



INSTRUCTION MANUAL

DL2 Distinction Stereo Controller

Includes Service Section

CROWN INTERNATIONAL, INC. 1718 W. MISHAWAKA RD. ELKHART, INDIANA 46514

The information furnished in this manual does not include all of the details of design, production, or variations of the equipment. It does not cover all the possible contingencies which may arise during operation, installation, or maintenance. Should special problems arise, or further information be desired, please contact the Crown International Customer Services Department.

Crown International
1718 W. Mishawaka Rd.
Elkhart, Indiana 46517
Ph: (219) 294-5571

WARNING

**TO PREVENT SHOCK OR FIRE HAZARD DO NOT EXPOSE TO
RAIN OR MOISTURE!**

CAUTION

**TO PREVENT ELECTRIC SHOCK DO NOT USE THIS
(POLARIZED) PLUG WITH AN EXTENSION CORD,
RECEPTACLE OR OTHER OUTLET UNLESS THE BLADES
CAN BE FULLY INSERTED TO PREVENT BLADE EXPOSURE.**

ATTENTION

**POUR PREVENIR LES CHOCS ELECTRIQUES NE PAS
UTILISER CETTE FICHE POLARISEE AVEC UN
PROLONGATEUR. UNE PRISE DE COURANT OU UNE AUTRIE
SORTIE DE COURANT, SAUF SI LES LAMES PEUVENT ETRE
INSEREES A FOND SANS EN LAISSER AUCUNE PARTIE A
DECOUVERT.**



TABLE OF CONTENTS

Section 1 General Introduction

1.1	Introduction	1-1
1.2	Purpose of Equipment	1-1
1.3	Unpacking and Inspecting Equipment	1-1
1.4	Units and Accessories Supplied	1-1
1.5	Service Policies	1-2
1.6	Glossary of Terms	1-2

Section 2 Specifications and Performance

2.1.1	Control Module	2-1
2.1.2	Phono Module A	2-2
2.1.3	Power Supply	2-2
2.2	Performance Graphs	2-3

Section 3 Installation and Operation

3.1	General Information	3-1
3.2	Installation	3-1
3.2.1	Rack Mounting	3-1
3.2.2	Mounting the Phono Module	3-1
3.3	Interconnecting Modules	3-2
3.3.1	Connecting the Phono Module	3-2
3.4	Rear Panel Control Center Connections and Functions	3-2
3.5	Front Panel Controls and Functions	3-4
3.6	DL2 Phono Module	3-6
3.7	DL2 Power Module	3-7
3.8	How to Use Audio Imaging Controls	3-7
3.9	Volume and Loudness Compensation	3-8
3.10	Remote Control	3-12
3.11	Computer Control	3-12

Section 4 Theory of Operation

4.1	General	4-1
4.2	Principles of the DL2 Controller	4-1
4.3	Block Diagram Circuit Theory	4-1
4.4	Detailed Circuit Theory	4-3
4.5	Schematic and Board Layout Drawings	4-6

Section 5 Accessories/Options

5-1

Section 6 Parts List

6.1	General Information	6-1
6.2	Standard and Special Parts	6-1
6.3	Ordering Parts	6-1
6.4	Component Part Listings by Module	6-2
	Gain Module	6-2

Logic Module	6-5
Loudness Module	6-7
Tone Output Module	6-8
Tone Switch Module	6-10
Display Module	6-11
Panorama Module	6-12
Tape Monitor Module	6-12
5V Regulator Module	6-12
Output Module	6-12
Output Switch Module	6-13
Input Selector Module	6-13
Input Module	6-13
General Miscellaneous	6-14
Power Supply Module	6-15
Phono Module	6-16

Section 7

Maintenance

7.1 Introduction	7-1
7.2 Required Test Equipment	7-1
7.3 Visual Inspection	7-1
7.4 Troubleshooting and Repair	7-1
7.5 Disassembly Procedure	7-4
7.6 Reassembly	7-9
7.7 Electrical Checkout	7-9
7.8 Changing AC Mains Voltage	7-11

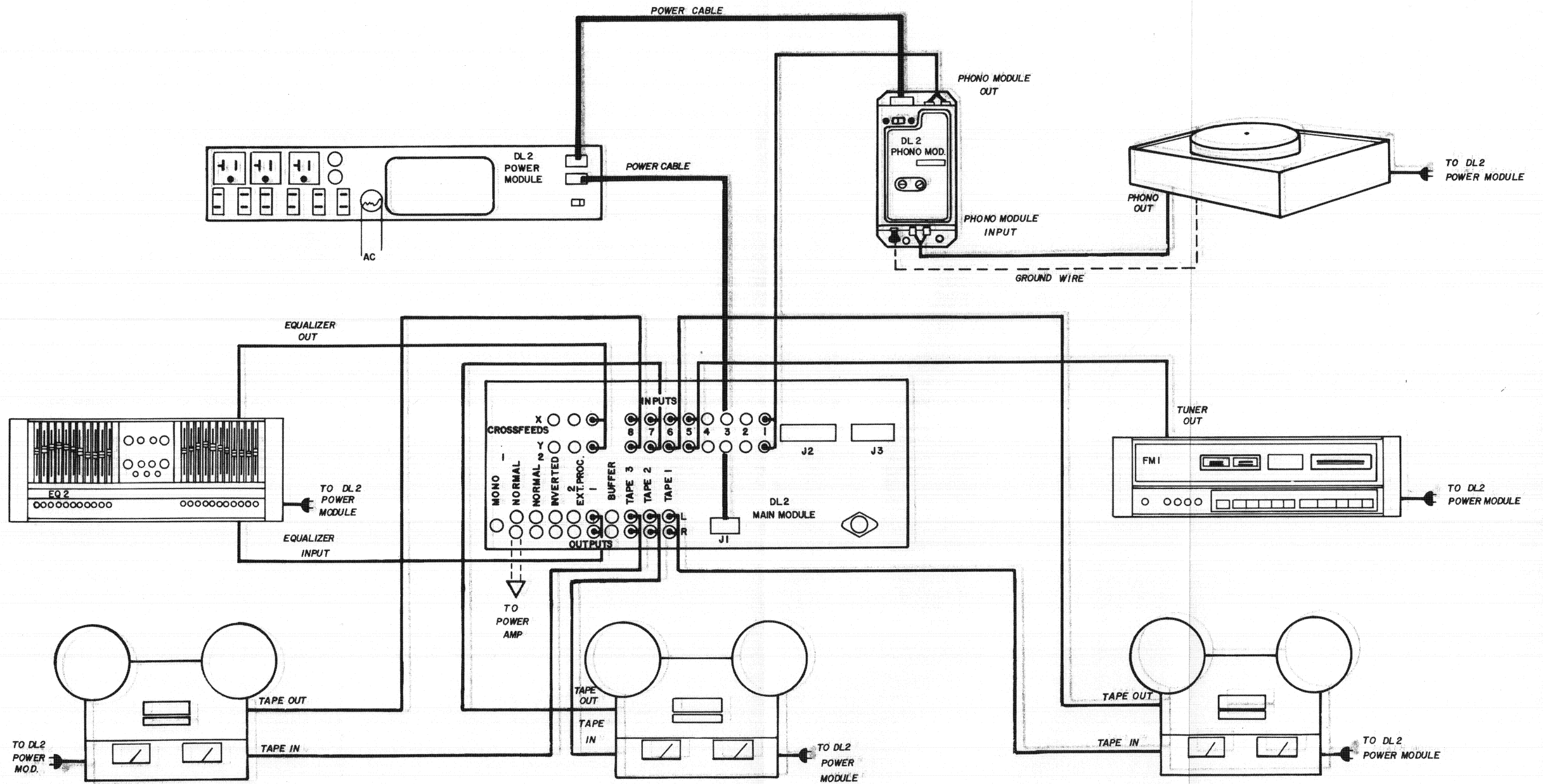


LIST OF ILLUSTRATIONS

Fig. 1.1	DL2 Control Center System	1-1
Fig. 1.2	Crown Care Card	1-2
Fig. 2.1	Maximum Output IHF Load	2-3
Fig. 2.2	Frequency Response, Main Control Module	2-3
Fig. 2.3	Phase Response, Main Control Module	2-4
Fig. 2.4	IM Distortion	2-4
Fig. 2.5	Low, High Filter Response	2-5
Fig. 2.6	Frequency Response, Headphone Amplifier	2-5
Fig. 2.7	Phase Response, Phono Module	2-6
Fig. 2.8	RIAA Phono Module Frequency Response	2-6
Fig. 2.9	Control Module Loudness Contour Curves	2-7
Fig. 2.10	20Hz, 400Hz, 5KHz Tone Control Response Curve	2-8
Fig. 2.11	40Hz, 800Hz, 10KHz Tone Control Response Curve	2-8
Fig. 2.12	80Hz, 1600Hz, 20KHz Tone Control Response Curve	2-9
Fig. 2.13	50Hz Square Wave Response	2-9
Fig. 2.14	10KHz Square Wave Response	2-9
Fig. 3.1	Mounting Dimensions - Main Control Module	3-1
Fig. 3.2	Mounting Dimensions - Power Module	3-1
Fig. 3.3	Mounting Dimensions - Phono Module	3-2
Fig. 3.4	DL2 Phono Module	3-2
Fig. 3.5	Tape #3 Front Panel Adapter Plug Wiring	3-2
Fig. 3.6	Rear Panel	3-3
Fig. 3.7	Balanced Output Adapter	3-3
Fig. 3.8	Front Panel	3-4
Fig. 3.9	Power Module Rear Panel	3-7
Fig. 3.10	Power Module Front Panel	3-7
Fig. 3.11	Simplified Audio Imaging Control Diagram	3-8
Fig. 3.12	Audio Imaging Controls - Normal, Mono, Reverse, Panorama	3-9
Fig. 3.13	Audio Imaging Controls - Image Placement	3-9
Fig. 3.14	Audio Imaging Controls - Crossfeed Input Mixing	3-10
Fig. 3.15	Normalized Loudness Correction Curves (ISO)	3-11
Fig. 3.16	Remote Control Wiring Diagram	3-12
Fig. 3.17	INPUTS and LEVELS with Hex #'s and Decimal Equivalents	3-13
Fig. 3.18	Machine Language Program	3-15
Fig. 3.19	Address Switch Layouts	3-16
Fig. 4.1	DL2 Block Diagram	4-2
Fig. 4.2A	DL2 Power Supply Schematic	4-5
Fig. 4.2B	Power Supply Board Layout	4-6
Fig. 4.3A	Output Switch Board Schematic	4-7
Fig. 4.3B	Output Switch Board Layout	4-7
Fig. 4.4A	Output Board Schematic	4-7
Fig. 4.4B	Output Board Layout	4-7
Fig. 4.5A	Input Selector, Input and Loudness Board Schematics	4-8
Fig. 4.5B	Loudness Board Layout	4-9

Fig. 4.5C	Input Board Layout.....	4-9
Fig. 4.5D	Input Selector Board Layout	4-10
Fig. 4.6A	Tone Switch Board Schematic.....	4-11
Fig. 4.6B	Tone Switch Board Layout	4-11
Fig. 4.7A	Tone Output Board Schematic	4-12
Fig. 4.7B	Tone Output Board Layout	4-12
Fig. 4.8A	Display, Logic and Regulator Schematics	4-13
Fig. 4.8B	Logic Board Layout	4-14
Fig. 4.8C	Regulator Board Layout.....	4-14
Fig. 4.8D	Display Board Layout.....	4-15
Fig. 4.9A	Main Board, Tape Monitor and Panorama Board Schematics	4-16
Fig. 4.9B	Main Board Layout	4-17
Fig. 4.9C	Tape Monitor Board Layout	4-17
Fig. 4.9D	Panorama Board Layout	4-17
Fig. 7.1	Interconnect Diagram	7-2
Fig. 7.2	Required Test Equipment.....	7-3
Fig. 7.3	Relay Action.....	7-5
Fig. 7.3	Relay Action.....	7-6
Fig. 7.3	Relay Action.....	7-7
Fig. 7.4	Input Selector Board Solder Joints.....	7-8
Fig. 7.5	Tape Monitor Truth Table	7-10
Fig. 7.6	Loudness Contour Attenuation Table	7-10
Fig. 7.7	Tone Control Table	7-11
Fig. 7.8	IM Distortion/Hum and Noise Test Set-up	7-12
Fig. 7.9	AC Voltage Mains Chart	7-13

THE FOLDOUT OPPOSITE THIS PAGE, IS A QUICK CONNECT DIAGRAM WHICH MAY BE HELPFUL IN CONNECTING TOGETHER YOUR NEW DL2 SYSTEM





SECTION 1

GENERAL INFORMATION

1.1 Introduction

This manual is intended to provide all the necessary technical information required to operate the CROWN DL2 Digilogic Control Center and to maintain the unit in optimum operating condition.

This manual is comprehensive, containing a physical description of the DL2 Control Module, Power Module, and the Phono Module Fig. 1.1. Electrical and mechanical specifications, complete installation and operating instructions, and a detailed circuit description are also included.



Fig. 1.1 DL2 System

Read this manual thoroughly noting especially the warranty information. If all the instructions, procedures and cautions are strictly followed, the DL2 will provide many years of professional quality service and listening enjoyment.

1.2 Purpose of Equipment

The CROWN DL2 is a digitally controlled preamplifier and control center designed to be sonically pure and versatile in control. It will control or mix up to nine separate inputs and two main outputs. Other outputs include mono, inverted, headphone, tape, processor and buffer. It contains a unique shelving tone control system, a variable loudness contour system, and adjustable high and low filters. The volume control is a very accurate, digitally switched and displayed (in one half dB increments) gain control. The DL2 applications consist of sonically superb home hi-fi usage and commercial broadcast automation systems when coupled to a mini-computer.

1.3 Unpacking and Inspecting Equipment

Immediately upon receipt of the DL2, inspect the unit for any damage incurred in transit. The unit was carefully inspected and tested and left the factory unmarred. Notify the transportation company immediately if any damage is found. Only the consignee may initiate a claim with the carrier for damage during shipment. However, Crown, will cooperate fully in any such event. Be sure to save the carton as evidence of damage for the shipper's inspection.

Crown recommends that you save the packing materials, even if the unit arrives in perfect condition. Both the carton and internal pack are specifically designed for protection during transit. They will prove valuable in preventing damage should there be occasion to transport or ship the unit. **DO NOT SHIP THE UNIT WITHOUT THIS FACTORY PACK!**

1.4 Units and Accessories Supplied

Your DL2 comes with an instruction manual and:

Control Module:

- 2 - 6 ft. pin to pin cables
- 2 - 6 ft. phone to phone cables
- 4 - mounting thumbscrews

Power Module:

- 1 - 20A-15A AC adapter
- 1 - dual-connect power cable
- 4 - mounting thumbscrews

Phono Module:

- 2 - 6 ft. pin to phone cables
- 2 - 9 in. pin to pin cables
- 1 - power cable

1.5 Service Policies

Due to the sophisticated circuitry of your unit, only qualified, fully trained technicians should be allowed to service it. Please observe the following label on the rear panel. **CAUTION: TO PREVENT ELECTRIC SHOCK DO NOT OPEN. NO USER SERVICABLE PARTS INSIDE. REFER SERVICING TO A QUALIFIED TECHNICIAN.**

For service, return the unit to the factory in the original packing or in replacement packing obtainable from the Crown factory. For warranty service, the unit must be returned to the factory or an approved service station (Amcron customers consult your local Crown representative). In either case, enclose a brief letter explaining the problem you are experiencing. This will help to insure a speedy and effective response.

Crown will pay shipping costs for warranty service upon receiving copies of all shipping receipts.

Before returning your unit to the factory for service, authorization should be obtained from the Technical Service Department. All shipments should be sent by either UPS or truck freight (insured). The factory will then return your serviced unit by one of the above methods.

Upon receipt of the warranty registration card from your dealer, Crown will automatically send you the Crown Care Card. This is a plastic, wallet-sized card. This is your warranty certificate.

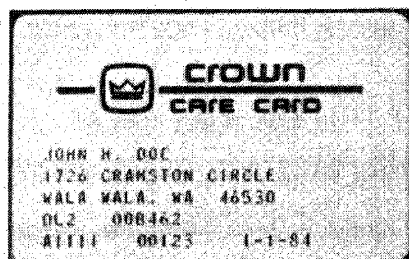


Fig. 1.2 Crown Care Card

Embossed on it will be your name and current address, the model number and serial number of your new Crown unit, and the expiration date of the warranty on that product.

Crown will replace lost Care Cards at no charge and will issue new cards for repurchased units.

When you need service for your unit from an authorized Crown Service Station, simply present your Crown Care Card. It is all the identification you need. With it, the service station can promptly initiate any needed paperwork. It will save you time and effort.

The Crown Care Card is also your proof of ownership should you need it for insurance or legal reasons. It is also a convenient place to look for the serial number should you want to write to the factory for any reason.

1.6 Glossary of Terms

A-B Test: Evaluating relative performance of two (or more) components or systems by changing quickly from one to the other. Most high fidelity dealers have A-B test facilities.

AGC: Automatic Gain Control. A type of circuit used to maintain the output volume of a receiver constant, regardless of variations in the signal strength applied to the receiver.

Attenuation: A decrease in signal magnitude from one point to another, or the process causing this decrease.

Balanced Input: A three wire input system where the voltages and currents in two of the wires are equal in magnitude but opposite in polarity with respect to ground which is the third wire. The impedance of a balanced input is usually low. (600 ohms or less)

Balun: A device used for the transformation from an unbalanced line or system to a balanced line or system, or vice versa. The term is derived from balance to unbalance transformer.

Bandpass Filter: A filter that allows transmission of alternating signals whose frequencies are between given upper and lower cutoff values, while substantially attenuating all frequencies outside this band.

Biamp: The use of independent amplifiers to feed the bass and treble portions of a loudspeaker or loudspeakers with a crossover network. The purpose is to help reduce intermodulation between bands when a system is overloaded. It also reduces the cost of implementing good, low loss precision crossover networks.



Butterworth Filter: A filter that exhibits the flattest possible response in the passband. The response rolls off smoothly into the stop band, where it approaches a constant slope of $6n$ dB/octave; where n is the number of poles in the network, i.e. a single pole is 6dB, two pole 12dB, 3 pole 18dB, etc.

Capture Ratio: The ability of a receiver to reject a weaker station whose frequency is the same as that of the desired station. This measurement will be in dB. The smaller the figure, the better the spec.

Ceramic Filter: A bandpass filter using a piezoelectric substrate material.

Channel: A channel is a complete sound path. A single channel, or monophonic system, has one channel. A stereophonic system has at least two full channels designated as left (A) and right (B). Monophonic material may be played through a stereo system; both channels will carry the same signal. Stereo material, if played on a monophonic system, mixes and emerges as a monophonic sound.

Channel Separation: Specified in dB, channel separation is the ratio of the measurable output of one channel to the unwanted output of the (undriven) opposite channel.

Clipping: The truncation of peaks of a signal due to exceeding the operating range of an electronic circuit. Normally, it refers to the result of voltage limitations in the circuit.

Corner Frequency: The frequency at which a filter goes from a condition of passing the signal unattenuated to "rolling off" or attenuating the signal according to its frequency. It is sometimes referred to as the "cutoff" frequency or the "break" frequency. It is also defined as 3dB below the unattenuated output level of the signal.

