dbx

Models 140A/180A

Professional
Noise-Reduction Systems

Instruction Manual

INSPECTION and INSTALLATION

Your unit was carefully packed at the factory in a protective carton. Nonetheless, be sure to examine the unit and the carton for any signs of damage that may have occurred during shipping. If there is such evidence, don't destroy the carton or packing material, and notify your dealer immediately.

It's a good idea in any case to save the carton and packing should you ever need to ship the unit.

In the event of initial problems, first contact your dealer; your unit was thoroughly inspected and tested at the factory.

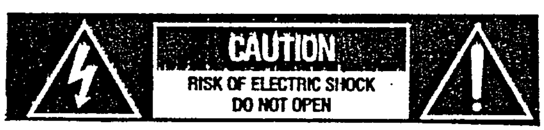
In addition to a model 140A or 180A and this owner's manual, the carton should contain a warranty/registration card. Please fill it out and send it to us.

The chassis has integral brackets ("ears") for mounting into a standard equipment rack (19" or 48.3 cm wide). No special cooling or ventilation is required in any installation; other components may be stacked above or below the unit provided they don't generate excessive heat.

WARNING

TO PREVENT FIRE OR SHOCK HAZARD,
DO NOT EXPOSE THIS COMPONENT
TO RAIN OR MOISTURE.

This triangle, which appears on your component, alerts you to the presence of uninsulated dangerous voltage inside the enclosure -- voltage that may be sufficient to constitute a risk of shock.



This triangle also appears on your component, and it alerts you to important operating and maintenance instructions in this accompanying literature.

CAUTION
To Reduce Further the Risk
of Shock, Do Not Remove
the Cover or Back. There Are
No User-Serviceable Parts
Inside; Refer All Servicing
to Qualified Personnel.

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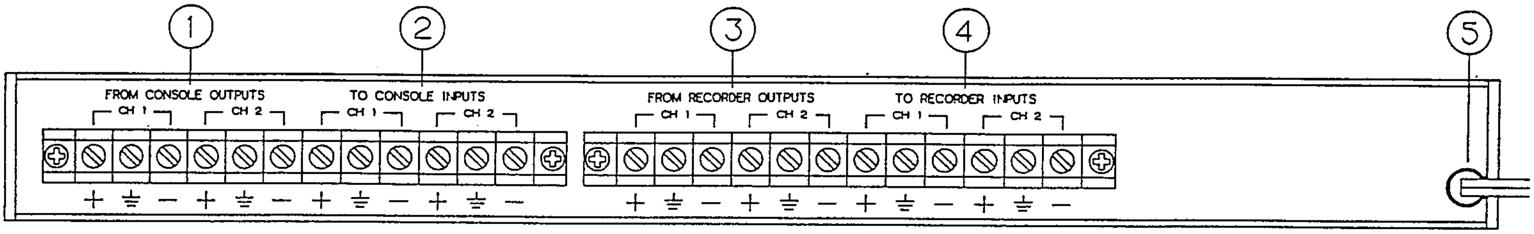
CONTENTS

SPECIFICATIONS

Effective noise reduction	40 dB or more, depending on transmission medium
Frequency response	
140A	±0.5 dB 40 Hz-20 kHz, -1 dB at 35 Hz, -2 dB at 30 Hz
180A	±0.5 dB 30 Hz-20 kHz, -1 dB at 20 Hz
Dynamic range	Greater than 115 dB
Equivalent input noise	-93 dBv
Total harmonic distortion (THD)	0.1% 100 Hz-20 kHz, less than 0.5% 30-100 Hz
Intermodulation distortion (IMD) IHF or SMPTE	0.2%
Maximum input and output levels	+23 dBv (11 V)
Input	75 k-ohms, differential
Output	Low-impedance, single-ended, designed to drive 600 ohms or greater
Optional output transformer	Jensen JE-123-SLPC
Level range for unity gain (level match)	Set at 1 V; adjustable from 200 mV to 4.9 V (-12 to +16 dBv)
Dimensions	17"w x 1-3/4"h x 7"d
Power requirements	See rear of unit

