MODEL 150X

Type I
Noise Reduction

dbx®

SERVICE MANUAL
(Preliminary)

CAUTION	AVIS
RISK OF ELECTRIC SHOCK DO NOT OPEN	RISQUE DE CHOC ELECTRIQUE NE PAS OUVRIR

DO NOT EXPOSE TO RAIN OR MOISTURE

CAUTION: TO REDUCE THE RISK OF ELECTRICAL SHOCK, DO NOT REMOVE COVER (OR BACK). NO USER SERVICEABLE PARTS INSIDE. REFER SERVICING TO QUALIFIED SERVICE PERSONNEL.

WARNING: TO REDUCE THE RISK OF FIRE OR ELECTRICAL SHOCK, DO NOT EXPOSE THIS APPLIANCE TO RAIN OR MOISTURE.

This symbol, wherever it appears, alerts you to the presence of uninsulated dangerous voltage inside the enclosure — voltage that may be sufficient to constitute a risk of shock.

This symbol, wherever it appears, alerts you to important operating and maintenance instructions in the accompanying literature. Read the manual.

Manufactured under one or more of the following U.S. patents: 3,377,792; 3,681,618; 3,714,462; 3,789,143; 4,097,767; 4,329,598; 4,403,199; 4,409,500; 4,425,551; 4,473,795. Other patents pending.

This dbx-branded product has been manufactured by AKG Acoustics, Inc.

This manual is part number 96017-000-02

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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.

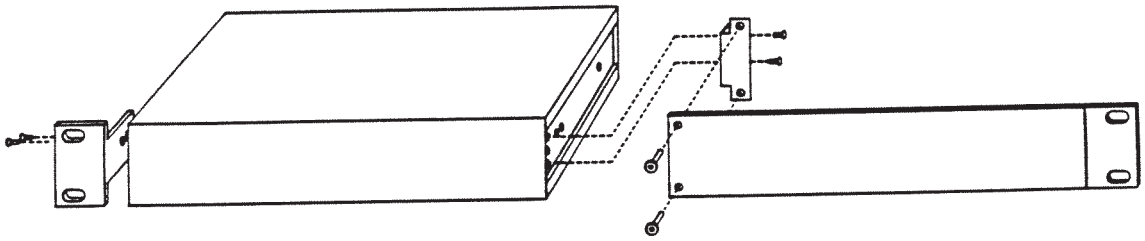
It's a good idea in any case to save the carton and packing should you ever need to ship the unit.

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

The carton should contain this owner's manual, a 150X of course, and a warranty/registration card. Please fill the card out and send it to us. The carton also should contain hardware for rack-mounting both a single unit (screws, a long ear [half-rack width], a small L-bracket, and a short rack ear) and two units together (side plates along with a screwdriven joiner). See below.

No special ventilation is required in any installation; other components may be stacked above or below your 150X provided they don't generate excessive heat.

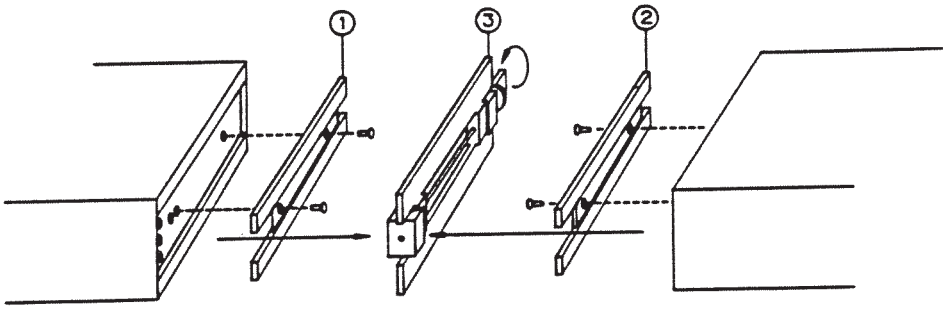
Here's rack-mounting for a single unit:



It may help to line everything up on a table as you tighten the screws.

Here's rack-mounting for a pair of 150Xes or for a 150X and dbx 163X Compressor/Limiter, 263X De-Esser, or 463X Noise Gate or other "-63X" series units:

1 & 2) Attach side panels;
3) Bring units together, lining up the side panels with the screw-joiner catches, and then gently tighten the screw to close the catches.