Cross Modulation: In FM signals, a type of intermodulation that occurs when the carrier of the desired signal is modulated by an undesirable signal. Each signal is of independent origin.

Crossover Frequency: The frequency at which a dividing network delivers equal power to the upper and lower frequency channels when both are terminated in specified loads.

Crossover Network: A selective network used to divide the audio frequency output of an amplifier into two or more bands of frequency. The band below the crossover frequency is fed the woofer loudspeaker while the high frequency band is fed to the tweeter. Also called dividing network and loudspeaker dividing network.

Crosstalk: Signal leakage from one signal source into another.

Damping: Controlling of vibrations, response, or resonances which if unchecked, would cause coloration of the sound.

Damping Factor: A numerical indication of an amplifier's ability to decrease unwanted loudspeaker movements. Damping factor can be found by dividing the load impedance by the amplifier's output impedance.

Decibel: A numerical expression of acoustic or electrical ratios, such as the relative intensity of a sound or the relative strength of a signal. One (dB) is about the smallest change in sound perceptible to the ear.

Decoder: A matrix of logic elements that selects one or more output channels according to the combination of input signals present. Can be used in FM to recover stereo signals of a stereo encoded multiplex transmission.

De-emphasis: In FM signals, restoring the pre-emphasized (for proper transmission characteristics) signal to result in a "flat" frequency response curve.

Distortion: Unwanted noise, or sounds which didn't exist in the studio when the original recording was made. Harmonic distortion produces tones harmonically related to a single, pure tone. Intermodulation distortion (IM) introduces new tones caused by mixing of two or more original tones. Phase distortion, or non-linear phase shift, disturbs timing sequence between a tone and its related overtones. Distortion which create new spectra are expressed in percentages and phase distortion in degrees of phase shift or seconds of group delay vs. frequency errors.

Dynamic Range: The difference between the most intense and the least intense levels in a sound system.

EAROM: EAROM stands for Electrically Alterable Read Only Memory which is an integrated circuit chip used to store information without power applied (used in the FM-1).

Equalization: Frequency response manipulation to meet the requirements of recording, and an inverse manipulation on playback to restore the original. Also known as compensation for acoustic problems of listening rooms.

Feedback: Sending a "part" of a system's output back to the input. Positive feedback may lead to unstable conditions such as PA system "howling" due to sound from the speakers being fed back thru the microphone and amplified again. Another form is low frequency interference created when vibrations from loudspeakers are picked up by the cartridge and amplified again. However, carefully controlled negative feedback in electronic circuits can be used to help reduce distortion and control stability.

Flutter: Rapid variation in the speed of a turntable or tape transport. Flutter causes a wavering of musical pitch.

Frequency Response: This term indicates any amplitude variations in a system output signal with respect to frequency. This measurement is made with a constant level input signal.

Gain: The ratio of an amplifier's output voltage to its input voltage.

Headroom: Stated in dB, headroom is the difference between the signal level and the limits of the sound system ie; the ratio of power available to power used.

Hertz: As in cycles-per-second, not rental agency.

High Pass/Low Pass Filter: High pass - a filter having a single transmission band extending from some critical, or cutoff, frequency other than zero, up to infinite frequency. Low pass - a filter network which passes all frequencies below a specified frequency with little or no loss but discriminates strongly against higher frequencies.

IF: An Intermediate Frequency is a frequency to which a signal wave is shifted locally as an intermediate step in FM transmission or reception.

Image Response: Response of a superheterodyne receiver to the image frequency, as compared to the response to the desired frequency.

Input Sensitivity: The input voltage required to drive an amplifier to its rated output.

IOC™: Stands for Input-Output Comparator. An extremely sensitive Crown circuit used to indicate, via a front panel LED, the fact that the operating limits of an amplifier are being exceeded resulting in output non-linearity.

LED: Light Emitting Diode. A PN junction that emits light when biased in the forward direction.

Limiter: A circuit in which the output amplitude is substantially linear with regard to the input up to a pre-determined value and substantially constant thereafter.

Load: A device that absorbs power and converts it into the desired form.

LSI: Large Scale Integration is an integrated circuit chip housing a large number of active devices.

Mixer: A device having two or more inputs, usually adjustable, and a common output, which operates to combine linearly in a desired proportion the separate input signals to produce an output signal.

Monitoring Amplifier: A power amplifier used primarily for evaluation and supervision of a program.

Multipath Delay: In FM, the existence of more than one signal path between transmitter and receiver. The two signals arriving at different times causes distortion.

Music Power: This rating expresses the ability of an amplifier to handle short duration power peaks, as opposed to sustained power levels. An amplifier may only be capable of putting out 45 watts if that level is continuous, but it may be able to handle 60 watt peaks (such as might occur in a musical passage), if the peaks do not last too long.

Muting Circuit: In FM, a circuit which cuts off the receiver output when the r-f carrier reaching the first detector is at or below a pre-determined intensity.

Pre-emphasis: A process in a system designed to emphasize the magnitude of some frequency components with respect to the magnitude of others, to reduce adverse effects, such as noise, in subsequent parts of the system.

PLL: Phase Lock Loop; a circuit for synchronizing a variable local oscillator with the phase of a transmitted signal. Can be used in FM as a synthesizer circuit as well as a stereo decoder in the multiplex system.

Polar Curve: A pattern used to show the directional characteristics of antennas, microphones or speakers.

Quartz Crystal: A complete assembly, comprising a piezoelectric quartz-crystal element mounted, housed, and adjusted to the desired frequency. Such a device is commonly employed for purposes of frequency control, frequency measurement and electric wave filtering.

Rotor: A motor driven assembly which turns an antenna so that it can be aimed in the direction of best reception.

SAW Filter: Surface Acoustic Wave; commonly used as a bandpass filter. Device transmits surface acoustic waves on a piezoelectric substrate. Used in the FMI for low FM distortion.



Selectivity: In FM, a measure of the extent to which a receiver is capable of differentiating between the desired signal and disturbances at other frequencies.

Sensitivity: A receiver specification indicating the smallest input signal strength required to produce an output signal whose characteristics are standardized and used as a reference.

Signal: A visible, audible or other conveyor of information.

Signal-To-Noise Ratio: Measured in dB, signal to noise ratio is a relative term meaning the ratio between the desired output signal and the interference or noise. A typical figure would be 60dB which stands for a ratio of 1000-1.

Superheterodyne: A method of receiving radio waves in which the process of heterodyne reception is used to convert the voltage of the received wave into a voltage of an intermediate, but usually superaudible frequency, that is then detected.

Synthesizer: A Phase Lock Loop (PLL) system which constructs an oscillator signal by phase locking it to a reference times (X) the ratio of two integers (reference is usually a quartz crystal oscillator).

Triamp: The use of three separate amplifiers to drive the high, middle and low frequency sections of a speaker system.

Varactor: A two terminal semiconductor device in which the electrical characteristic of primary interest is a voltage-dependent capacitance. Used in FM tuner inputs for voltage controlled timing.

Wow: Distortion caused in sound reproduction by slow variation in speed of the turntable or tape. (See flutter).



SECTION 2

SPECIFICATIONS AND PERFORMANCE

2.1 Specifications

2.1.1 Control Module

Frequency Response (10K load): $\pm 0.1\text{dB}$, 10Hz-50KHz, $\pm 0.5\text{dB}$, 1Hz-100KHz.

Phase Response (10K load): within $\pm 8^\circ$, 20Hz to 20KHz.

Hum and Noise (inputs shorted, 20Hz-20KHz, below rated output of 2.5V):

	Unweighted	A weighted
Max gain	97dB	101dB
Unity gain	103dB	107dB
Min gain	110dB	118dB

Intermodulation Distortion - IMD (10K load, SMPTE analyzer): Less than .0003% at 10 volts out or below.

Total Harmonic Distortion - THD (10K load):

	Rated output (2.5V)	10 volts out
1KHz	less than .0003%	less than .0006%
20Hz-20KHz	less than .0008%	less than .0025%

Noise Transfer IM Distortion - NTIM (10K load): Less than .002% at 2.5V peak equiv. level. Less than .001% at 10.0V peak equiv. level. Test signal 10KHz-26KHz white noise (crest factor 3.9); distortion products measured below 10KHz; NTIM testing is highly sensitive to audible products of high frequency distortion mechanisms such as S.I.D. (T.I.M.).

Inputs: Eight dual channel high level inputs pushbutton selectable with LED indicators (three with tape monitor and copy functions). Two dual channel inputs for external signal processors (such as an equalizer and/or noise reduction unit). One dual channel mixable input (crossfeed) with separate level controls for any other external input.

Input Gain and Impedance: $20\text{dB} \pm 0.2\text{dB}$, 100K ohm (impedance constant regardless of level setting unless inputs are paralleled).

Outputs: 11 volts RMS maximum before overload; 2.5 volts rated with 50 ohm output impedance. 22 volts RMS maximum on balanced outputs; 5.0 volts rated with 100 ohm output impedance. 1 dual channel buffered output available to feed signal processors or other preamp circuits of a complex system.

Headphone Output: Connector; stereo 3 circuit $\frac{1}{4}$ " phone jack. Impedance: less than 1 ohm (less than 220 milliohms in series with less than 6 microhenries). Max Level: 17 volts RMS. Frequency Response: $\pm 0.1\text{dB}$ 10Hz-50KHz at 17 volts RMS.

Gain-Balance Controls: Pushbutton controlled, digitally stepped in 0.5dB increments (129 positions); the Level-Balance controls are active, seven section precision switched gain relays with 63.5dB (and "off") dynamic range; the seven segment LED displays show gain in dB above minimum level; channels can be independently adjusted, or ganged over the full range with a tracking accuracy of $\pm 0.2\text{dB}$.

Loudness Compensation: Precision wide-range ISO curves from 100 to 55 phons in 5-phon steps with exclusive three-pole compensation; rotary switch controlled.

Tone Control: Shelving bandpass type bass, mid, and treble controls provide precision-stepped $\pm 15\text{dB}$ boost and cut; normally out of the audio path, the tone controls are switch activated for flat-vs-compensated comparisons. Center frequency selections are as follows:

	X.5	X1	X2
Bass:	20Hz	- 40Hz	- 80Hz
Mid:	400Hz	- 800Hz	- 1600Hz
Treble:	5KHz	- 10KHz	- 20KHz

3Filters: Precision stepped adjustable active filters with individual activation switches. Low Filter: -3dB at 20Hz, 30Hz, 50Hz, or 100Hz; -18dB/octave. High Filter: -3dB at 4KHz, 7KHz, 12KHz, or 20KHz; -18dB/octave.

Tape Monitors and Copy: Three inputs include monitoring functions for taping. The Tape Copy switch allows any or all three tape recorder positions to copy from the master input buss or from one recorder to any two other recorders.

External Signal Processors: Two dual channel input and output jacks to provide for two external processors in the audio chain; individually switched in or out. Prerecord switch allows insertion of the processors ahead of the tape outputs to permit pre-tape processing such as equalization or noise reduction.

Audio Imaging: Unique stepped 31 position switched attenuators for precision (± 0.2 dB over 50dB range) adjustment of stereo image. Six separate controls allow mono, reverse stereo, and mixing of an external two channel signal (crossfeed), as well as exact control of the channel 1 - channel 2 mix.

Muting: Relays provide 5-7 second delay in opening output channels at turn-on, to protect speakers from large transients.

Remote Functions: A wireless remote control module will be available for the DL2 soon. The following functions will be remote controlled:

- Input selection
- Power on/off
- Mute/Gain/Balance

Two or more DL2's can be ganged together for control of more than 2 channels of sound; all original specs will be maintained between any such combination of channels.

Computer Interface: The DL2 logic has been designed for computer control; a computer interface connector appears on the rear panel of the Main Control Unit.

Dimensions and Weight/Mass: 17" x 7½" x 14" behind front panel (43.18cm x 19.05cm x 35.56cm). Front panel fits standard 19" rack (48.26cm).

Chassis Finish: All satinized aluminum front panel with charcoal Lexan inlay; other surfaces anodized.

2.1.2 Phono Module A

Frequency Response (10K load): $\pm .25$ dB of RIAA, 20Hz-20KHz (switch in RIAA position). ± 0.1 dB, 10Hz-30KHz (switch in flat position).

Phase Response (10K load): within $\pm 5^\circ$, 20Hz-20KHz (RIAA). Within $\pm 2^\circ$, -12°, 20Hz-20KHz (flat).

Hum and Noise (inputs shorted, 20Hz-20KHz, below 10mV input):

	Unweighted	A weighted
RIAA	88dB	94dB
Flat	84dB	89dB

Intermodulation Distortion; -IMD (10K load, SMPTE analyzer): less than .0005% at 2.5 volts out (switch in flat position).

Total Harmonic Distortion - THD (10K load): less than .002% (20Hz-20KHz) at 2.5 volts out.

Noise Transfer IM Distortion - NTIM (10K load): less than .0004% at 2.5V peak equivalent level. (See Control Module specs for test conditions.)

Input Gain and Impedance: Gain adjustable from 30-50dB. Impedance switch selectable 47K ohms/100K ohms. Parallel capacitance at input jacks less than 5pf. .75mV at 1KHz for rated output at max gain.

Input Overload: 33-330mV at 1KHz depending on gain. 100mV at 40dB gain.

Output Impedance and Voltage: 600 ohm with max 11 volts RMS, 20Hz-20KHz, 10K load.

Dimensions and Weight: 1⅞" x 3½" x 6⅞" (3.49cm x 8.89cm x 16.83cm).

12oz (337gm).

Finish: Black chrome steel.

2.1.3 Power Supply

AC Outlets: Seven switched (3-20A, 4-15A). Two unswitched.

DC Outputs: ± 28 volts at 500MA. ± 18 volts at 500 MA. +10 volts at 2 amps.

Power Requirements: Less than 40 watts 120 volts AC.

Dimensions and Weight: 17" x 3½" x 7½" (43.18cm x 8.89cm x 19.05cm). Fits 19" standard rack mount (48.26cm). 9 lb. (4.05kgm)

Finish: Satinized aluminum front panel with black anodized chassis.



