advanced music systems

dmx 15-80 S

Users Manual

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1 SYSTEM OVERVIEW AND SPECIFICATIONS

1.1 SYSTEM OVERVIEW

The dmx15-80S is a true stereo microprocessor controlled digital delay line. Originally designed to meet specifications laid down by the British Broadcasting Corporation for equipment to be supplied to them, it offers two completely independently delayed channels with precisely controlled delay times. The delay times are adjustable without any sacrifice in the 18KHz bandwidth which remains constant irrespective of the amount of delay selected. A total delay of over thirty two seconds is possible with this unit, split over the two channels.

The design is highly modular, allowing incorporation of new technologies and techniques; thus the system will not be rendered obsolescent as developments take place.

The design employs effective 15 bit digital encoding, achieved by basic 12 bit conversion with three instantaneously switched gain ranges of 6dB each. This allows a 90dB dynamic range without the use of analogue compression/expansion techniques with their inherent distortion problems, such as noise breathing.

Memory back up is supplied as standard to ensure that entries to the microprocessor are not lost on power down.

Both input and output levels are adjustable to accommodate units sending or requiring non-standard signal levels.

An intelligent, glitch splicing pitch change option is available which can be incorporated on either or both channels. It will allow pitch shifting as much as one octave either side of the original frequency.

If the pitch shift option is incorporated, the 'lock in' function, which normally just allows indefinite storage and playback of both delay line contents (without deterioration), is transformed into a tape loop editing system. The then 'locked-in' audio may also be audio triggered if required.

An 'add-on' digital reverberation system is also available. This add-on unit, the DMX15R, is physically the same size as the DMX15-80S. It provides nine preset programs with an 18KHz bandwidth, 90dB dynamic range and typical distortion of 0.03%. Independent control of variables, such as pre-delay, decay time, high frequency decay profile and low frequency decay profile, is also provided.

Regeneration controls are provided on both channels of the dmx15-80S and the input may be switched to allow single line input (channel 'a' feeds both delay lines). A switch is also provided to allow the outputs to be mixed for special effects.

An important feature of the dmx range is the use of 'Nudge buttons'. These buttons when used will cause the delay to sweep up or down in steps of only 25uS making the sweep virtually silent.

A facility for switching channel 'a' in phase or out of phase with the original and channel 'b' is provided.

A reset switch is provided for initialising the computer. A complete store clear instruction is also provided (DØ).

A flight case is available for the DMX15-80S if required. Flight cases that accommodate multiples of A.M.S. systems are also available.

The unit is engineered to the same high standard as the rest of the dmx range and offers similar microprocessor controlled front panel facilities ie. programmability, repeatability and storage capability not available with other forms of data entry, putting the engineer in complete control of the system functions.

The dmx15-80S is designed as a rack mounting unit 3.5"/2U high and 11" deep, excluding knobs and connectors. The choice of components and quality of construction are of a high standard and because of the choice of mother board system with plug in circuit cards and ribbon cables, maintenance problems are reduced to a minimum.

There is no doubt that the unit's small size and rugged construction, coupled with the unquestionable performance characteristics represents the best value in delay line technology available in the world today.

1.2 SPECIFICATIONS

1.2.1 ELECTRICAL

Input Impedance (Zin):	10 K ohm : Electronically balanced
Input Sensitivity:	-10 dBV : Ref. 0.775 Volts
Inputs:	Two independent inputs
Maximum Gain:	20 dB
Output Impedance:	150 ohm : Symmetrical Electronically balanced.
Maximum Output Level:	+24 dBV (balanced mode)
Outputs:	Two independent outputs
Distortion:	Less than 0.035% at 1 KHz; full output
Dynamic Range:	90 dB / 15 bit equivalent
Frequency Response:	20 Hz to 18 KHz : - 3 dB , + 0 dB
Input Connectors:	Two XLR-3-31 (female)
Output Connectors:	Two XLR-3-32 (male)
Power Consumption:	30 VA maximum
Power requirements:	110/220/240 VAC : Internally adjustable.
Power Connector:	Standard IEC mains connector

DELAY:

Basic unit:	There are three basic increments:
	[1] The 4K RAM unit giving 1mS to 102.4mS.
	[2] The 16K RAM unit giving 1mS to 409.6mS.
	[3] The 64K RAM unit giving 1mS to 1.638 Seconds.

RAM cards can be added to increase the basic delay times each 4K RAM card increases the delay range by 102.4mS with no change in resolution or bandwidth whilst each 16K RAM card increases the delay range by 409.6mS and each 64K RAM card by 1.638 Seconds, again with no change in response.

Channels may have differing delays and may use different types of RAM cards; further each channel may use a different number of RAM cards. However, different types of RAM card are not permissible on the same channel.

At present the maximum achievable delay is 29.5 seconds for stereo operation (this allows 3.2 seconds of delay for the second channel) and 32 seconds using single channel only.

Fractions of a millisecond are not programmable.

NUDGE:

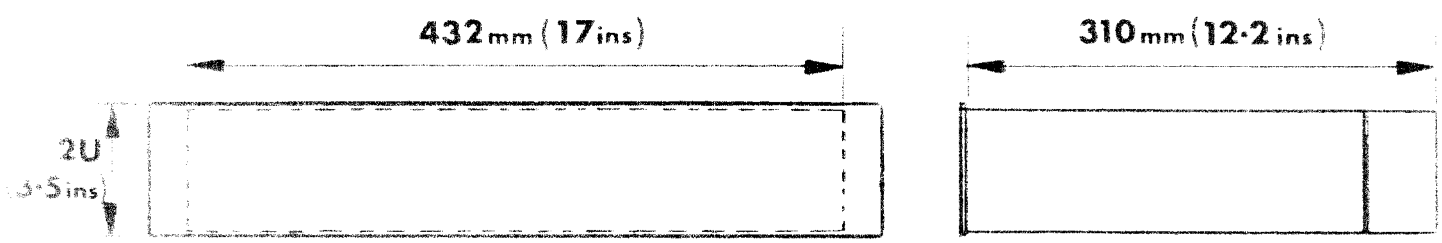
Nudge control resolution:	25uS
Nudge rate (approximate):	2mS/Second

VCO:

VCO control (speed):	Ø.Ø5Hz to 2ØHz
VCO frequency shift (depth):	Ø to 2Ø% continuously variable

COMPUTER CONTROL:

Time Entry:	By numeric keypad; least significant digit = 1mS.
Display:	7 Segment digital readout of selected delay time.
Memory:	9 Memory locations capable of storing 'a', 'b' and 'c' entries.
Reset.	The computer is initialised when this is depressed. NOTE: The computer is also initialised on switch on.



1.2.2 PHYSICAL DIMENSIONS

The unit is designed for 19" rack mounting and has the following dimensions:

Height:	2U (3.5")
Width:	432mm (17"); Behind front panel
Depth:	31Ømm (12.2"); Behind front panel
Weight:	12Kg (including packaging)

1.2.3 OPTIONS

[1] RAM OPTIONS:

Numerous RAM options are available for the dmx 15-80S and it would be pointless to try to incorporate all of this data within the Owner's Manual. If RAM extension is necessary decide on the maximum delay required (on each channel) and from this information A.M.S. or their official representative can recommend the RAM card configuration that would best suit your particular needs.

[2] A pitch change option is available for both channels of this unit allowing pitch shifting as much as one octave either side of the original frequency. This option also allows use of the tape loop editing system.

[3] A reverberation option (DMX15R) is also available.

1.2.4 IMPROVEMENTS

The designs of all AMS products are subject to continuous development and improvement, consequently Advanced Music Systems must reserve the right to alter specifications or modify their products without prior notification.

2 OPERATING INSTRUCTIONS

2.1 INTRODUCTION

This section of the manual contains information regarding installation and operation of the dmx15-80S Stereo Digital Delay Line. It is recommended that the contents of this section are read and understood before attempting to operate the dmx15-80S. Should any difficulties arise during operation contact your nearest A.M.S. representative or contact:

ADVANCED MUSIC SYSTEMS.
WALLSTREAMS LANE,
WORSTHORNE VILLAGE,
BURNLEY,
LANCASHIRE,
ENGLAND.

OR TELEPHONE: 0282 36943
TELEX: 63108

2.2 SHIPPING INFORMATION

2.2.1 The dmx15-80S is packaged in a specially designed container for the best possible protection. Upon receipt of the equipment a thorough inspection should be made to reveal any possible shipping damage. If damage is found a claim should be made against the shipping company immediately or at least on the following working day.

2.2.2 If the unit is returned for service or modifications etc., the original container should be used. If the original container is not available, a new container can be obtained from Advanced Music Systems. Please specify the model number when requesting the new container.

2.3 INSTALLATION

2.3.1 The model dmx15-80S can be operated with a line input voltage of 110, 220 or 240 Volts, adjustable internally. Before connecting the equipment to primary power check that the line voltage setting is correct (see the label on the back of the unit); if the line voltage is not specified assume, for safety, a line voltage setting of 110 Volts. If it is found that the line voltage setting is incorrect use the following procedure to change it.

[a] Disconnect the line power cord from the unit and remove the top cover plate.

[b] With the front panel facing you, you will see the line voltage selection connector on the right hand side near the front of the unit.

[c] Remove the cover over the mains wiring; then remove the brown wire and insert it into the correct position.

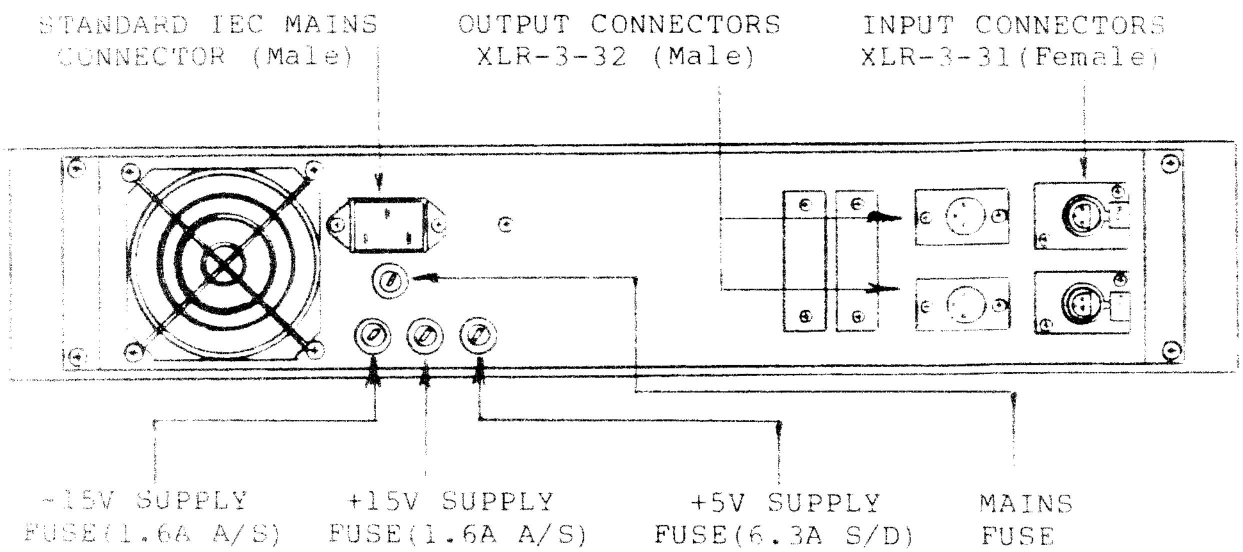
[d] Replace the cover over the mains wiring, the top

2.3.2 The outputs of the dmx15-80S are balanced electronically and therefore they are not floating. If using unbalanced lines ensure that pins 3 (hot) and 1 (ground) are connected as normal and pin 2 (cold) is left floating on the outputs of the unit. DO NOT GROUND PIN 2 AT THE OUTPUT. Pin 2 however, should be grounded on the inputs.