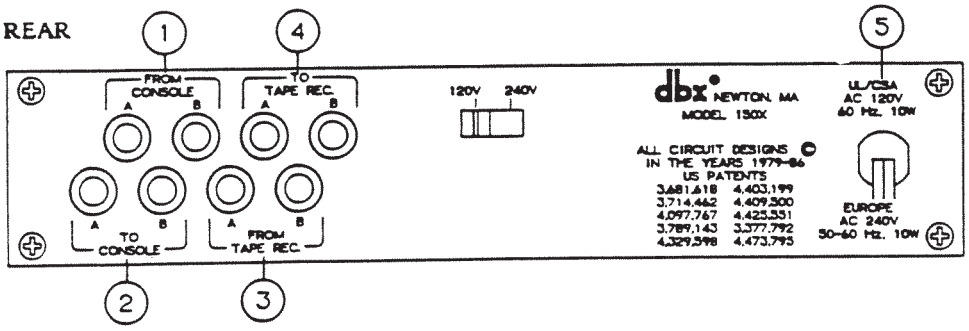


SPECIFICATIONS
(for single-ended, "unbalanced" 150X operation except as noted)

Effective noise reduction	40 dB or more, depending on transmission medium
Frequency response	±0.5 dB 30 Hz-20 kHz, -1 dB at 20 Hz
Dynamic range	115 dB balanced, 112 dB single-ended
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	12 V; will drive 600 ohms to +24 dBv balanced, +21 dBv single-ended
Input	75 k-ohms, balanced
Output	Low-impedance (22 ohms), designed to drive 600 ohms or greater
Level range for unity gain (level match)	Set at 316 mV, adjustable 50 mV-2V (-24 to +10 dBv)
Power requirements	See rear of unit

Notes
1) Specifications are subject to change.
2) All voltages are rms (root-mean-square). 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."
3) Dynamic range is defined as the difference between the maximum 1 kHz rms signal and unweighted noise. Other noise figures are for 20 Hz-20 kHz, also unweighted. A-weighting will improve all of these figures by a few dB.
4) Frequency-response figures are for pink noise.
5) THD and IMD measurements are for total encode/decode processing. SMPTE IMD is measured with 60 and 7k Hz mixed 4:1; IHF (difference-tone) IMD is measured with 19 & 20 kHz mixed 1:1; output 1 V.
6) Inputs and outputs have identical polarity.

REAR



1 FROM CONSOLE

These are the Encoder Inputs. Connect them to the appropriate bus, line, or other outputs on the mixer/console or tape deck.

2 TO CONSOLE

These are the Decoder Outputs. Connect them to the appropriate inputs on the mixer/console.

Note that with some mixing equipment Tape Out may be called Tape Rec and Tape In called Tape Play or Monitor. There are other nomenclature variations as well.

3 FROM TAPE REC(ORDER)

These are the Decoder Inputs. Connect them to the tape deck's outputs.

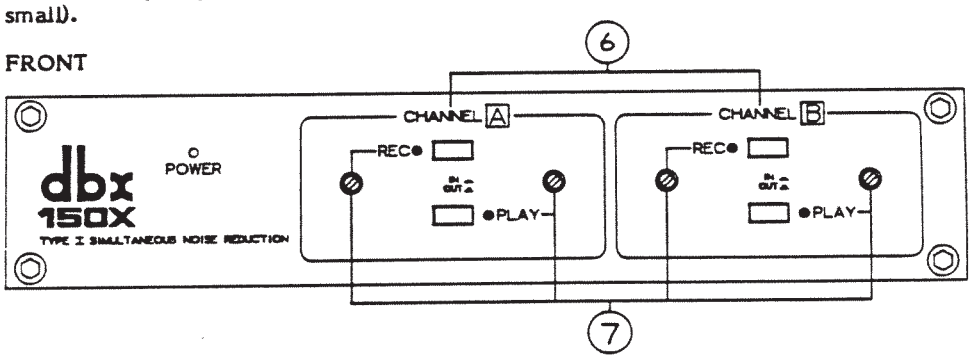
4 TO TAPE REC(ORDER)

These are the Encoder Outputs. Connect them to the tape deck's line inputs.

5 AC POWER

Connect this cord to the approriate power source; check that the 120/240 V switch is correctly set. Note that your 150X is not equipped with an On/Off switch, so you may plug it into either a switched outlet on your equipment rack (be advised that the unit will make a thump on power up or down) or, better, an unswitched outlet (the power draw is small).

FRONT



6 CHANNEL A (or B): REC and PLAY, IN or OUT

Pushing the Rec button in engages the encoder of that channel. The Out position is a hardwire bypass, outputs connected directly to inputs.

Pushing the Play button in engages the decoder of that channel. The Out position here is also a hardwire bypass.

7 TRIMPOTS (LEVEL ADJUSTMENTS)

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

HOOKUPS and CABLING

Inputs and balanced and unbalanced sources

The 150X is a 600-ohm, balanced (differential) unit designed for nominal +4 levels; inputs and outputs are tip/ring/sleeve phone jacks. 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. The outputs are also electronically balanced, making it possible to use the 150X with either balanced or unbalanced loads (see next page).