2.2 Performance Graphs

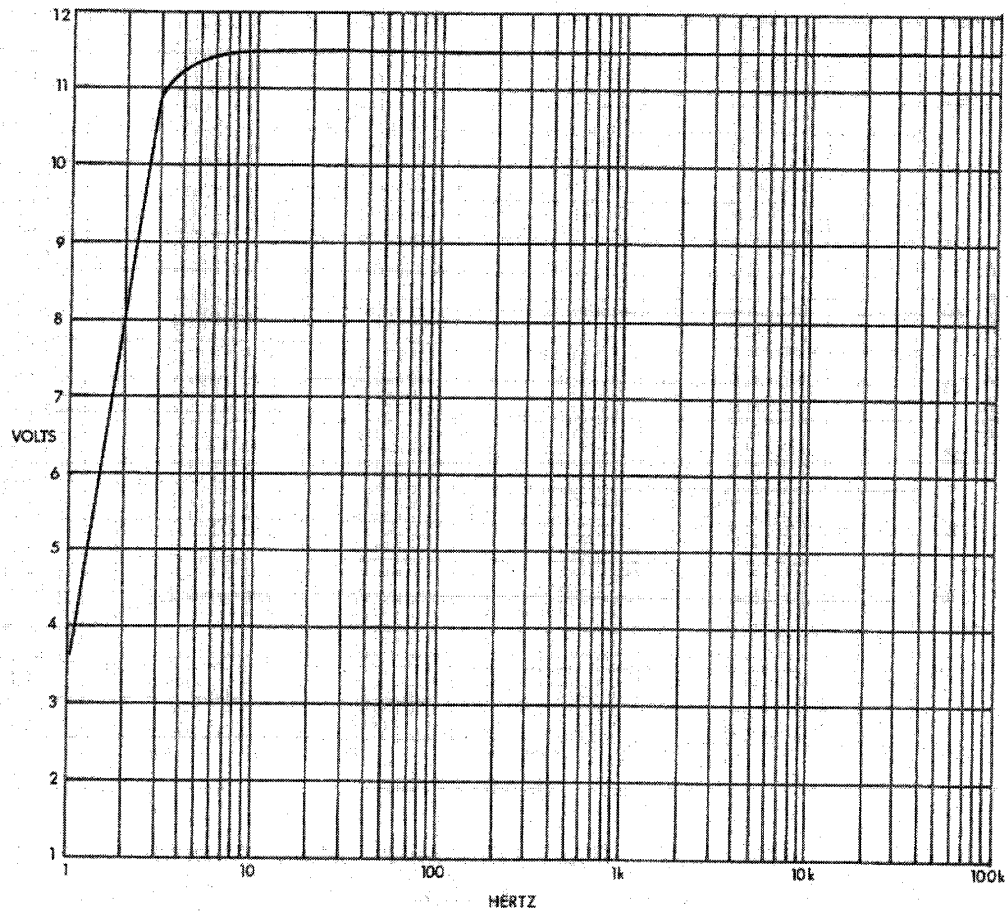


Fig. 2.1 Maximum Output IHF Load

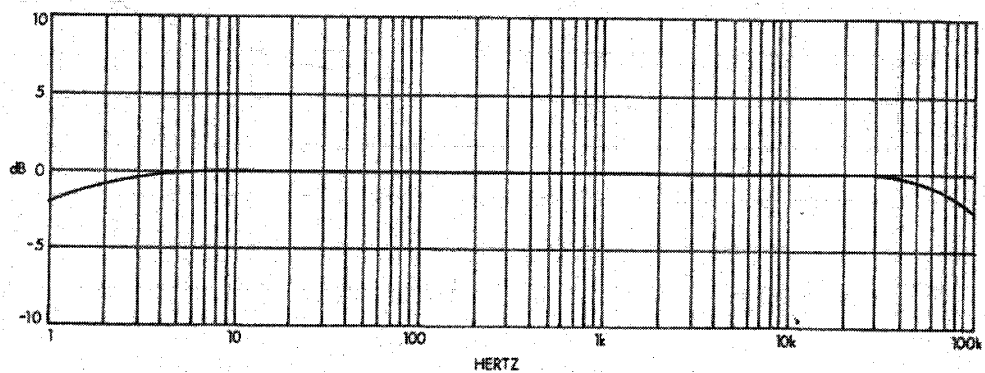


Fig. 2.2 Frequency Response, Main Control Module

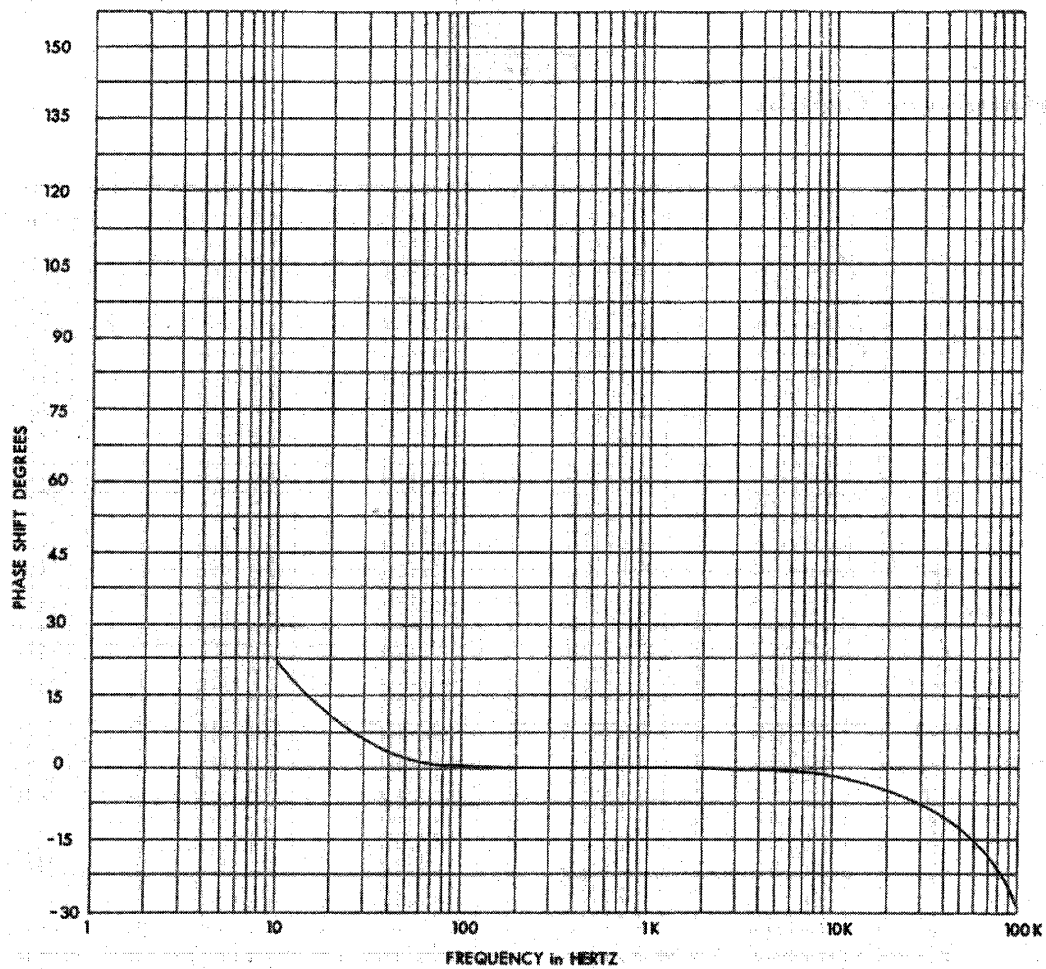


Fig. 2.3 Phase Response, Main Control Module

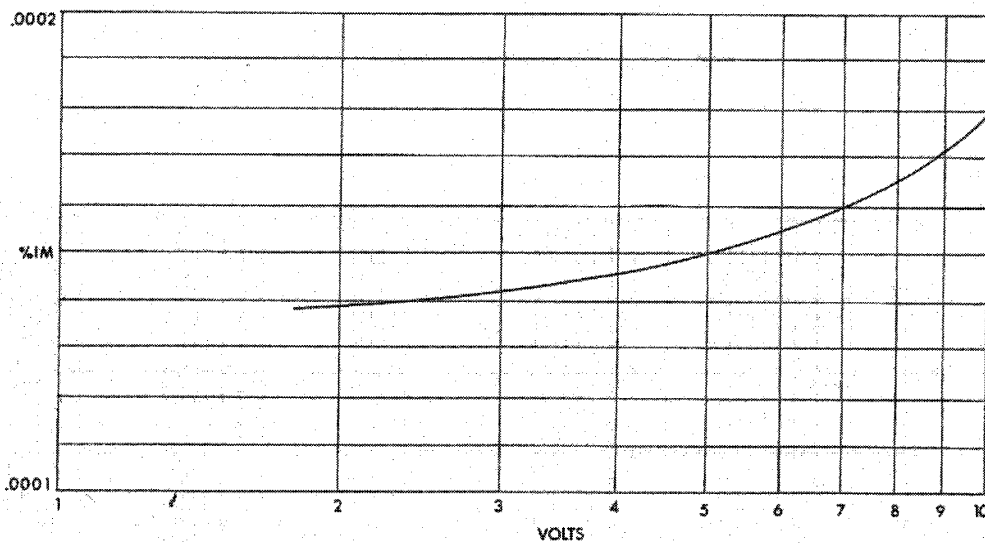


Fig. 2.4 IM Distortion

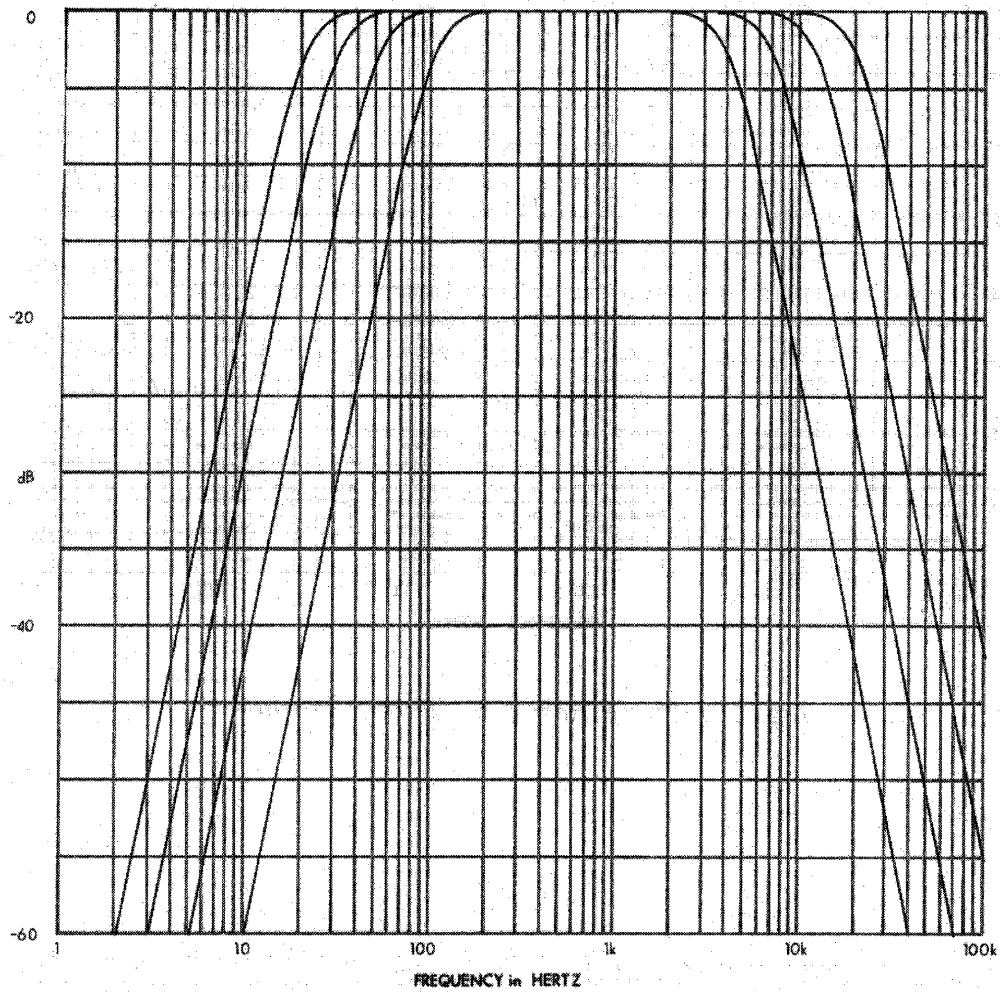


Fig. 2.5 Low, High Filter Response

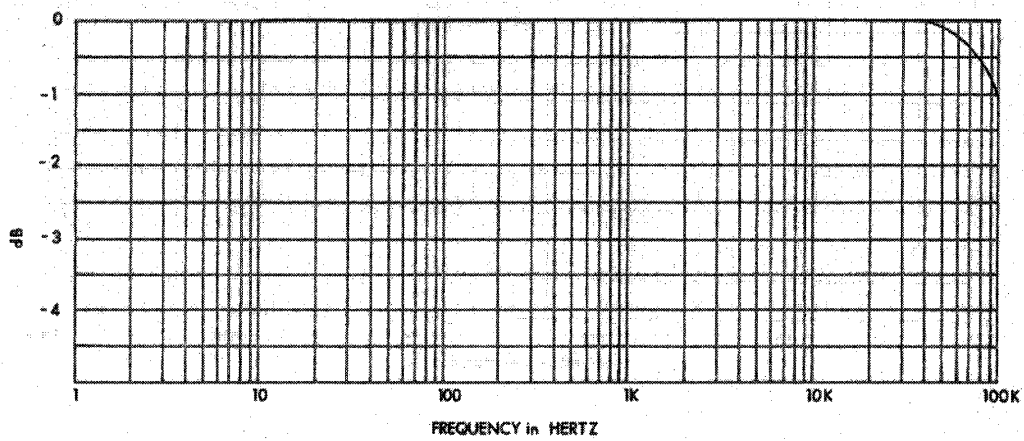


Fig. 2.6 Frequency Response, Headphone Amplifier

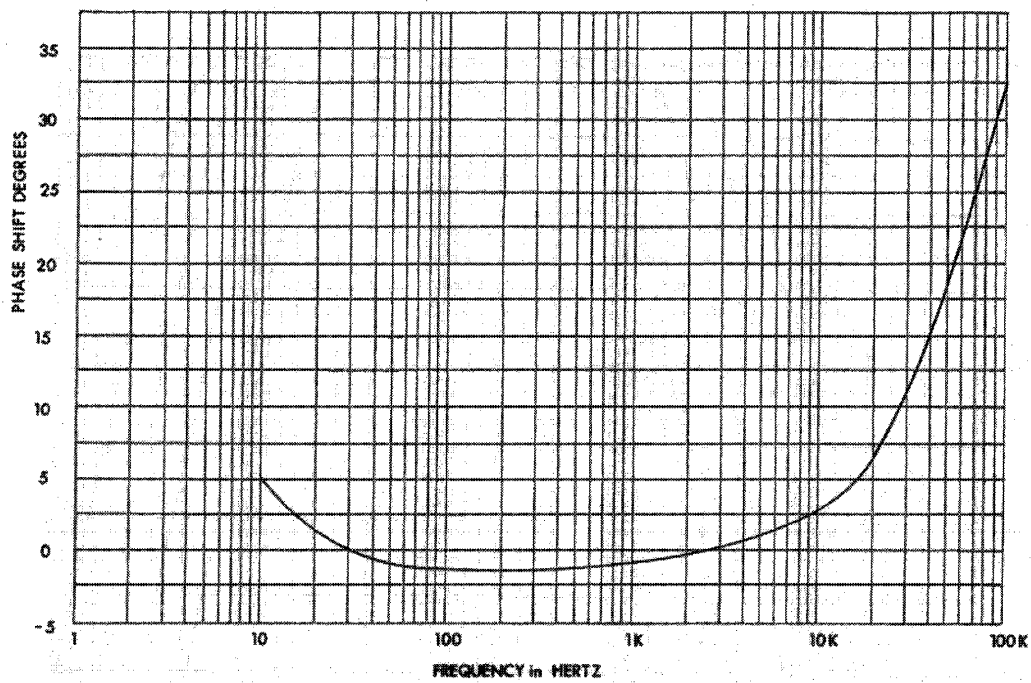


Fig. 2.7 Phase Response, Phono Module

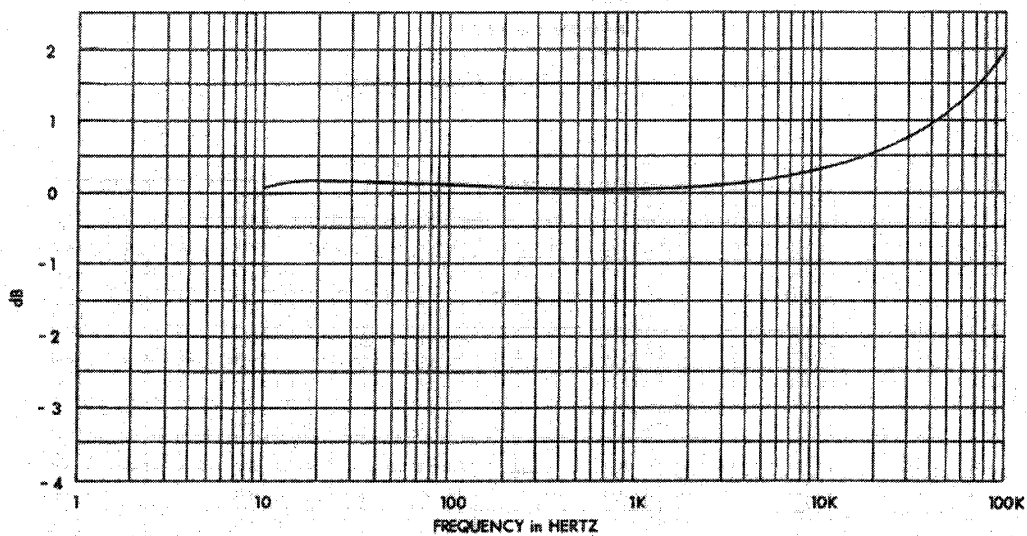


Fig. 2.8 RIAA Phono Module Frequency Response

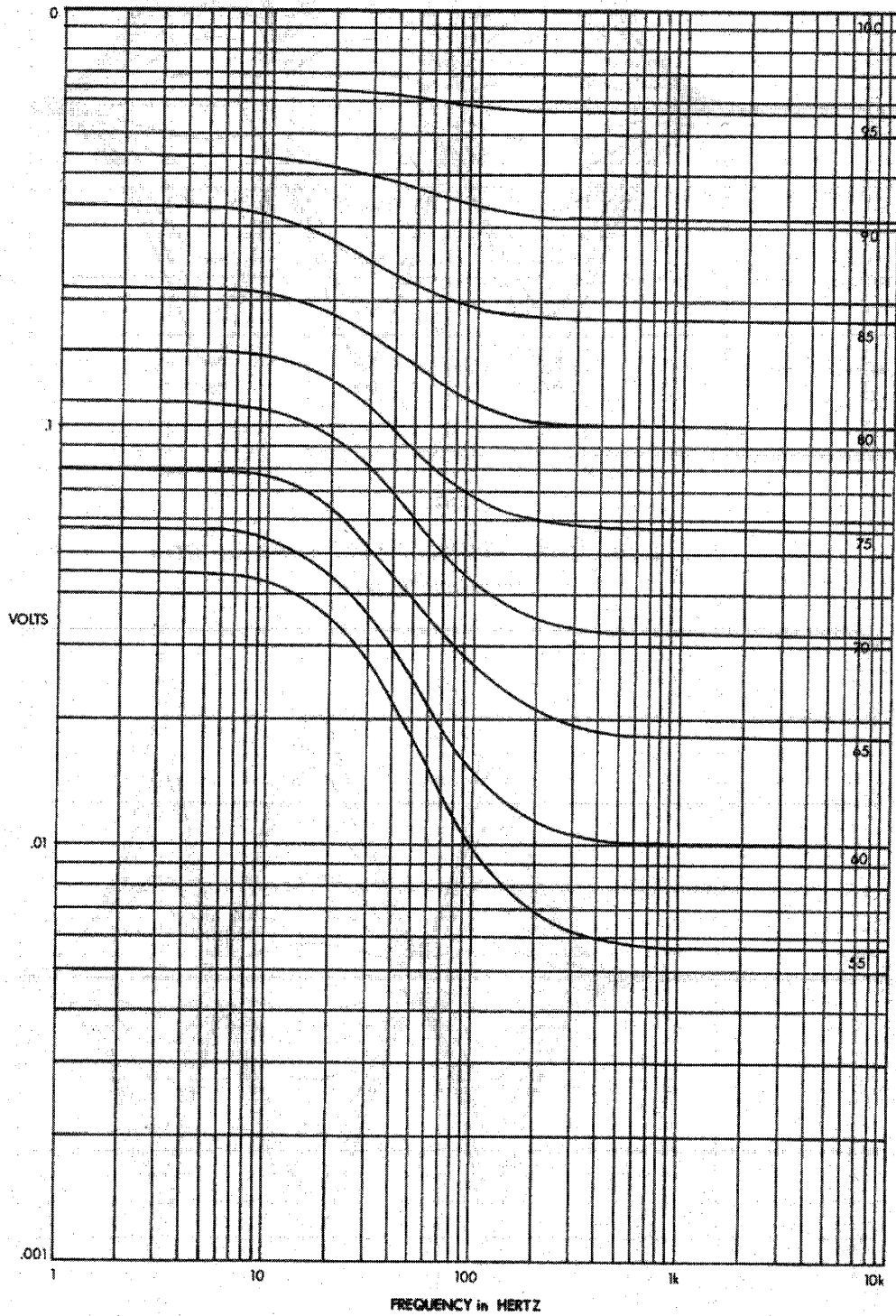


Fig. 2.9 Control Module Loudness Contour Curves

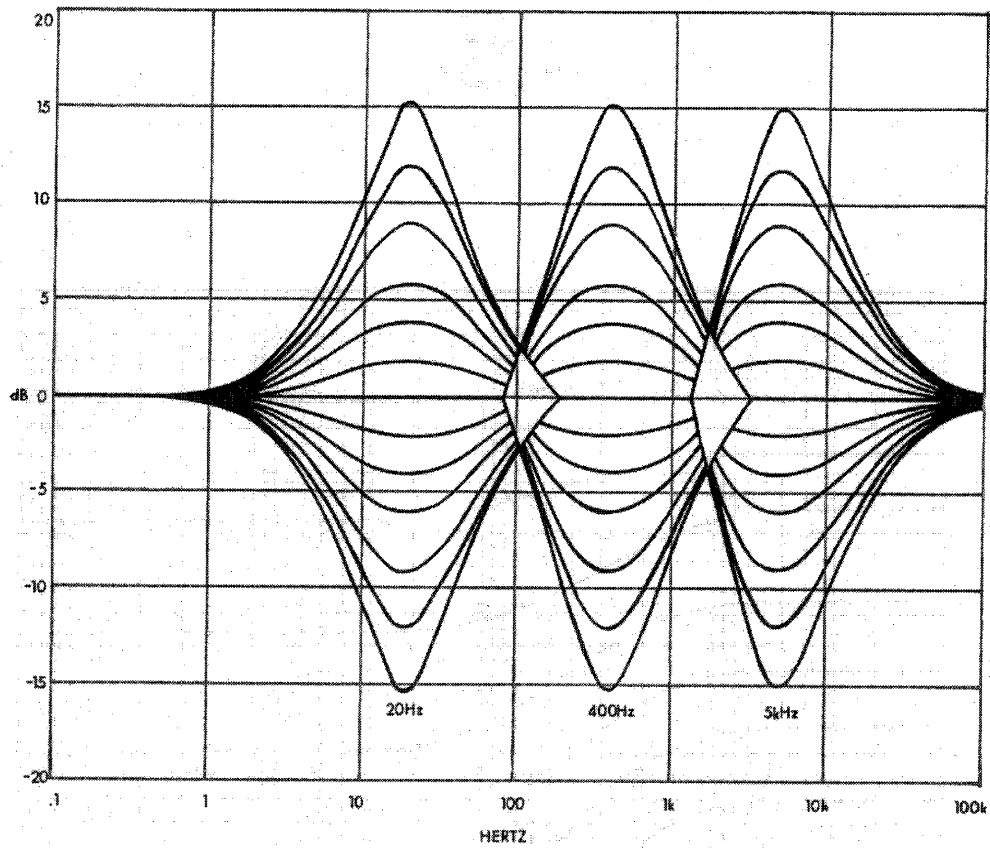


*Fig. 2.10 20Hz, 400Hz, 5KHz Tone Control
Response Curve*

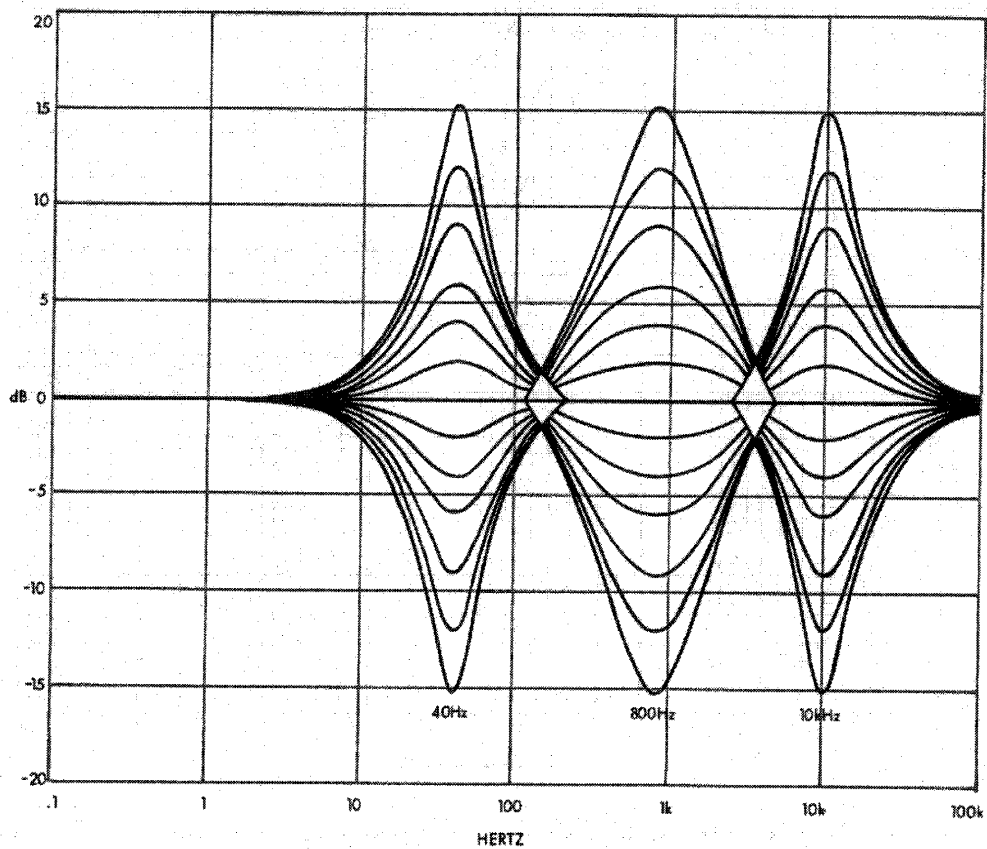


*Fig. 2:11 40 Hz, 800 Hz, 10kHz Tone Control
Response Curve*

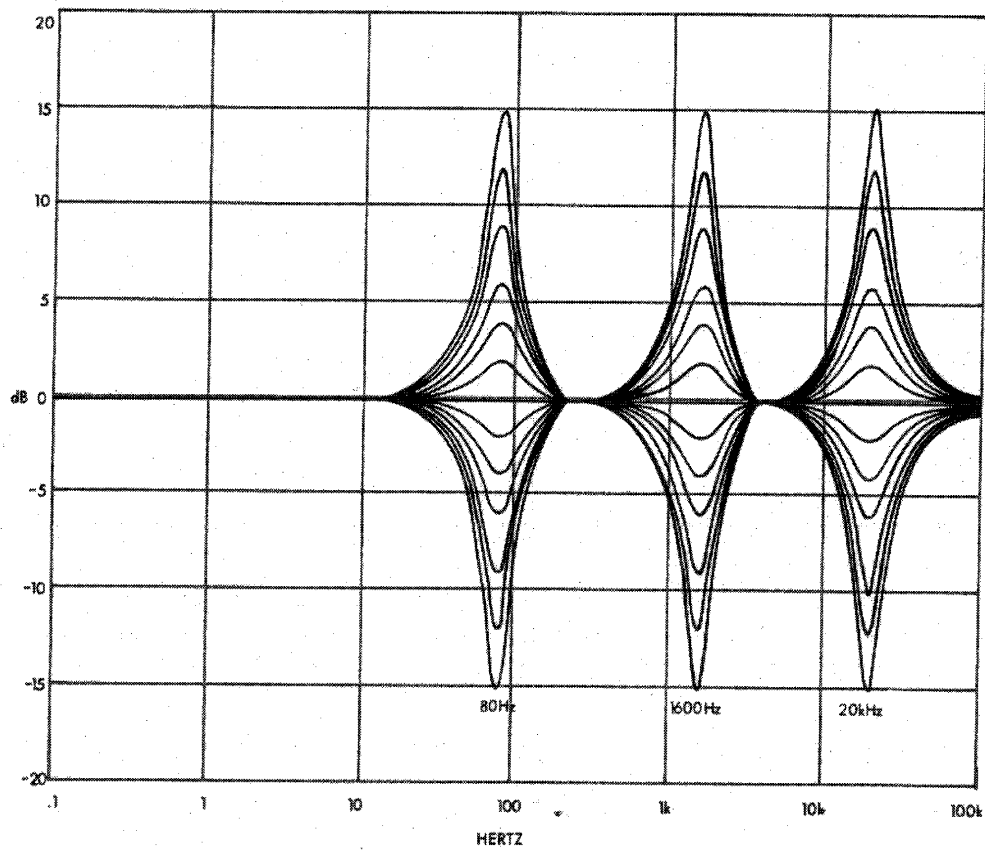


Fig. 2.12 80 Hz, 1600 Hz, 20 kHz Tone Control Response Curve

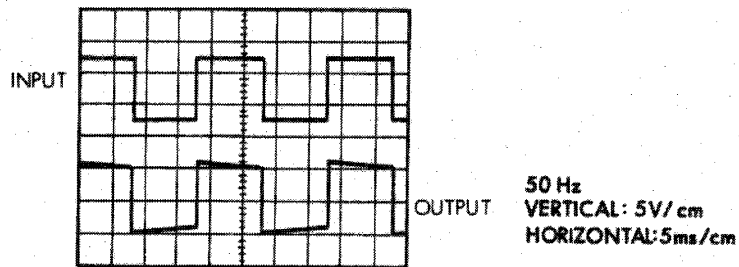


Fig. 2.13 50Hz Square Wave Response

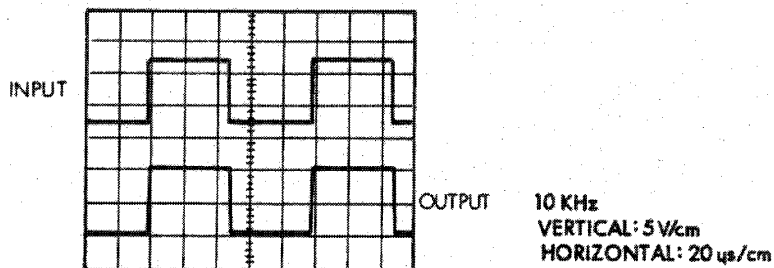


Fig. 2.14 10KHz Square Wave Response



SECTION 3

INSTALLATION AND OPERATION

3.1 General Information

This section of the manual is to familiarize you with the installation and operational practice of the DL2 Control Center/Pre-Amplifier. Due to the limited size of this manual, every contingency is not covered. In actual application, the installation and operation is only limited by your imagination.

3.2 Installation

The DL2 control center may be mounted in a standard 19" rack or in one of the custom wood cabinets available from Crown at additional cost. For a custom installation, see Fig. 3.1 mounting dimensions.

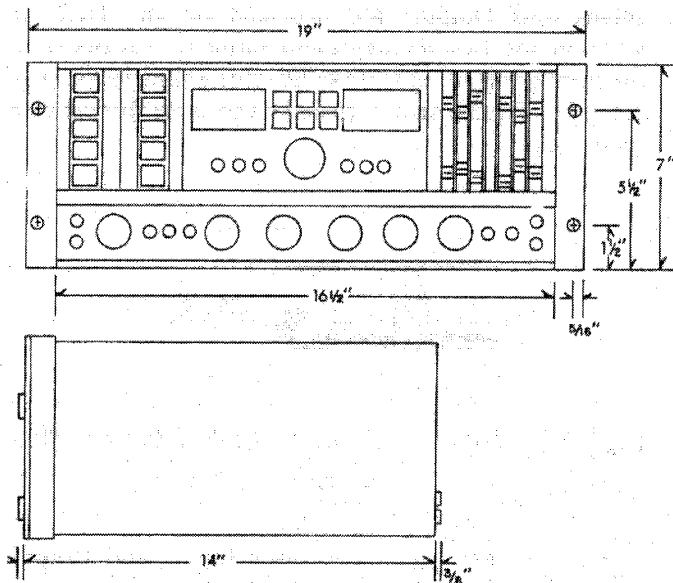


Fig. 3.1 Mounting Dimensions - Main Control Module

3.2.1 Rack Mounting

Consideration must be given for the depth (14 1/2") of the cabinet interconnecting cables, then for front panel clearance and accessibility.

Hold the DL2 with one hand and install the two lower thumb screws with the other (do not tighten), then you can relax your hold and install the two remaining top screws. Position the DL2 and tighten the thumb screws securely. The thumb screws are provided in the accessory kit packed with the DL2. The rear of the case need not be supported unless the equipment will be mobile or portable.

The Power Module may be rack mounted or hidden on the floor or shelf (see Fig. 3.2 for mounting dimensions). In any case, be sure the module gets plenty of air for cooling purposes. **DO NOT STACK IT ON TOP OF A POWER AMP.** The rising heat from the power amp may cause thermal shut-off in the Power Module. If this happens, move the module to another location. The module will return to normal operation after cooling.

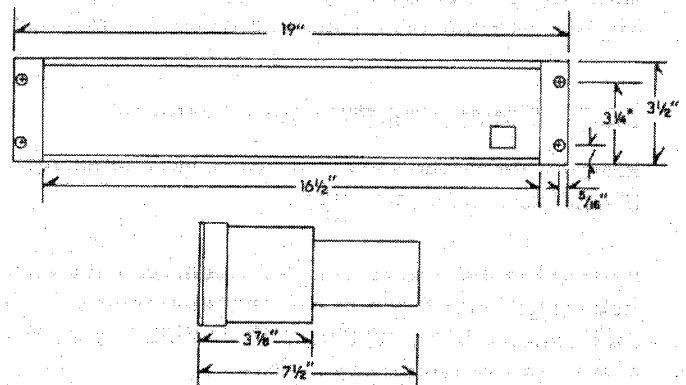


Fig. 3.2 Mounting Dimensions - Power Module

3.2.2 Mounting the Phono Module

The Phono Module is designed to be mounted in close proximity to the turntable. The low impedance, signal level, and special design of the module, practically eliminate RF interference and high frequency loss. By mounting the module close to the turntable, shorter cables may be used between the cartridge and phono

module, thereby reducing cartridge loading. Since the module itself may be fastened to just about any surface, you can place your turntable in any convenient location. See Fig. 3.3 for physical dimensions.

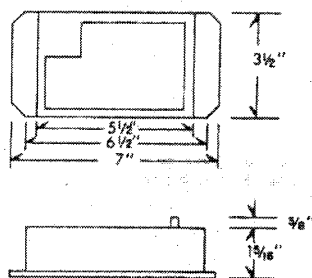


Fig. 3.3 Mounting Dimensions - Phono Module

3.3 Interconnecting Modules

After the equipment is mounted, connect two power cables to the Power Module (connectors marked J1). Then connect the heavier cable to the Control Center and the "Y" cable to the Phono Module. The third connector on this cable is to "daisy chain" additional cables and Phono Modules.