2.3.3 MAINS WIRING

All mains leads supplied with equipment manufactured at A.M.S. are wired in accordance with the European (I.E.C.) colour code. The code is as follows:

BROWN .... LIVE
BLUE .... NEUTRAL
GREEN/YELLOW .... EARTH



2.3.4 CONNECTIONS TO THE REAR OF THE UNIT ARE SHOWN ABOVE.

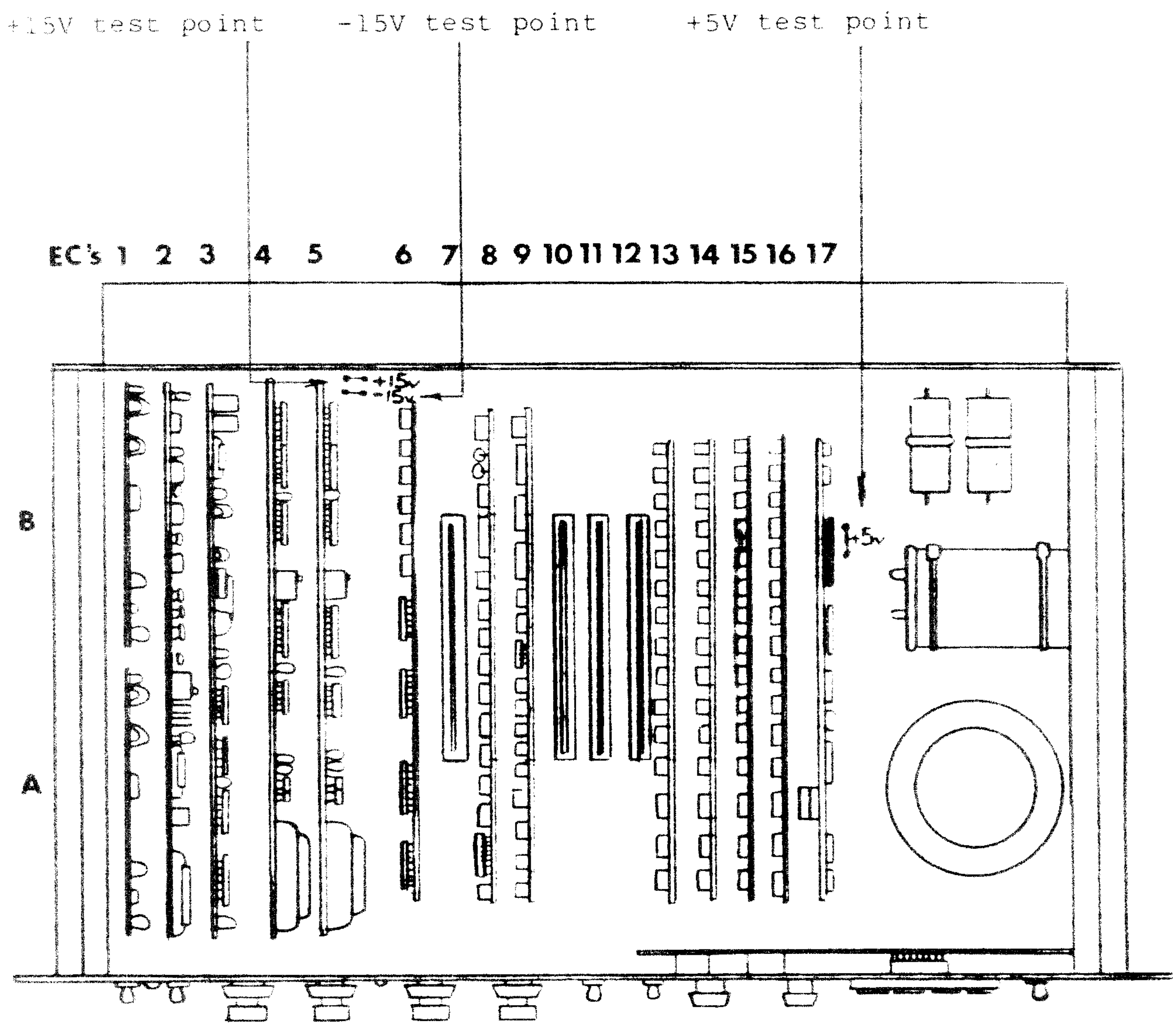
2.3.5 FUSE REPLACEMENT

All power supply rails within the dmx15-80S are protected by current limiting and also against overvoltage. If for any reason the voltage on a particular rail exceeds certain limits, a thyristor connected across that power rail is activated, thus blowing the fuse and protecting the unit's electronics. It should be noted that this may (very occasionally) happen when no fault is present, but a sharp and possibly dangerous mains surge has occurred.

There are four fuses contained within the dmx15-80S. All of which are immediately accessible on the rear panel of the unit. The three power supply test points are marked below:

2.3.6 BOARD POSITIONING

Board positioning within the dmx15-80S is mainly dependent on how many pitch shift processors are incorporated within the actual unit ie. none, one or two. Slots one to six inclusive and slot seventeen are standard positions but the board allocation for slots seven through sixteen varies according to the type of unit in use. As noted above there are three different types of unit:



(a) Stereo ddl, NO pitch shift processor:

With no pitch shift processor involved RAM cards may fill all of the remaining slots i.e. seven to sixteen inclusive. In all cases RAM is fitted from slot sixteen downwards; the slots must be filled in order and there must be no gaps between the RAM cards.

(b) Stereo ddl, ONE pitch shift processor:

With one pitch shift processor (channel A), slot eight is usually used for dmx9 and slot nine for dmx10. As long as dmx9 is always slotted lower than dmx10 then in theory the two pitch shift cards may be placed anywhere between slots seven and nine inclusive. Slots eight and nine are preferred because of the increased airflow across dmx9.

(c) Stereo ddl, TWO pitch shift processors:

The first processor board set (dmx9+dmx10 for channel A) is positioned as above. The second processor set (dmx9+dmx10 for channel B) must not be situated lower than slot nine but may be positioned anywhere above this position up to the end of the RAM.

```
       ----------------------------------------------------------------------
       |                           SLOT POSITIONS                           |
       |--------------------------------------------------------------------|
       | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 |10 |11 |12 |13 |14 |15 |16 |17 |
-------|---|----------------------------------------------------------------|
SDDL   |dmx|dmx|dmx|dmx|dmx|dmx|   .   .   .   .   .   .   .   .   .   |dmx|
       | 1 |15 |14 |12 |12 |16 |   .   .   .   .  RAM  .   .   .   .   |7  |
no     |   |   |   |   |   |   |                                       |   |
pitch  |   |   |5  |dmx|dmx|   |   .   .   .   .   .   .   .   .   .   |dmx|
shift  |   |   |MHz| 5 | 5 |   |   .   .   .   .   .   .   .   .   .   |23 |
---------------------------------------------------------------------------
SDDL   |dmx|dmx|dmx|dmx|dmx|dmx| Channel A .   .   .   .   .   .   .   |dmx|
       | 1 |15 |14 |12 |12 |27 |  dmx9+10  .   .   .   .   .   .   .   |7  |
mono   |   |   |   |   |   |   |   .   .   .   .  RAM  .   .   .   .   |   |
pitch  |   |   |10 |   |dmx|   |   .   .   .   .   .   .   .   .   .   |dmx|
shift  |   |   |MHz|   | 5 |   |   .   .   .   .   .   .   .   .   .   |23 |
---------------------------------------------------------------------------
SDDL   |dmx|dmx|dmx|dmx|dmx|dmx| Channel A .   .   .   .   .   .   .   |dmx|
       | 1 |15 |14 |12 |12 |27 |  dmx9+10  . Channel B .   .   .   .   |7  |
ster.  |   |   |   |   |   |   |   .   .   .  dmx9+10  .   .   .   .   |   |
pitch  |   |   |10 |   |   |   |   .   .   .   .   .   .   .  RAM  .   |dmx|
shift  |   |   |MHz|   |   |   |   .   .   .   .   .   .   .   .   .   |23 |
---------------------------------------------------------------------------
       | R | R | R | R | R | L | L | L | L | L | L | L | L | L | L | L | R |
       |--------------------------------------------------------------------|
       |    Direction components face with front panel facing engineer      |
       ----------------------------------------------------------------------
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R = Right L = Left

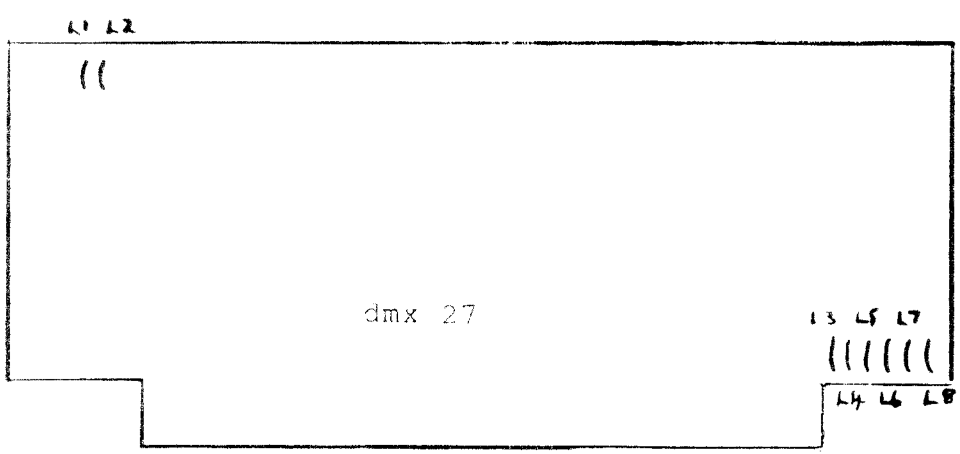
2.3.7 ON-BOARD OPTIONS

Slot 6 - dmx27 only:

Link L1 should always be in. Link L2 should be out if one pitch shift processor is fitted and in if two pitch shift processors are fitted. The other links are as follows:

RAM CARD TYPE	CHANNEL A	CHANNEL B
	L3	L4
4K (102mS)	in	in
16K (408mS)	out	out

NUMBER OF RAM CARDS	CHANNEL A		CHANNEL B	
	L5	L7	L6	L8
1	in	in	in	in
2	out	in	out	in
3	in	out	in	out
4	out	out	out	out



Slot 17 - E2 connections (both dmx7+23):

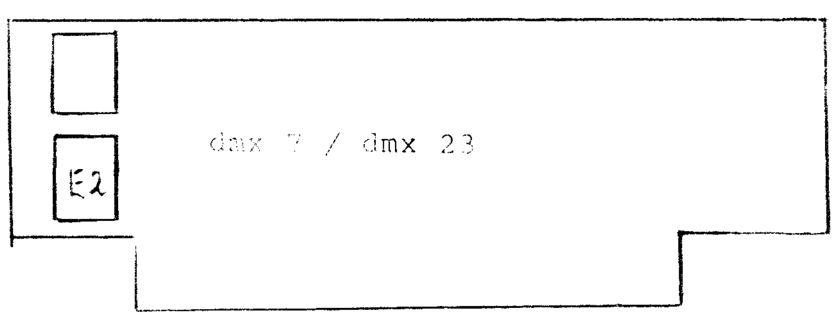
If dmx23 is fitted link Pin9 to Pin16. If dmx7 is fitted link Pin 9 to Pin5.

NUMBER OF RAM CARDS	CHANNEL A		CHANNEL B	
	Link Pin4 to	Link Pin5 to	link pin6 to	link Pin7 to
1	Pin 1	Pin 1	Pin 1	Pin 1
2	Pin 1	Pin 16	Pin 1	Pin 16
3	Pin 16	Pin 1	Pin 16	Pin 1
4	Pin 16	Pin 16	Pin 16	Pin 16

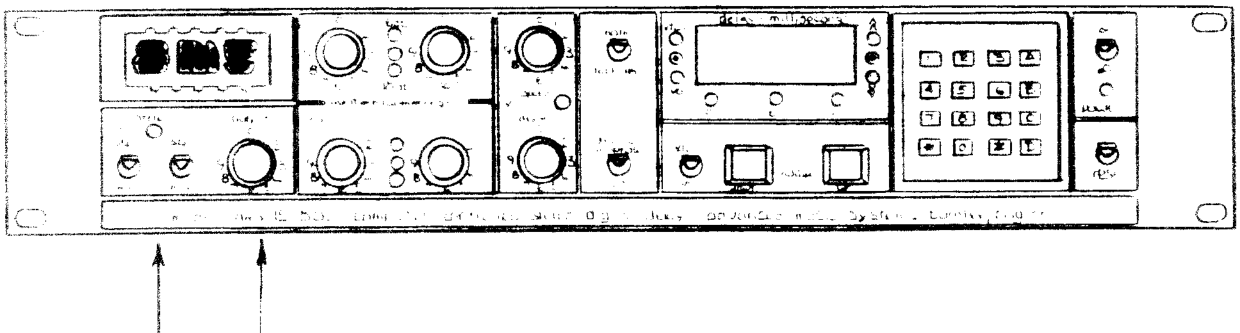
E2 connections continued:

RAM CARD TYPE	CHANNEL A Link Pin11 to	CHANNEL B Link Pin10 to
4K (102mS)	Pin 1	Pin 1
16K (408mS)	Pin 16	Pin 16

NUMBER OF PITCH CHANGE SETS	Link Pin 8 to
1	Pin 1
2	Pin 16



2.4 OPERATION



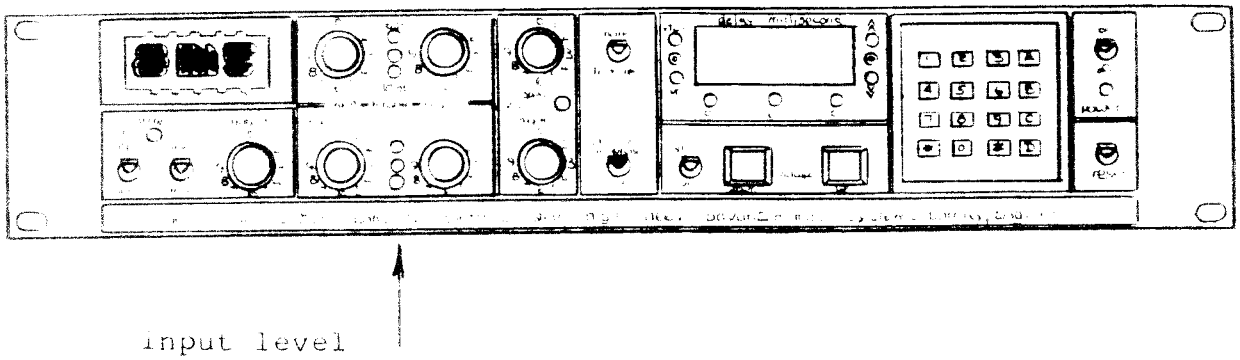
2.4.1 GENERAL INTRODUCTION

Looking at the front panel, it can be seen that it is divided into various sections. Probably the most effective way of learning how to operate the unit is to become familiar with the front panel controls.

2.4.2 OUTPUT LEVEL AND MIXING SECTION

The input may be switched between two channel operation (stereo input) and single input, dual output operation (mono input using channel 'a' as source). The two outputs may also be mixed if desired to create or enhance various effects. If the input is two channel and the outputs remain unmixed the stereo LED will illuminate to indicate true stereo operation.

The output level control is a ganged control which governs the output of both channel 'a' and channel 'b'. After the input levels have been set for the best working conditions this control should be adjusted to give a good level match on the mixing desk.

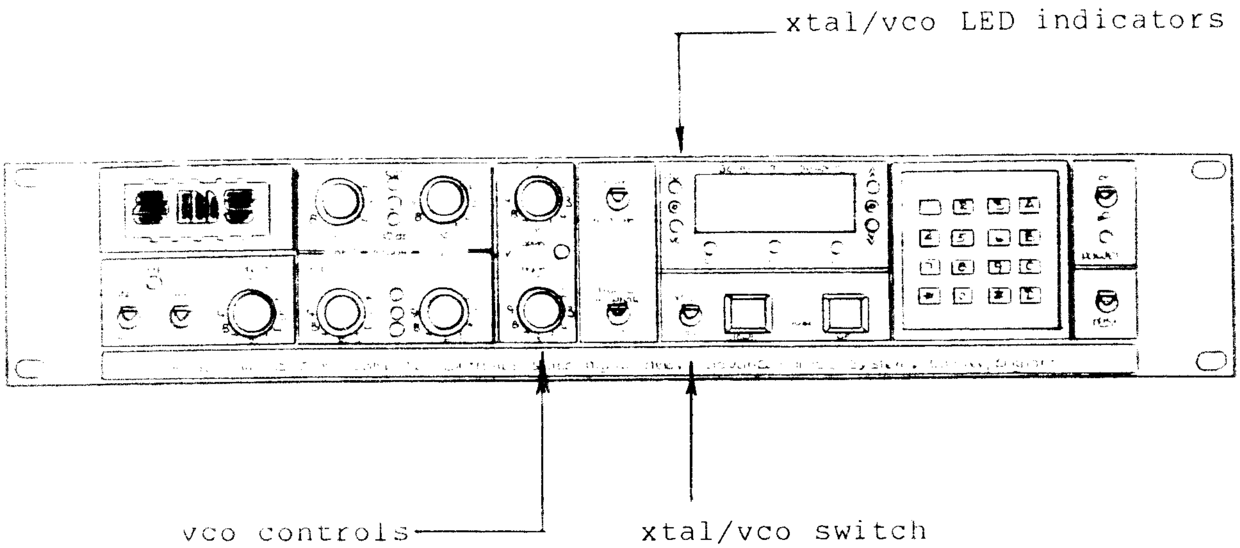


input level
control section

2.4.3 INPUT LEVEL CONTROL SECTION

Individual input signal levels can be varied using the two input control knobs, an indication of the signal levels is given by "traffic signal" light emitting diodes (LEDs). The red LEDs are illuminated 6 dB before clipping, the yellow LEDs 12 dB and the green LEDs 18 dB. In normal operation, with a signal present the input levels should be adjusted so that the green LEDs are illuminated all the time, the yellow LEDs most of the time and the red LEDs illuminate only when the highest programme peaks occur.

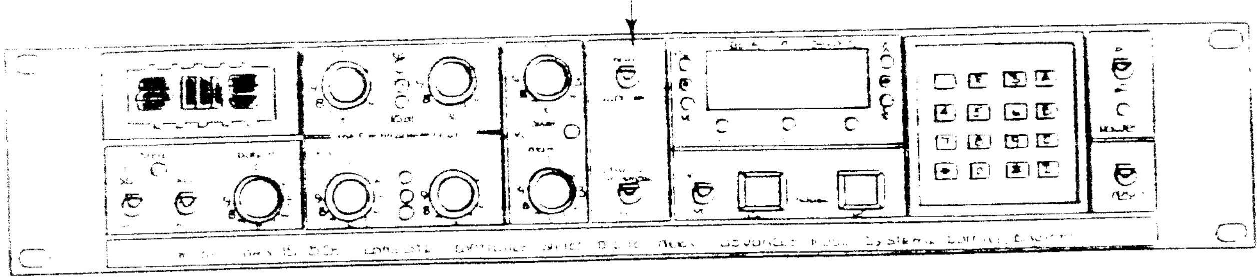
A proportion of the output of a channel may be fed back to that channel's input by adjustment of the respective regenerative control knob.



2.4.4 VCO CONTROL SECTION

The VCO control section is adjustable both in speed and depth of modulation, the yellow LED indicates the speed and illuminates once per cycle. To use the VCO the crystal must be switched out and the VCO switched in; the switch can be found under the display. At the left hand side of the display can be found two LEDs indicating whether the crystal or the VCO has been selected.

the lock in
switch



2.4.5 THE 'LOCK IN' FUNCTION

When the 'lock in' switch is thrown the word LOC will be displayed and the contents of both delay lines at that instant will be preserved indefinitely, or until the unit is switched off or the 'lock in' switch is switched back to 'norm'. With the 'lock in' mode selected and no editing done, each delay line will loop round its maximum capability: hence with one 4K RAM card the loop size will be 102mS, with five 4K RAM cards 512mS, with one 16K RAM card 409mS, whilst with four of these cards 1.638Sec., etc.. This will result in two continuously repeated signals, one on each channel, the durations of which are governed by the maximum delay capability of the individual delay line.

Think of it as two tape recorders with the ability to create instant tape loops, the initial loop sizes being governed by the maximum storage capability of the delay lines.

Even with the pitch change option on channel B (stereo pitch change only), channel B's tape loop is fixed at this maximum loop size and therefore no editing of the channel B loop is possible. The incorporation of a pitch change option (single or stereo) does however allow editing of channel A's loop. This is accomplished by using the "lock in" facility in conjunction with either the keypad or the nudge buttons.

If we consider the channel A loop as having a start at zero delay and an end at the maximum delay then we may cut off portions of the loop either at the beginning or at the end. To remove part of the loop we simply redefine either the beginning of the loop or the end of the loop. (for an actual example see Section 2.5.11).

Editing with the keypad:

To remove a portion from the beginning of the loop first press the 'A' key. When the 'A' key is pressed the present starting point of the loop is displayed. Now enter the new start point in milliseconds; the 'a' channel identifier LED will flash until the splice point has actually been entered (ie. until the # key has been pressed).

To remove a portion from the end of the loop first press the 'B' key. When the 'B' key is pressed the current 'loop end' position will be displayed. The new loop end point must now be keyed in again in milliseconds; the 'b' identifier LED will flash until the enter key (#) is pressed.

Editing using the nudge keys:

Editing can also be accomplished by using the nudge keys. The nudge keys allow the ends of the loop to be incremented as well as decremented in 5mS steps. This is a quick way of 'fine tuning' the loop before use, it should be noted however that during the nudging process some audio interference will occur.

Other effects:

If the 'C' key is pressed it is possible to vary the pitch of the loop. The effect obtained can be likened to using varispeed on a tape loop system. This may be accomplished by direct entry, using the key pad, or by use of the nudge buttons. Unlike nudging the ends of the loop; nudging pitch does not cause audio interference.