Notes
1) Specifications are subject to change.
2) All voltages are rms (root-mean-square).
3) 0 dBv is defined as 0.775 V regardless of load impedance. Subtract 2.2 from the dBv figure to convert to dBV (i.e., referred to 1 V). When the load impedance is 600 ohms, this particular dBv is also known as "dBm."
4) Dynamic range is defined as the difference between the maximum rms signal and unweighted noise. Other noise figures are for 20 Hz-20 kHz, also unweighted.
5) Frequency-response figures are for pink noise (or music).
6) THD and IMD measurements are for total encode/decode processing. SMPTE IMD is measured with 60 Hz and 7 kHz mixed 4:1; IHF (difference-tone) IMD is measured with 19 kHz and 20 kHz mixed 1:1; output 1 V.
7) Inputs and outputs have identical polarity.
8) Units for use with line voltages other than nominal 117 V ac are available outside the USA; contact dbx.

REAR



1 FROM CONSOLE OUTPUTS

These are the encoder inputs. Connect them to the appropriate bus, line, or other outputs on the mixer/console or originating tape deck. If you're using a 140A to reduce noise in an STL (studio-transmitter link, i.e., microwave, land line, etc.), connect the studio outputs to these inputs. L and R only -- no composite signals.

2 TO CONSOLE INPUTS

These are the decoder outputs. Connect them to the appropriate bus, line or other inputs on the mixer/console. In STL use, connect these outputs to the transmitter input. If you wish to monitor at the studio end the signal sent to the STL, this is where you connect an audio monitor.

3 FROM RECODER OUTPUTS

These are the decoder inputs. Connect them to the tape deck's or broadcast cart's outputs. For STLs, connect the output of the receiver to these inputs. To monitor at the studio end, these inputs should be connected directly to the To Recorder Inputs terminals.

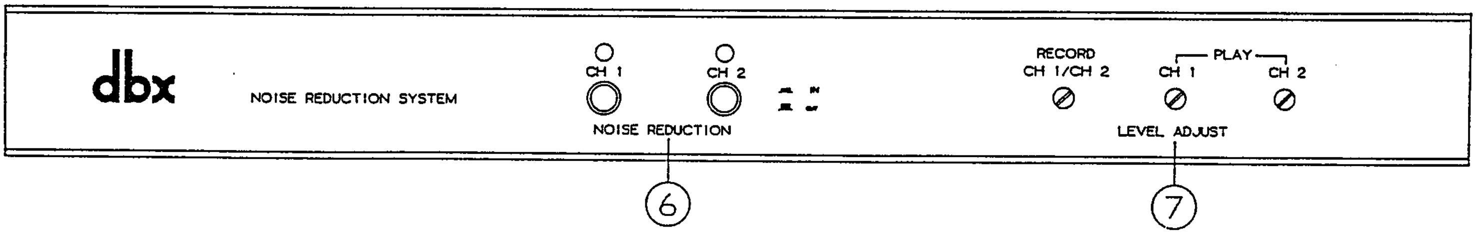
4 TO RECORDER INPUTS

These are the encoder outputs. Connect them to the tape deck or broadcast cart's line inputs or, for STL applications, to the STL inputs. For monitoring as noted above, again connect these terminals directly to the From Recorder Outputs.

5 AC POWER

Connect this cord to a nominal 117-V 50/60-Hz ac source only. Models for use with other power sources are available outside the United States; contact us for information. Note that your unit is not equipped with an On/Off switch, so you may plug it into either a switched outlet on your equipment rack or, since the ac-power draw is small, an unswitched outlet.

FRONT



6 NOISE REDUCTION:CH 1/CH 2 IN and OUT

Pushing the button In engages both the encoder and the decoder of that channel only. The Out position is a hardwire bypass, outputs connected directly to inputs.

7 LEVEL ADJUST:RECORD CH 1/CH 2 and :PLAY CH 1 and CH 2

These screwdriver-adjustable trims control the overall (full-bandwidth) gain of the encoder (channels together) and the decoder (channels separate). They allow you to achieve unity (0) gain throughout the record/play (encode/decode) process and to alter the playback (decode) interchannel balance as necessary. The settings are not critical, and need to be done usually only once in a given installation. See the discussion on p. 7.

TYPICAL HOOKUPS

Inputs and balanced and unbalanced sources

Your unit's two sets of inputs are balanced electronically by differential amplifiers. They won't unbalance a balanced source, and they may be used with unbalanced sources as well.

