEMS."

CAUTION: Make sure that the power is OFF on all equipment when installing the Boom Box. As a precaution, turn down your amplifier VOLUME control prior to switching on the Boom Box for the first time. Excessive bass boost of already bass-heavy programs at high volume levels can damage your speakers.

**To simplify this manual, we may refer to any of these units as a PREAMPLIFIER; in such cases, the text also applies to a receiver or integrated preamp/amplifier. To further define these terms, a PREAMPLIFIER (preamp) is a control unit that usually accepts inputs from a record player (turntable), one or more tape recorders, a tuner (radio) and any other equipment (auxiliary); the preamp has input selector controls plus volume and tone controls. Since the preamp output is "line level," adequate for headphones but not speakers, the preamp output is fed to a POWER AMPLIFIER which can drive speakers. When the power amplifier is contained in the same chassis as the preamp, the combined unit is known as an INTEGRATED AMPLIFIER. A tuner is a radio, and is typically designed to pick up the FM broadcast band (nominally 88-108MHz) and the AM broadcast band (nominally 530kHz to 1600kHz). The tuner output is "line level" and is fed to a preamp (although it may be fed directly to a power amp if a preamp is not used). A RECEIVER is an integrated amplifier (preamp and power amp) that also contains a tuner.*

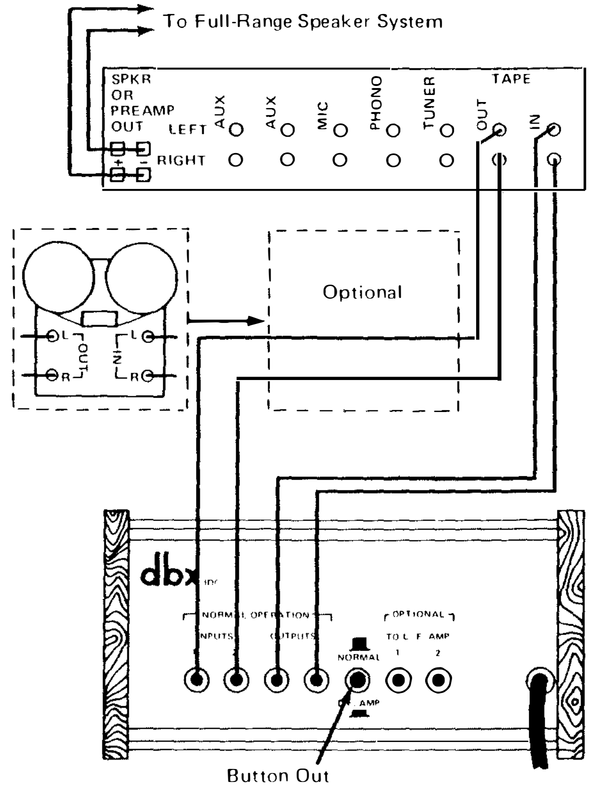


Fig. 3 – Connecting the Boom Box in the Tape Monitor Loop of any preamplifier, receiver or integrated preamp/amp.
(Dotted Lines show where to connect a tape recorder.)

SETUP & OPERATION

Step 1

With your Hi-Fi system turned OFF, connect the Boom Box as shown in Figure 3. The rear-panel NORMAL/L.F. AMP pushbutton should be out for normal operation.

Step 2

Plug in the Boom Box a suitable 117VAC, 50-60Hz power outlet; this turns the unit ON. The Boom Box has no ON/OFF switch, and since it draws only 10 watts, the Boom Box may be left ON continuously. If your Hi-Fi amplifier has an unswitched AC accessory outlet, use it for the Boom Box. It is best to leave the Boom Box ON at all times so that turn-on transients will not be created and thus cannot damage the speakers.

Step 3

Set the OPERATE/BYPASS switch to OPERATE mode (button out); set the LOW FREQUENCY BOOST at minimum (fully counterclockwise), set SUB HARMONIC LEVEL at about mid rotation.

Step 4

Temporarily turn down your preamp's volume. Turn on the preamp, and select a musical program source—preferably a record with good bass content. Set the preamp's Tape Monitor switch so you can hear its Play Input (i.e., the preamp input to which the Boom Box cables are connected).

With music playing the Boom Box's SYNTHESIZE L.E.D. normally should be fully illuminated whenever there are loud drum beats, strong bass notes, or other passages containing mid to low bass frequencies. The L.E.D. may flicker at other times.

Step 5

Adjust the SUB HARMONIC LEVEL (the big knob) to obtain a pleasing amount of ultra-low bass response. The setting is non-critical. Usually if the program itself has a lot of low bass, the SUB HARMONIC LEVEL can be decreased (turned counterclockwise). For programs with very little deep bass content, turn up SUB HARMONIC LEVEL. *See "PRECAUTIONS" on page 8.*

Step 6

Adjust the LOW FREQUENCY BOOST control (the little knob) to "even out" the bass response. Clockwise rotation increases the overall bass content of the existing program, even if no synthesis is used (i.e., even if SUB HARMONIC LEVEL is at minimum). The LOW FREQUENCY BOOST setting is non-critical, but excessively high settings may result in a muddy, poorly defined sound. *See "PRECAUTIONS" on page 8.*

From time to time, as you play different musical selections, you may wish to readjust SUB HARMONIC LEVEL and/or LOW FREQUENCY BOOST to achieve the most pleasing sound. For a dramatic demonstration of what the Boom Box is doing for the sound, press in the OPERATE/BYPASS switch; you will hear the music just as it used to sound before you had the Boom Box. Then press the switch again so the button is out, and hear what a difference the Boom Box makes.

PRECAUTIONS

(**NOTE**: Additional Precautions listed on page 17.)

1. **TO PROTECT YOUR SPEAKER SYSTEM & AMPLIFIER, FIRST TURN THE VOLUME DOWN, THEN GRADUALLY INCREASE IT.**

The Boom box not only boosts the existing bass frequencies in a program, it generates new bass at even lower frequencies. Very low bass uses up more amplifier power than mid bass frequencies, which can cause amplifier clipping (distortion). Woofers (low frequency speakers) are more easily damaged by very low bass, particularly at high amplifier volume levels or if clipping is present. Therefore, use extra caution, especially when the "raw" unprocessed program contains a lot of bass. *If distortion, cracking, or popping sounds are heard, either lower the volume or decrease the amount of LOW FREQUENCY BOOST and/or SUB HARMONIC LEVEL.*

2. **IF YOU LIVE IN AN APARTMENT, BE CONSIDERATE OF YOUR NEIGHBORS.**

The ultra-low bass frequencies generated by the Boom Box are more readily transmitted through walls, floors, and ceilings than mid-bass or midrange frequencies. Depending on the nature of the program, the effect heard in a neighbor's apartment might resemble a passing train, construction work, or a restless elephant.

3. **LOWER THE TONEARM CAREFULLY; NEVER DROP IT ON A RECORD.**

If a tonearm falls hard onto a record, it produces a very strong low-frequency "spike" or transient which, even without the Boom Box, could damage your woofers. Since the Boom Box can magnify low frequency information, use extra care when lowering the tonearm while the Boom Box is in use. If you use an automatic record changer, first test it's behavior at low volume levels; if you hear a loud thump when the arm comes down, cue records manually.

4. **AVOID SWITCHING TRANSIENTS (NOISES).**

Pops and thuds created by noisy ON-OFF or function switches in preamps and tape machines can cause speaker damage for much the same reasons cited in precaution #3 above. To safeguard your speakers, always turn your power amplifier *ON last*, after all other equipment, and turn the power amplifier *OFF first*. It is also a good idea to keep the main Volume control set low when first turning a system ON.

5. **dbx ASSUMES NO RESPONSIBILITY FOR ANY DAMAGE TO YOUR AMPLIFIER, SPEAKER SYSTEM, OR OTHER HI-FI COMPONENTS RESULTING FROM THE USE OF THE BOOM BOX.**

HINTS: FOR OPTIMUM RESULTS WITH THE BOOM BOX

1. Set your Hi-Fi's BASS tone control for "flat" response (no effect) at first. Then, after adjusting the Boom Box for the desired sound, you may wish to use the Hi-Fi tone control, but add only a moderate amount of bass boost, if any. This will avoid distortion due to excessive boost from the combined effects of the Boom Box and the tone control.

2. Make sure your speakers are "in phase," that the wires from the amplifier to the speakers are correctly polarized. This is important whether or not you are using the Boom Box. (Refer to the speaker instruction manual or the Appendix of this manual for speaker phasing instructions.)

3. Experiment with the placement of your speakers. Sometimes moving a speaker only a few inches can result in a noticeable difference in tonal balance. Placement is important whether or not you use the Boom Box, but a speaker location which worked well without the Boom Box may no longer be ideal. As a rule, the closer a speaker is located to a corner (the junction of two walls, the wall and floor, all three, etc.) the more bass will be heard. If a speaker is placed in the middle of a room, away from walls and up in the air, the bass output is minimized. Refer to the speaker instruction manual or the Appendix of this manual for speaker placement instructions.)

4. If your speakers have built-in level controls for the midrange and/or tweeter, it may be desirable to readjust these. Once the Boom Box is set up and adjusted, experiment with different speaker settings.

5. Be sure your turntable or changer (record player) is acoustically isolated. That is, the unit should be installed so that it has maximum immunity to vibrations caused by the speakers. A very massive base is helpful (such as a heavy table or book case). Also, locate the turntable as far away from the speakers as practical.