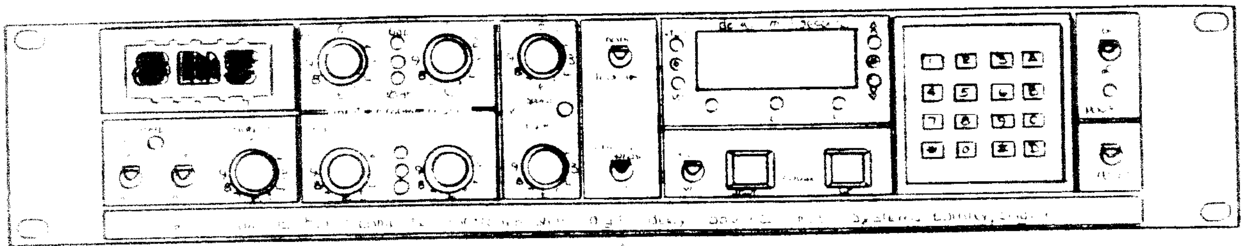
The loop may also be read out singly. This is accomplished by pushing the # key. After the # key has been pressed the edited loop will be read out once and will not be read out again until the # key is pressed again.

The loop may also be audio triggered. In this case the locked-in audio will be read out if an audio signal of sufficient level to illuminate the green LED is applied to channel A.

Cautionary notes:

Having come out of the lock in mode there is no point re-entering until the maximum tape loop time has elapsed. If re-entry prior to this is attempted the resulting audio signal will contain a mixture of the previously locked loop and new audio data.

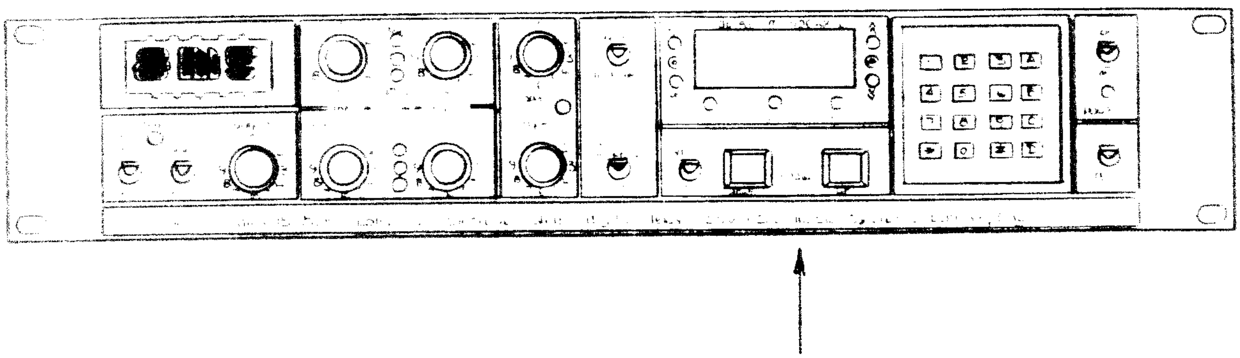
DO NOT switch the power on with the lock in function selected since random data in the RAMs will be locked in and sent to the audio chain.



in phase/out
switch

2.4.6 'IN PHASE/OUT' SWITCH

When the 'in phase/out' switch is depressed the output of channel 'a' is out of phase with the input, a useful device for creating various 'tunneling' effects if used in conjunction with the regenerative controls, the nudge controls and the output mix switch.



Nudge controls

2.4.7 NUDGE CONTROLS

Altering delays:

The nudge buttons may be used to edge the selected delay up or down, an indication of the direction is given by the LEDs to the right of the display. These buttons when used change the displayed delay, and also the value in the respective store location, by approximately 2mS/Second in 25uS steps producing a smooth, virtually silent change in the delay time.

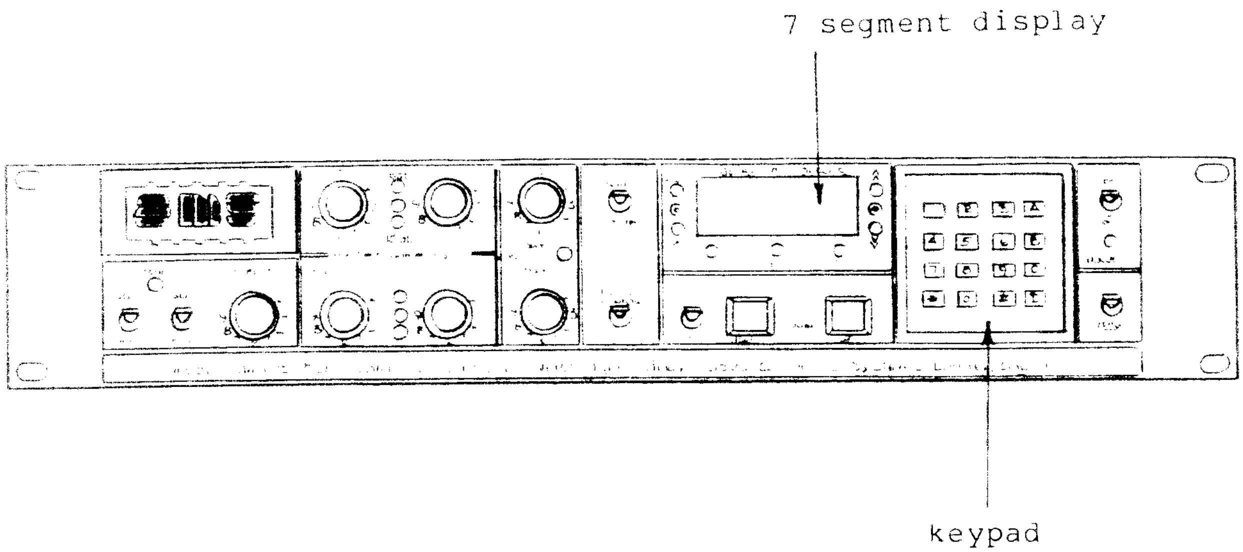
Pitch shifting:

The nudge buttons may also be used for pitch shifting and are most effective for pitch searching and tuning.

Loop editing:

When in the lock-in mode the nudge buttons may also be used for loop editing. In this case the beginning and the end of the loop may be adjusted (i.e. increasing or decreasing the length of either end of the loop is possible, within the limits) in 5mS increments.

Pitch shifting in the lock-in mode is also possible; the effect obtained can be likened to varispeed on mechanical tape loop systems.



2.4.8 COMPUTER CONTROL

Computer control is effected by the use of the keypad:

KEYS A and B These select the audio channel to be processed either 'a' or 'b', depressing either will cause the delay setting for that channel to be displayed {in milliseconds} and the relevant identifier LED immediately below the display to be lit (a or b). These keys are also used for loop editing in the lock in mode when the pitch change option is included.

KEY C This key is inoperative unless the pitch change option is included within the unit. If this option is included then pressing the 'C' key after selecting the appropriate channel (a or b) results in the pitch ratio information being displayed and allows the operator to change the pitch of the signal. In this case LED c glows in addition to the channel identifier LED a or b.

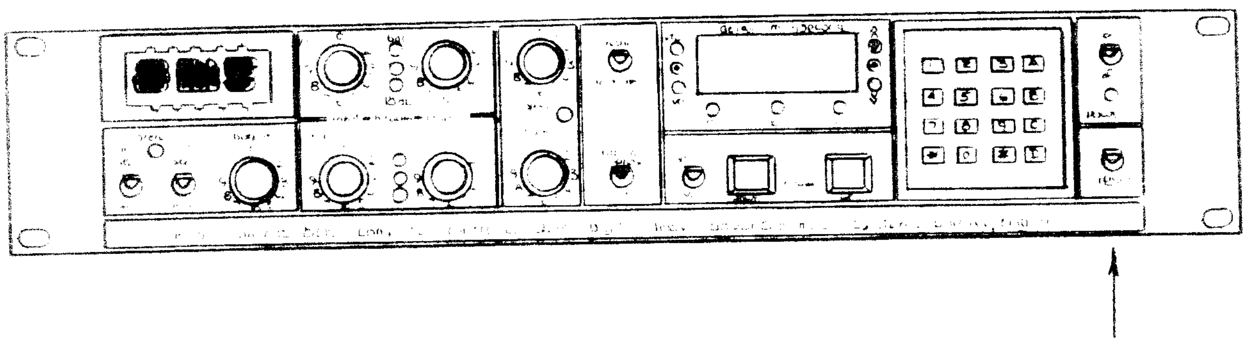
KEYS 0 to 9 Keying in a delay setting will cause entry into the display only and will not affect the programme until the data is entered into the unit by depressing the # key. Until the # key is depressed the channel identifier LED will flash indicating temporary data. The display only reflects the contents of the ddl once the # key has been depressed and the channel identifier LED has stopped flashing.

KEY * This key is used for decimal point entry when using the pitch shift option.

KEY D Depressing this key causes the current store number to be displayed in the format 'Str.5'. To change the current store a single numeral is typed in the range 1 to 9. No enter keystroke (#) is required, the system will immediately enter the delays programmed within that store and all subsequent entries will be retained in that store. If instead of a numeral, A or B is then depressed the system will display the delay contents for that channel (see above) and store will remain unaltered. If the 'Ø' key is pressed immediately after the 'D' key then all of the store locations will be cleared - be careful.

KEY # This is the 'Enter' key which must be depressed to enter a delay into the unit. It is also used for single shot looping in the lock in mode.

GENERAL If more than four numeric characters are typed, or the user attempts to enter a delay outside the limit of the unit an error message 'Err' will be displayed to warn the operator. The correct entry may then be typed.



power 'on/off'
and reset

2.4.9 POWER SWITCHING AND RESET

When the unit is first switched on or if 'reset' is depressed the unit will be initialised. Initialising the unit does not clear any of the store locations. It does however re-start the micro and therefore the display and the output will reflect the contents of the last store location selected and also the last function selected.

If you wish to clear all of the store locations simply enter 'DØ'.

To summarise:

Do not be put off by the unit's apparent complexity. Whilst the operation of the unit may at first sight appear to be complicated it will take little time to become familiar with the controls and the sequences of operation.

2.5 A USER'S GUIDE TO THE dmx15-80S STEREO DIGITAL DELAY LINE

2.5.1 INTRODUCTION

This section of the manual describes in more detail how the dmx15-80S stereo digital delay line can be used to create various effects.

2.5.2 BACK TO BASICS

A digital delay line (ddl) as its name implies, is a device that receives a given signal at its input and after a defined lapse of time (programmed by the user) reproduces that exact same signal at the output. The ddl, as in the case of the dmx15-80S, must not alter or modify the signal no matter what delay is programmed. The dmx15-80S has two such delay lines which are independently programmable.

The output of channel 'a' can be switched in phase or 180 degrees out of phase with the original signal whilst the output of channel 'b' is always in phase with the original.

The dmx15-80S, as stated above, does not modify the signal when applying a delay; however, regenerative effects and VCO or vibrato effects can be introduced if required by adjusting the relevant controls on the front panel.

Both Pitch Change and Reverberation options are available and a section on Pitch Shifting has been included for reference.

It is obvious that a great deal can be accomplished using the dmx15-80S besides simple delay, and the following paragraphs have been written to help the studio engineer get the best possible use from the unit.

2.5.3 ORIGINAL SETTINGS

With the power switched on and a signal present ensure that both input level control knobs are set correctly, ie. ensure that both regen control knobs are set to zero and adjust the input level control knobs so that the red LEDs illuminate on programme peaks only. The output control knob should be adjusted to ensure a good match between the ddl and the mixing desk. The following should now be set as indicated:

'in sep./mix' to in sep.
'out sep./mix' to out sep.
'vco speed and depth' controls to zero.
'regen' controls still set to zero.
'norm/lock-in' switch to norm.
'chan. a in phase/out' switch to chan. a in phase.
'xtal/vco' switch to xtal

SWITCH ON : When power is switched on, the computer is initialised and the output of the ddl is governed by the last store location selected and also the last function selected. If the 'D' key is pressed followed by the '0' key all store locations will be cleared. In real terms this means that outputs 'a' and 'b' will keyed in. Also the DDL will reflect the contents of channel 'a' store 1 and therefore the channel 'a' identifier LED will be illuminated. If the 'D' key is pressed Str.1 will be displayed indicating that the outputs are governed by entries in this store location; if now the 'B' key is pressed zero will be displayed indicating no delay on channel 'b'; the 'A' key should now be pressed, the display will again show no delay on channel 'a'.

2.5.4 CHANNEL 'a' DELAY