A balanced line is defined as two-conductor shielded cable with the center conductors carrying the signal and of opposite polarity and equal but opposite potential difference from ground. An unbalanced line is generally a single-conductor shielded cable with the center conductor carrying the signal and the shield at ground potential.

Figure 1 shows the connection of balanced signal sources to the two sets of input terminals, and Figures 2A and 2B show unbalanced sources connected to these inputs. Note that for proper operation from an unbalanced source, each minus terminal at the inputs must be connected to a ground terminal, as shown. Also note that other terms for plus, minus, and ground are high [or hot], low, and shield.

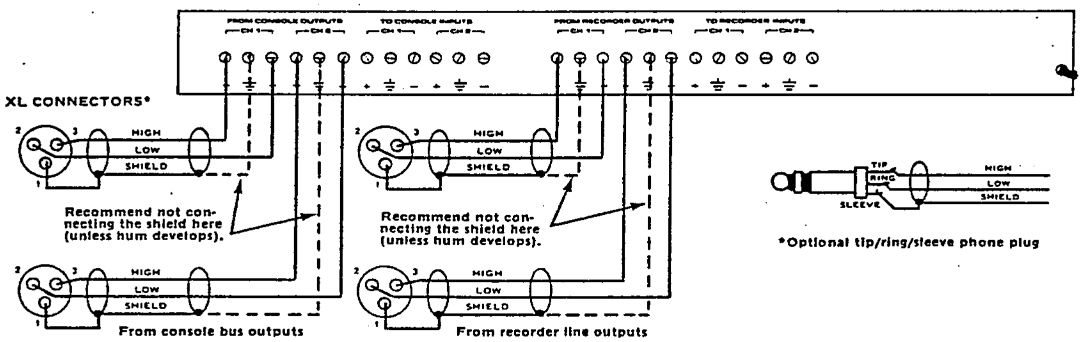


Figure 1: Balanced sources

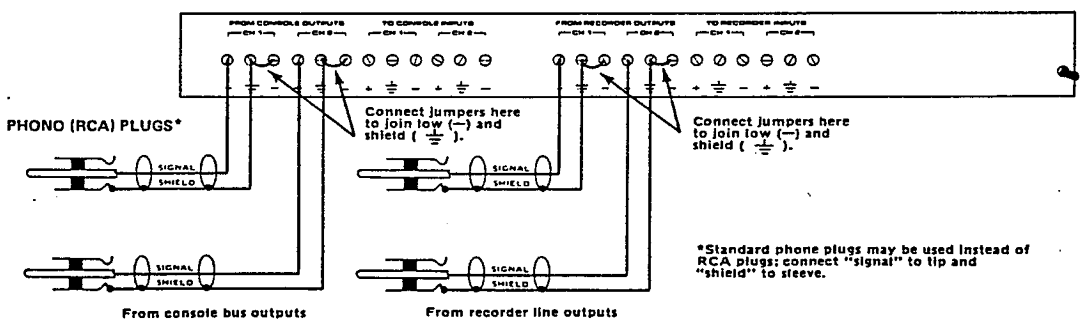


Figure 2A: Unbalanced sources, single-conductor shielded cable

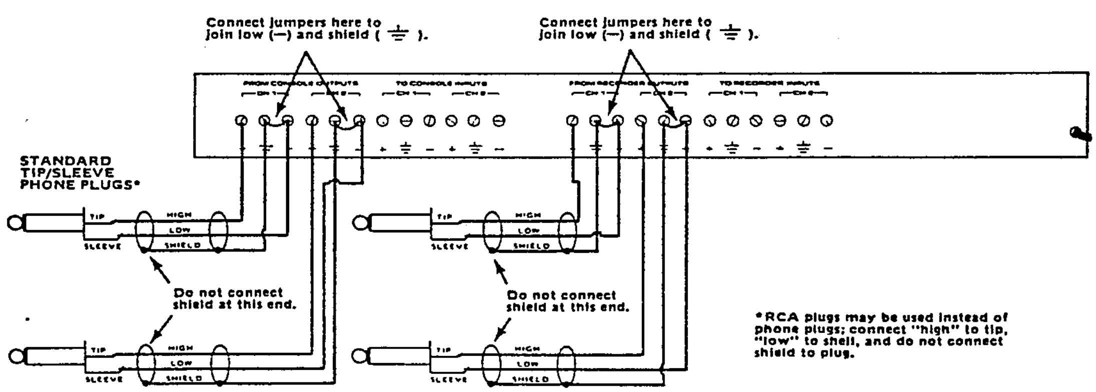


Figure 2B: Same, dual-conductor shielded cable

Outputs (unmodified) and balanced and unbalanced loads

The two sets of outputs are driven by unbalanced, single-ended line amplifiers whenever the unit is in Tape mode -- that is, whenever the encoders and decoders are processing the signal. These outputs are suitable for connection to most studio equipment, balanced or unbalanced.