In other words, you can mix and match balanced and unbalanced equipment with the 150X — it will work with virtually anything, provided you use the proper cabling. The only rule is to be sure not to short out the outputs, as we'll stress presently.

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

For balanced connections, any cable that's called "balanced mike cable" should do, e.g., Belden's part no. is 8412 and Radio Shack's is 278-1276 (the Radio Shack plug nos. are 274-285 or 274-1546). Switchcraft sells cable and plugs assembled: 10BD10 (2'), 10BF10 (3'), 10BK10 (6'), 10BN10 (10'), or 10BU10 (25').

Figures 1a & b show the connection of balanced signal sources, and Figures 2a, b, & c (next page) show unbalanced sources connected. Note that for proper operation from an unbalanced source, each minus terminal at the inputs must be connected to a ground terminal. (Also note that other terms for plus, minus, and ground are high [or hot], low, and shield.)

Fully balanced 150X operation with consoles OR tape decks

Figure 1a: 3-circuit ("stereo") plugs and dual-conductor shielded cable

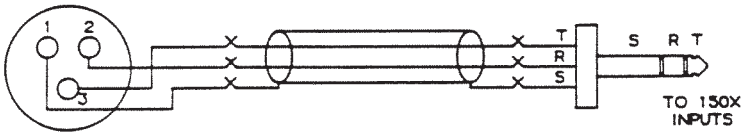


Figure 1b: As above but with XLR jack (shown from rear)

Single-ended operation with 150X inputs only

Figure 2a: 2-circuit ("mono") plugs and single-conductor shielded cable

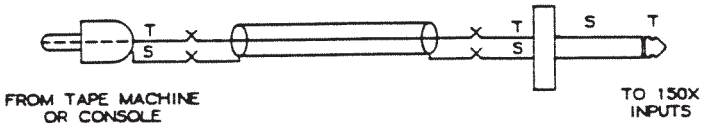


Figure 2b: As above but with RCA plug at source

Figure 2c: As in 2a but with 3-circuit ("stereo") plug at 150X input

Outputs and balanced and unbalanced loads

The two sets of outputs, as mentioned, are driven by electronically balanced line amplifiers whenever their channel is In. It's possible to use the 150X to drive either balanced or unbalanced loads.

Figures 3a & b show the connection to balanced inputs. Figures 4a & b show the connection to unbalanced inputs; either the tip or the ring can be used (tip "+" is normal), depending on the desired polarity ("phase") of the output. Again be sure not to short the unused output to ground.

In fact, you must take care NEVER to short out either polarity of output to ground. For example, if you're running unbalanced and inadvertently reverse or switch the wires (center conductor and shield) somewhere internally — say, when you're making up a new plug at the other end — you'll short out the 150X. Do not make this mistake. In the case of the 150X output jacks, for example, all wiring MUST use tip/ring/sleeve phone plugs.

Note 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. However, when mixing balanced and unbalanced inputs and outputs, pushing Bypass may change the connections -- so the easiest course may be to maintain either balanced or unbalanced connections throughout.

Fully balanced 150X operation with consoles OR tape decks

Figure 3a: 3-circuit ("stereo") plugs and dual-conductor shielded cable

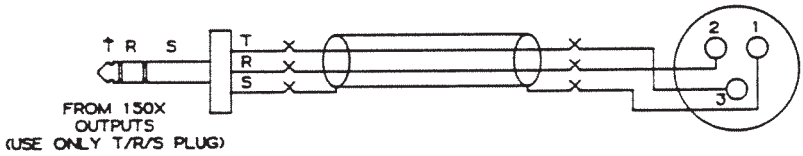


Figure 3b: As above but with XLR plug (shown from rear)

Single-ended operation with 150X outputs only

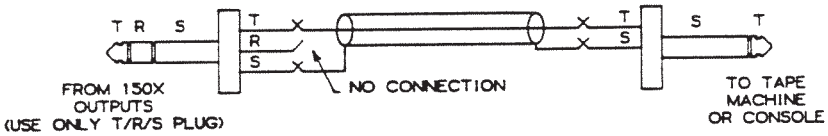


Figure 4a: 2-circuit ("mono") plugs and single-conductor shielded cable

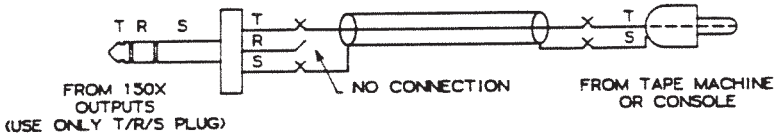


Figure 4b: As above but with RCA plug

ABOUT YOUR UNIT AND dbx TYPE I (AND II) NOISE REDUCTION

The model 150X is a 2-channel record/play (simultaneous encode/decode) noise-reduction unit for linear transmission media with 20 Hz-20 kHz ±1 dB response up to high input levels. Principally this means tape decks operating at 15 inches per second (ips).*