When connecting two Phono Modules to the DL2, use the extra power plug provided on the power cable (CPN42248).

The DL2 Phono Module hook-up is different from conventional preamps in which the Phono Module is inherently part of the system. With a separate Phono Module, external interconnecting cables are required.

3.3.1 Connecting the Phono Module

Basically, the Phono Module is treated like another piece of equipment such as a tuner.

Refer to Fig. 3.4. The cartridge wires connect to the input jacks. (Fig. 3.4A) A pair of RCA phono-to-phone patch cords connect the output (Fig. 3.4B) to a DL2 input. Any of the first five inputs may be used.

3.4 Rear Panel Control Center Connections and Control Functions

Refer to Fig. 3.6 for the following discussion. The letters refer to the labels on Fig. 6. Note that all inputs are RCA Phono type jacks; all outputs are standard 1/4" phone jacks. J1 connects to the power module. The other two connectors (J2 and J3) are for external control (either mini-computer or remote control).

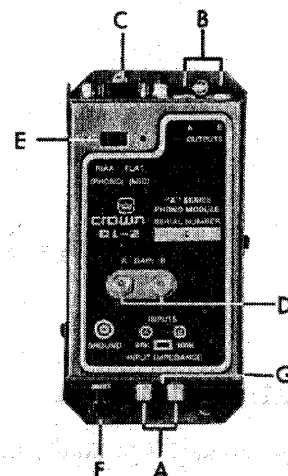


Fig. 3.4 DL2 Phono Module

A. Main Inputs - These five inputs are identical and will accept any high-level signal such as a tuner output, phono module output, or microphone preamp output. Each is selected by a corresponding input selector button on the front panel. See Section 3.5F for correct selector button - input identification.

B. Tape Inputs and Outputs - Three stereo Tape Inputs and Outputs are provided on the DL2. In addition, the Tape #3 inputs and outputs are available on the front panel using three-conductor phone jacks. (Fig. 3.8G) See Fig. 3.6 on how to make an adapter for use with these jacks.

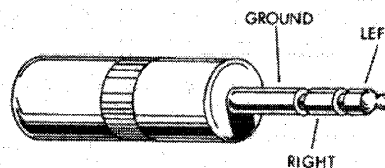


Fig. 3.5 Tape #3 Front Panel Adapter Plug Wiring

C. External Signal Processor Inputs and Outputs Provisions are made for two external signal processors (equalizers, tape noise reduction units, etc.).

Each processor has a left and right channel input and output. Processor operation is selected by the two push buttons #1 and #3 (Fig. 3.8J).

D. X and Y Crossfeed Inputs - Any signal connected to the X and Y crossfeed inputs is controlled by the Cross-Feed Audio Imaging Controls and fed into the left and

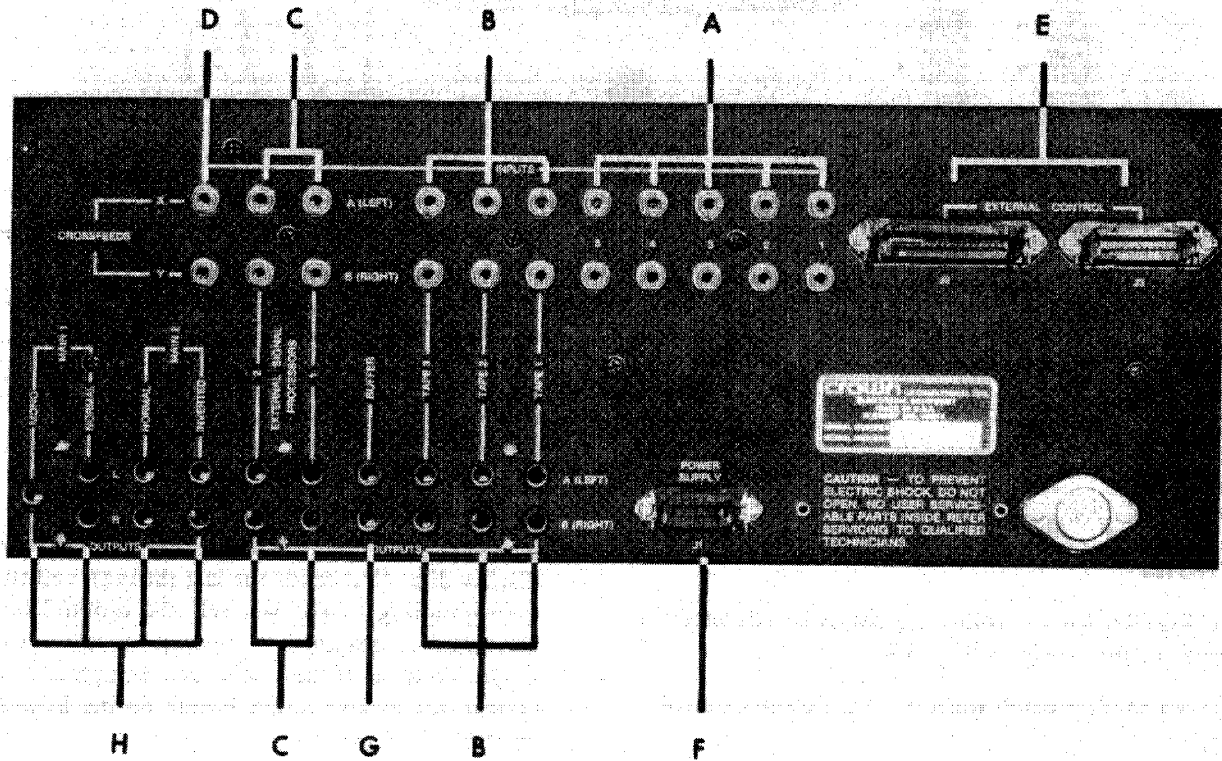


Fig. 3.6 Rear Panel

right channels. See Section 3.5N and 3.8 for complete instructions.

E. External Control Connectors - The DL2's on/off mode, level setting and input selection can be controlled by a remote control unit or by a home computer. At the factory this capability is used to help test the unit's operation. The DL2 test bench uses an S-100 (8080) computer with an IMSAI PIO-6 board to communicate with the unit under test. There is further information on this topic in Section 3.10, 3.11.

F. Power Supply Connector - J1 is the power connection to the Power Module.

G. Buffer Outputs - The output of the input; selector is fed to the buffer amplifier. Its output can be used with most peripheral equipment (reverb units, special effect devices, etc). When used with crossfeed inputs a total of three processors can be used in addition to the tape monitor loop. The only controls affecting the Buffer Outputs are the input selector and low filter. **Note:** The buffer output level is determined by the device selected at the input.

H. Main Outputs - There are two main stereo output sections from the DL2 that can be switch-selected from the front panel (Fig. 3.8O). Main output #1 has a Normal stereo output and a Mono output. Main output #2 provides a Normal output plus an Inverted output. A balanced, low-impedance output is available by using both the Inverted and Normal outputs. See Fig. 3.7 for a balanced output adapter. All Main Outputs (Normal, Inverted and Mono) remain shorted until the appropriate output selector switch is engaged.