USING THE BOOM BOX IN MORE COMPLEX SOUND SYSTEMS

A SEPARATE AMPLIFIER & SPEAKER SYSTEM FOR LOW FREQUENCIES ONLY

Theory

It is possible to extend the bass response of a Hi-Fi system by using a sub woofer speaker system. Sub woofers are simply speakers which are designed to operate at very low frequencies, generally from a low of 20Hz or 25Hz to a high of between 60Hz and 125Hz. By feeding the boosted and synthesized low frequencies to a sub woofer instead of a full-range speaker system, two advantages can be realized: improved deep bass response and less overall distortion.

There is less distortion in the bass region because the sub woofer, by taking over the longest cone excursions, reduces the strain on the main speaker system's woofer cone and thereby lessens the chances of cone "break up," "doubling" (where the cone bottoms out against the speaker's magnetic structure), or over-excursion (which can damage the cone's suspension system). Sub woofers, by definition, are capable of reproducing lower frequencies than standard woofers. There is less distortion in the mid and higher frequency region because better headroom is maintained (as explained below).

The woofers in home speaker systems of average dimensions tend to be less efficient, particularly at very low frequencies, than the midrange drivers and tweeters. This is because speaker design engineers often trade off efficiency to obtain extended low frequency response; an alternative is to use a very large enclosure. Thus, the regular woofers require a large proportion of the amplifier's power in order to reproduce the extra low frequencies created by the Boom Box (and even then, the woofers may not be able to reproduce the lowest of the synthesized bass). Because the additional lows "use up" a lot of amplifier power, there is less reserve amplifier power left to reproduce musical peaks in the midrange, which is where most of the program frequencies are centered.

The reserve power available for peaks (transients), over and above the average power used to reproduce a program, is known as "headroom"—the power margin before distortion occurs. If a sound system has one amplifier and is required to reproduce the extra octave created by the boom box, that amplifier probably would need to be about twice as powerful in order to maintain the same headroom as before. On the other hand, if one power amplifier and one sub woofer are used only for the boosted and synthesized low frequencies, and another amplifier and full-range speakers are used for the original program the overall amplifier power needed will probably be much less than the overall power requirements for a single larger amplifier. (The reasons why two amplifiers can maintain the same headroom with less total power capacity than one large amplifier are technically and mathematically complex. For further information, see "POWER: How Much is Enough" by Chris Foreman, in the April 1977 issue (Vol. 8, #2) of "Recording engineer/producer," especially the discussion on page 86.)

A further advantage to using a separate power amplifier and sub woofer(s) to handle only low frequencies is that any harmonic distortion generated by that amplifier will probably occur at a frequency which is in the region where the sub woofer produces little sound (above its roll-off point), so the amplifier's electrical distortion is never converted to sound and remains inaudible. On the other hand, were a single amplifier and speaker system is used, any higher frequency electrical distortion produced by bass notes would be audibly reproduced by the regular woofer or would go through the speaker system's crossover network and be reproduced by the midrange driver or the tweeter.

Two sub woofers are not necessary because low bass gives no useful directional information to the humar ear. One sub woofer fed by a monaural mix of the two channels will work quite well, and you can save the cost and space required for two sub woofers; you can also use a monaural power amplifier for a single sub woofer. (When using a single sub woofer and a monaural amplifier, either the #1 or #2 TO L.F. AMP output may be used, since both contain the identical signal). In either case, whether you use one or two sub woofers, use of a separate power amplifier to drive the sub woofers still provides the best overall sound.

NOTE: The separate power amplifier and sub woofer(s) described here constitute a pseudo "biamplified" sound system, offering many of the advantages of true biamplification without the expense of a separate low-level (electronic) crossover unit. (For a further discussion of biamplification, see page 13.)

Connections

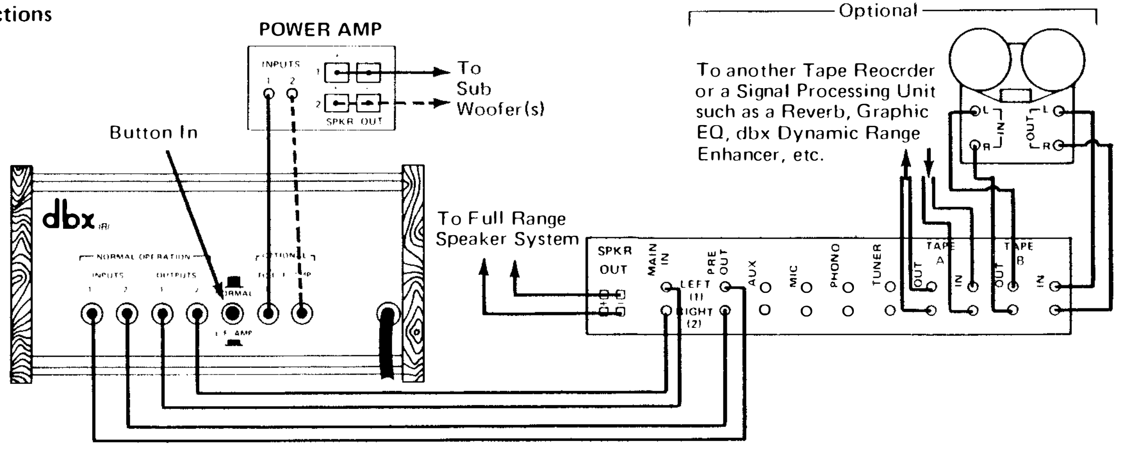


Fig. 4A – Driving a Separate Power Amplifier (L.F. Amp) and Speaker System (Sub Woofer) from the Boom Box's TO L.F. AMP outputs.

The Boom Box is usually located between the Preamp Out/Main Amp In connectors of the receiver or integrated amp, or between a preamp and two separate power amps.

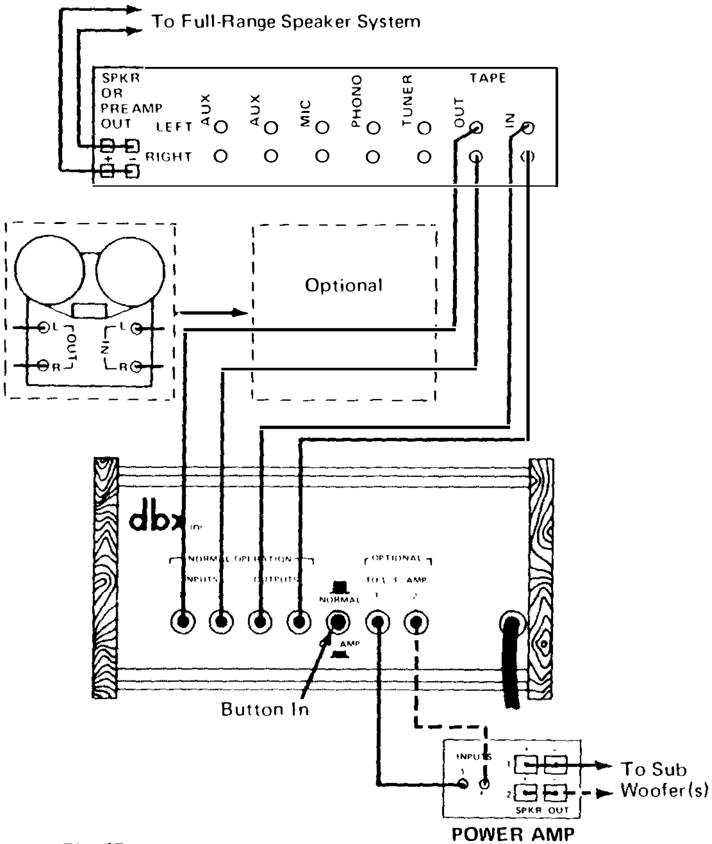


Fig. 4B —

The Boom Box may be located in the Tape Monitor Loop of a preamp; this setup will require separate adjustment of the preamp Volume and the L.F. amplifier Volume controls whenever any volume change is required. Also, any preamp Mute, Filter or Tone Control functions do not

Operation

All Boom Box controls and indicators work just as described in the callouts in Figures 1 and 2, and in SETUP & OPERATION on page 7. However, the rear-panel NORMAL/L.F. AMP switch should be pushed in.* With the preamp Volume set at a level which produces an average listening level for the full range speaker system, the Volume of the L.F. power amplifier should be adjusted so the boosted and synthesized bass frequencies are properly balanced. The balance is only as critical as your ears, and no one Volume setting is "ideal." What sounds good to you is appropriate, so long as your speakers can handle the power safely. It may be desirable to change the power amp Volume with different program materials or as the Boom Box's LOW FREQUENCY BOOST and SUB HARMONIC LEVEL controls are adjusted.

**It is possible to operate the Boom Box in this setup with the NORMAL/L.F. AMP switch out (in NORMAL mode). The NORMAL outputs will then contain the full frequency program plus the boosted and synthesized low frequencies, while the TO L.F. AMP outputs contain only the boosted and synthesized low frequencies. Thus, the enhanced low frequencies are being fed redundantly to two amplifiers and two sets of speakers, offering the potential for very powerful bass sound. This mode of operation should be approached very carefully. It would be easy to damage the woofers in the "normal" full range speaker system by using too much bass boost or too much synthesis while adjusting the Boom Box to realize the full potential of the sub woofers. Also, there may be some loss in sound definition and clarity because two speaker systems are producing the same sounds, hence it is possible for phase cancellation to occur. (The sound energy can add together or subtract, depending on the precise spacing between the two speaker systems, the phasing of all amplifier-to-speaker cables, and the specific frequencies involved.) Therefore, if you use two amplifiers and two sets of speakers with the NORMAL/L.F. switch in Normal mode, pay close attention to speaker phasing and spend some time experimenting to find the best possible speaker positions.*

BIAMPLIFIED OR TRIAMPLIFIED SOUND SYSTEMS

Theory

With a conventional sound system and two-way speakers, the speaker cable from the power amplifier carries the full-range program. It is connected to a passive, high-level cross-over network within the speaker, and the crossover network divides the program; low frequency material is fed to the woofer, and high frequency material to the tweeter. In a three-way speaker system, there are three divisions, so the crossover network divides the sound three ways.