To delay the output on channel 'a' with respect to the original, enter the required delay (in milliseconds) into the display via the keypad. The 'a' channel identifier LED will now flash, indicating temporary data in the display, until the data is entered into the store by depressing the # key. Immediately it is entered into the store location channel 'a' will be delayed with respect to the original by the amount programmed and the 'a' LED will stop flashing. If the user attempts to enter a delay outside the limit of the unit an error message 'Err' will be displayed to warn the operator; the correct entry may then be typed.

2.5.5 CHANNEL 'b' DELAY

To delay the output of channel 'b' with respect to the original, first depress the 'B' key. The 'b' LED will now illuminate indicating that the display now reflects the contents of channel 'b'. To delay the output, follow the procedure as outlined for channel 'a'; this time the 'b' LED will flash until the # key is depressed.

2.5.6 STORING DELAYS

If the 'D' key is depressed, the display will indicate the current store in use (at the present store 1, 'Str.1').

Let us say that we do not wish to lose the effects of the delays just entered in channels 'a' and 'b' but wish to try a longer delay, say on channel 'a', to see what effect this will have. If the '2' key is depressed immediately after the 'D' key, the outputs will now be governed by the contents of the second store location (Str.2; zero at present). We can now enter the same delay for channel 'b' and the longer delay for channel 'a' in Str.2. We now have a combination of delays in two store locations; there are nine store locations in all, therefore nine different combinations may be tried if so desired. To compare the different settings all we need do is keep the 'D' key depressed whilst keying through the store locations 1 to 9.

This ability to store different settings, and also the ability to instantly recall a setting, is very useful to the Studio Engineer and Producer alike. For example, let us say that during a session the Producer asks for echo on vocals during the verse of a song, but during the chorus he would like to reduce this to a close double tracking effect. Possibly a Guitarist wishes to have a 'distant' sound during the vocal parts of a song but during the solo he would like a 'dry' sound. Both of the above are readily accomplished using the dmx15-80S:

Using channel 'a' only for this exercise and mixing this signal with the original on the mixing desk we first program a delay of say 80mS in store 1 and a shorter delay of about 18mS in store 2 leaving store 3 clear. With the present settings the 'wet' mix will consist of just one repeat signal 80mS or 18mS respectively away from the original. This will give a rather false sounding echo and to improve this the 'regen' control knob should be adjusted until a more natural voicing is achieved.

For the first example (assuming the song starts with a verse) prior to the start of the song store 1 is selected giving the echo effect required; during the verse the 'D' key is depressed in preparation for the chorus, as the chorus starts the '2' key is depressed immediately giving the close ADT effect required. The ADT effect can be improved using either the VCO section, or the second channel and/or the pitch change facility of the dmx15-80S; this is discussed in more detail later. During the chorus the 'D' key is again depressed in preparation for the verse and as the verse starts the 1 key is depressed resulting in the immediate recovery of the original echo effect etc.,etc.,etc..

For the second example, prior to the start of the song store 1 is again selected and during the vocal parts of the song the 'D' key is depressed in preparation for the solo; just prior to the beginning of the solo the '3' key is depressed giving the dry sound required. In this case store 2 could be selected to see whether the guitarist would like the signal slightly 'wet', whichever is chosen, during the solo the 'D' key must be depressed in preparation for the 'backing' section and as the vocals begin the '1' key must be depressed to reproduce the echo effect.

The case above is taken for simplicity, obviously the dmx15-80S is more flexible than the example shows; the example is only used as a guide to the techniques involved.

2.5.7 MIXING

When using the dmx15-80S alone, mixing of the two delayed channels can be accomplished by the use of the output mix control on the front panel of the ddl. A much larger variety of effects are attainable when using the SDDL in conjunction with a mixing desk since the original signal may also be mixed with the outputs of the delay lines.

2.5.8 ADT

Automatic Double Tracking is improved if the secondary signal is varied in pitch by a small amount. This can be accomplished by adjustment of the vco controls or by incorporation of the pitch change option.

When using the vco section to improve ADT remember that the pitch variation is dependent upon both the vco speed and depth control settings, and also the delay times between the original signal and the outputs of the ddl. To obtain this improved ADT effect first switch the 'xtal/vco' switch to vco. For an ADT setting of 18mS the best effect is obtained by setting the depth control to '9' and the speed control between '2' and '3'; as the delay setting is increased however, the depth setting will have to be decreased to retain a reasonable ADT effect.

Improving ADT using the pitch change option is best accomplished with a pitch ratio of approximately 1.015. Set the pitch ratio by first depressing the 'c' key. If the pitch change option is not incorporated nothing will happen, but if the pitch change option is included the 'c' LED will flash, together with the respective channel identifier LED. On entry of the data (depression of the # key) the two LEDs will stop flashing and the ADT effect will be audible.

2.5.9 VCO

The vibrato effect created by the VCO can, if used properly, enhance the effect of certain instruments and is a useful tool in its own right.

2.5.10 CONTROLLED FLANGING

The following method can be used if flanging is required using the dmx15-80S outputs alone:

First return all the front panel controls to their 'original' settings. Now switch the 'out sep./mix' control to mix; if now the nudge buttons are used around zero delay, flanging should be audible.

By using the mixing desk other methods can be employed to achieve flanging using the basic principles outlined above.

2.5.11 THE 'LOCK IN' FUNCTION

As mentioned in Section 2.4.5 when the 'lock-in' switch is thrown the word LOC will be displayed and the contents of both delay lines at that instant will be locked in.

Channel B's loop is always fixed at the maximum loop size but if the pitch change option is contained in the unit then by using the 'lock-in' facility the loop on channel A may now be edited.

As an example we have chosen to remove 100mS from the beginning and end of a loop. We have chosen a loop size of 1.638 Seconds and have assumed for clarity that no previous editing has been done:

Editing with the keypad:

To remove 100mS from the beginning of the loop first press the 'A' key. When the 'A' key is pressed the present starting point of the loop (ie. 0) will be displayed. Now enter the new start point in milliseconds (ie. 100); the 'a' channel identifier LED will flash until the splice point has been entered by pressing the # key.

A loop start point of zero can not be entered via the keypad. Once the loop start point has been set away from zero the lowest time entry via the keypad is 1mS. The nudge buttons may be used to enter zero if so desired (see below).

To remove 100mS from the end of the loop first press the 'B' key. When the 'B' key is pressed the current 'loop end' position will be displayed. The new loop end point must now be keyed in again in milliseconds (ie. 1503). The 'b' identifier LED will flash until the enter key (#) is pressed.

Notice that the end of the loop is 35mS less than the RAM storage size would suggest (in our case 1.603S and not 1.638S). This is the amount of time taken to complete the splicing operation. In fact, after entry of a start or finish point for the loop the high speed microprocessor analyses the frequency and amplitude content of the subsequent 30mS for two suitable splicing points to enable the join to be as free from interference as possible. Thus the actual splice points may be slightly different from those displayed and incrementing the start or finish points by only a few milliseconds may thus have no effect.

Editing using the nudge keys:

Editing can also be accomplished by using the nudge keys. The nudge keys allow the ends of the loop to be incremented as well as decremented in 5mS steps (see Section 2.4.5). The loop start point may be nudged to zero if the keyed delay is an exact multiple of 5.