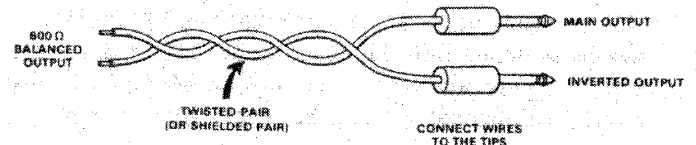


Fig. 3.7 Balanced Output Adapter

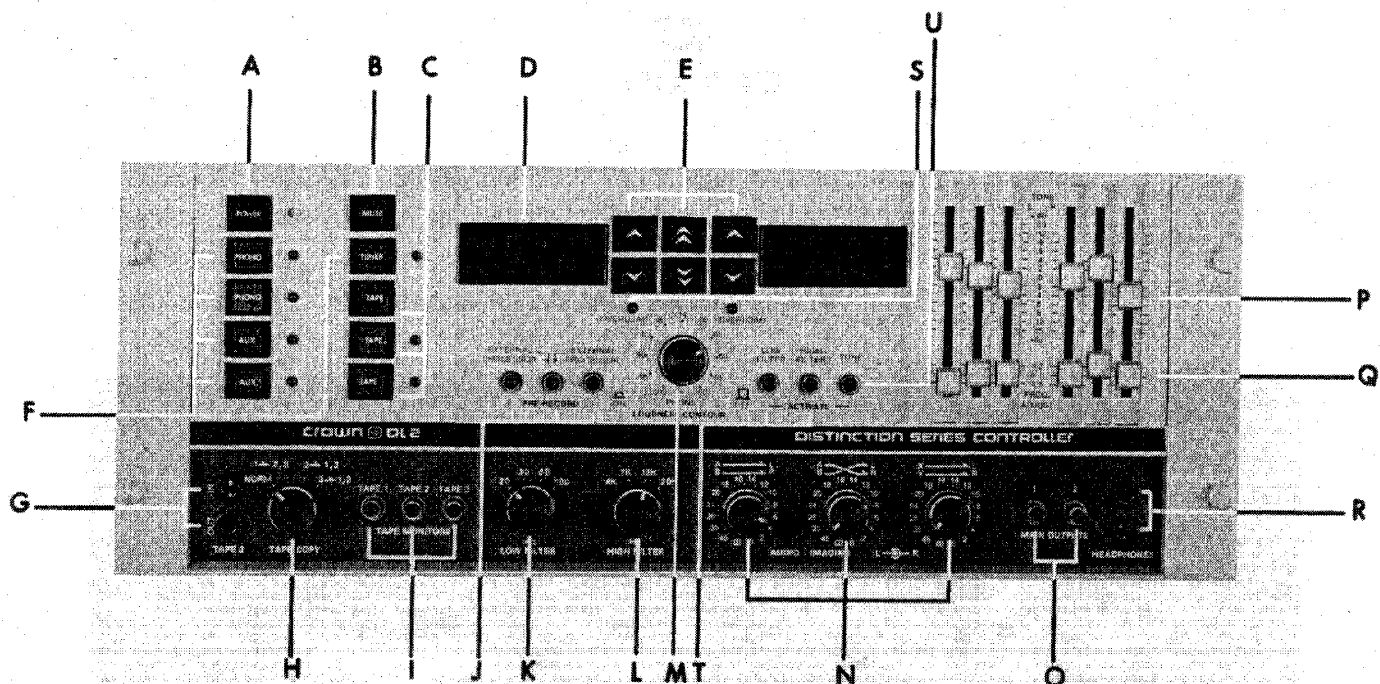


Fig. 3.8 Front Panel

3.5 Front Panel Controls and Functions

Refer to Fig. 3.8 for the following controls. All letters below refer to the labels on Fig. 3.8.

A, B, (C, F) Mode Switch Buttons (AC Power Control Circuits).

With all power cables connected, switching the master power switch (Power Module front panel) to ON puts the DL2 into the Standby and Mute modes. The amber Power-on LED on the Power Module comes on, and logic voltages are supplied to the DL2. The volume display lights, and reads 0. The yellow mute LED also is lit.

To place the DL2 in the operate mode, touch one or more Input Selector switches (C, F, Fig. 3.8). This "enables" that input(s) and the corresponding LED(s) lights. Also the amber Power LED lights and power is provided to all equipment connected to the Power Modules switched outlets. The mute delay circuitry is activated and the audio signal will be delayed for a few seconds after turn-on. When the mute relay switches out of mute, the yellow Mute light extinguishes.

Now by touching the Mute button (B, Fig. 3.8) the system can be switched into the Mute mode. This disables the main outputs and lights the yellow Mute LED. By retouching an input selector the operate mode is instantly reinstated.

Touching the Power button (A, Fig. 3.8) will place the DL2 into the Standby and Mute modes, in addition to

removing all service to the Power Modules switched outlets. The amber Power LED goes off, the volume display dims (but reads the last setting), and the LED(s) corresponding to the last selected input(s) remains lit. The DL2 is designed to operate continuously in this mode. To turn off the display and completely disable the system, the master power switch on the Power Module must be switched.

C.1, 2, 3, Tape Input Selectors

These three switch selector buttons perform the same power-on and other functions as the first five input select buttons, but have additional capabilities as outlined in Sections 3.5H and I.

D. LED Gain Displays

These seven-segment displays indicate pre-amp gain in dB. The gain is stepped precisely in .5dB steps, over a range of 0 - 63.5dB. (-43.5dB to +20dB actual gain). At a reading of 43.5dB, the preamp is operating at unity gain. In other words, the output level is the same as the input level.

Each display consists of four digits with the upper and lower segments on the digits closest to the center of the unit being used to indicate upper and lower limits. When the lower limit is reached, the DL2 switches to the Mute mode. (The yellow mute LED does not light, however.)

The display operates at full brilliance during the Operate mode and at reduced brilliance during the Standby mode.

E. Gain Control Buttons

The six buttons (four discrete channel, two masters) control the level of the DL2. The two center buttons step the level of both channels up or down (the arrows indicate the proper direction). The other buttons control each individual channel. For different levels on each channel, set them as desired individually. The center buttons will keep that balance.

The Gain buttons are speed conscious. In other words, the level will change slowly when the button is initially pushed. If the button is held, the level will change with increasing rapidity.

If the two channels are set to different levels for balancing purposes, a quick way to restore equal settings is to step the Gain to the lowest level (underange). One channel will reach the limit before the other, the second channel will then be stepped until it reaches that limit. The two then can be stepped to the desired level.

F 1, 2, 3, 4, 5 Input Selector Switch Buttons

The DL2 is supplied with Input Selector switch buttons with the following markings: (2) Phono, (2) Aux, (1) Tuner. These five buttons may be interchanged or arranged in any fashion. Replacements are available for any combination. So, for example, all the buttons could be marked Phono, or Aux.

The five input selector buttons correspond to the five main inputs as follows: (proceeding top to bottom, left column, then right column) Phono 1 - input #1, Phono 2 input #2, Aux 1 - input #3, Aux 2 - input #4 and tuner input #5.

As stated above, touching one or more Input buttons turns on the DL2. LED's beside each button indicate which input(s) has been selected. By selecting more than one input simultaneously, several inputs can be fed into the input buss.

G. Tape #3 Inputs and Outputs

These two jacks are three-conductor, 1/4" phone jacks. Electrically, they parallel the Tape 3 input and output jacks on the rear panel. See Fig. 3.5 for information on making an adapter cable for use with these jacks.

H. Tape Copy Switch

The Tape Copy switch provides a wealth of flexibility for systems requiring tape duplicating. The switch allows direct duplicating from one of three tapes recorders directly onto the remaining two. This operation is independent of the other five inputs. So, for example, it would be possible to be listening to a disc through the speakers while simultaneously playing a tape on one tape machine and copying it onto two others. To hear the tape playback from any one of the three recorders, press the respective Tape Monitor switch.

With the Tape Copy switch in the Norm (normal) position, the selected input signal is fed to all three tape recorders.

I.1, 2, 3 Tape Monitor Switches

These switches select the output from a recorder and feed it directly to the preamp input buss, bypassing the input selector switching. So with a three-head recorder, you may compare the input to the recorder and the output off-tape from the recorder, by engaging the appropriate Tape Monitor switch.

Another feature of these switches is an electrical lockout which prevents oscillations when a Tape Input is selected and that same input is switched into the monitor mode.

The Tape #1 switch has priority over the Tape #2 switch which has priority over the Tape #3 switch. So Tape #1 must be released in order to monitor tape recorder #2 or #3, etc.

J.1, 3 Signal Processor Switches

Up to two processors may be selected by the two pushbuttons #1 and #3. These pushbuttons correspond to the External Processor inputs and outputs #1 and #2 on the rear panel. The processor switching is independent from, and additional to the Tape Monitor switching, thus the problem of using the Tape Monitor facilities for signal processing is avoided.

Any processor such as an equalizer or a noise reduction unit may be used.

J. 2 Signal Processor Switch

The #2 switch controls the placement of the processors in the output circuits of the preamp. When the switch is out, the processors occur after the tape outputs (affect playback only). With the switch in, the processors are inserted before the Tape Outputs and can be used for special effects while recording.

K. L. High and Low Filters

The High and Low filters are activated by the filter activate switches (T1, 2). The control switches (K, L) select from four possible cutoff frequencies for each filter. The four selections are as follows:

Low: 20Hz, 30Hz, 50Hz, and 100Hz.

High: 4KHz, 7KHz, 12KHz, 20KHz.

M. Loudness Contour Switch

This switch allows the loudness contour to be adjusted over a wide range of personal preference. But before discussing the Contour switch function, a brief review of volume and loudness compensation is in order. See Section 3.9 for this discussion and how to use the Loudness Contour switch.

N. 1, 2, 3. Audio Imaging Controls

The Panorama control made famous in the IC-150 is again implemented in the DL2, but now in an expanded concept, called "audio imaging", offering even greater flexibility. The system uses three precision-stepped switches, Normal (#1), Reverse (#2), and Crossfeed (#3).

Since the three controls are dual-concentric (one switch for each input, six switches total), minute balancing is possible in any of the three modes. See Section 3.8 for detailed instruction.

O. 1, 2. Main Output Switches

The two Main Outputs are controlled by the two front panel switches #1 and #2. Engaging either or both switch buttons activates the corresponding output(s). With the switches disengaged, the respective output jacks are shorted thus eliminating transients on the input lines to following equipment.

P. 1, 2, 3. Tone Controls

The DL2 Tone Control system is a hinge-point, shelving system providing precision-stepped bass, midrange, and treble adjustments for both channels. The three controls (per channel) provide 15dB boost or cut at the selected frequency.

Q. 1, 2, 3. Center Frequency Select Switches

The three lower slide switches select the frequency at which the tone action takes place. Note the three positions are labeled X.5, X1 and X2. The center frequencies (X1) for the bass, midrange and treble are 40Hz, 800Hz, and 10KHz respectively. Thus the Tone Controls (P1, 2, 3) can be adjusted for the following center frequencies:

Bass x .5 = 20Hz

x 1 = 40Hz

x 2 = 80Hz

Midrange x .5 = 400Hz

x 1 = 800Hz

x 2 = 1600Hz

Treble x .5 = 5KHz

x 1 = 10KHz

x 2 = 20KHz

R. 1, 2. Headphone Output Jacks

Two stereo Headphone Jacks are provided on the front panel. These jacks remain on all the time irrespective of the output selector switches.

Most high quality headphones may be used in these jacks. Electrostatic, and certain other types have a very low sensitivity and will not function properly. They require a separate power amplifier.

S. Overload Indicators

Front panel LEDs indicate when the input signal is driving the preamplifier into clipping. If clipping is indicated, reduce the input level at the source.

T. 1, 2. High/Low Filter Switches

These switches simply activate the High and Low filters.

U. Tone Activate Switch

The Tone controls are inactive until this switch is engaged.

3.6 DL2 Phono Module

Refer to Fig. 3.4 for the following controls.

The DL2 Phono Module has a gain of 20dB and is variable controlled via two top panel potentiometers labeled GAIN A and B (Fig. 3.4D). The unit is shipped from the factory set at unity gain.

Input Impedance of the module is determined by the corresponding slide switch (Fig. 3.4G). The 47K position is used with most system cartridges. However, 100K is available when needed (as determined by the cartridge manufacturer).

When using the Phono Module as a microphone preamplifier, remove the RIAA phono equalization by moving the remaining slide switch (Fig. 3.4E) from "RIAA" position to "FLAT".

A ground terminal (Fig. 3.4F) is accessible for ground wire connection from your turntable. To eliminate annoying hum noise, it is necessary that this wire be connected!

3.7 DL2 Power Module

The Power Module provides nine AC outlets; seven switched, and two unswitched (Fig. 3.9A). Total current capacity is limited by the 20 amp power cord rating. The master power switch (Fig. 3.10A) and the amber power on LED (Fig. 3.10B) are on the Power Module front panel. The Power switch places the DL2 into the Mute and Standby modes.

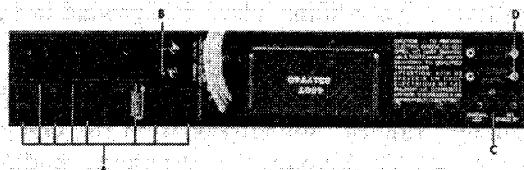


Fig. 3.9 Power Supply Rear Panel

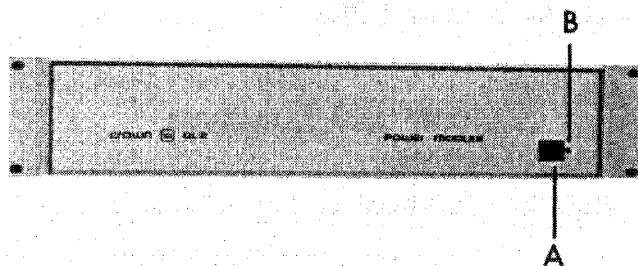


Fig. 3.10 Power Supply Front Panel

By activating one or more of the Input Selector switches on the DL2 control unit, a relay is latched in the power module. This activates the preamplifier by sending DC voltages to the other modules, the control unit and supplies power to seven switched AC outlets. This type of design minimizes interference from power line noise and unwanted outside signals. Rear panel labels identify the two unswitched and seven switched AC receptacles. The

two fuses (logic and audio, both .5 amp; (Fig. 3.9B) are also identified.

The switch located in the lower right corner (Fig. 3.9C) allows the Phono Module to be powered independently. If the switch is moved to the NO STANDBY position with the MASTER POWER switch on, the phono preamp will be powered, and can be used as a separate phono or mic preamp. The DL2 Controller will operate as previously described except that the audio power and switched outlets will always be energized.

3.8 How to Use the Audio Imaging Controls

The DL2 Audio Imaging system is truly a study of flexibility. The following discussion can only serve as an "inspiration" because the range of possible applications and adjustments is probably inexhaustible. Since the system is somewhat complex, a few minutes spent studying these illustrations will provide a good working knowledge of the system. That alone will be well worth the effort and will be the beginning of many "enjoyable hours" using the DL2 control center.

First notice the simplified Audio Imaging system drawing, Fig. 3.11. The system consists of four input busses, A, B, X, and Y fed into two output busses L and R. The four inputs are "mixed" onto the two outputs via six attenuators; two each Normal, Reverse, and Crossfeed. Each attenuator is step-adjustable from off (full counterclockwise), to full level (full clockwise).

The two attenuators labeled Normal independently adjust the A and B signals onto the L and R busses; while the controls labeled Reverse independently assign the A and B signals onto the R and L busses respectively.

Note: When using the Audio Imaging controls, it may be helpful to think in terms of the center portion of each imaging control affecting the output of the LEFT channel speaker system and the outer ring of each control affecting the output of the RIGHT channel speaker system.

This is an appropriate time to examine Fig. 3.12A. For normal operation, the Normal control is full on, and the Reverse and Crossfeed controls are full off. This provides the conventional A-B, left-right stereo imaging. Now, by rotating the Reverse control clockwise, both A and B

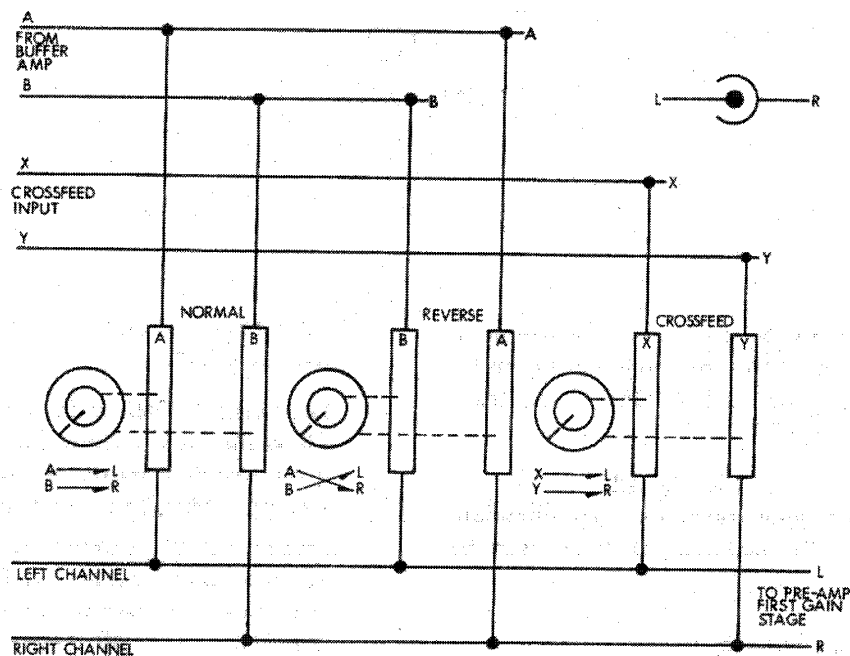


Fig. 3.11 Simplified Audio Imaging Control Diagram

signals will start converging toward the center. When both the Normal and Reverse controls are full clockwise, the A and B inputs are completely centered (Mono) as shown in Fig. 3.12B. (Note that any time the Normal and Reverse controls are in the same position, at any level, that input will be centered, or in Mono.) If the Normal control is now rotated counterclockwise, the A and B inputs will pan toward the right and left channels. Reverse stereo will occur when the Normal control is fully counterclockwise and the reverse controls are clockwise, Fig 3.12C.

Since the six Audio Imaging controls are attenuators, a very useful feature inherent in the system is input balancing. In Fig. 3.12D, E, F, notice that all controls are partially open. The result is a lower level depicted by the smaller image for these drawings. Since the A and B inputs are independently adjustable, intricate balance or image placement changes are possible.

Fig. 3.13A, B, C illustrates image placement. For simplicity, the B input is omitted. Notice how the image can be placed anywhere from left to right. Add the B input and precision balancing is possible. Since the attenuators are precision-stepped switched, it is quite simple to obtain exact resettability.

Fig. 3.14A, B, C illustrates the crossfeed function. By rotating the Crossfeed control clockwise, the X and Y signals are fed to the Left and Right channels, totally independent of the A and B inputs. The normal-reverse function for the A and B inputs is still intact. Fig. 3.14D, E, F illustrate the "mixing" capability of the system. Shown is a panorama function with A and B at a lower level than X and Y.

The applications for this mixing feature are unlimited. The following examples present only a few ideas.