Each channel's encoder and decoder can be used independently at the same time, so full (decoded) monitoring is possible, as is separate use of the encode and/or decode halves for mastering with a second machine. The full record/play cycle results in a doubling of usable dynamic range with or without signal present, to a maximum of 112 or 115 dB. This dynamic range exceeds 16-bit PCM systems and all other analog noise-reduction systems, even the new Dolby SR compander. Depending on the individual channel noise of the tape machine, this 150X performance translates into at least 40 dB of noise reduction. With a good enough deck, any last distinction between analog and digital is eliminated.

Such a feat is achieved by compressing the signal by a 2:1 ratio and applying a carefully tailored frequency-response preemphasis during record, and expanding 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 noise reduction is for professional use, as mentioned, with tape machines of flat frequency response and full headroom maintained at high frequencies. Type II, the format of the dbx 140A, 224X, and the 941A/942A (and their predecessors), 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 since 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 (as in the 150X) 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 HF-headroom limitations. The frequency response of the processing circuitry does <u>not</u> restrict the bandwidth of the audio signal passed.

(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.)

The benefit of the 2:1 compression, of course, is that the signal becomes easier 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 <u>not</u> 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,

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*It's possible to use the 150X with a 7-1/2 ips deck if it is a <u>very</u> good one (no significant bass head bumps, for example) <u>and</u> EE tape is used, giving increased headroom. The 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 wouldn't mistrack with Type II), as might "flat" decks that are misbiased or have azimuth problems.

the Blackmer rms detector and voltage-controlled amplifier (VCA). The former enables optimum decode tracking and transient response despite the phase shifts typical of tape recorders. The latter affords precise gain control over an extremely wide dynamic range while maintaining very low noise and distortion.

It's sometimes claimed that 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 measurements (sine waves), yield performance that looks identical, they'll mistrack with transient/dynamic material. The success of the dbx systems in coping with imperfections in the storage and transmission media is due largely to our proprietary rms-level detectors; do not be misled 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. 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 as necessary) and to keep levels about the same with and without Type I 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 input jacks. 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 Rec In and turn the 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 lineup in your studio, you can continue to do so, but use the same-frequency tone throughout. Since the record trims are separate, you can also use this procedure before the tape deck's own recording-level meters to balance the recording levels to each other or to compensate for a channel-imbalanced source being encoded.

Play

With the dbx unit in Bypass, send a 1-kHz reference-level tone (from an alignment tape or other source) into the From Tape Recorder terminals and monitor the level on the console's meters. Presumably they will read 0. Push in the Play button for the appropriate channel and again adjust the trim until the console 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.

Recording Levels

Using 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 and tape of modern manufacture and Type I, recording levels should always be as high as is consistent with clean (undistorted, unsaturated) sound. This means that peaks almost invariably should be well above your deck's nominal 0 — indeed, above its +3, which for many decks is the end of the meter range — depending solely on the dynamic range and especially the spectrum of the program material.

Music with great high-frequency energy, transients, and the greatest peak/average ratios — synthesizer, female chorus, brass, percussion, for example — naturally will require close attention to meters and more prudent settings. On the other hand, electric guitar, chamber music and small-ensemble jazz, piano, strings, male vocals, plus material that's been compressed or limited beforehand, may be put on the tape at good healthy 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? Disregard the meter scale, then; keen monitoring is the key, and the only real choice. As mentioned, the program's spectral content and dynamic range will govern success in choosing levels more than any other factor. And note that as fast, so-called "peak"-reading meters become more common on consoles and decks, your recording-level numbers will change. dbx-encoded peaks on such meters should never exceed the medium's headroom.

The reason for recording at the hottest feasible level is that doing so keeps the signal as far as possible above the noise floor of the medium. And 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. (With Type II, transmission lines, audio cassette decks, and VCRs, including U-Matics, are considerably more likely to exhibit such problems than open-reel recorders with Type I, because the former's inherent dynamic-range figures are often so poor — 45-55 dB, for example.)

Fortunately, the vast majority of the material that might otherwise present noise-modulation problems can be recorded with Type I on a good machine at high levels without saturating the tape. Note that if levels are hot enough and if your deck is properly biased and quiet enough, there will be no audible noise modulation on <u>any</u> material. And fortunately, the majority of the musical material recorded today — ensemble pop/rock, for one example — can be recorded perfectly cleanly with Type I 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 sounds is all that's important. Just try starting out hitting the tape higher than you may have been accustomed to in the past, with other noise-reduction systems, or none. Back off only if you hear distortion or saturation, not as a matter of "policy." Do realize 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," 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 — the decoder will not track them properly. Likewise, nothing must come between the 150X and your deck, for example, no equalization or other signal processing.