A biamplified sound system utilizes two power amplifiers, one which feeds the woofer directly and one which feeds the tweeter directly. The speaker has no built-in crossover network. Instead, a low-level crossover network (usually electronic) divides the full-range program into low and high frequencies *before* it is fed to the power amplifiers. Hence, the low level crossover network operates at preamp levels rather than high levels (speaker levels). Similarly, in a tri-amplified system, the low-level crossover network divides the program into low, mid and high frequencies and feeds three power amplifiers, which in turn feed the low, mid and high frequency sections of the speaker system.

Biamplified or triamplified sound systems usually cost more than similar traditional systems due to the added cost of the low-level crossover network and the need for one or two more power amplifiers. The extra cost and complexity are offset by several advantages, such as higher efficiency, better effective damping factor, and less audible distortion. Efficiency and damping factor are improved because there are no crossover components between the speaker and the power amplifier (high-level crossover components cause power losses and increase the apparent output impedance of the amplifier as "seen" by the woofer). In some cases, the improved damping offered by the biamplified system will better control the woofer cone by preventing over-shoot (by controlling the inertia of the speaker cone). Audible harmonic distortion is decreased because the harmonics generated by the low frequency amplifier are too high in frequency to be reproduced by the woofer and because they have no way to reach the midrange or tweeter. Intermodulation distortion in the power amplifiers will tend to be reduced because each amplifier handles a narrower range of frequencies. There are also headroom advantages (see THEORY discussion on separate L.F. amplifier, page 10). In very large systems with many sets of speakers, the bi-amplified approach can save money because only one low-level crossover is needed instead of one high-level crossover per speaker.

When the Boom Box is used with a biamplified or tri-amplified system, there are several possible connection schemes. Generally, it is best to place the Boom Box before the low-level crossover network (between the preamp and the crossover), switch the Boom Box to NORMAL mode, us use only the NORMAL outputs, and allow the low-level crossover to divide the sound for feeding the various drivers. However, if the main speaker/amplifier system is already biamplified or triamplified and you wish to add sub woofers, it is possible to use the Boom Box NORMAL outputs to feed the low-level crossover, switch the Boom Box to L.F. AMP mode, and use the TO L.F. AMP outputs to feed the amplifier for the sub woofers, as shown in Figure 6.

THIS TYPE OF SYSTEM IS RECOMMENDED FOR DISCO INSTALLATIONS.

Connections

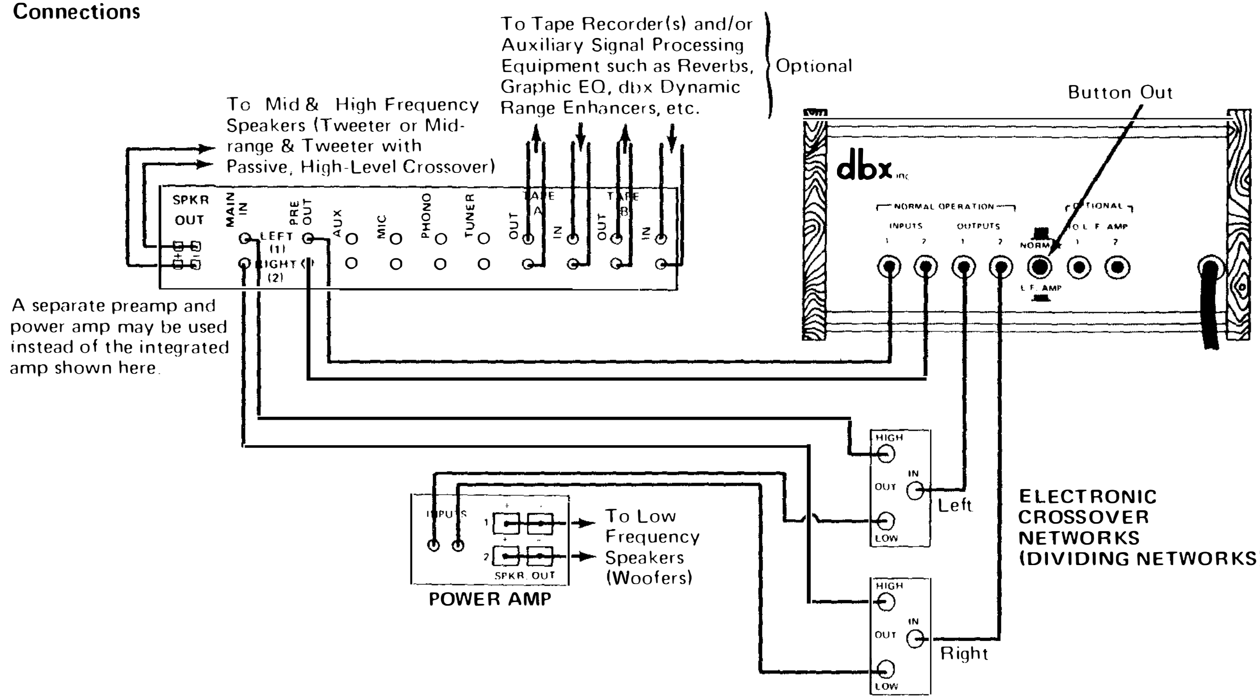


Fig. 5 – Using the Boom Box with a Biamplified Sound System

(A triamplified system would be set up nearly the same, with the addition of another power amplifier to feed the midrange speakers.)

The Boom Box is usually located between the Preamp Out connectors of the preamp, receiver or integrated amp and the low level crossover input. (It may be placed in the preamp's Tape Monitor Loop, and the preamp output would then be connected directly to the crossover input.) In most installations, only the NORMAL outputs are utilized.

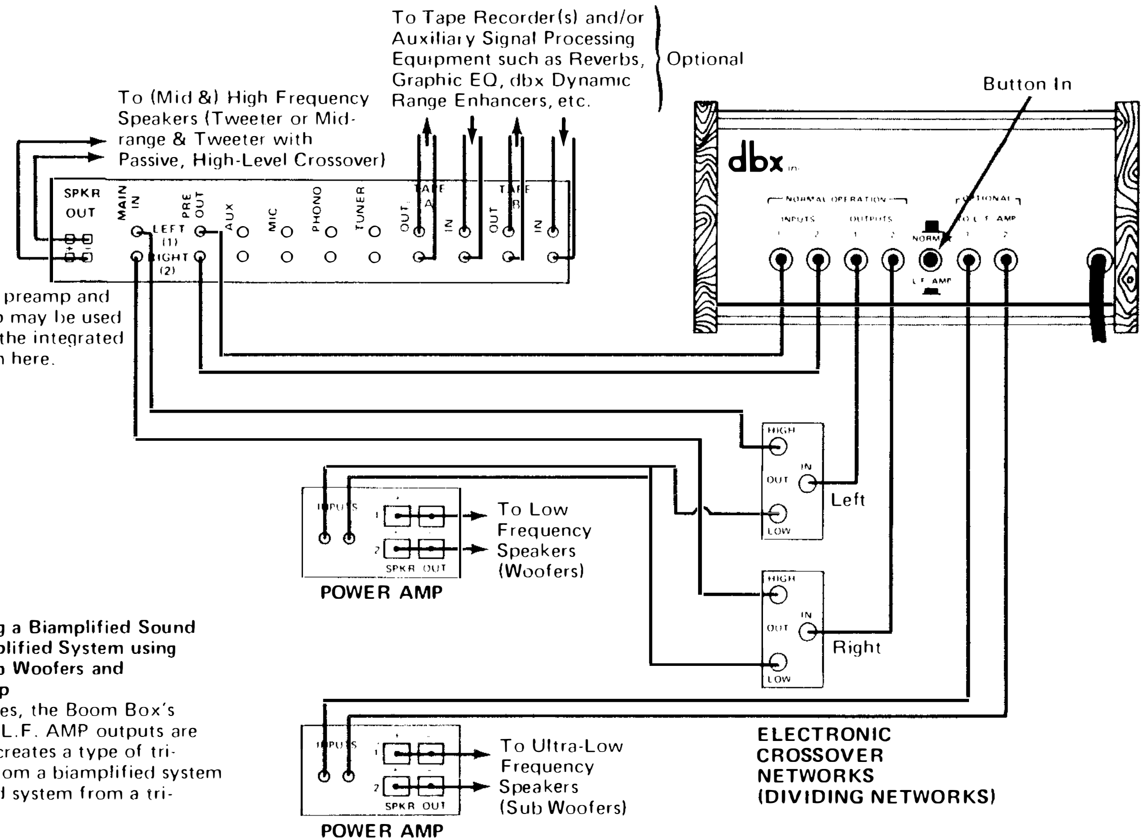


Fig. 6 – Converting a Biamplified Sound System to a Triamplified System using the Boom Box, Sub Woofers and another Power Amp

In some instances, the Boom Box's NORMAL and TO L.F. AMP outputs are both utilized; this creates a type of triamplified system from a biamplified system (or a quadamplified system from a triamplified system)

Operation

All Boom Box controls and indicators work just as described in the callouts in Figures 1 and 2, and in SETUP & OPERATION on page 7. The biamplified or tri-amplified system is set up for an appropriate balance of levels between the various power amp/speaker combinations, just as it would be without the Boom Box; the basic set up and balancing can be done with the Boom Box's OPERATE/BYPASS switch in BYPASS mode. Then the Boom Box can be placed in OPERATE Mode and adjusted for the desired effect.

dbx DYNAMIC RANGE ENHANCERS

Theory

You have probably noticed that much of the excitement of a live performance is missing in a recorded or broadcast performance. Some of this may be due to the lack of extremely low bass frequencies, absent for the reasons described in the introduction to this manual. However, the major reason for the loss of excitement is that the dynamic range* of the recorded or broadcast performance has been purposely restricted to fit the dynamic range limitations of the recording or broadcast mediums.

dbx dynamic range enhancers, including the Models 3BX, 117, 118 and 119, make it possible to restore dynamic range through a process called "expansion." By expanding dynamic range, these units lower the characteristic noise level of a tape, phonograph record or FM broadcast. They restore the "punch" of loud passages and the whisper of quiet ones. They can add new life to an old record collection and make FM broadcasts worth listening to. The use of a dbx dynamic range enhancer with a dbx tape noise reduction system, such as our 120 or 150 series, lets you make tapes that actually sound better than the original (these functions are already combined in the dbx Model 128).