Long cables connected to the outputs in very high RFI/EMI environments or in tricky grounding situations may make balanced, isolated outputs necessary. Your unit's circuit board comes drilled for user installation of such isolation/balancing transformers; see the next page.

Figure 3 shows the unbalanced connection of the set of output terminals to unbalanced inputs, and Figure 4 to balanced inputs.

Note again that each output has signal (+), (-), and ground, like the balanced inputs. The outputs are connected directly to the inputs in Bypass mode, so a balanced input remains balanced at the output when the unit is bypassed. Further, when output-isolation transformers are installed, the terminals provide standard balanced connections. Otherwise, as delivered, the minus and the ground terminals of each output are internally connected whenever the unit is in Tape mode.

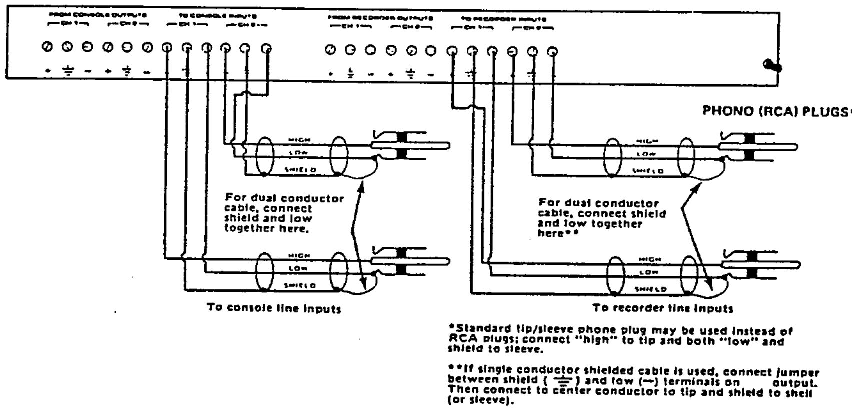


Figure 3: Unbalanced inputs

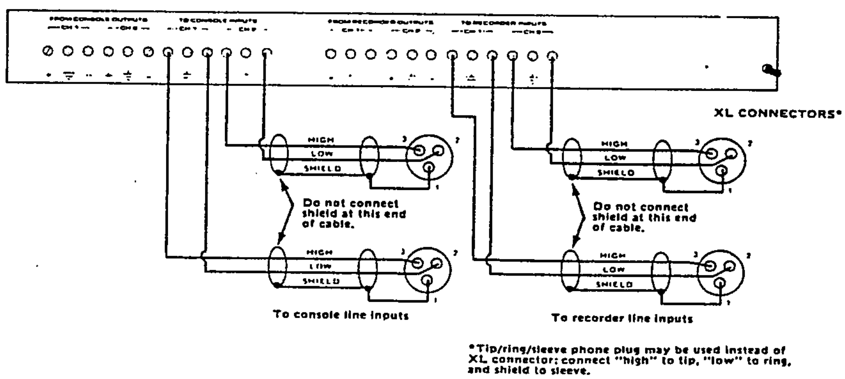


Figure 4: Balanced inputs

Transformer installation, to obtain balanced, isolated outputs

The circuit board is designed to accept four Jensen Model JE-123-SLPC transformers. These were selected because they do not degrade performance. The transformer has a 1:1 turns ratio and a nickel core to keep distortion very low. Under worst-case conditions (saturation at 20 Hz), the JE-123-SLPC handles +24 dBm maximum level with 1% THD; below saturation, THD at 20 Hz drops below 0.03% and is further cut roughly in half for each octave higher. The bandwidth is greater than 350 kHz and there is less than 200° of phase shift at 20 kHz. See your Jensen distributor or order directly: 10735 Burbank Boulevard, North Hollywood, California 91601 (telephone 213-876-0059, Monday-Thursday 9:30-4:30 USA West Coast time).