For other effects and a few cautionary notes on the operation of the lock-in function see section 2.4.5.

2.5.12 THE PITCH CHANGE OPTION

In the case of the stereo ddl, where fitted, the Pitch Change Option is available on both channel 'a' and channel 'b' and therefore the following notes will apply to both channels. Remember that any effect selected using the 'C' key will always be in addition to the programmed delay. However an inherent effect of the pitch shifting process is an inbuilt delay which varies continually. To this end, the pitch shift software incorporates an algorithm which ignores programmed delays below 30mS when pitch changing and makes allowance for the inherent delay offset above this value.

Programming Pitch Change - Channel 'a':

First set all the controls to the 'ORIGINAL SETTINGS' outlined in Section 2.5.3 and press the 'D' key followed by the '0' key. Now depress the 'C' key, both the channel 'a' LED and the 'c' LED will illuminate and the display will read 1.000 indicating a pitch ratio of one ie. no change in pitch from the original, or 'unison'. The lowest pitch available is an octave down from the original, that is half the original frequency or a ratio of '0.500'. To obtain this depress the '*' key followed by the '5' key, the display will now read '0.5' and the 'a' and 'c' LEDs will flash indicating temporary data in the display. Entering this ratio by depressing the '#' key will cause the 'a' and 'c' LEDs to stop flashing and the display to read '0.500' indicating an octave decrease in pitch on channel 'a'. If we now look at the contents of channel 'a' (this is accomplished by depressing the 'A' key) the letter H will be flashing on the left of the display indicating that the original and output 'a' are not in unison. If a delay is now required on channel 'a' it can be programmed in exactly the same way as in 2.5.4.

The highest pitch available is an octave increase in the original, that is twice the original frequency or a ratio of two, and can be programmed by depressing the 'C, 2 and #' keys respectively.

Pitch ratios of less than half or greater than two are not programmable and if tried an error (Err) message will be displayed to warn the operator of his mistake. Pitch changes between these two extremes are programmable to three decimal places.

The 'Nudge' buttons can be used to alter the pitch up or down and are most effective for pitch searching and tuning.

Programming Pitch Change - Channel 'b':

This is achieved exactly as for channel 'a' except that the 'B' key must be depressed prior to programming the pitch ratio. In this case the 'b' and 'c' LEDs will flash until the pitch ratio is entered.

Understanding And Using The Pitch Changer:

The AMS Pitch Changer employs a very fast microprocessor which examines the digitised musical signal within the memory and performs a number of tests to isolate two points in time which have the highest correlation with each other in terms of waveshape. The pitch shift algorithm contained in the programme memory uses this information in conjunction with the required pitch ratio to minimise the 'glitch' problem normally associated with real time pitch changing. Since no fade-in, fade-out is employed, with suitable program material almost perfect pitch shifting can be obtained without the characteristic 'fluttering' of other units on the market. The exceptionally wide bandwidth and excellent noise performance of the basic ddl is also retained when changing pitch, as is the vco facility. Combining all these features, and the ability to key in a pitch ratio of crystal-locked stability means the engineer may create automatic double tracking of hitherto unobtainable quality and effectiveness. In certain cases, feedback enhances the 'fullness' of the sound.

Feedback, used with care, can be used to great effect; for instance, at a pitch ratio of 2.0, feedback generates harmonics of the input, and single notes on a bass guitar for instance can be made to sound as if they were played on an organ. Chords, or beat frequencies between two notes on the bass guitar will make it impossible for the microprocessor to identify the frequency of the incoming signal, and will cause the effect to be marred. Nevertheless, the lack of the usual 'wobble' on the output means that it is now possible to use a pitch-changer musically at pitch ratios other than 1.01.

NOTES:

[1] Interesting arpeggio effects can be obtained by programming delays longer than 200-300mS and applying feedback to just short of instability. This effect is especially useful if musical intervals are programmed as the pitch shift, such as 1.26 (see [6] below). When applying feedback whilst pitch changing, care should be taken when shifting the frequency down by significant amounts, since if instability is reached successive cycles of feedback will cause any input frequency to tend towards zero. On the way, the output will pass the resonant frequency of most things in the control room, including the engineers and the monitors; so beware!

[2] The inherent delay mentioned above is less when shifting down in pitch than up; thus, 0.985 produces an output closer in time to the original than 1.015.

[3] A 'straight' delay programmed in channel 'b' may be used to effectively eliminate the inherent delay by programming (say) 25mS and mixing the two outputs at the desk, or splitting the outputs across the stereo image. ('a' to the left, 'b' to the right, for instance)

(4) The unit performs at its best on programme material with recognisable pitch and amplitude; material with high elements of attack does not allow perfect splicing, since two compatible points in the memory do not exist.

(5) A by-product of the pitch shifting software means that a pitch ratio entered in the downward direction always starts off at zero delay. Try applying feedback to just short of instability and entering 0.999 as a pitch ratio, with no pre-delay, successive depressions of the 'enter' button will produce a sweeping 'flange' or 'tunnel' effect (dependent on the setting of the 'in phase/out' switch) at a rate dependent on the pitch ratio. The special effect is useful down to about 0.98.

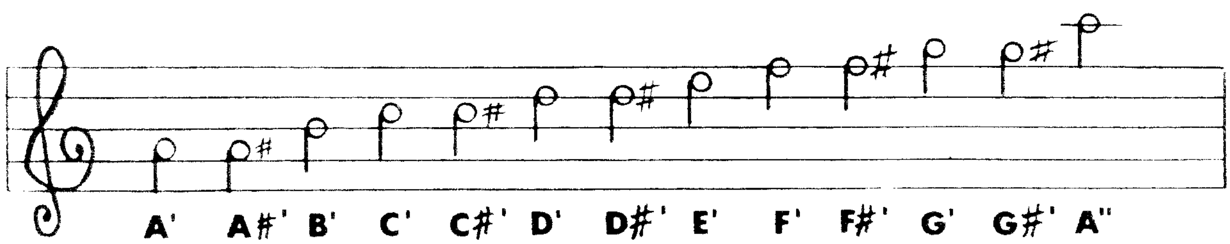
(6) In an attempt to ensure that all ddl users are familiar with 'pitch ratio', not only as an effect but also as a musical tool, we have included the following data. We have done this knowing full well that most of the people who are reading this manual will already be familiar with the theories expounded below. However for those who are uninitiated the following may help:

Each note in the chromatic scale is separated from the next by a 'semitone' increase in pitch; the lowest common musical interval. This 'semitone' increase is not a fixed frequency. If for example we move from A' to A#' we will need to increase the frequency by 26Hz to effect a 'semitone' increase in pitch; if we now add 26Hz to G#' we will not arrive at A''. To move from G#' to A'' we will need to add 49Hz to effect the same semitone increase. This is because the semitone is a frequency ratio or pitch ratio and is therefore dependent on the frequency of the note being played.

The pitch ratio has the following geometric progression:

$$\left(\sqrt[12]{2}\right)^{n}$$

, where n is the number of semitones above unison. Let us use the chromatic scale of A' as our example:



NOTE	FUNDAMENTAL (Hz) x PITCH RATIO	FREQUENCY (Hz)	FREQUENCY INCREASE
A'	$440 \times \left(\sqrt[12]{2}\right)^{0} = 440 \times 1$	440	26.1638
A#'	$440 \times \left(\sqrt[12]{2}\right)^{1} = 440 \times 1.059463$	466.1638	27.7195
B'	$440 \times \left(\sqrt[12]{2}\right)^{2} = 440 \times 1.122462$	493.8833	29.3678
C'	$440 \times \left(\sqrt[12]{2}\right)^{3} = 440 \times 1.189207$	523.2511	31.1141
C#'	$440 \times \left(\sqrt[12]{2}\right)^{4} = 440 \times 1.259921$	554.3652	32.9644
D'	$440 \times \left(\sqrt[12]{2}\right)^{5} = 440 \times 1.334840$	587.3296	34.9244
D#'	$440 \times \left(\sqrt[12]{2}\right)^{6} = 440 \times 1.414214$	622.2540	37.0011
E'	$440 \times \left(\sqrt[12]{2}\right)^{7} = 440 \times 1.498307$	659.2551	39.2014
F'	$440 \times \left(\sqrt[12]{2}\right)^{8} = 440 \times 1.587400$	698.4565	41.5324
F#'	$440 \times \left(\sqrt[12]{2}\right)^{9} = 440 \times 1.681793$	739.9889	44.0020
G'	$440 \times \left(\sqrt[12]{2}\right)^{10} = 440 \times 1.781798$	783.9909	46.6185
G#'	$440 \times \left(\sqrt[12]{2}\right)^{11} = 440 \times 1.887749$	830.6094	49.3906