<u>PCM digital</u>

Some recordists may wish to investigate using their 150X with a 16-bit or (especially) a 14-bit PCM digital recorder. In effect, the 150X will adequately dither such a machine, as well as increase its usable dynamic range some 20 dB and thereby change any so-called "digital sound" its noise floor has. In such a combination recording system there will be no mistracking and no possibility of audible noise modulation.

dbx model 150x test and alignment procedure.

I. Pre-calibration procedure.

Plug the unit to be tested into a rated source of ac mains voltage and frequency.

Cycle the noise reduction in/out switches and confirm that the noise reduction "in" led-indicators illuminate when the switch is pushed to the "in" position and extinguish when the switch is in the "out" position. Verify that the operation of the switches is smooth and free of any unusual binding etc. Verify that the power indicator is illuminated.

monitor the output voltages at TP 3 and TP 4 and verify that they are within +/- 600 mV of the nominal +/- 15 vdc, respectively.

Place all calibration trim pots to the center of their rotation. Rotate each front panel level match control through its entire range and verify that the operation of the control is smooth and free of any unusual binding, etc. Place each front panel control to its center position.

II. Decoder calibration proceedure

2.1 Connect a low output impedance signal generator to the J1 input of the unit to be tested. Set the output of the signal generator to a magnitude of 2 vrms ac at a frequency of 50 hz. Monitor the output of U4 at pin 7, (TP 1) using a suitable oscilloscope having a minimum vertical sensitivity of 5 mV/c.m. Adjust R41 for the best possible symmetry of the 100 hz waveform at TP 1. set the signal generator to a frequency of 1 Khz and Decrement the output level of the signal generator in 10 db steps from 2.0 v rms to 200 mV rms. Verify that proper rms detector tracking follows at +6 mV/db. verify that the output of U3 is within +/- 12 mv dc of 0.000 volts dc at an input amplitude of 200 mV rms. Continue to decrement the output of the signal generator from 200 mV rms to 2 mV rms and again verify that the rms detector properly tracks at 6 mV/db.

2.2 Dis-connect the oscilloscope from TP 1 and connect TP 1 to TP 5 with a suitable clip lead (thus shorting the output of U4 to ground). Monitor the output of U3 and U5 at J2 using a suitable a.c. voltmeter. Perform spot checks of the frequency response at test frequencies of 30 hz, 300 hz, and 3 khz and verify that the measured response is within +/- 1 db of the tabulated nominal response indicated in table 1.

1.5db

2.3 Dis-connect the signal generator from the input of the unit under test. Set the output of the signal generator to an amplitude of 84 mV rms at a frequency of 100 hz and re-connect the signal generator to TP 6. (* the dc output offset of the signal generator used for this part of the proceedure must be less than 3 mV) Monitor the output of U3 with respect to ground at J2 using an ac coupled oscilloscope set to a minimum vertical sensitivity of 5 mV/c.m. Adjust R15 for minimum control voltage feedthrough. Remove the cliplead from TP 1 to TP 5. measure the dc ouput offset voltages at the outputs of U3 and U5 with respect to ground and verify that they are less than +/- 10 mV dc.

2.4 Set the output of the signal generator TO 5KHZ ∞ an amplitude of 316 mV rms. Adjust R9 for 316 mV rms at the output of U3 at J2 with respect to ground.

2.5 Repeat paragraphs 2.1 through 2.4 for the alternate decode channel.

III. Encoder calibration proceedure

3.1 Connect a low output impedance signal generator to the J1 input of the unit to be tested. Set the output of the signal generator to a magnitude of 2 vrms ac at a frequency of 50 hz. Monitor the output of U4 at pin 7, (TP 1) using a suitable oscilloscope having a minimum vertical sensitivity of 5 mV/c.m. Adjust R41 for the best possible symmetry of the 100 hz waveform at TP 1. set the signal generator to a frequency of 1 Khz and Decrement the output level of the signal generator in 10 db steps from 1.0 v rms to 100 mV rms. Verify that proper rms detector tracking follows at + 3 mV/db. verify that the output of U3 is within +/- 12 mv dc of 0.000 volts dc at an input amplitude of 100 mV rms. Continue to decrement the output of the signal generator from 100 mV rms to 1 mV rms and again verify that the rms detector properly tracks at 3 mV/db.