The addition of these transformers provides a balanced, floating output stage. We recommend installing them only when such an output is required, and only on the necessary outputs.

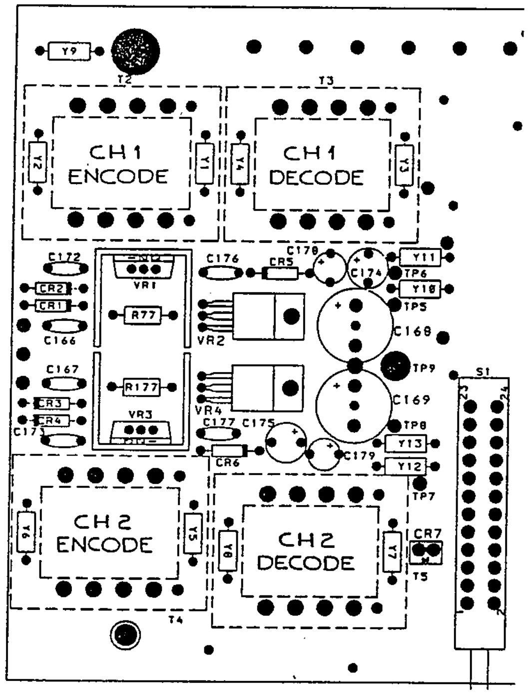
Here's the procedure:

1) Unplug the unit from ac power. Don't fail to do this.

2) Remove the four screws on each side (they secure the rack-ear bracket). Then firmly slide out the top and bottom covers (they're a snug fit -- take care not to scrape your fingers on the bottom of the PC board).

3) Referring to Figure 5, find the locations on the board for those outputs that require transformers. Within each of the mounting areas are jumpers; cut or unsolder before installing the transformer.

4) Insert the transformer into the top of the PCB and solder its pins in place on the bottom. The Jensen transformer design is symmetrical (primary and secondary windings are identical), so orientation on the board doesn't matter: if the pins line up, the transformer is installed correctly.



5) Slide the covers back on, taking care not to snag the foam tape along the extrusion flanges, and screw the side brackets on. The outputs with transformers are now balanced and floating, with no ground reference, so label the rear panel accordingly.

Figure 6 shows the typical connection of transformer-isolated outputs to balanced or unbalanced loads.

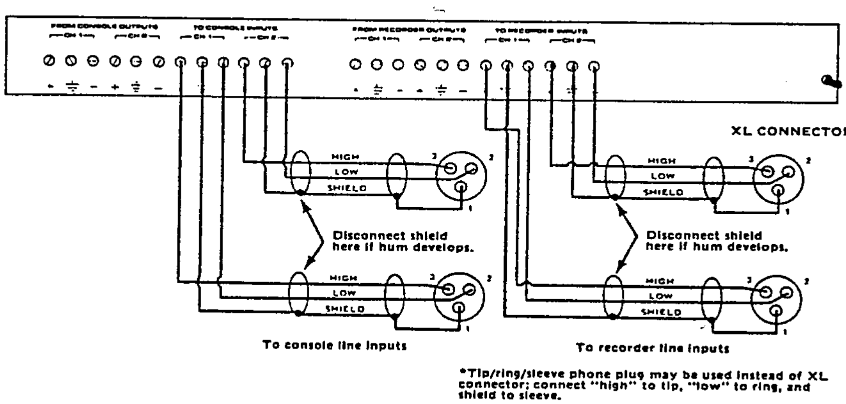


Figure 6: Connecting transformer-isolated outputs

ABOUT YOUR UNIT AND dbx NOISE REDUCTION

The Model 180A is a 2-channel record/play (encode/decode) professional component using dbx Type I noise reduction for professional-quality tape recorders. The Model 140A is identical to it — including the amount of noise reduction -- except that the 140A uses dbx Type II noise reduction, and thus (as we shall explain) is suited for tape decks of nonlinear frequency response, broadcast cartridge machines, VCRs, microwave links or land lines, and other frequently noisy consumer-grade or broadcast/transmission media.

Each unit doubles the dynamic range of the transmission medium to greater than 115 dB. Depending on the individual channel noise of the medium, each unit therefore can reduce the noise of the medium 40 dB or more. This is achieved by compressing (encoding) the signal by a 2:1 ratio and applying a carefully tailored frequency-response preemphasis during record, and expanding (decoding) the signal 1:2 with a precisely complementary deemphasis during playback. The companding is linear over a 100-dB range and requires no pilot tones or special calibration.