When a dbx dynamic range enhancer (expander), such as the Model 3BX, is combined with the Boom Box, the results can be astounding. Usually, the enhancer is placed ahead of the Boom Box, where it increases the dynamic range. The increased dynamics serve to heighten the effect

**Dynamic range is the difference in level between the loudest and the quietest portions of a program. Since the quietest parts of a recorded program are usually restricted by noise, the dynamic range of a recording is usually defined as the difference in level (in dB) between the loudest parts of the program and the noise level.*

of the Boom Box, which itself "stretches" the frequency range of the program. Combining low frequency synthesis plus dynamic range enhancement can yield a highly realistic effect or, at more extreme settings, the program may actually sound "bigger than life."

Instructions and additional theory regarding the dbx dynamic range enhancer will be found in that unit's instruction manual. Bear in mind that the expansion process increases the power during program peaks, and that alone this extra peak power constitutes a potential threat to speakers. With the addition of the ultra-low frequency synthesis and low frequency boost from the Boom Box, extreme caution must be exercised in order to protect woofers.

Precautions (also see page 8)

1. Be sure the speakers have a power rating that is commensurate with the amplifier being used to drive them.
2. Lower the record player tone arm slowly and carefully.
3. Use the preamp's sub-sonic filter (20Hz High Pass Filter).
4. If the woofers begin to distort or make popping or scraping noises, either turn down the amplifier power, reduce the expander's EXPANSION ratio setting, and/or lower the amount of Boom Box SUB HARMONIC LEVEL and LOW FREQUENCY BOOST.

Connections

See Figures 7A, B, C and 8 on the following pages.

Operation

All Boom Box controls and indicators work just as described in the callouts in Figures 1 and 2, and in SETUP & OPERATION on page 7. The dynamic range enhancer's EXPANSION and THRESHOLD settings can be adjusted with the Boom Box's OPERATE/BYPASS switch in BYPASS mode. Then the Boom Box can be placed in OPERATE Mode and adjusted for the desired amounts of LOW FREQUENCY BOOST and SUB HARMONIC LEVEL.

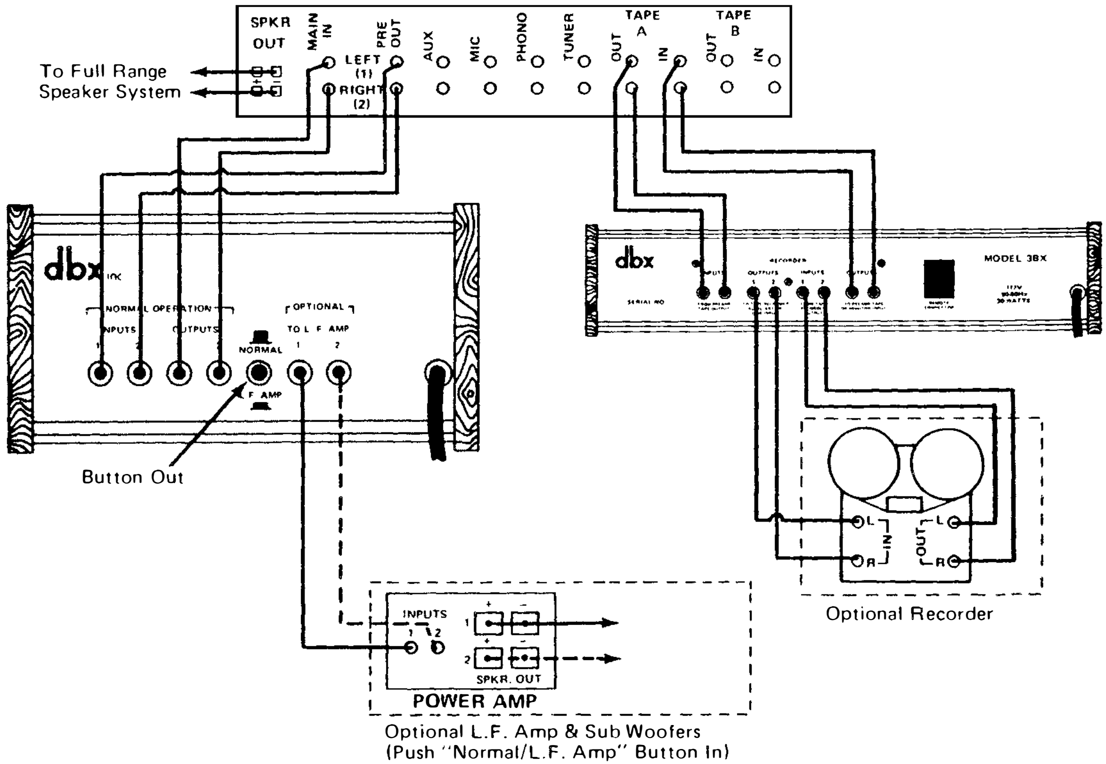


Fig. 7 – The Boom Box with a dbx Model 3BX Dynamic Range Enhancer

(The TO L.F. AMP outputs could be used with a second, power amplifier, as further described in Figures 4 and 5 and their accompanying text.)

Fig. 7A –

The Model 3BX usually is located in the preamp's Tape Monitor Loop, and the Boom Box usually is located between the Preamp Out/Main Amp In connectors of the receiver or integrated amp.

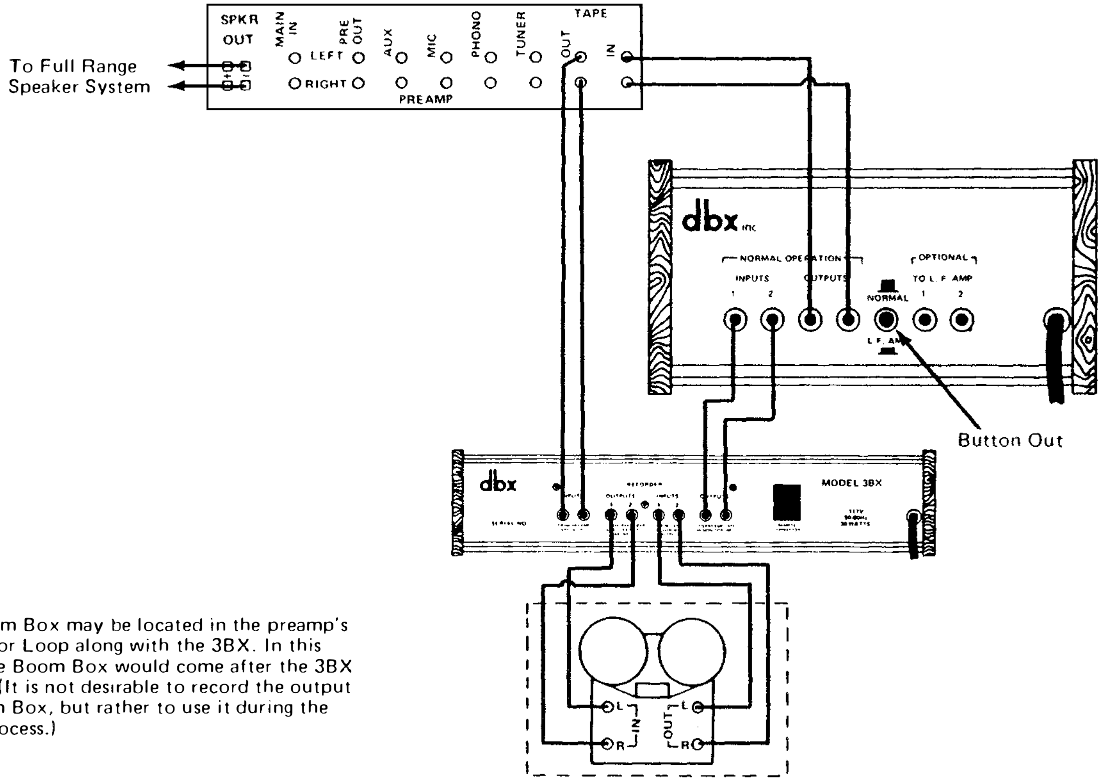
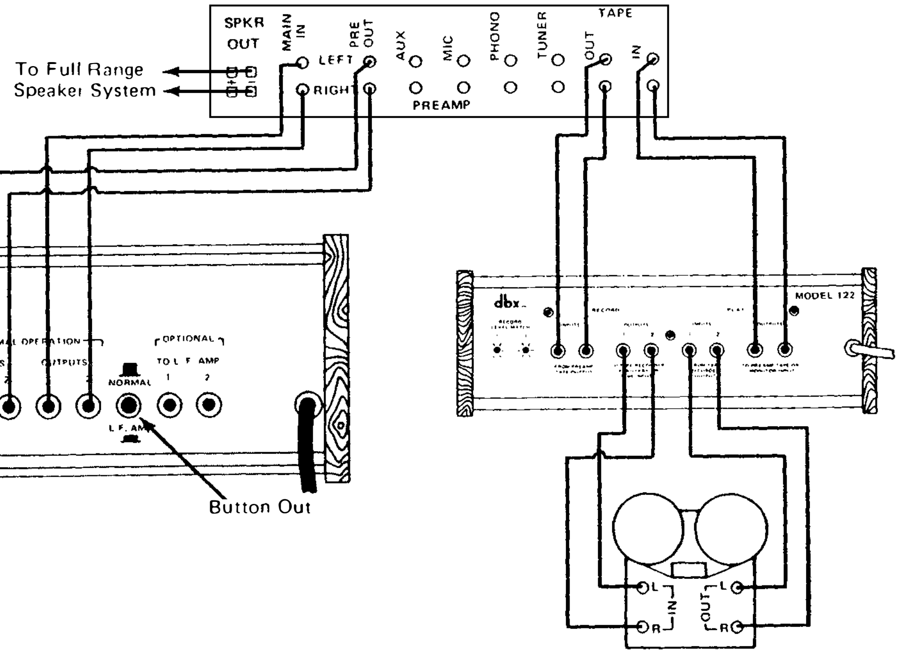


Fig. 7B —
The Boom Box may be located in the preamp's Tape Monitor Loop along with the 3BX. In this instance, the Boom Box would come after the 3BX processing. (It is not desirable to record the output of the Boom Box, but rather to use it during the playback process.)