3.2 Dis-connect the oscilloscope from TP 1 and connect TP 1 to TP 5 with a suitable clip lead (thus shorting the output of U4 to ground). Monitor the output of U3 and U5 at J2 using a suitable a.c. voltmeter. Perform spot checks of the frequency response at test frequencies of 30 hz, 300 hz, and 3 khz and verify that the measured response is within +/- tk db of the tabulated nominal response indicated in table 1.

3.3 Dis-connect the signal generator from the input of the unit under test. Set the output of the signal generator to an amplitude of 84 mV rms at a frequency of 100 hz and re-connect the signal generator to TP 6. (* the dc output offset of the signal generator used for this part of the proceedure must be less than 3 mV) Monitor the output of U3 with respect to ground at J2 using an ac coupled oscilloscope set to a minimum vertical sensitivity of 5 mV/c.m. Adjust R15 for minimum control voltage feedthrough. Remove the cliplead from TP 1 to TP 5. measure the dc ouput offset voltages at the outputs of U3 and U5 with respect to ground and verify that they are less than +/- 10 mV dc.

3.4 Set the output of the signal generator to 5KHZ @ an amplitude of 316 mV rms. Adjust R9 for 316 mV rms at the output of U3 at J2 with respect to ground.

3.5 Repeat paragraphs 3.1 through 3.4 for the alternate encode channel.

IV. Back-to-back Performance tests, reference encoder/decoder performance tests.

4.1 the following connections will be made utilizing standard 1/4 " two conductor phone jacks and cabling. Connect the encoder outputs designated J2 and J4 to a reference decoder. Connect the encoder inputs designated J1 and J3 to the output of a tone burst generator. Apply a toneburst of 8 cylcles on (at 0 dbV) followed by 128 cycles off (less than -40 dbV) at a test frequency of 1 Khz. Monitor the outputs of the reference decoder and verify that there is not more than 20 % overshoot on the first cycle of the decoded waveform and no overshoot on the remaining cycles.

4.2 Connect the decoder inputs designated J1 and J3 to the output of a reference encoder. Connect the input of the reference encoder to the tone burst generator as described in paraggraph 4.1. Monitor the outputs of the decoder under test at J2 and J4. Verify that there is not more than 20 % overshoot of the output waveform for the first cycle, and no overshoot on the remaining cycles

4.3 Connect the encoder output designated J2 to the decoder input designated J1. Connect the encoder output designated J4 to the decoder input designated J3. Connect the encoder inputs to a suitable low distortion signal generator. Monitor the decoder output designated J2 with a sui table distortion analyzer. Measure the

thd at test frequencies of 100 hz and 10 kHz and verify that it is not greater than .15 % at a test amplitude of 1.0 v rms. Repeat this measurement for the decoder output designated J4.

4.4 Temporarily bypass the noise reduction circuitry utilizing the noise reduction in/out switches and verify that the signal bypasses both the encoder and decoder functions. Re-instate the noise reduction circuitry to the "in" mode. Sweep the signal generator from 40 hz to 20 kHz and verify that the amplitude vs frequency response is flat within +/- 1 dB.

4.5 Terminate the encoder inputs designated J1 and J3 with 1 K ohm resistors to ground. Measure the output noise of the decoder channels at J2 and J4 in a 20 to 20kHz "A" weighted bandwidth and verify that it is not greater than -90 dBv, RE: .775 VRMS

.150X TEST PROCEDURE
TABLE 1

+.75
-.25

File: 150X.ENC 18:29:28 04-22-1986 ENCODER
Output node: 0

Part	Nodes	Val	Tol

Freq, Hz	Mag, dB Calc	Mag, dB Des	Mag, dB Err	Phase, deg Calc	Phase, deg Des	Phase, deg Err
30	11.61	11.61	0.00	64.90	64.90	0.00
300	12.98	12.98	0.00	32.34	32.34	0.00
3000	22.48	22.48	0.00	11.78	11.78	0.00

Rms = 0 dB Best = 0 dB Itera = 0

3.8 +.342/-.107
4.46 +.402/-.126
13.3 +1.2/-.377

File: 150X.DEC 18:30:02 04-22-1986 DECODER
Output node: 0

Part	Nodes	Val	Tol

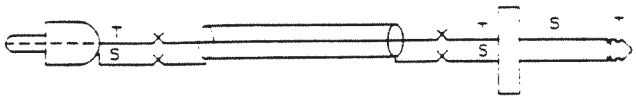
Freq, Hz	Mag, dB Calc	Mag, dB Des	Mag, dB Err	Phase, deg Calc	Phase, deg Des	Phase, deg Err
30	10.73	10.73	0.00	5.52	5.52	0.00
300	6.78	6.78	0.00	-27.34	-27.34	0.00
3000	-0.76	-0.76	0.00	-17.01	-17.01	0.00

Rms = 0 dB Best = 0 dB Itera = 0

150X

Field Service Alignment Procedure

Equipment required:	Oscillator	Krohn Hite 4200 or equivalent
	Voltmeter	Simpson 464 or equivalent
	Oscilliscope	Philips PM 3232 or equivalent

From Oscillator output  To 150X Input

Single-ended operation with 150X inputs only

1. Power Supply Test

   +15vdc +/- 600 mv at TP3
   -15dvc +/- 600 mv at TP4

2. RMS Symmetry Alignment
   Set Oscillator to 316 mv rms (-10dbv) at 50 Hz

   Connect Oscillator to J1 and J3 of the play (decode) board.
   Engage both channel A and channel B play switches.