Type I is intended for tape machines with flat frequency response (±1 dB 20 Hz-20 kHz) and running at 15 ips or greater,* with full headroom maintained at high frequencies. Type II was developed for media where the high-frequency response is not as flat and headroom is reduced because of tape saturation, 75-us preemphasis, or other reasons. The two systems are incompatible, because the filters and preemphases used in the rms detectors are different (although the signal preemphases are virtually the same).

For example, Type I's detectors respond from about 22 Hz to 21 kHz, whereas Type II's respond from around 30 Hz to 10 kHz. Type II's filters prevent mistracking due to the frequency-response errors (head bumps, rumble, high-frequency rolloff, and the like) at the ends of the audio band that are common in consumer and broadcast equipment. Additionally, the detector preemphasis in Type II is more severe than in Type I, which causes the Type II compressor to reduce gain more when high-frequency energy is large; this too makes Type II more tolerant of high-frequency-headroom limitations. Note in the performance specifications that the frequency response of the processing circuitry hardly restricts the bandwidth of the audio signal itself.

The benefit of the 2:1 compression, of course, is that the signal becomes easier for any medium to handle. Its dynamic range has been cut in half, with the hottest levels considerably reduced and the softest passages boosted. On decoding, the signal is precisely expanded back, and the original dynamic range of the program is retrieved without hiss, saturation distortion, or degradation of frequency response. There is none of the noise buildup normally encountered in transferring information from one recorded medium to another. Noise present in the original, naturally, is not reduced in this process.

Although simple in theory, classic 2:1:2-compander noise reduction could not be achieved before the development by dbx in the early 1970s of two patented circuits, the Blackmer rms detector and voltage-controlled amplifier (VCA). The former enables optimum decode tracking and transient response despite the phase shifts typically induced by tape recorders. The latter affords precise gain control over an extremely wide dynamic range while maintaining very low noise and distortion. In quietness and dynamic range, in fact, dbx NR is markedly superior to any 16-bit PCM digital audio system.

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*It <u>is</u> possible to use Type I at 7-1/2 ips if the deck is a <u>very</u> good one (no bass head bumps, for example) and EE tape is used; the main criteria are flatness of frequency response and high-frequency headroom. The point is that a deck of less than perfect flatness may well mistrack with Type I (and would not do so with Type II). Note: so might flat decks that are over- or underbiased, or that have azimuth problems or, especially, head bumps.

It is occasionally claimed that other (non-dbx) 2:1:2-compansion systems with similar pre/deemphasis but with average, peak, or peak/average detection are compatible (or "compatible enough") with dbx Type I or II. These claims are false. While such detectors may, with steady-state (sine-wave) measurements, yield performance that looks identical, they will mistrack with transient/dynamic material. The success of the dbx systems in coping with various imperfections in the storage and transmission media is due largely to our (relatively expensive) rms-level detectors; don't be fooled into thinking otherwise.

APPLICATION NOTES (including advice on levels)

The Level-Adjust controls

For convenience in listening and in matching levels of other equipment, there are screwdriver-adjustable trim controls accessible through holes in the front panel, as noted earlier. These allow you to keep the Record and the Play levels about the same (achieving unity gain through the unit's processing and achieving the proper interchannel balance on playback) and to keep levels about the same with and without noise reduction.

Since these trims adjust only gain for the full bandwidth, please note that their settings are not critical to proper performance. Linearity and frequency response, for example, are completely unaffected.

Record

1) With the dbx unit in Bypass (Noise Reduction Out), feed a 1-kHz tone at your "0" reference level to the From Console Outputs terminals. Go through your console if you wish. Set the tape recorder's recording-level meters to their nominal calibration point (e.g., 0). Then switch the dbx unit to Noise Reduction In and turn the Record Level Adjust trim as necessary to achieve 0 again on the tape-deck meters. Don't expect to read the same levels on the deck's meters at frequencies other than 1 kHz, owing to the preemphasis in the dbx unit. If you normally use 400 Hz or some other mid-frequency tone for line-up in your studio, you can continue to do so, but use the same-frequency tone throughout.