- Practice your electric bass with a favorite recording artist. Feed the output of the phono preamp into the A and B inputs; adjust A and B levels appropriately. Feed output of bass preamp into the Crossfeed inputs. Adjust bass level to your liking; place background in normal or reverse stereo.
- Your local ensemble is doing a nightclub performance. Feed the "canned" soundtrack from the tape recorder into the crossfeed inputs. Use the phono module "Flat" for two mic preamps; feed into A and B inputs. Adjust all levels accordingly. You will have full stereo separation in the soundtrack, and can balance or pan the two vocal mics for any desired effect.

3.9 Volume and Loudness Compensation

(How to use the Loudness Contour switch)

Audio Engineers spend substantial amounts of time designing audio equipment with uniform frequency response over the entire audio spectrum. One intentional departure from this "perfectly flat" response is the introduction of loudness compensation.

The theory of loudness compensation is based on psychoacoustical research which demonstrates that human hearing responds in a frequency dependent manner. Low frequency sensitivity decreases as sound pressure decreases.

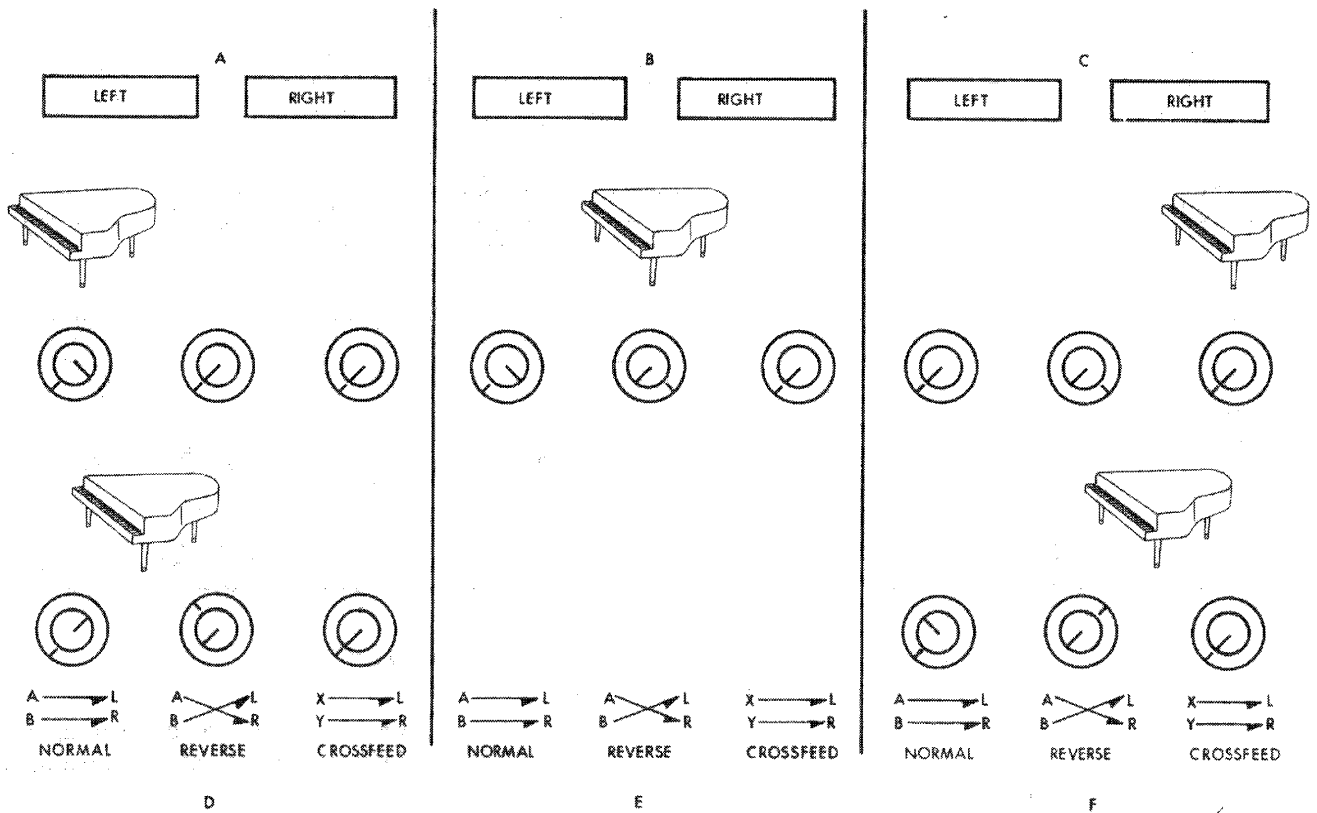


*Fig. 3.12 Audio Imaging Controls - Normal,
Mono, Reverse, Panorama*

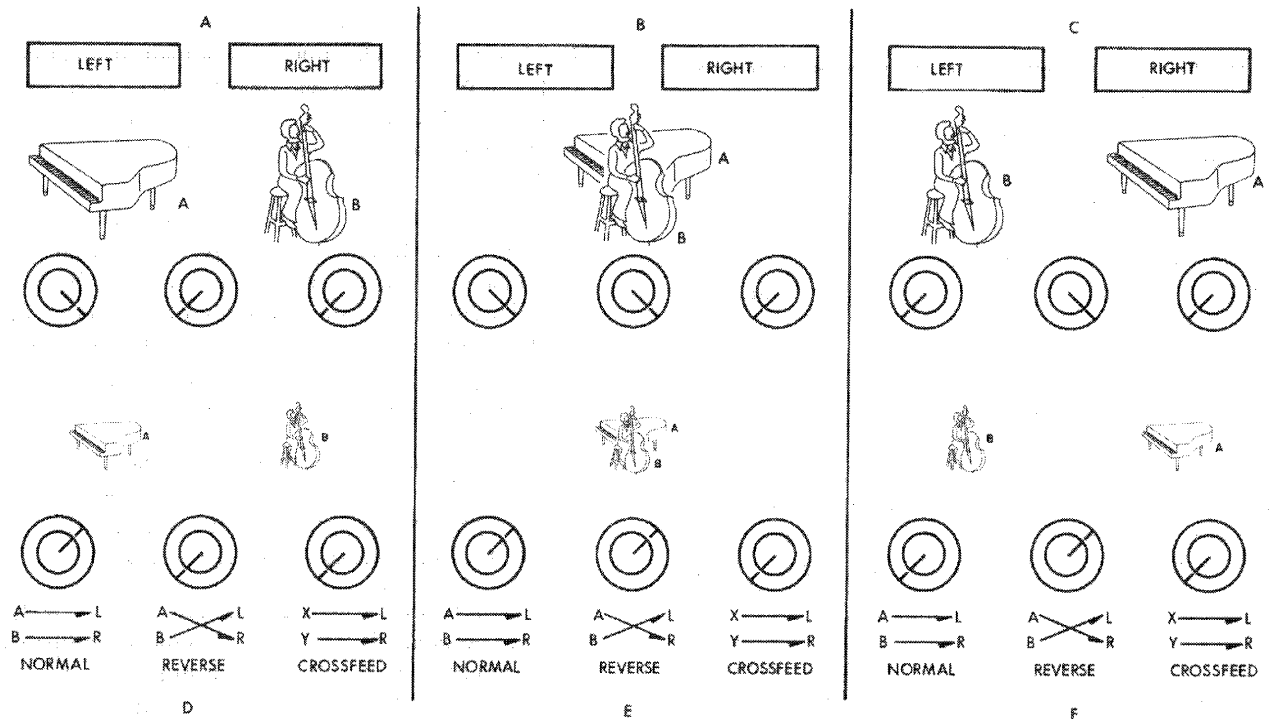


*Fig. 3.13 Audio Imaging Controls - Image
Placement*

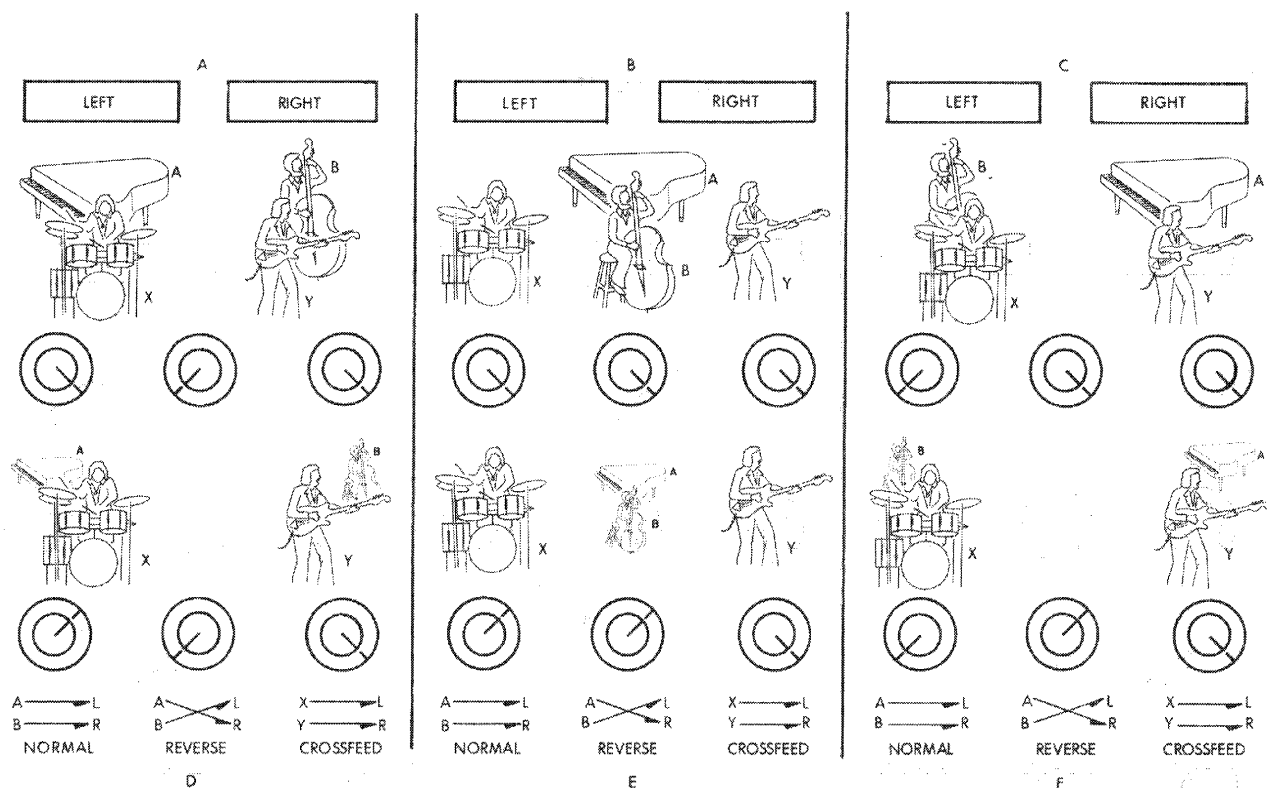


Fig. 3.14 Audio Imaging Controls - Crossfeed Input Mixing

Many experiments have been made in which listeners have been asked to rate the apparent loudness of many steady tones throughout the normal audio range. Under carefully controlled laboratory conditions, these listeners were asked to compare tones of high and low frequencies with a 1KHz reference tone. From this the researchers plotted a set of equal-loudness contour curves.

The work of two research teams has been widely accepted throughout the audio field. The findings of Fletcher and Munson (1933) has been used as a basis for loudness compensation for many years. More recently the I.S.O. (International Standards Organization) has accepted the work of a second team, Robinson and Dadson (1956), as being more accurate. As a result, the trend in the audio industry is to use the I.S.O. curves as a basis for loudness compensation.

The design objective then, is a compensation circuit which adjusts the system response to compensate for hearing deficiencies at low sound levels. The starting point is to pick a "normal listening level". A sound pressure level of 100dB is widely accepted as this "normal listening level". Now we make the assumption that the net auditory perception is "flat" at this "normal" level. (This isn't quite true, of course, but remember that hearing is increasingly uniform at louder levels. 100dB is fairly loud!) Now the response curves for the lower listening levels are adjusted with respect to the "flat" 100dB curve and a set of normalized loudness correction curves is plotted. These are illustrated in Fig. 3.15. These

Normalized curves serve as the basis for designing the volume control/loudness compensation in preamplifiers.

As a particular example, observe that the 40 phon contour (which intersects 40dB SPL at 1KHz) increases at low frequencies until it reaches the 63dB level at 50Hz. This indicates that a 50Hz tone would have to be boosted 23dB to maintain equal loudness with the 1KHz tone at this sound level. The loudness compensation circuitry of a preamplifier provides this boost in the low (and sometimes the high) frequencies in order to make the loudness of each frequency match the loudness of the 1KHz tone.

Not every designer uses the high-frequency correction shown in the I.S.O. curves (Fig. 3.15) because of the discrepancy between the findings of Fletcher-Munson and Robinson-Dadson in the high frequencies. Because of this controversy it is questionable whether high frequency emphasis is necessary or desirable. The DL2, for example, does not use high frequency emphasis.

Most preamplifier designs combine the loudness compensation and gain circuitry in one control, the volume control. So at low volume settings, the loudness compensation is greatest and vice versa.

The DL2 is unique in that the gain control is digitally stepped independent of the loudness compensation. Additionally, the Loudness Contour control is a dual-function control. It steps the gain in 5dB steps as well as changing the contour. See Fig. 2.9. So in the technical sense, it is really a volume control.)

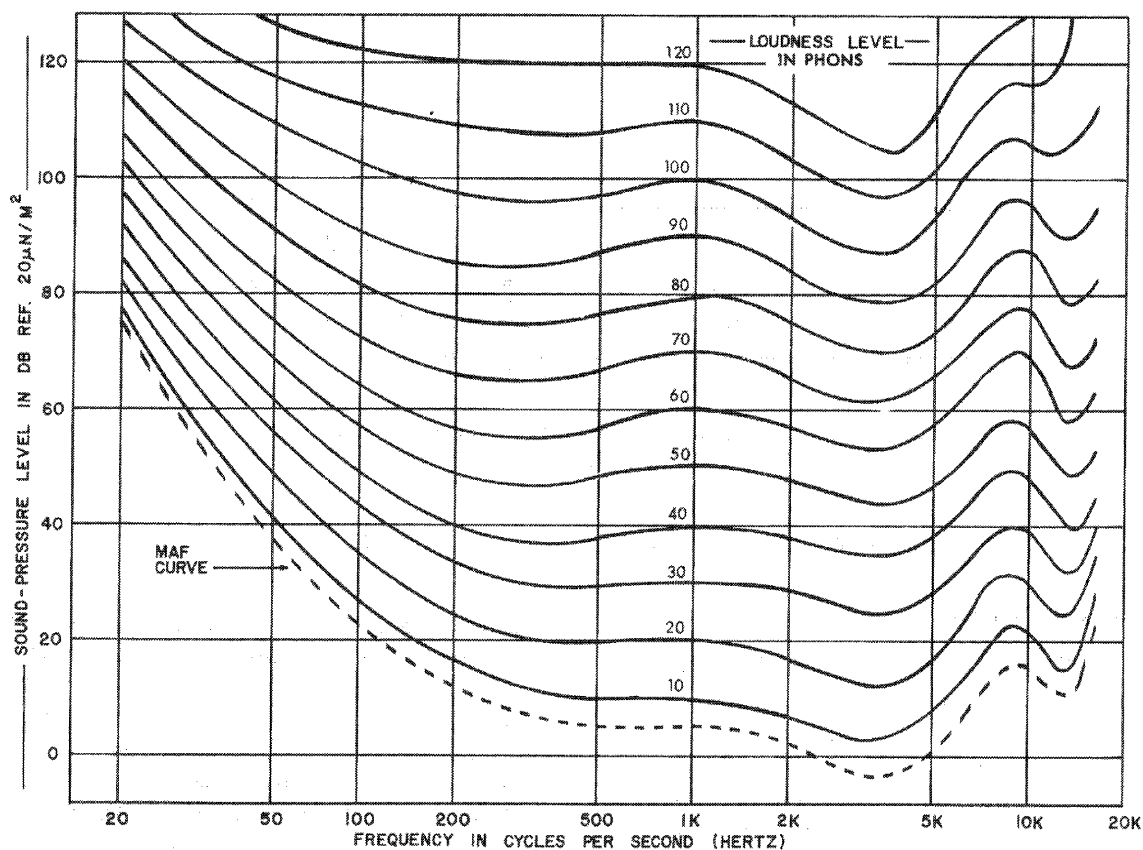


*Fig. 3.15 Normalized Loudness Correction
Curves (ISO)*

With this degree of flexibility, precise adjustments are possible. In fact, with test equipment and the normalized loudness correction curves (Fig. 3.15) it would be possible to predetermine a correct loudness contour setting. However, since Fig. 3.15 represents "average" hearing and few people own test equipment, a more practical method is required.

The simplest approach would be to set the Loudness Contour switch to 100 phons. Next adjust the program level with the digital gain controls to full or maximum listening level. Now the Loudness Contour may be used as a volume control. As the level is decreased, loudness compensation is automatically provided. As per Fig. 2.9 the pre-amp response is flat at a setting of 100 phons and achieves maximum compensation at a setting of 55 phons.

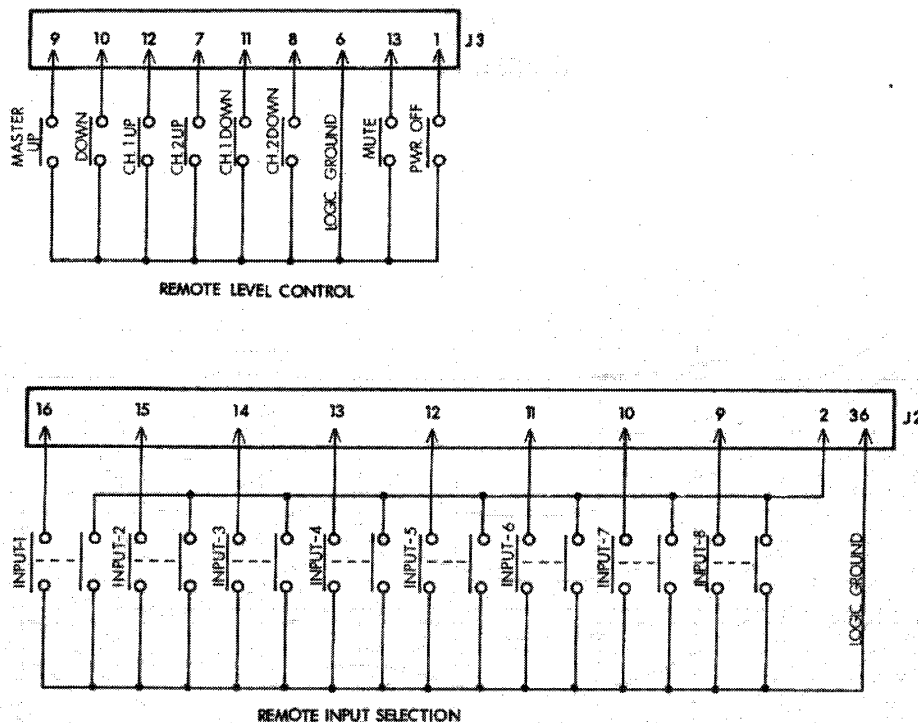


Fig. 3.16 Remote Control Wiring Diagram

3.10 Remote Control

It is possible to control the On/Off mode, gain level and input selection functions of your DL2 at a remote location. This is done by constructing a remote control box consisting of several momentary Double Pole-Single Throw switches and wiring them through a cable harness to the External Control Jacks (J2, J3) located on the rear panel. A suggested wiring method for such a circuit is shown in Fig. 3.16.