   Connect Oscilloscope to TP1.
   Adjust R41 for even RMS waveform peaks. (see figure A)

   Connect Oscilloscope to TP2.
   Adjust R86 for even RMS waveform peaks.

   Connect Oscillator to J1 and J3 of the record (encode) board.
   Engage both chanel A and channel B play switches.

   Connect Oscilloscope to TP1.
   Adjust R43 for even RMS waveform peaks.

   Connect Oscilloscope to TP2.
   Adjust R90 for even RMS waveform peaks

3. Level Match Alignment

Set Oscillator to 316 mv RMS (-10dbV) at 1K Hz

Connect Oscillator to J1 and J3 of the play board.
Adjust channel A play level pot (R9) for 316 mv RMS at pin 7 of U3.
Check that the signal at pin 6 of U5 is 316 mv RMS +/- 15mv.

Adjust channel B play level pot (R54) for 316 mv RMS at pin 7 of U7.
Check that the singal at pin 6 of U9 is 316 mv RMS +/- 15mv.

Connect Oscillator to J1 and J3 of the record board.
Adjust channel A record level pot (R11) for 316 mv RMS at pin 7 of U3.
Check that the signal at pin 6 of U5 is 316 mv RMs +/-15mv.

Adjust channel B record level pot (R58) for 316 mv RMS at pin 7 of U8.
Check that the signal at pin 6 of U10 is 316mv RMS +/- 15mv.

4. VCA Symmetry Alignment

NOTE: The dc output offset of the Oscillator used for this part of the procedure must be less than 3mv.

Disconnect the Oscillator from the unit.
Set Oscillator to 84 mv rms at 100 Hz.

Play board

Connect a jumper between TP1 and TP5.
Connect Oscillator to TP6.
Connect Oscilloscope to pin 7 of U3.
Adjust R15 for flattest possible waveform. (see figure B)
Remove jumper between TP1 and TP5.
Check that the dc offset at pin 7 of U3 is 0vdc +/-10mv.

Connect a jumper between TP2 and TP5.
Connect Oscillator to TP7.
Connect Oscilloscope to pin 7 of U7.
Adjust R60 for flattest possible waveform.
Remove jumper between TP2 and TP5.
Check that the dc offset at pin 7 of U7 is 0vdc +/-10mv.

Record board

Connect a jumper between TP1 and TP5.
Connect Oscillator to TP6.
Connect Oscilloscope to pin 7 of U3.
Adjust R16 for flattest possible waveform.
Remove jumper between TP1 and TP5.
Check that the dc offset at pin 7 of U3 is 0vdc +/-10mv.

Connect a jumper between TP2 and TP5.
Connect Oscillator to TP7 of U8.
Adjust R63 for flattest possible waveform.
Remove jumper between TP2 and TP5.
Check that the dc offset at pin 7 of U8 is 0vdc +/-10mv.