Connections



The Boom Box with a dbx Tape Noise Reduction System

These setups are applicable to any of the dbx 120 Series (Type II) and 150 Series (Pro) equipment. (While these setups show only one amplifier in use, the TO L.F. AMP outputs could be used with a second, power amplifier, as described in Figure 4 and the accompanying text on pages 10 through 12.)

Fig. 10A –

The dbx Tape Noise Reduction System usually is located in the preamp's Tape Monitor Loop, and the Boom Box usually is located between the Preamp Out/Main Amp In connectors of the receiver or integrated amp.

Operation

All Boom Box controls and indicators work just as described in the callouts in Figures 1 and 2, and in SETUP & OPERATION on page 7. The dbx Tape Noise Reduction System is adjusted and operated in the normal fashion, as described in its Instruction Manual. Since substantial dynamic range is available from dbx-encoded Tapes (or Discs), it is sometimes difficult to select an appropriate Volume setting when the program is first turned ON. We therefore recommend initially setting the amplifier Volume lower than normal. Then search the encoded tape or disc for a very loud passage, and adjust the Boom Box's LOW FREQUENCY BOOST and SUB HARMONIC LEVEL as desired. Finally, readjust the amplifier's Volume for a normal listening level, making sure that there are no pops, crackles, and other signs of amplifier clipping or woofer overdrive. This procedure will prevent an unexpectedly loud passage from damaging the woofers. When decoding a dbx-encoded disc and using the Boom Box, special care must be exercised to avoid acoustic feedback; be sure the turntable is on a very solid, massive base, and do everything within reason to separate the turntable from the speakers. In extreme instances, it may be necessary to increase the tracking force on the tone arm; alternatively, lower the amplifier Volume, LOW FREQUENCY BOOST and/or SUB HARMONIC LEVEL.

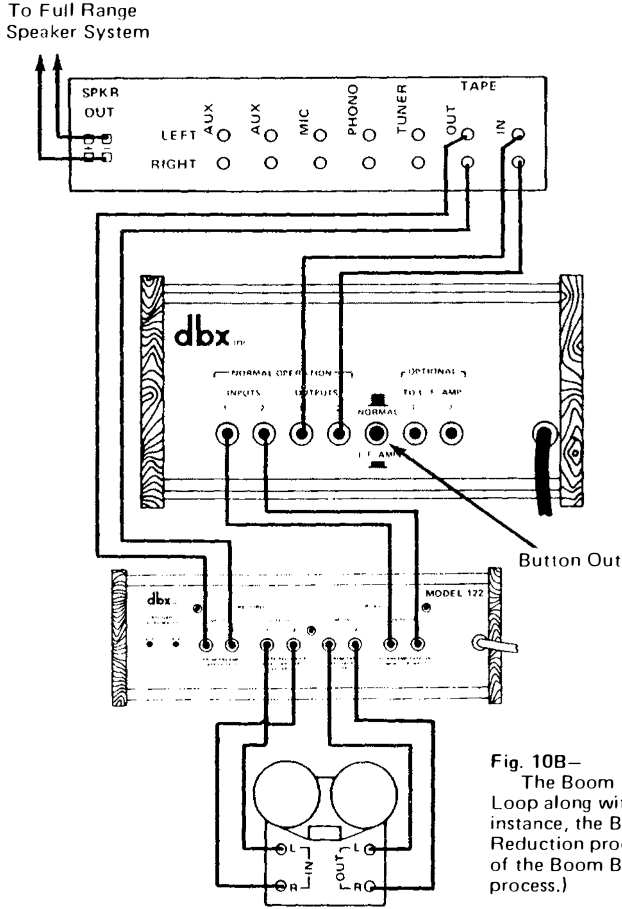


Fig. 10B–
The Boom Box may be located in the preamp's Tape Monitor Loop along with the Tape Noise Reduction System. In this instance, the Boom Box would come after the Tape Noise Reduction processing. (It is not desirable to record the output of the Boom Box, but rather to use it during the playback process.)

A SPEAKER EQUALIZER SUCH AS THOSE REQUIRED FOR BOSE OR E.V.–INTERFACE SERIES SPEAKERS (OR A GRAPHIC EQUALIZER)

Theory

A few types of speaker systems are designed to be used in conjunction with special Speaker Equalizers. The equalizer usually boosts low frequencies in a particular fashion that, together with the inherent frequency response of the speaker, produces "flatter" response at bass frequencies. We recommend using extreme caution if the Boom Box is connected to an equalized speaker because a speaker which has been designed to give smooth, extended bass response with an equalizer may already be "pushing" its woofers to their mechanical limits and nearing the "clipping" point of the power amp. Thus, the extra bass boost available from the Boom Box could quickly lead to woofer damage. A safer alternative would be to use the Boom Box in L.F. AMP Mode, to feed a separate power amp and sub woofers from the OPTIONAL TO L.F. AMP outputs. Feed the Boom Box Input from the Speaker Equalizer output, and feed the NORMAL OPERATION outputs to the full range speakers.

NOTE: Speaker systems which are designed for use with Speaker Equalizers must be used with the equalizers; the Boom Box is not a substitute for the Speaker Equalizer.

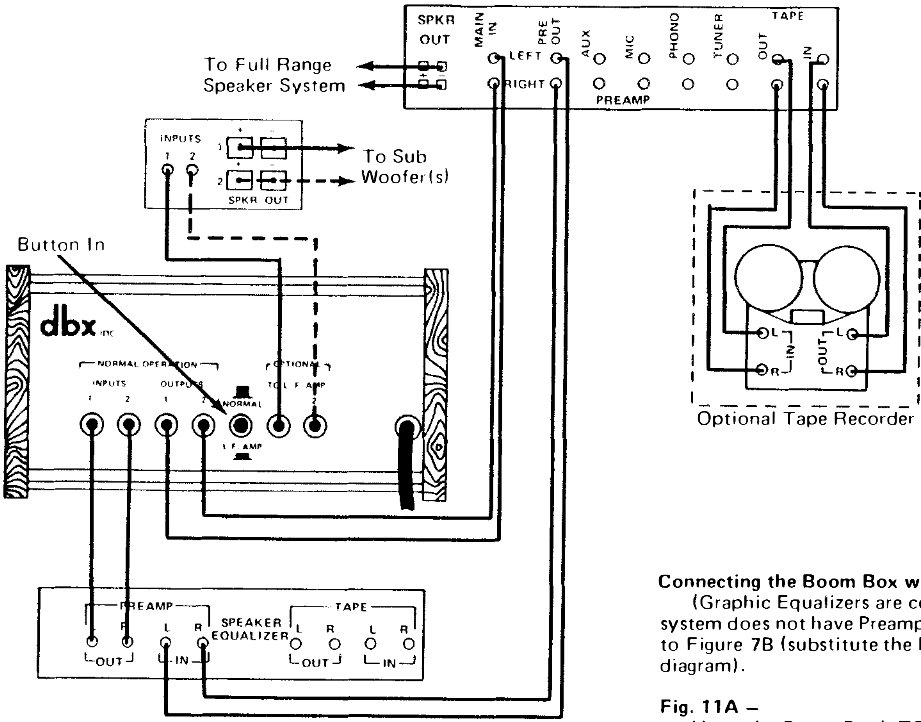
It makes little difference whether the Boom Box comes before or after the Speaker Equalizer. Thus, if there are Preamp Out/Main Amp In jacks, both the Speaker Equalizer and the Boom Box may be inserted at this point in the circuit, as shown in Figure 11. If there are no Preamp Out/Main Amp In jacks, the Boom Box and the Speaker Equalizer can both the inserted in the Tape Monitor Loop; the connections shown in Figure 7 are appropriate here, substituting the Speaker Equalizer for the 3BX.

NOTE: Graphic Equalizers may be used with any speaker system. Insofar as connection and operation with the Boom Box, Graphic Equalizers may be treated the same as Speaker Equalizers.

Operation

All Boom Box controls and indicators work just as described in the callouts in Figures 1 and 2, and in SETUP & OPERATION on page 7. The Speaker Equalizer's Tape Monitor Loop should be bypassed unless a tape machine connected in that loop is serving as the program source.

Connections



Connecting the Boom Box with a Speaker Equalizer
(Graphic Equalizers are connected the same way.) If the system does not have Preamp Out/Main Amp In jacks, refer to Figure 7B (substitute the Equalizer for the 3BX in that diagram).