3.11 Computer Control

The DL2's level setting and input selection can be controlled by a home computer. At the factory this capability is used to help test the unit's operation. The DL2 test bench uses an S-100 (8080) computer with an IMSAI PIO-6 board to communicate with the unit under test.

The IMSAI PIO-6 board is a six port programmable input/output board. Three of these parallel eight bit ports are needed in order to fully communicate with the DL2's control busses; however, if a DL2 is the only device on this buss, or if there are several DL2's all using the same device address, only two and one half ports will be needed. This is accomplished by "hard-wiring" in the device address.

One output port is used for the address buss. The address buss carries the information needed to tell the DL2 which of its input ports the operator is trying to control, either input selection, Ch. 1 level set, Ch. 2 level set, or power off. If the user will always set Ch. 1 and Ch. 2 levels to the same level, he may want to set their ports to the same address. This will then require only one pass through a control program in order to set the levels in both channels.

The second output port is used for the data buss. The data buss carries the information needed to tell the DL2 which input is selected or what level is to be set. In Fig. 3.17 is a list of inputs and levels and the corresponding hexadecimal numbers and decimal equivalents for each. These hex numbers, when written on the data buss to the correct address, will accomplish the level or input selection.

The third port is split in half, four bits as input and the other four bits as output, only one bit of each is used. The input bit will monitor the BSY, which is low when the DL2 or any other controller is using the data buss. The output bit is used to drive a transistor that will pull down the BSY line if high, thus denying any other controller access until the computer has finished its operation. When the computer is finished, it will allow the BSY line to go high, which will then make it possible for the DL2 (etc.) to use the data buss.



INPUTS	DECIMAL	HEX						
1	254	FE1	16.5	33	21			
2	253	FD	17.0	34	22	40.5	81	51
3	251	FB	17.5	35	23	41.0	82	52
4	247	F7	18.0	36	24	41.5	83	53
5	239	EF	18.5	37	25	42.0	84	54
6	223	DF	19.0	38	26	42.5	85	55
7	191	BF	19.5	39	27	43.0	86	56
8	127	7F	20.0	40	28	43.5	87	57
			20.5	41	29	44.0	88	58
			21.0	42	2A	44.5	89	59
Mute and Underage Level:			21.5	43	2B	45.0	90	5A
	128	80	22.0	44	2C	45.5	91	5B
			22.5	45	2D	46.0	92	5C
LEVEL	DECIMAL	HEX	23.0	46	2E	46.5	93	5D
0	0	00	23.5	47	2F	47.0	94	5E
.5	1	01	24.0	48	30	47.5	95	5F
1.0	2	02	24.5	49	31	48.0	96	60
1.5	3	03	25.0	50	32	48.5	97	61
2.0	4	04	25.5	51	33	49.0	98	62
2.5	5	05	26.0	52	34	49.5	99	63
3.0	6	06	26.5	53	35	50.0	100	64
3.5	7	07	27.0	54	36	50.5	101	65
4.0	8	08	27.5	55	37	51.0	102	66
4.5	9	09	28.0	56	38	51.5	103	67
5.0	10	0A	28.5	57	39	52.0	104	68
5.5	11	0B	29.0	58	3A	52.5	105	69
6.0	12	0C	29.5	59	3B	53.0	106	6A
6.5	13	0D	30.0	60	3C	53.5	107	6B
7.0	14	0E	30.5	61	3D	54.0	108	6C
7.5	15	0F	31.0	62	3E	54.5	109	6D
8.0	16	10	31.5	63	3F	55.0	110	6E
8.5	17	11	32.0	64	40	55.5	111	6F
9.0	18	12	32.5	65	41	56.0	112	70
9.5	19	13	33.0	66	42	56.5	113	71
10.0	20	14	33.5	67	43	57.0	114	72
10.5	21	15	34.0	68	44	57.5	115	73
11.0	22	16	34.5	69	45	58.0	116	74
11.5	23	17	35.0	70	46	58.5	117	75
12.0	24	18	35.5	71	47	59.0	118	76
12.5	25	19	36.0	72	48	59.5	119	77
13.0	26	1A	36.5	73	49	60.0	120	78
13.5	27	1B	37.0	74	4A	60.5	121	79
14.0	28	1C	37.5	75	4B	61.0	122	7A
14.5	29	1D	38.0	76	4C	61.5	123	7B
15.0	30	1E	38.5	77	4D	62.0	124	7C
15.5	31	1F	39.0	78	4E	62.5	125	7D
16.0	32	20	39.5	79	4F	63.0	126	7E
			40.0	80	50	63.5	127	7F

Fig. 3.17 INPUTS and LEVELS with Hex #'s
and Decimal Equivalents

An 8080 machine language program that will communicate with the DL2 is shown in Fig. 3.18. This program is written as a subroutine and could be accessed by performing a CALL statement in a BASIC language or a CALL statement in a larger machine language program. In the control subroutine, the address is in the D register and the data is in the E register.

The sample control subroutine starts at address 0000, but can be moved to any place in available memory by adding a constant to each address. All JUMP statements and other statements using addresses that will change if the subroutine is moved must also have the constant added to them. (Remember in 8080 language addresses used in statements are recorded with the low order byte first.) All numbers in the subroutine are recorded as HEX values.

The first thing done in the control subroutine is to push PSW and H. By pushing PSW onto the stack, the accumulator's last contents and the state of any flags will be saved. Since register H is used, it is also pushed onto the stack.

The next operation is to test the BSY line for a low state which indicates the buss is busy. This program uses a multiple test for BSY. The number of times to test BSY is loaded into the H register, a loop is then entered that decrements H, inputs the port containing the BSY line (BSY is a bit 0 of a four bit port), and performs an ANI (AND IMMEDIATE) with 0 Hex. If the zero flag is set, indicating unit is busy, a jump back to the point where H was initialized will be executed and the loop will be reentered. If the flag is not set, indicating the unit is not busy, an ADD H will be performed (add register H and accumulator). If the result is zero, a jump back to DCR H (decrement H) will be performed. When the result is zero, the program will then take control of the DL2. This first part of the program just described continues to test the BSY line, resetting the H register to FF Hex each cycle until BSY is high, indicating the DL2 is not busy. It proceeds to test the BSY 256 times in order to insure that the DL2 buss is not busy.

The next step is MVI A, 10 Hex (MOVE IMMEDIATE TO A, 10 Hex) which supplies the accumulator with the address needed to set the BSY line low. Out is the output to the 4 bit output port used to pull down BSY line (in this case bit 4 of port C). MVI A, 81 HEX, OUT - these two instructions set the accumulator to the address needed to configure ports A and B as outputs (these two ports must be initialized as inputs before this program is run) and then outputs this information to the PIO-6 board. MOV A, D moves the address information for the DL2 from register D to the accumulator. Out then asserts this address on port A. MOV A, E moves the data for the DL2 from register E to the accumulator. OUT then asserts this data on port B. This completes the command delivery to the DL2.

Next, a 30ms delay is started. This delay is needed in order to hold the data on the buss long enough for the DL2 to recognize the data and make sure it is valid. To make sure the data is valid, the DL2 has a time delay between the time the address has been asserted and the time when it starts looking at the data buss. The time delay is accomplished by going through a loop enough times to use up 30ms. The loop is initiated by loading into the H register the number of times the loop must be gone through (2500 times), LXI H, C409. The loop then starts by decrementing H, MOV A, H (move register H into the accumulator), ORA L (OR the accumulator with register L), JNZ DLY (if the result of the previous statement is not zero, a jump back to the beginning of the loop DLY is performed). When the loop has been completed the accumulator is set to zero, XRA A. The accumulator contents are then output to port A, (OUT). This releases the address on the DL2. MVI A, 93 Hex moves the address, needed to change the configuration of the A and B ports back to inputs, into the accumulator. OUT sends this information to the PIO-6 board. XRA A is used to zero the accumulator. OUT releases the BSY line by writing zeros into the C port. The only thing left to do now is to return register H, the accumulator, and the flags to the state they were in just before entering this subroutine. This is accomplished by first POP H and the POP PSW. To get back to the calling program, a RET statement is needed.

Using this program, a computer can change the levels and the input. As mentioned before, the operation performed depends upon the address entered and the setting depends on the data entered. The address occupies the high order bits of the DE register pair and the data occupies the lower order bits.

The address is split into two parts. The high order bits, A4-A7, are used to select the desired operation. The lower order bits, A0-A3, are the device address. The addresses are preset at the factory to the following: Device, E; Level Ch. 1, B; Level Ch. 2, 7; Power off, D; Load input, E.

The full address for each operation would then be as follows:

Operation	Hex	Binary (A7-A0)	Decimal
Level, Ch. 1	BE	1011 1110	190
Level, Ch. 2	7E	0111 1110	126
Power off	DE	1101 1110	222
Load input	EE	1110 1110	238

If the addresses need to be changed for some reason, it can be done by changing the appropriate jumpers on the logic board. The logic board is layed out for SPDT rocker switches to set the addresses, so if for some reason the addresses need to be changed often, the user can install switches at locations S1-S5. Fig. 3.19 shows the

Label	Address	Machine Code	Op Code	Remarks
		3 E 9 3 D 3 A 3		Initialization statements
START	0000	F 5	PUSH PSW	Address in D Data in E
	01	E 5	PUSH H	
ACCESS	02	2 6 F F	MVI H FF	# Tests of <u>BSY</u>
COUNT	04	2 5	DCR H	
	05	D B A 2	IN A2	
	07	E 6 0 1	ANI 01	Test C0 for <u>BSY</u>
	09	C A 0 2 0 0	JZ ACCESS	
	0C	8 4	ADD H	
	0D	C 2 0 4 0 0	JNZ COUNT	
	10	3 E 1 0	MVI A 10	
	12	D 3 A 2	OUT A2	Assert <u>BSY</u> on C4
	14	3 E 8 1	MVI A 81	
	16	D 3 A 3	OUT A3	Engage Output Ports
	18	7 A	MOV A, D	
	19	D 3 A 0	OUT A0	Assert address on Port A
	1B	7 B	MOV A, E	
	1C	D 3 A 1	OUT A1	Assert data on Port B
	1E	2 1 C 4 0 9	LXI H C409	Delay (30MS)
DLY	21	2 B	DCX H	
	22	7 C	MOV A, H	
	23	B 5	ORA L	
	24	C 2 2 1 0 0	JNZ DLY	
	27	A F	XRA A	Zero Acc
	28	D 3 A 0	OUT A0	Release address
	2A	3 E 9 3	MVI A 93	Change Ports A&B to IN
	2C	D 3 A 3	OUT A3	
	2E	A F	XRA A	Zero Acc
	2F	D 3 A 2	OUT A2	Release <u>BSY</u>
	31	E 1	POP H	
	32	F 1	POP PSW	
	33	C 9	RET	

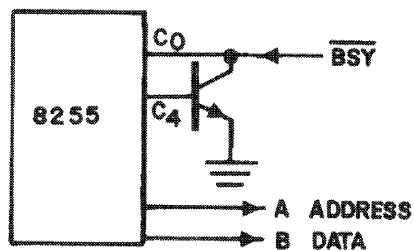


Fig. 3.18 Machine Language Program

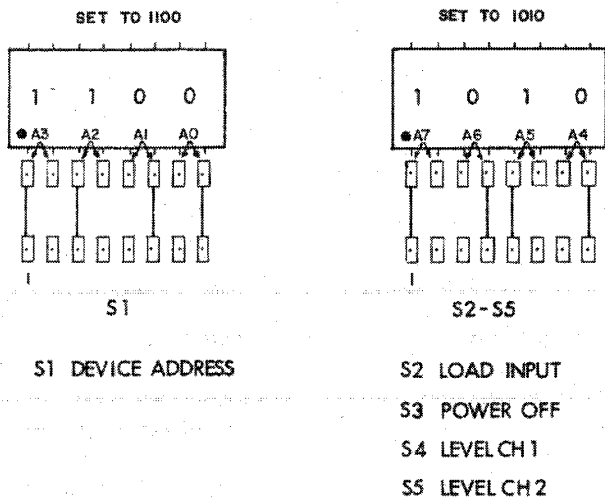


Fig. 3.19 Address Switch Layouts

address switch layouts and what function each performs, also two examples showing the jumper locations for two different addresses are given.

To demonstrate better how to access the subroutine and get it to perform the desired operation, the following example will be worked out.

To set Ch. 1 level to 42.0, the first information needed is the data for 42.0. Looking at Fig. 3.17 the data is found to be 54 Hex and 84 decimal. For the CALL statement, the decimal number is needed. The next information needed is the address for Level Ch. 1 in decimal. Looking in the table, the address is found to be 190. Since this needs to be the high order number when translated into the machine language DE register pair, the 190 must be multiplied by 256. This makes the correct number 48,640. The number for the data is 84 and the number for the address is 48,640. In order to get the single number that will translate into the DE register pair, the address and data numbers must be added together, giving 48,725. The last information needed is the address of the machine language subroutine. In the sample program, the beginning address is 0000 HEX which translates to 0 decimal.

All the information to fill out the CALL statement has now been determined. For example, Processor Technology Extended Cassette Basic's CALL format is as follows:

CALL (exp 1, exp 2) Calls a routine at address exp
 passing optional exp 2 in
 registers D and E
 numerical expression between 1 & 65535
 numerical expression between 0 & 65535

The CALL statement for this example will then be as follows:

CALL (0, 48725)

When this statement is executed it will go to address 0000 and start executing after register D is loaded with HEX (BE) or 190 decimal and register E is loaded with HEX (54) and decimal 84.

This example program is just that, an example. From the use of the DL2 as only a manually controlled stereo preamp at one extreme, to its use as the audio controller section of a totally computer controlled home environment at the other, the DL2 is a very versatile audio tool. The limiting factors in its use as a computer controlled instrument are the user's expertise in programming and the versatility of his computer system. As these things grow, so will the possible uses for the DL2.



SECTION 4

THEORY OF OPERATION

4.1 General

The DL2 is a precision, high-quality audio preamplifier. The Crown designed and developed circuitry represents a level of quality and performance presently unequaled in the field of audio preamplifier design.

4.2 Principles of the DL2 Controller

The DL2 uses digital logic to control the analog signal path with the use of high quality reed relays. The main signal path is not affected by filters, tone circuits or external signal processors unless the function is switched manually. By using this straight signal path in conjunction with very fast, high quality IC audio amplifiers, the DL2 produces the most sonically pure signal that technology will allow. All forms of distortion including slew-induced distortion (TIM) in the DL2 are not perceptible by the human ear and most test equipment.

The tone circuits are Crown's famous hinge point shelving circuits similar to the EQ-2 equalizer. The high and low cut filters have selectable center frequencies. The loudness contour switch provides selectable contours for various loudness levels.

By using an external phono module, the DL2 system realizes many advantages over the in-board preamp. The resulting sonic purity is truly "state of the art". The module is designed to be mounted as close to the turntable as possible. This allows short leads to the cartridge reducing cartridge loading, high frequency loss, and RF interference.

The power supply module is a separate module that can be placed on the floor out of sight or rack mounted with the DL2. By separating the power supply from the circuitry, Crown has again improved signal noise and hum rejection.

4.3 Block Diagram Circuit Theory

The signal source is fed to the input control section. This section is controlled by both the digital logic control and manual switches. See Fig. 4.1. Notice that the main signal path is straight through with no circuits that will color or alter the signal in line. Circuits that are used to alter the signal, such as the tone system, filters, etc., are switched in and out of the signal path manually.

The signal goes to the buffer amplifier where several outputs occur. The low filter system is switched into the buffer amp. A buffered output is provided from this point to the rear panel and the external processor control gets its output. The external processor control can return the signal after the buffer amp or return it to the input control so that the signal is before the tape outputs.

From the buffer amp the signal goes to the panorama control system. Here the signal is switched to effect channel balance, normal, mono and reversed stereo modes and mixing with the X and Y input feeds. From the panorama system the signal goes to the first gain amplifier section. The loudness contour system is switched at this point. From here the main signal proceeds to the second, third and fourth gain amps. At this point the high filter is switched in or out. Note that the digitally controlled volume control system primarily controls the gain of the four gain amps.

The signal now flows to the output section. Here is where the Crown hinge point shelving tone control systems is switched. This is a precision switched system that allows the user to return to an exact setting. The signal continues to the output amp, inverting amp, mixing amp and to the headphone amp. From these amplifiers, the signal goes to the rear panel through the muting relay and output control switches.

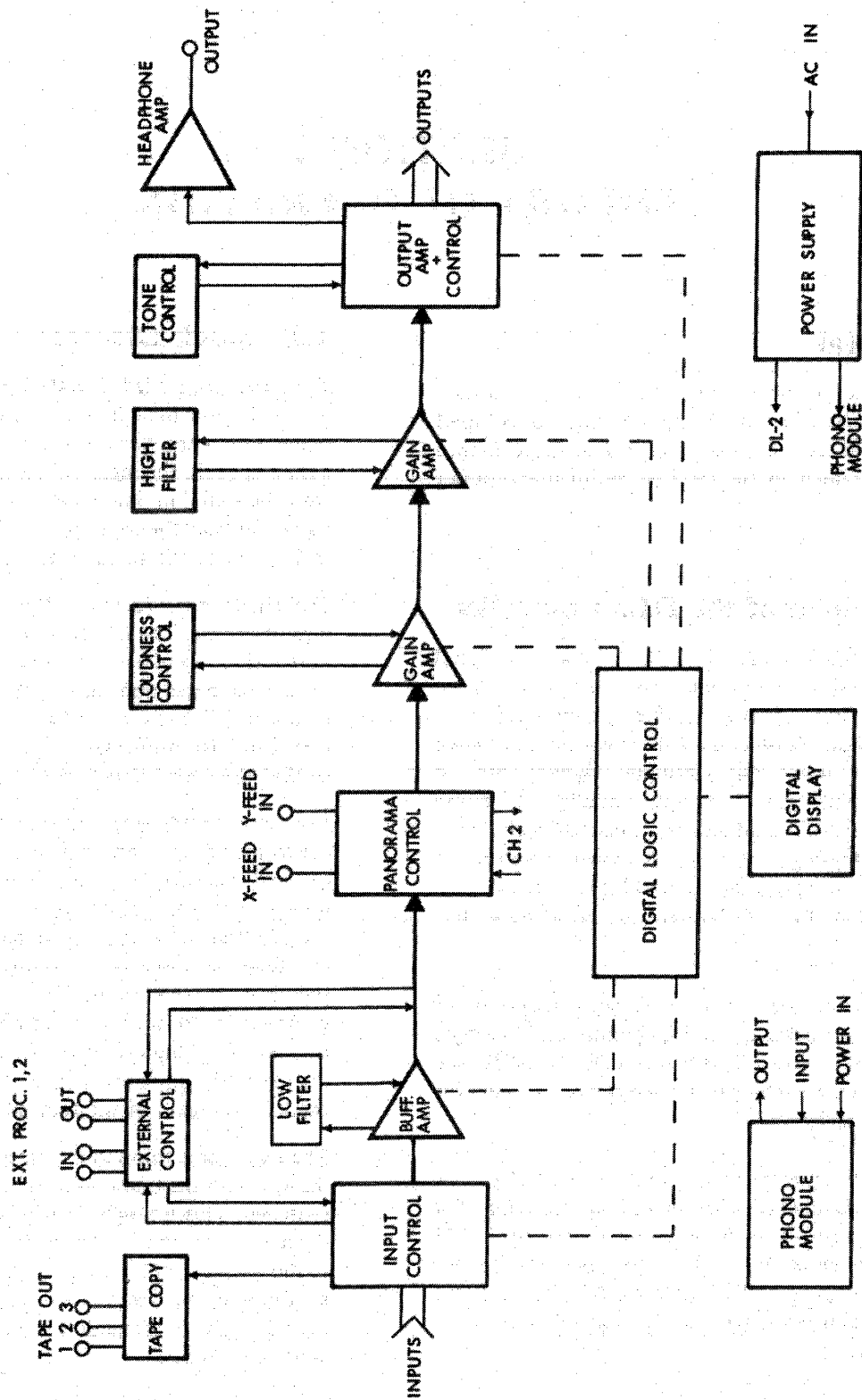


Fig. 4.1 DL2 Block Diagram



SECTION 5

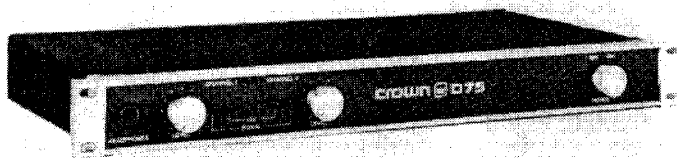
ACCESSORIES/OPTIONS

INTRODUCTION

This section contains information concerning Crown products, accessories used with these products, and optional equipment available at an additional cost to the consumer.