Play

With the dbx unit in Bypass, send a 1-kHz reference-level tone (from an alignment tape or other source) into the From Recorder Outputs terminals and monitor the level on the console's meters. Presumably they will read 0. Alternatively, connect a high-impedance voltmeter (VM) across the dbx unit's To Console Inputs (+) and (-) terminals to check the levels. Push in the Noise Reduction button for the appropriate channel and again adjust the Play trim until the console or VM reads 0. Since the play trims are separate, you can also use this procedure to balance the play channels to each other, or to compensate for a channel-imbalanced source being decoded. (Should you ever need to rebalance the record channels, there is an internal trim, R111, for Channel 2. See sector E7 of the schematic.)

Recording Levels

With today's hotter tapes and faster meters, we believe there are no longer any hard and fast numbers about maximum recording levels with dbx noise reduction. Generally, with decks of modern manufacture and good tape, recording levels should always be as high as is consistent with clean sound. This means that peaks almost invariably should go well above the deck's nominal 0 -- indeed, above its +3, the end of the meter range for many decks -- depending on the dynamic range and especially the spectrum of the program material. Synthesizer, female chorus, brass, percussion -- music with considerable high-frequency energy, transients, and the greatest peak-to-average ratios -- naturally will require close attention to the meters and more prudent settings. On the other hand, electric guitar, chamber music and small-ensemble jazz, piano, strings, male vocals, and any material that has been compressed or limited beforehand may usually be put on the tape at healthy, high levels. Since so many decks' meters stop at +3 (and have different 0 levels) and since meter time constants vary so much, we can't even suggest a number --

+6? +8? +10? Keen monitoring is the key (and your only choice, really). As mentioned, the program's spectral content will govern success in choosing levels more than any other factor. Note that as fast, "peak"-reading meters become more common on consoles, decks, etc., your recording-level numbers will change. dbx-encoded peaks on such meters shouldn't exceed the medium's headroom.

The reason for recording at the hottest level possible (or, in broadcasting, maintaining high signal levels short of overmodulation) is that doing so keeps the signal as far as possible above the noise floor of the transmission medium. This in turn minimizes the only potential drawback of aggressive, full-bandwidth companding that produces as extremely quiet results as dbx does: audible noise modulation in the absence of masking. This phenomenon, sometimes a faint "sshhh" sound accompanying a low-level sound with little or no high-frequency content (e.g., male speech, or solo, dryly miked bass or piano), is most likely to occur when the transmission medium is very noisy and/or the noise is close to the signal. Transmission lines, audio cassette decks, and VCRs (including U-Matics) are considerably more likely to exhibit such problems than open-reel recorders, because their inherent dynamic-range figures are often so poor -- 45-55 dB, for example. (Note that C-type VTRs, using open reels of 1-inch tape and usually having considerable headroom above a relatively low-set 0, are much less prone to noise modulation if recording levels are kept high.)

Fortunately, the vast majority of the material that might otherwise present noise-modulation problems can be recorded at healthy high levels without saturating the tape, and if the deck is quiet enough, there will be no audible noise modulation on any material. Finally, most of the musical material recorded and broadcast today -- ensemble pop/rock, for one example -- can be recorded perfectly cleanly on a wide variety of decks without a trace of noise added.

The point of all this, again, is to use and trust your ears in setting recording levels. How a recording or a broadcast sounds is more important than any specific level figures -- in fact, it's the only important criterion. Just try starting out hotter than you may be accustomed to in the past, with other noise-reduction systems (or none) -- and back off only if you hear something you don't like, not as a matter of "policy." Do note that your meters will <u>deflect</u> less than you are used to, because the signal hitting the tape is compressed 2:1.

Don't forget to mark your tapes "Encoded with dbx I [or II]," since undecoded playback of encoded tapes or decoded playback of unencoded tapes is not much fun.

<u>Alignment tones</u>

Any tones you customarily use at the head of a tape (for HF or LF EQ, azimuth adjustment, levels, etc.) should <u>not</u> be encoded.

<u>Mixing</u>

All mixing must be done with decoded tracks only. Never mix encoded channels together -- the decoder will not track them properly.

<u>Applications other than recording and broadcasting</u>

dbx Type II often is well-suited to reducing the noise of echo/delay/reverberation lines (bucket-brigade, digital, "ambience" systems, etc.). Experimentation is called for; frequently the result is a startling improvement in quietness.