Fig. 11A –
Using the Boom Box's TO L.F. AMP outputs (recommended) with a separate power amp and sub woofer(s).

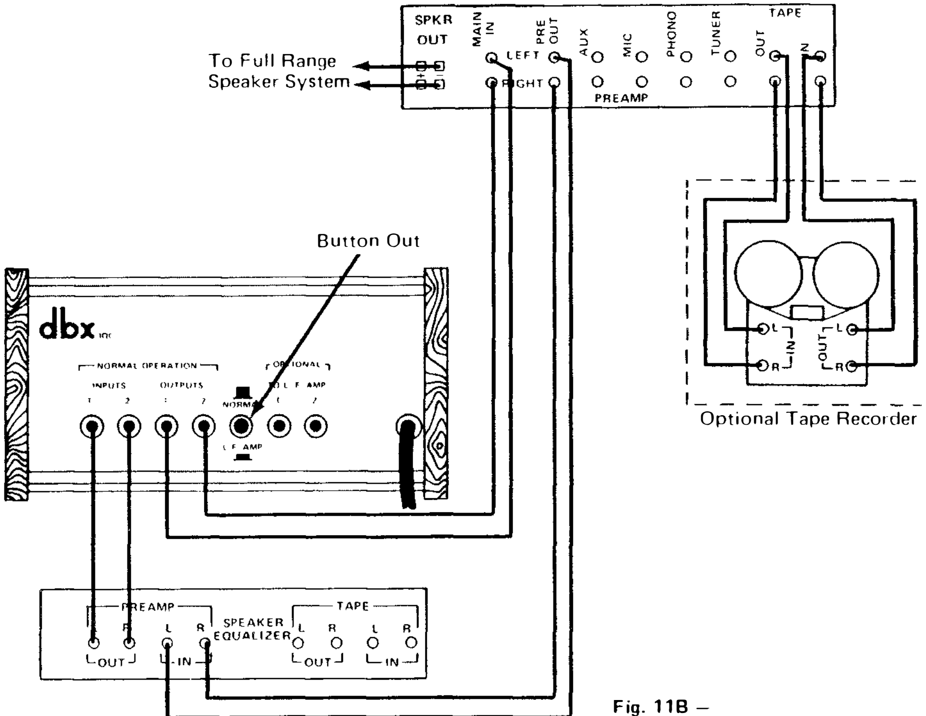


Fig. 11B –

Using only the Boom Box's NORMAL outputs (not recommended in most cases).

A FOUR CHANNEL SOUND SYSTEM

Theory

Bass frequencies do not provide much useful psychoacoustic information about the location of the sound source. Recording engineers take advantage of this fact and usually mix the low bass frequencies near the center of a stereo or four channel recording, positioning the bass so that the disc can be cut at higher levels, and so that there is less tendency for the stylus to skip upon playback. Even if the original tape has the bass in an extreme corner position, it is usually remixed during the record mastering operation so that frequencies below about 100Hz end up monauralized.

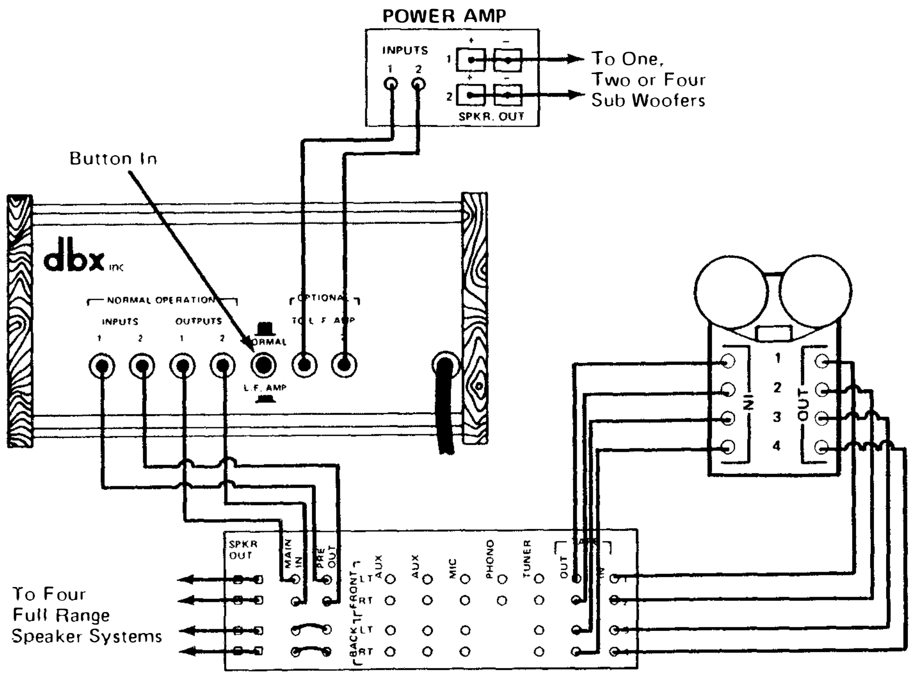
Recognizing the validity of this principle, dbx designed the Boom Box so that the bass boost and low frequency synthesis is created from a blend of the bass information from both of the Boom Box's input jacks. Thus, the enhancement created by the Boom Box is identical in all the outputs in NORMAL mode, and is identical in both the TO L.F. AMP outputs in L.F. AMP mode.

It follows that one can use a single Boom Box for enhancement of any 4-channel sound system. In most cases it is easier to use sub woofers and a separate power amplifier for the enhancement (OPTIONAL mode), rather than mixing the enhancement together with the full range program (NORMAL mode). It is sufficient to feed the Boom Box input with a stereo signal, either the front two channels of a discrete four channel program, a 4:2 mix of the discrete four channel program, or the matrix quad program before it has been decoded. (The front channels of a discrete quad program probably contain sufficiently accurate bass information, without the rear channels, to guide the Boom Box's enhancement circuitry.) The Boom Box's TO L.F. AMP output may then be fed to one, two, four or more sub-woofers with an appropriate number of power amplifiers. (If necessary, use Y-adapter cables to split the Boom Box's TO L.F. AMP outputs for feeding more than two amplifier inputs.)

Operation

All Boom Box controls and indicators work just as described in the callouts in Figures 1 and 2, and in SETUP & OPERATION on page 7. If only one sub woofer is utilized, it should be near the center of the sound field (i.e., located in the middle of the area between the four speakers). If two sub woofers are utilized, they may be placed at the sides or at the front and rear of the sound field. If four sub woofers are utilized, they may be placed adjacent to each full-range speaker or, they may be stacked in one central location; the configuration depends on the size of the room, the placement of the full range speakers, and the floor space available. (Locating sub woofers in the corners of the room multiplies their bass response through acoustic coupling to the walls, but so does stacking the sub woofers, due to acoustic coupling between the speakers themselves.)

Connections



Four Channel Amp/Preamp or Receiver

Connecting the Boom Box in a Four Channel Sound System

Fig. 12A – A Discrete Four-Channel System

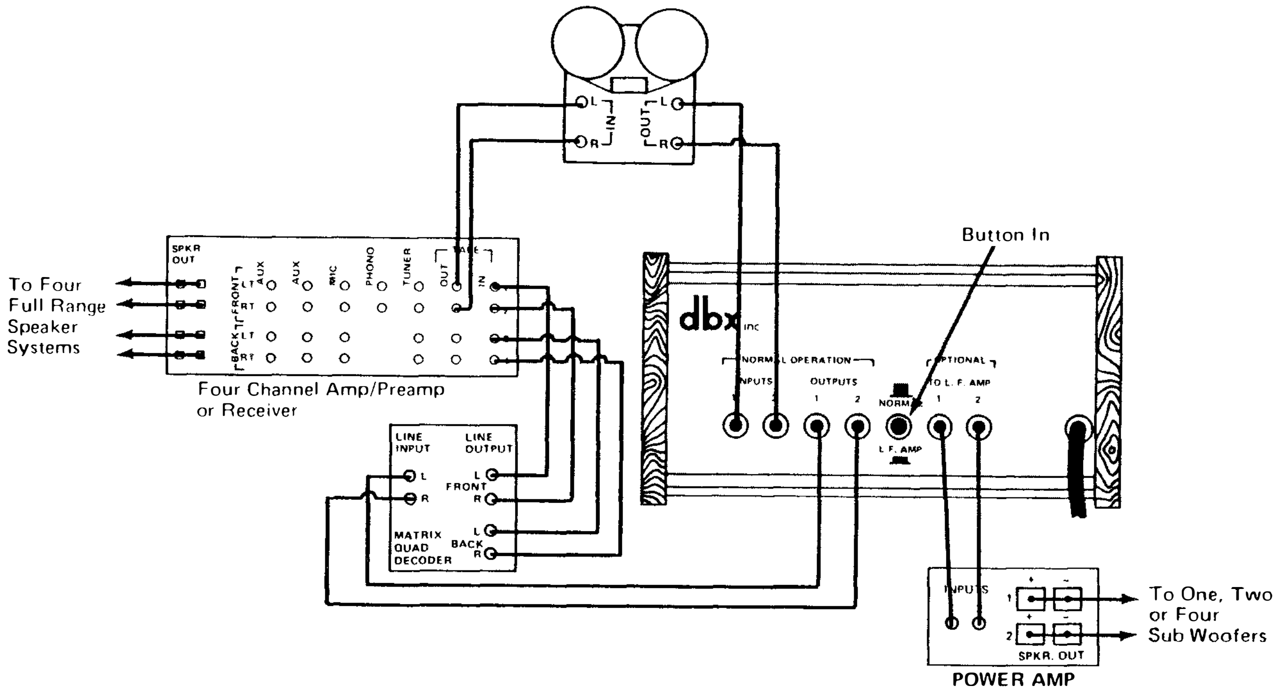


Fig. 12B – A Matrix Four-Channel System

HOW THE BOOM BOX WORKS:
A Brief Functional Description

SUB HARMONIC SYNTHESIZER

The sub harmonic synthesizer portion of the Boom Box is similar to a device which musicians call an "octave divider." Essentially, a digital circuit--a sort of micro-computer--looks at all input signals between 55Hz and 110Hz. The synthesizer then creates a signal of corresponding volume, but at half the frequency (i.e., between 27Hz and 55Hz).