DIAGRAM A

UNCALIBRATED RMS WAVEFORM

PROPERLY CALIBRATED RMS WAVEFORM

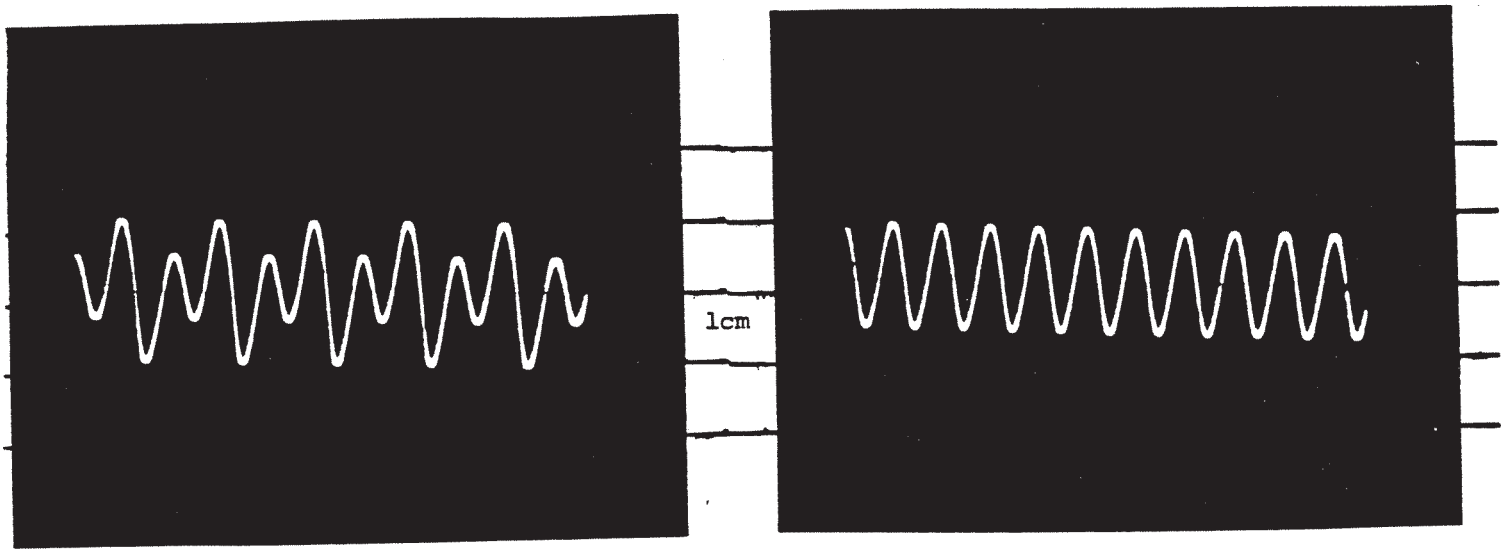
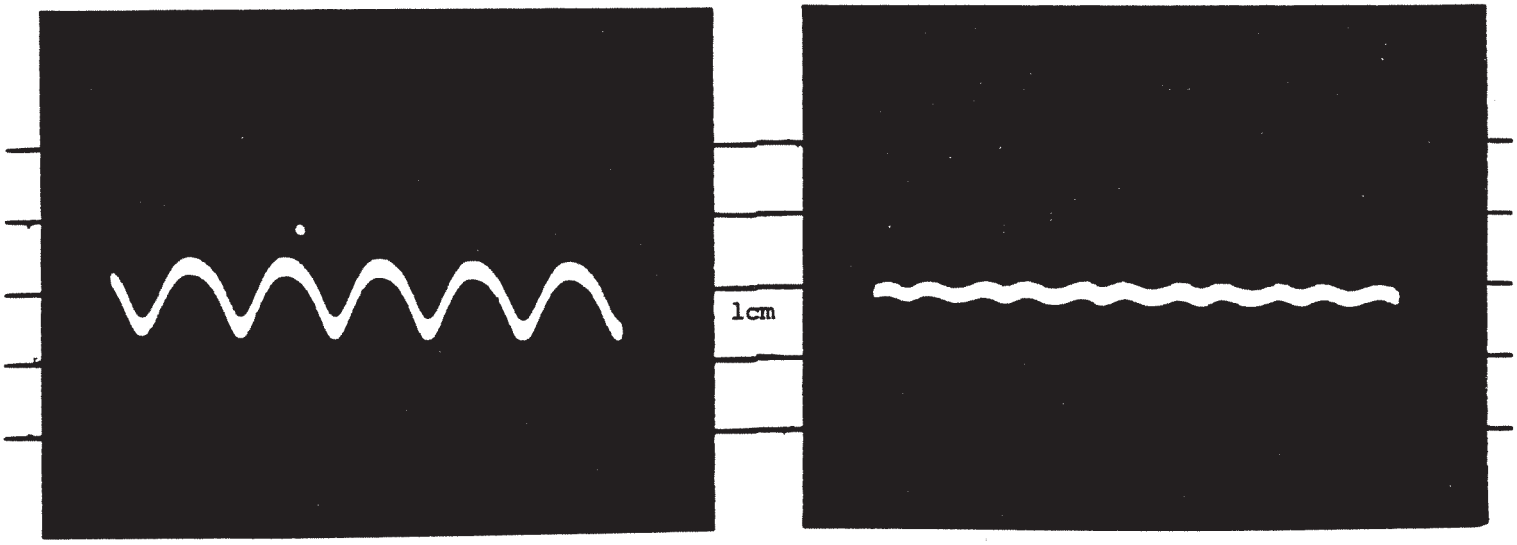


DIAGRAM B

UNCALIBRATED VCA SYMMETRY

PROPERLY CALIBRATED VCA SYMMETRY



OSCILLOSCOPE SETTINGS FOR
VCA SYMMETRY ADJUSTMENTS
TIME = 5ms/cm
AMPL = 2mv/cm