A''	$440 \times \left(\sqrt[12]{2}\right)^{12} = 440 \times 2$	880	

It is important only that you remember the relevant pitch ratios since the fundamental frequency can be any value; these are summarised below:

CHROMATIC SCALING	FRENCH NAME (Tonic Solfa)	PITCH RATIO
unison (1)	Do	1
2		1.059
3	Re	1.122
4		1.189
5	Mi	1.260
6	Fa	1.335
7		1.414
8	So	1.498
9		1.587
10	La	1.682
11		1.782
12	Te	1.888
octave (1')	Do'	2

To obtain the familiar do, re, mi, fa, so, la, te, do' (the major scale) we are all so conversant with; first feed in a constant signal, possibly a sine wave, to the pitch change channel and ensure that the pitch ratio in store 1 is 1. Now set store 2 to 1.122, store 3 to 1.26, store 4 to 1.335 etc.,etc., culminating in entering 2 in store 8. If we now return to store 1 and keep the 'D' key depressed whilst thumbing through the eight store locations in sequence the major scale will be audible.

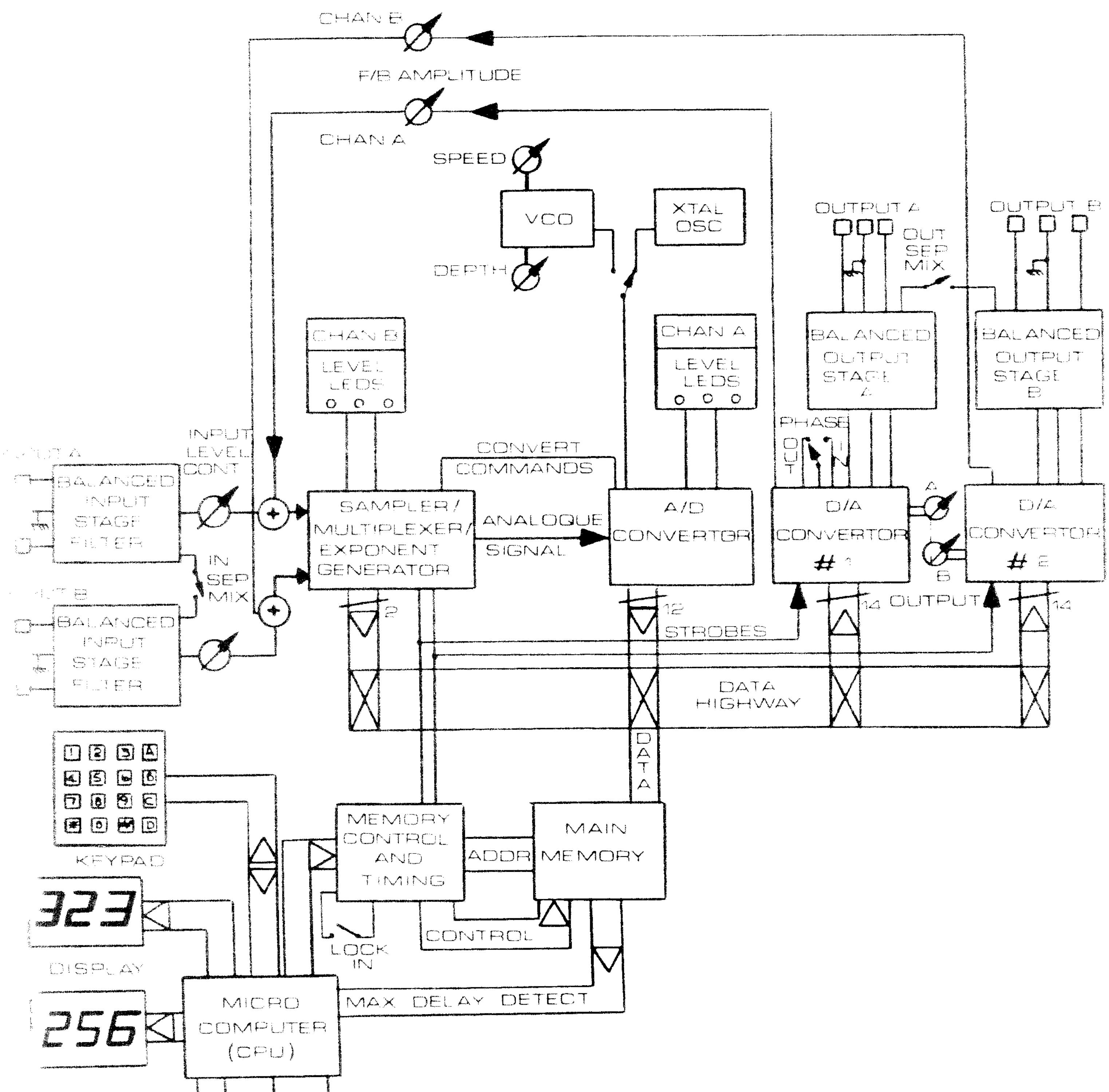
The above can be modified to suit any scale characteristics and is only restricted by the operator's imagination.

3 THEORY OF OPERATION

3.1 INTRODUCTION

This section of the manual contains a complete description of how the dmx15-80S Stero Digital Delay Line works together with a functional block diagram of the unit (Fig 3.1).

Fig. 3.1



3.2 OVERALL FUNCTIONAL DESCRIPTION

The balanced inputs to the unit are fed to the input cards which process the signals into unbalanced form, these signals are then fed through low pass filtering to prevent aliasing components appearing at the outputs of the ddl. Buffered drive signals are then fed to the input level controls on the front panel and the outputs from these front panel controls are fed to preset gain stages on the input cards, the outputs of which are fed to the stereo sampler card.

The stereo sampler card accepts the outputs from the input cards together with the feedback signals and mixes them into balanced stages to avoid common mode interference pickup. Each channel is fed to a monolithic sample and hold device. These devices are strobed in antiphase at a sampling rate of 40KHz with the crystal switched in, but variable on vco. The outputs of these are fed to a comparator/gain switching section alternately by means of a FET switch. The signal is then scaled by 1, 2, 4 or 8 to optimise the signal to noise ratio, and a two bit exponent number is generated to note the scaling factor used. The signals for driving the 'traffic signal' LEDs for channel B are also derived from this board.

Control and timing signals for this board are derived from the memory control board, the clock-inputs being generated by either the crystal oscillator or the voltage controlled oscillator (vco); the ADC conversion command is then decoded, and the resulting response, 'conversion complete' is used to control the latching of the correct exponent for that sampling period.

The sampled analogue output from the sampler card is fed into the ADC via a buffer amplifier. Timing signals from the sampler card cause a conversion to be initiated. The twelve-bit result is latched into a set of registers/data highway drivers for storage in memory. The level indicating drivers for channel A are also located on the ADC card. The vco is also generated on the ADC card by first generating a triangular waveform whose frequency is varied by the vco speed control on the front panel. This triangular waveform besides feeding the drive circuitry for the vco speed LED, also feeds a shaping chip which generates a sine wave. This sine wave is buffered and fed to the vco depth control on the front panel; the output from this is then mixed with a preset d.c. voltage and is fed to the voltage controlled oscillator. The d.c. voltage thus gives the nominal output frequency of the vco, the amplitude of the sine wave gives the range of swing (depth) from the nominal frequency, and the frequency of the sine wave increases and decreases the rapidity of the swing (speed). The vco has the effect of altering the sampling rate.

The memory control card provides timing signals for the random access memory (RAM) array and also controls which addresses in the array are written to and read from. These addresses are loaded into the memory control board by the microcomputer board, via a ribbon cable.

In each sampling period, the following sequence occurs:

[1] Write channel A data into RAM.

[2] Read channel A data from RAM address offset 'A', into DAC A.

[3] Convert channel B / sample channel A

[4] Write channel B data into RAM.

[5] Read channel B data from RAM address offset 'B', into DAC B.

[6] Convert channel A / sample channel B

Delays are altered by changing address offsets 'A' and 'B'.

The RAM card address lines are multiplexed, and the current RAM card being accessed is selected on the baseboard as a result of the three highest address bits from the memory control board and the channel selected (0=A, 1=B).

Each 4K RAM card is used to retain 4,096 samples of the analogue signal in the form of 12 bits of amplitude data (mantissa) and two bits of range data (exponent). At the normal sampling rate of 25uS per sample, this corresponds to a delay of 102.4mS. Apart from the RAM devices themselves, the card provides data highway buffering and write control enabling when the card is selected.

The digital signal from the RAM is converted back to analogue form on the DAC boards (2 off: one for channel 'A' and one for channel 'B'). These cards accept the 'mantissa and exponent' digital input, perform shift left operations on the mantissa dependent on the value of the exponent (multiplication), and reconstitute the original signal by means of the two DAC devices on each board. The outputs of the two DACs are summed, buffered and filtered to remove sampling steps and fed to the baseboard, then to the output controls, the outputs of which are fed back to the DAC cards where the balanced outputs are generated.

The interface with the delay circuitry, the keypad, the display, the storage of preset values, the 'Nudge' button controls, the status LED controls and also the decimal to binary conversion for RAM control are performed by the microprocessor board. This board holds the Intel 8085 microprocessor; 2 kByte of Programmable Read Only Memory (PROM); 256 Bytes of CMOS RAM and two programmable I/O devices.

The basic back wiring of the unit is accomplished by the use of a mother board or baseboard. This board contains all the power supplies for the unit together with the buffer amplifiers and steering logic. The supplies are as follows:

VOLTAGE	CURRENT RATING
+5V	5A
+15V	1A
-15V	1A
+12V	Ø.5A
-5V	Ø.5A

Communication to and from the central processor is accomplished via the display board which contains the drive for the 7 segment display together with the display itself and the 'Nudge' buttons. It is driven by I/O ports on the microprocessor board and is connected by ribbon cables.

4 MAINTENANCE

4.1 INTRODUCTION

This section contains maintenance information for the dmx15-80S and includes general maintenance procedures and trouble shooting information.

4.2 SERVICE INFORMATION AND WARRANTY

4.2.1 Each dmx15-80S is warranted for a period of one year upon delivery to the original purchaser. Details of the warranty are given in Section 6 of this manual.

4.2.2 A factory service is available for the dmx15-80S on request. Shipping information is given in the 'Operating Instructions Section' of this manual. If required an estimate can be provided to the customer prior to work being carried out.

4.3 GENERAL MAINTENANCE

4.3.1 ACCESS

All printed circuit board assemblies can be accessed by removing the top cover plate, four DZUS fasteners hold the plate in place. For cleaning purposes it will be necessary to remove the bottom plate and the two side panels. The bottom plate is also held on by four DZUS fasteners whilst four M3 x 6mm pan head Posidriv screws hold each side plate in place. The front panel is affixed by four M5 x 12mm countersunk socket head screws (black) and can be removed using a 3mm AF Allen key. The display board is held on to the front panel by three M3 x 20mm countersunk socket head screws (black) together with three 9mm plastic spacers, plastic washers and M3 nuts; these can be removed using a 2mm AF Allen key. The back panel is held on by four M5 x 12mm Pan Head Posidriv screws (+ shakeproofs). The fan, the blanking panel and the fan guard are also held on by four M5 x 12mm Pan Head Posidriv screws.