A Light Emitting Diode (L.E.D.) is illuminated whenever the synthesizer is functioning. The L.E.D. is somewhat level-dependent, becoming brighter as the signal which triggers the synthesizer becomes stronger.

The synthesized sub harmonics (which we have also called ultra-low bass throughout this manual) may be assigned to the OPTIONAL TO L.F. AMP outputs only or to these outputs and to the NORMAL OPERATION outputs. The amount of synthesized sub harmonics that actually appear at these outputs is adjustable by means of the SUB HARMONIC LEVEL control.

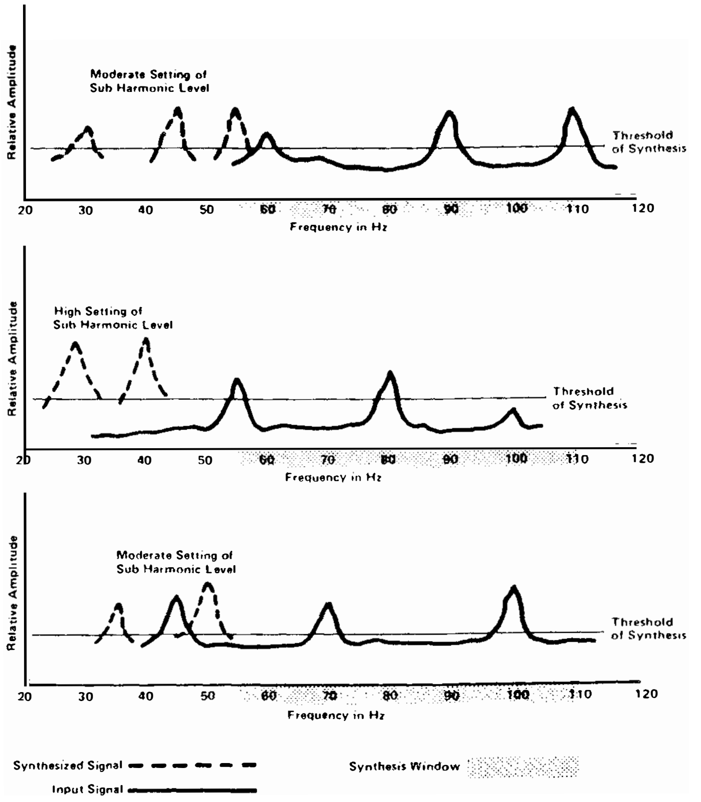


Fig. 13 – Input Signal (Solid Lines) and Synthesized Sub Harmonics (Dotted Lines) at Several Points in Time and with Different Boom Box Rotary Control Settings

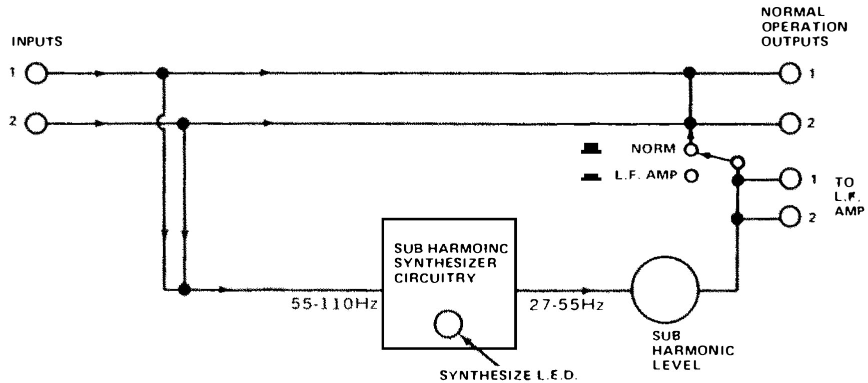


Fig. 14 — Where the SYNTHESIZE L.E.D. & SUB HARMONIC LEVEL are Located Relative to the Signal Flow.

LOW FREQUENCY BOOST

The Low Frequency Boost is a form of equalization, a specially derived curve which serves a number of functions. Primarily, the L.F. Boost is useful for "filling in" the low-mid bass frequencies. A sort of gap can occur between the synthesized sub harmonics and the input signals an octave higher which trigger the sub harmonics. The L.F. Boost circuits peak at between 50Hz and 80Hz, the actual peak shifting lower in frequency as more boost is added (see the curves in Figure 15A). L.F. Boost can be used much the same as the Bass tone control on an amplifier, although the audible effect differs because tone controls generally have *shelving* characteristics, rather than the *peaking* characteristic of the Boom Box (see the curves in Figure 15B).

When a typical Bass tone control is adjusted for a large amount of boost, say 10dB, the audible effect may be desired at 50Hz. However, the nature of the shelving curve is such that this setting will also produce from 10 to 12dB of boost at 20Hz. Because the Boom Box has a peaking L.F. Boost curve, when it is set for 10dB of boost at 50Hz, there is only about 5dB of boost at 20Hz. Since the chances are good that the only 20Hz signal present in the input program is noise and rumble, the less these frequencies are boosted, the better. Thus, the Boom Box's L.F. Boost will probably sound better than a bass tone control, given both are set for the same maximum amount of boost. (Remember that although the Boom Box's Sub Harmonic Synthesizer creates new bass in this frequency range, the synthesis has been triggered by bass at higher frequencies, not by noise.)

The boosted low frequencies may be assigned to the OPTIONAL TO L.F. AMP outputs only or to these outputs and to the NORMAL OPERATION outputs. The amount of Low Frequency Boost is controlled only by the LOW FREQUENCY BOOST control (the small knob),

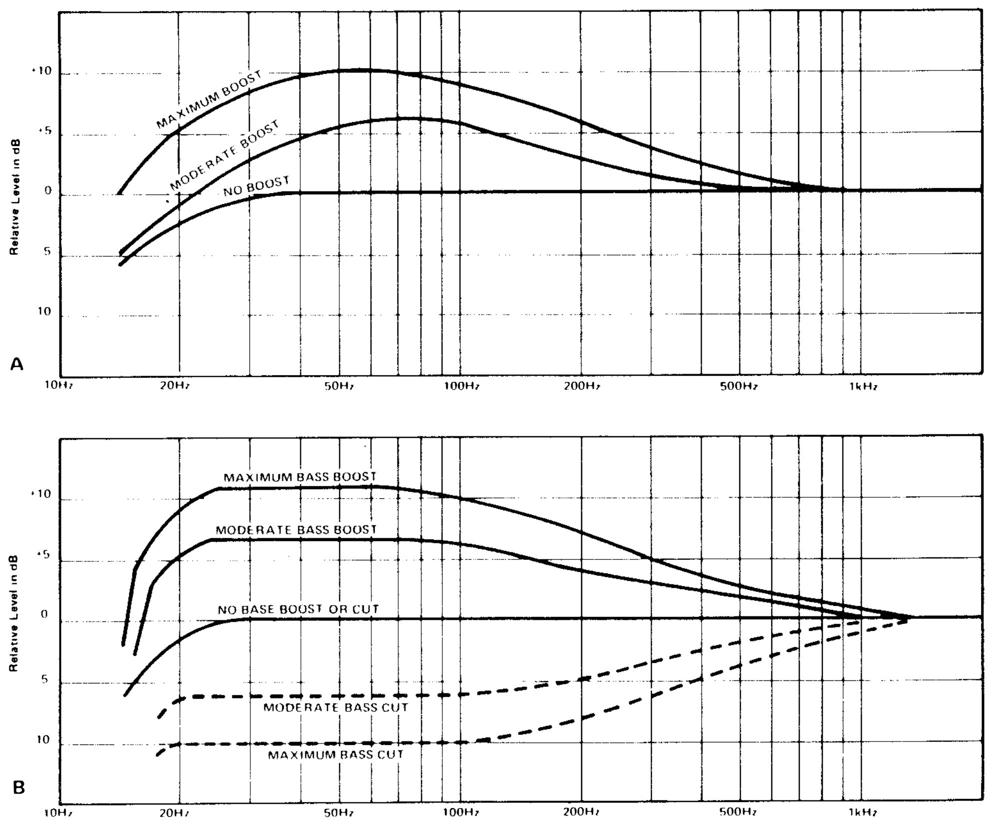


Fig. 15 — Comparison of the Boom Box's L.F. Boost Characteristics with Typical Bass Tone Control Characteristics

A. Boom Box L.F. Boost characteristics at three different control settings.

B. Typical Bass Tone Control Boost characteristics at settings comparable to "A" above. (Bass cut, which has no counterpart in the Boom Box, is designated by the dotted lines.)

and is not affected by any other controls on the Boom Box. The L.F. Boost is also independent of the SYNTHESIZE L.E.D.

BOOST & SYNTHESIS ARE MONAURAL FUNCTIONS

As explained in the "Theory" discussion for four-channel sound systems, on page 29 of this manual, bass frequencies do not provide much useful psychoacoustic information about the location of the sound source. Therefore, there would be no audible advantage in synthesizing and boosting the bass separately for the left and right channels of a stereo sound system; it would merely add a lot of cost to the Boom Box's circuitry. Therefore, dbx designed the Boom Box so that the bass boost and low frequency synthesis is created from a blend of the bass information from both of the Boom Box's input jacks.