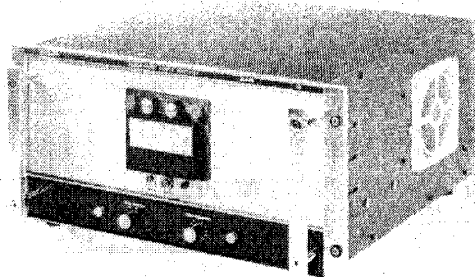
D-75 AMPLIFIER

The CROWN D-75 single or dual channel amplifier (35 watts per channel, 8 ohms; 45 watts per channel 4 ohms) may be used for driving efficient speaker systems, as a headphone amplifier, or as an amplifier for the ambience channels in a four channel system.



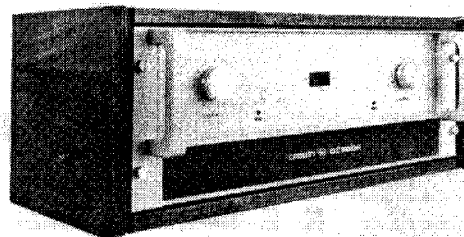
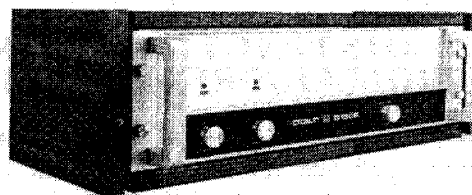
M-600 AMPLIFIER

The CROWN M-600 power amplifier provides 600 watts of continuous power into 8 ohms, and 1000 watts into 4 ohms. An adaptable input module format fits the varied need and operating environments of industrial and commercial audio uses.



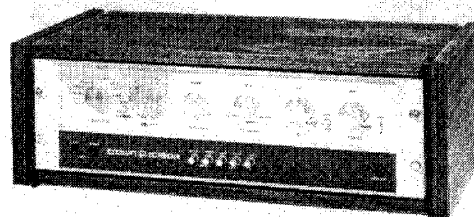
D-150A, DC-300A AMPLIFIERS

The CROWN D-150A (80 watts per channel into 8 ohms; 125 watts per channel into 4 ohms) and DC-300A (155 watts per channel into 8 ohms; 250 watts per channel into 4 ohms) are single or dual channel amplifiers designed for precision amplification of frequencies from DC to 20KHz. These amplifiers also provide extremely low harmonic and intermodulation distortion with very low noise.



IC-150A CONTROL CENTER

The CROWN IC-150A is an input control center and preamplifier designed for the professional user and the sophisticated audiophile.



STRAIGHT LINE ONE PREAMPLIFIER

The CROWN SL-1 is a sophisticated high performance pre-amplifier. The superior technology and "straight wire" approach provide precise signal control for any sound system.

POWER LINE ONE AMPLIFIER

The CROWN PL-1 is a high quality audio power amplifier (50 watts per channel into 8 ohms; 80 watts per channel into 4 ohms) designed to compliment the SL-1 stereo preamplifier.



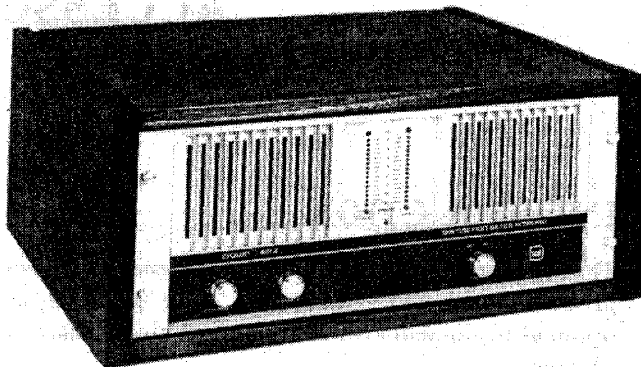
DL-2 DISTINCTION SERIES CONTROLLER/PREAMPLIFIER

The CROWN DL2 is a digital control center and preamplifier designed for the highest sonic qualities attainable by today's technology. It incorporates numerous unique features including 3 module construction; phono preamp, control unit and power supply.



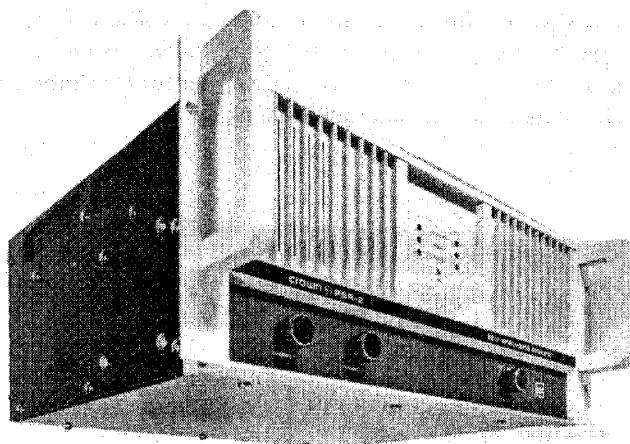
SA2 AMPLIFIER

The CROWN SA2 is a 220 watts per channel power amplifier designed for the highest sonic quality and reliability attainable by today's technology. It employs the Crown Self-Analyzing output system, which uses analog computers to continually adjust the amplifier's output limits.



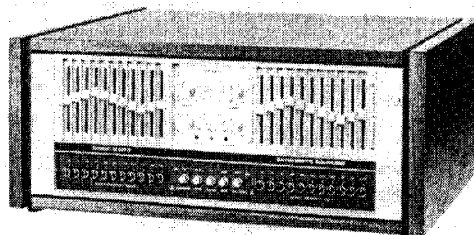
PSA-2 AMPLIFIER

The CROWN PSA-2 is the professional counterpart of the SA2 power amplifier. It incorporates many unique features for use in today's demanding sound systems.



EQ2 EQUALIZER

The CROWN EQ2 is designed to take care of frequency amplitude problems due to room acoustics, poor loudspeaker response, and poor cartridge response. A flexible tone control system is combined with 22 half octave width filters with movable center frequencies.





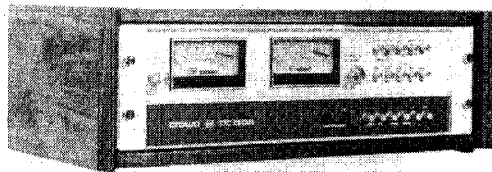
FM-1 TUNER

The CROWN FM-1 is a precision Stereo FM Tuner including such features as digitally-synthesized tuning, superior sound quality, precise tuning, and ease of use make the FM-1 unique.



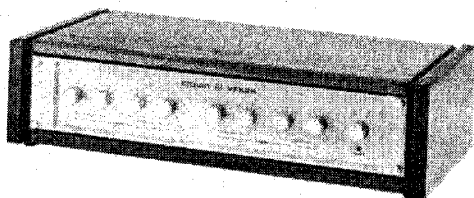
OC-150A CONTROL CENTER

The CROWN OC-150A is designed for precision output monitoring and amplifier/speaker switching.



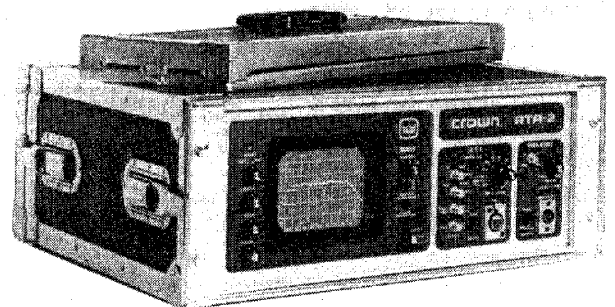
VFX-2A CROSSOVER/FILTER

The CROWN VFX-2A provides two sets of continuously variable filters which can be used to perform either crossover or bandpass functions.



RTA-2 REAL TIME AUDIO ANALYZER

The CROWN RTA-2 is an audio band Real Time Analyzer designed to measure acoustical and actual frequency responses in third octave bands from 16Hz to 20KHz.



6' PIN TO PHONE CABLES — CPN 3339



6' PHONE TO PHONE CABLES — CPN 4363



6' PIN TO PIN CABLES — CPN 3338



1 $\frac{3}{4}$ " RACK EARS — CPN 4800

Use with the D-75.

3" RACK EARS — CPN 41667

Used with the VFX-2A and DL2 POWER module.

5" RACK EARS — CPN 4802

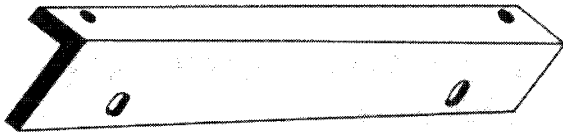
Used with the D-150A, IC-150A, OC-150A, FM1.

7" RACK EARS — CPN 20037

Used with the DC-300A.

7" RACK EARS — DISTINCTION SERIES — CPN 4267

Used with the DL2, SA2, EQ2 and RTA-2.



MOUNTING SCREWS — CPN 20032

Used with Crown equipment to secure equipment to rack mounts.



ALLEN WRENCH — CPN 3454

Used with Crown equipment to remove end bars and secure rack ears to equipment.



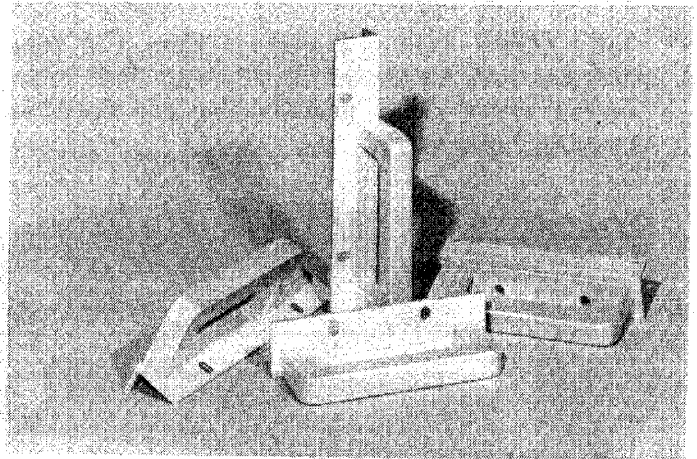
HANDLES

5" — CPN 41855 (D-150A, FM-1)

7" — CPN 41887 (DC-300A)

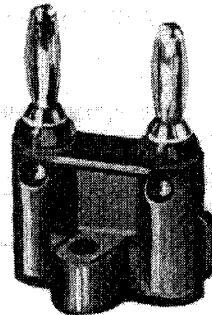
DISTINCTION — CPN 41889 (EQ2, DL2, and SA2)

These handles are used with Crown equipment for easier handling.



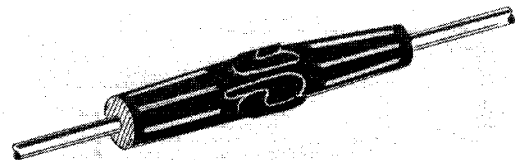
DUAL BANANA PLUGS — CPN 2981

Used with the D-75, D-150A, DC-300A, PL-1, OC-150A, and SA2 as part of the HiFi kit.



FUSEHOLDER — CPN 4245

Used with the D-75, D-150A, DC-300A, OC-150A, PL-1, and SA2 as part of the HiFi kit.



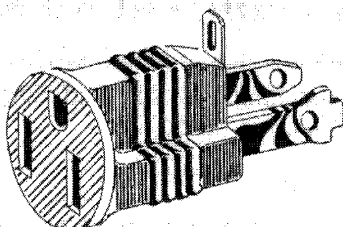


WIRE NUTS — CPN 3069

Used with the D-75, D-150A, DC-300A, PL-1, and SA2 as part of the HiFi kit.

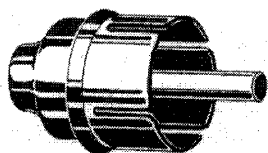


3-2 ADAPTERS — CPN 2939



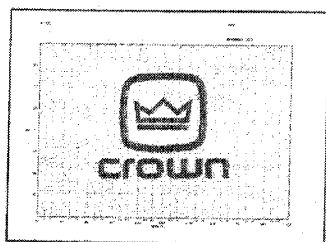
SHORTING PLUGS — CPN 3230

Used with the IC-150A, DL2 and SL-1 to short UNUSED input jacks for noise control.



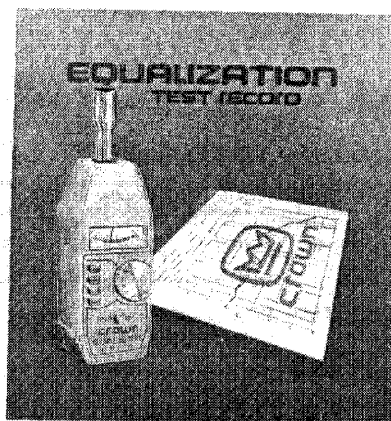
GRAPH PAPER — CPN 4383

Used with the EQ2 for charting room response.



TEST RECORD — CPN 4416

Used with the EQ2 for testing room response.

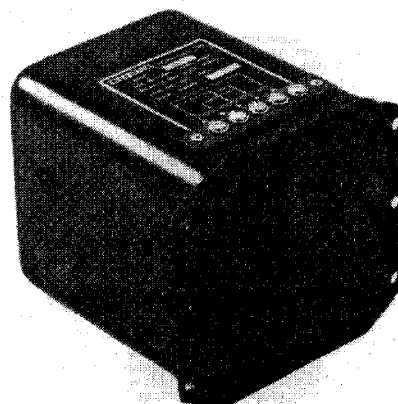


HI FI ADAPTER KIT — CPN 40377

Used with the D-75, D-150A, DC-300A, PL-1, and SA2 amplifiers.

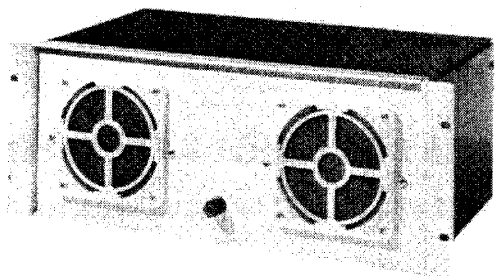
UMX-300 TRANSFORMER

The CROWN UMX-300 transformer offers a maximum 300 watts of power with IM distortion of less than .015%. Frequency response is rated +0 to -1dB, 20Hz to 20KHz at 300 watts. The UMX-300 can be used with any Crown amplifier.



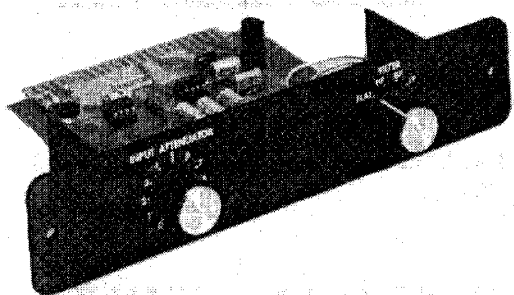
FN-1 FAN PACKAGE — CPN 41682

The CROWN FN-1 Fan Package is meant to provide cool ambient air for cooling any Crown amplifier which requires additional cooling due to operating conditions.



M-600 PA MODULE — CPN 41744

The CROWN M-600 PA Module is a special plug-in board developed to fit into the M-600 amplifier front panel. The module, developed for commercial sound public address systems, provides balanced inputs, filtering, and a continuously adjustable gain from 0 to 20dB at the front panel.



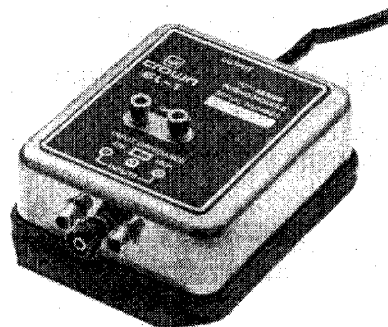
DL2 "A" PHONO MODULE

The CROWN DL2 "A" Phono Module is a separate phono preamp designed to eliminate RF interference and the problems of cable capacitance loading on phono cartridges.



SL-1 "C" PHONO MODULE

The CROWN SL-1 "C" Phono Module is a separate phono preamp similar in function and performance to the DL2 phono module, however designed specially for the SL-1.



*MOVING COIL PHONO MODULE

The CROWN Moving Coil Phono Module is specifically designed for matching a low output moving coil cartridge with the DL2 and/or the SL-1.

BLANK ALUMINUM FILLER PANELS

Standard 1P (1.75"), 3P (3.75"), and 5P (5.25") aluminum panels are available in "satin bead" or brushed aluminum finish.

CABINETS

Designed and specifically built for Crown is a complete line of fine audio furniture. Write for your free brochure.

*Available Fall 1979



MANUAL POLICY

An excellent way to evaluate a new or different Crown product is to purchase an owner's manual. After you've had a chance to read and study the manual, it may be returned for a **complete refund**. No questions asked. Under this special arrangement, the manuals for the IC-150A, OC-150A, D-75, D-150A, DC-300A, SL-1, PL-1, FM-1 and VFX-2A will cost \$2.00 (normal retail is \$5.00). Manuals for the M-600, RTA-2, PSA-2, and the distinction series components will cost \$6.00 (normal retail is \$15.00).

TEST REPORTS

Read what independent test centers and popular magazines critics say about Crown equipment. Free upon request are reprinted articles from major magazines. Read their critical opinions at no expense to you.

CROWN SHIRTS

Crown T-shirts are available in white with colored "ringers" and have the Crown logo on the front and a two-color design (see below) on the back. Contact your dealer or the Crown Parts Department.