4.3.2 CLEANING

The dmx15-80S should be cleaned periodically to remove dust, grease and other contaminants. The surface of all the printed circuit boards should be cleaned using dry air at low pressure (<20 psi). If grease is to be removed use Arklone F or Freon TF and remove grime with clean dry air at low pressure. Clean the front panel with a soft cloth dampened with a mild solution of detergent and water. DO NOT USE ARKLONE ON THE DISPLAY FILTER as this will cause damage.

4.3.3 FUSE REPLACEMENT

The fuses are located on the rear panel of the dmx15-80S Stereo Digital Delay Line. Spare fuses are included in the package of goods sent with your unit. If replacement is necessary ensure that the correct fuse is used; markings on the rear panel indicate the type of fuse in use.

4.4 TROUBLE SHOOTING

The following section has been written as a guide for fault finding in case of malfunction during service. If the dmx15-80S malfunctions whilst under warranty, contact the engineering department at A.M.S. or an approved service organisation immediately. If an attempt is made to service the unit whilst it is still under warranty without guidance or permission from one of the above bodies warranty may well be invalidated.

Before embarking on the checking procedures outlined below, ensure that all the power supplies are at their correct potential and that all the fuses are intact; also check the input wiring external to the DDL for shorts.

Possible causes are listed in order of probability and therefore should be checked in strict order.

SYMPTOM	POSSIBLE CAUSE
loss of output on channel B	1. input short circuit on channel B: check both the input cable and the internal wiring of the unit. 2. check input card. 3. output short circuit on channel b: check both the cable connected to the unit and the internal wiring of the unit. 3. check DAC board channel B 4. check IC15 on baseboard 5. check the memory control board
loss of output on channel A	1. input short circuit on channel A: check both the cable connected to the unit and the internal wiring of the unit. 2. check input card. 3. output short circuit on channel A: check both the output cable and the internal wiring of the unit. 3. check DAC board channel A 4. check IC16 on baseboard 5. check the memory control board
loss of output on both both channels	1. check the sampler board 2. check the ADC board
intermittent output on one channel	faulty RAM board
intermittent distortion on one channel	faulty RAM board
clicks on one channel with no delays prog'd	faulty RAM board

4.4 TROUBLE SHOOTING (CONTINUED)

SYMPTOM	POSSIBLE CAUSE
constantly distorted output	1. DAC board : distortion will be isolated to associated channel 2. ADC board 3. input board : distortion will be isolated to associated channel 4. sampler board
distortion or clicks only when delay prog'd ie. no distortion or clicks at zero delay	faulty memory control board
programmed delay does not correspond to displayed delay	1. faulty micro. board 2. faulty memory control board 3. faulty pitch change card (if fitted)

SWITCH THE POWER OFF WHEN REMOVING OR INSERTING CARDS and please take extreme care in ensuring that the cards are replaced in the correct slot with the correct orientation.

5 WARRANTY

5.1 LIMITED LIABILITY

EDENDECK LIMITED, TRADING AS ADVANCED MUSIC SYSTEMS AND HEREIN AFTER KNOWN AS THE MANUFACTURER, GUARANTEES THIS EQUIPMENT FROM DEFECTS IN MATERIALS AND WORKMANSHIP UNDER NORMAL USE AND SERVICE FOR A PERIOD OF ONE YEAR. THIS GUARANTEE EXTENDS TO THE ORIGINAL PURCHASER ONLY AND DOES NOT APPLY TO FUSES OR ANY PRODUCT OR PARTS SUBJECTED TO MISUSE, NEGLECT, ACCIDENT OR ABNORMAL CONDITIONS OF OPERATION. THE GUARANTEE BEGINS ON THE DATE OF DELIVERY TO THE ACTUAL PURCHASER OR TO HIS AUTHORISED AGENT OR CARRIER.

IN THE EVENT OF FAILURE OF A PRODUCT COVERED BY THIS GUARANTEE, THE MANUFACTURER OR THEIR CERTIFIED REPRESENTATIVES WILL REPAIR AND CALIBRATE EQUIPMENT RETURNED PREPAID TO AN AUTHORISED SERVICE FACILITY WITHIN ONE YEAR OF THE ORIGINAL PURCHASE AND PROVIDED THAT THE GUARANTORS EXAMINATION DISCLOSES TO HIS SATISFACTION THAT THE PRODUCT WAS DEFECTIVE, EQUIPMENT UNDER THIS GUARANTEE WILL BE REPAIRED OR REPLACED WITHOUT CHARGE. ANY FAULT THAT HAS BEEN CAUSED BY MISUSE; NEGLECT; ACCIDENT, ACT OF GOD, WAR OR CIVIL INSURRECTION; ALTERATION OR REPAIR BY UNAUTHORISED PERSONNEL; OPERATION FROM AN INCORRECT POWER SOURCE OR ABNORMAL CONDITIONS OF OPERATION, WILL NOT FALL UNDER THIS GUARANTEE. HOWEVER AN ESTIMATE OF THE COST OF THE REPAIR WORK WILL BE SUBMITTED BEFORE WORK IS STARTED.

THE MANUFACTURER SHALL NOT BE RESPONSIBLE FOR ANY LOSS OR DAMAGE, DIRECT OR CONSEQUENTIAL, RESULTING FROM MACHINE FAILURE OR THE INABILITY OF THE PRODUCT TO PERFORM.

THE MANUFACTURER SHALL NOT BE RESPONSIBLE FOR ANY DAMAGE OR LOSS DURING SHIPMENT TO OR FROM THE FACTORY OR ITS DESIGNATED SERVICE FACILITY.

THIS GUARANTEE IS IN LIEU OF ALL OTHER GUARANTEES, EXPRESSED OR IMPLIED, AND OF ANY OTHER LIABILITIES ON THE MANUFACTURERS PART.

THE MANUFACTURER DOES NOT AUTHORISE ANYONE TO MAKE ANY GUARANTEE OR ASSUME ANY LIABILITY NOT STRICTLY IN ACCORDANCE WITH THE ABOVE.

THE MANUFACTURER RESERVES THE RIGHT TO MAKE CHANGES OR IMPROVEMENTS IN THE DESIGN AND CONSTRUCTION OF THIS UNIT WITHOUT OBLIGATION TO MAKE SUCH CHANGES OR IMPROVEMENTS IN THE PURCHASER'S UNIT.

5.2 WHAT TO DO IF A FAULT IS FOUND

If a fault develops in the unit, notify Advanced Music Systems or their nearest service facility giving full details of the difficulty. On receipt of this information service or shipping instructions will be forwarded to you. No equipment should be returned under the warranty without prior consent from Advanced Music Systems or their Authorised Representative.

5.3 SHIPPING INFORMATION

Authorised returns should be prepaid and must be insured. All AMS products are packaged in specially designed containers for the best possible protection. If the unit is returned the original container should be used. If this is not possible, a new container can be obtained from Advanced Music Systems; please specify the model number when requesting a new container.

If the specially designed container is not used ensure that a suitable rigid container of adequate size is used, wrap the instrument in paper and surround it with a good thickness of shock absorbing material.

5.4 CLAIM FOR DAMAGE DURING TRANSIT

The instrument should be thoroughly inspected immediately upon delivery to the purchaser. If the instrument is damaged in any way a claim should be filed with the carrier immediately. A quotation to repair shipment damage can be obtained from Advanced Music Systems or their Certified Representative. Final claims and negotiations with the carrier must be completed by the customer.

5.5 APPLICATIONS PROBLEMS

Advanced Music Systems will be happy to answer all applications questions to enhance your use of this equipment. Please address all correspondence to:

ADVANCED MUSIC SYSTEMS.
WALLSTREAMS LANE,
WORSTHORNE VILLAGE,
BURNLEY,
LANCASHIRE,
ENGLAND.

OR TELEPHONE: 0282 36943
TELEX: 63108

EQUIPMENT QUALITY

The following page should be torn out and returned to :

THE QUALITY ASSURANCE DEPARTMENT.
ADVANCED MUSIC SYSTEMS,
WALLSTREAMS LANE,
WORSTHORNE,
NR. BURNLEY,
LANCASHIRE,
ENGLAND.

It has been included so that you may comment on the equipment you have just purchased. The inherent problem with this type of customer feedback is that one generally tends to receive data only if the customer has a complaint to make. At A.M.S. we would like to hear comments from everyone who has purchased one of our units whether their comments be good or bad. Only with a broad cross section of replies can we make a proper response to your suggestions.

QUALITY CHARACTERISTICS

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1 Was shipping damage evident ?
YES| | NO| | Describe:

---

2 Was shipment complete ?
YES| | NO| | Describe:

---

3 Were adjustments or replacements necessary for satisfactory performance ?
YES| | NO| | Describe:

---

4 General Appearance ?

---

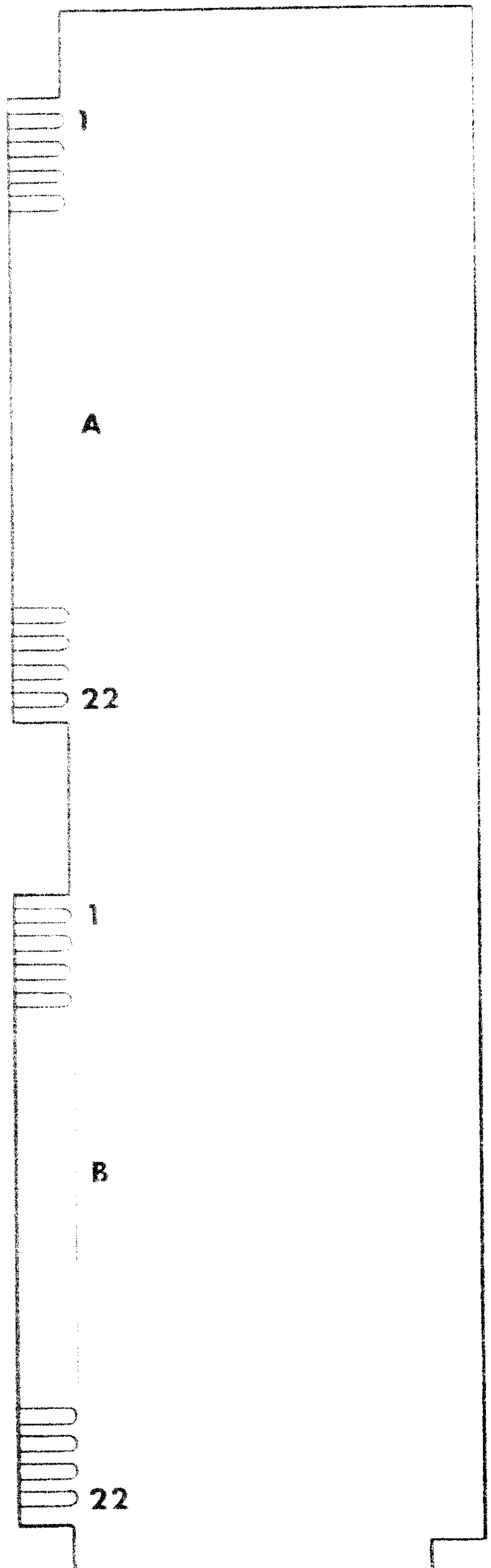
5 Additional Comments ?

EQUIPMENT	
SERIAL NO.	ACCEPTANCE TESTED BY:
	FINAL INSPECTION BY:

Q.A. or CUSTOMER ENGINEER :	
SITE LOCATION :	TEL:

BOARD EDGE CONNECTOR DETAILS

With component side facing and the edge connectors on the left the following applies:



AA:

Component side on edge connector A

AB:

Solder side on edge connector A

BA:

Component side on edge connector B

BB:

Solder side on edge connector B