In NORMAL operating mode, the combined L.F. Boosted and/or Sub Harmonic Synthesized signal is mixed equally into the left and right channels of the full frequency range input program and appears at the #1 and #2 NORMAL OPERATION output jacks. The Boosted/Synthesized signal also appears at the OPTIONAL TO L.F. AMP jacks. Stereo separation is thereby maintained through the midrange and higher frequencies, where it is necessary. In L.F. AMP mode, the same L.F. Boosted and/or Sub Harmonic Synthesized signal appears at the OPTIONAL TO L.F. AMP jacks, while the NORMAL OPERATION output jacks carry the same full-range program that appears at the input to the Boom Box.

dbx PRODUCT WARRANTY & FACTORY SERVICE

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

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, Telex: 92-2522, 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 U.S.A. — contact your nearest dbx dealer for name and address of the nearest authorized repair center.

SPECIFICATIONS

NUMBER OF CHANNELS	Two
INPUT IMPEDANCE	47k-ohms (actual), unbalanced
MAXIMUM INPUT LEVEL	7 volts
OUTPUTS	Main stereo composite outputs (NORMAL), and Synthesis/Boost only outputs (TO L.F. AMP).
OUTPUT IMPEDANCE	470-ohms (actual); designed to drive loads of 5,000-ohms or higher impedance.
MAXIMUM OUTPUT LEVEL	7 volts into a 5k-ohm load
FREQUENCY RESPONSE (Main Signal Channel)	±1dB, 25Hz to 20kHz
SYNTHESIS FREQUENCY RANGE	27Hz to 55Hz (derived from 55Hz to 110Hz input signal).
HARMONIC DISTORTION (Main Signal Channel)	0.05% T.H.D. @ 30Hz – 20kHz
HUM & NOISE OUTPUT	– 74dBV unweighted, with no input signal.
OPERATING TEMPERATURE	From 32°F to 122°F (0°C to 50°C).
CONNECTORS	6 RCA-type pin (phono) connectors.
DIMENSIONS	3-3/4" high x 7-5/16" wide x 15-5/16" deep; (9.525 cm x 18.57 cm x 26.19 cm).
NET WEIGHT	4 pounds, 5 ounces (1.96 kg)
POWER REQUIREMENTS	117VAC, 50-60Hz, 10 Watts; optional supply for 240V AC outside U.S.A.

These specifications are subject to change without notice or obligation.

HINTS: FOR OPTIMUM RESULTS WITH THE BOOM BOX

1. Set your Hi-Fi's BASS tone control for "flat" response (no effect) at first. Then, after adjusting the Boom Box for the desired sound, you may wish to use the Hi-Fi tone control, but add only a moderate amount of bass boost, if any. This will avoid distortion due to excessive boost from the combined effects of the Boom Box and the tone control.

2. Make sure your speakers are "in phase," that the wires from the amplifier to the speakers are correctly polarized. This is important whether or not you are using the Boom Box. (Refer to the speaker instruction manual or the following subsection for speaker phasing instructions.)

3. Experiment with the placement of your speakers. Sometimes moving a speaker only a few inches can result in a noticeable difference in tonal balance. Placement is important whether or not you use the Boom Box, but a speaker location which worked well without the Boom Box may no longer be ideal. As a rule, the closer a speaker is located to a corner (the junction of two walls, the wall and floor, all three, etc.) the more bass will be heard. If a speaker is placed in the middle of a room, away from walls and up in the air, the bass output is minimized. (Refer to the speaker instruction manual or the next page of this manual for speaker placement instructions.)

4. If your speakers have built-in level controls for the midrange and/or tweeter, it may be desirable to readjust these. Once the Boom Box is set up and adjusted, experiment with different speaker settings.

5. Be sure your turntable or changer (record player) is acoustically isolated. That is, the unit should be installed so that it has maximum immunity to vibrations caused by the speakers. A very massive base is helpful (such as a heavy table or book case). Also, locate the turntable as far away from the speakers as practical.

SPEAKER PHASING

Improper speaker phasing can be heard as lack of bass and/or a stereo image which appears to be unfocused, not localized between the speakers. The following instructions should yield proper phasing, but if in doubt, turn off the amplifier and reverse the two leads on the rear of one of the speakers (not both speakers). Then repeat the listening test.*

Perhaps the simplist way to check phase is to listen to a program with good bass content, and to reverse the leads on one speaker. If the bass increases, leave the speaker as it is. If the bass decreases, return the leads to their original connection.

Another way to properly phase the speakers is to connect the black (–) terminal on the rear of the left and right speakers to the amplifier's corresponding speaker ground terminals (sometimes indicated by black color, *common, com* or *gnd*). Then connect the red (+) terminals of each speaker to the corresponding 8-ohm (or suitable impedance) terminals of the amplifier (sometimes indicated by red color, *output*, (+) or *high*). To facilitate identification of the wire conductors at either end, most 2-conductor wire is coded. Look for: two wire colors (i.e., copper and

**If you are using two sets of speakers, a full range and a sub-woofer system, each set must be correctly phased alone, one pair at a time, as described above. Then turn on all the speakers and check for the stereo image and bass response. If in doubt, reverse the leads on only one pair of speakers (either both sub woofers or both full-range speakers).*

silver), a trace cord wrapped with one conductor, or a ridge in one half of the insulation.

SPEAKER PLACEMENT

Preferably, full range speakers should be placed at ear-level on a sturdy, vibration-free surface. Sub woofers, if used, need not be at ear-level. For stereo listening, the full range speakers should be placed approximately six to eight feet apart.

Due to acoustic coupling between the speaker, the air and the room, effective bass output tends to increase as the speaker is moved closer to the room boundaries. Thus, moving a speaker from the center of a wall toward one corner audibly increases the bass. Similarly, moving a speaker from ear-level to the floor or ceiling also increases the bass output without much effect on the high frequencies. Be sure to use a variety of records to check each speaker location. For the most critical listeners, symmetrical placement of furniture and speakers will help to achieve a more perfectly balanced stereo image.

Regardless of the specified response of a speaker system, the actual balance of bass to higher frequencies is dependent upon speaker placement. Experiment to find the optimal placement for the tonal balance you prefer, keeping the amplifier tone controls and any speaker tone controls at "flat" or "normal" position and the Boom Box in BYPASS mode. When a good tonal balance is achieved, place the Boom Box in OPERATE Mode and adjust it for the desired bass response. Finally, make any desired adjustments of the amplifier tone controls and any speaker level controls.

IN CASE OF DIFFICULTY

A. Check all connections and switch settings.

B. Make sure that an input signal is being fed to the Boom Box.

C. Make sure that AC power is applied and everything is ON.

D. Make sure the preamp Selector and Tape Play/Source switch are set correctly.

E. Press the Boom Box's OPERATE/BYPASS switch IN; if sound is heard only when the switch is IN (BYPASS mode), something may be wrong with your Boom Box.

F. For assistance, contact the dbx factory or your dbx dealer or service center. (See page 35 of this manual.)

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 auxillary 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.

Biamplified
Descriptive of a sound system which utilizes a low level crossover 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 oscilliscope, 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 oscilliscope 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 biamplified or triamplified 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

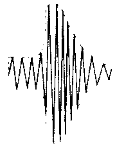
The dynamic range of a program is the range of signal levels from the lowest to the highest level. In equipment, the dynamic range is the "space," in dB, between the residual noise level and the maximum undistorted signal level. A program with wide dynamic range has a large variation from the softest 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 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 SHELVING. 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 levles.

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 impluse 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 overreaction 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 preamplifier) 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

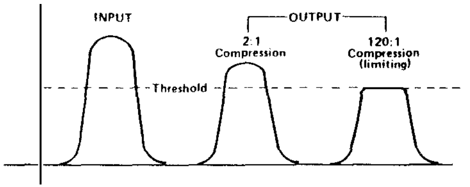
There is a maximum amount of energy that can be recorded on any given type of magnetic tape. When a recorder "tries" to record more energy, the signals become distorted, but are not recorded at any higher levels. This phenomenon is called tape saturation because the magnetic oxide particles of the tape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

Woofer

A loudspeaker which reproduces only low frequencies.

NOTES

Many dbx Model 100 sub harmonic synthesizers have a rear panel screwdriver adjustment labeled "Low Level Threshold." This control is pre-set at the factory and should require no adjustment. If for some reason this control should need adjusting or if you wish to verify that it is properly adjusted for your system, please align as follows:

1. Select a phonograph record with a lot of low frequency information.
2. Connect the Model 100 into your music system.
3. Place the OPERATE/BYPASS switch in the OPERATE position.
4. Set the LOW FREQUENCY BOOST control to the full counterclockwise position.
5. Set the SUB HARMONIC LEVEL control to the 9:00 position.
6. Play the record with the low frequency information.
7. Whenever there is low bass information the red L.E.D. above the SUB HARMONIC LEVEL control should flash.
8. Place a screwdriver in the slot of the low level threshold adjustment control on the rear panel.
9. *NOTICE:* This control will rotate a full 360°.
10. When this control is rotated between 5:00 and 7:00 the L.E.D. should not operate. Before the control reaches this dead zone the L.E.D. will be at maximum brightness. When the dead zone has been passed, the L.E.D. will be at minimum brightness.
11. Adjust the control for the maximum brightness of the L.E.D. and then reverse the control 1/8 of a turn.
12. The Model 100 "Boom Box" is now properly adjusted. You now have the full flexibility of the front panel controls in order to provide a new dimension in listening with musical sub harmonics.

dbx, Incorporated • 296 Newton Street • Waltham, Massachusetts 02154 • 617/899-8090