

**Operating
and service
information
DN716
Digital Delay Line**



**Audio engineering
like no other
in the world**

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Thank You for using Klark-Teknik Product

To obtain maximum performance from this precision electronic product, please study these instructions carefully. Installing and operating the delay line is not complicated, but the flexibility provided by its operating features merits familiarisation with its controls and connections. This unit has been prepared to comply with the power supply requirements, that exist in your location.

Precautions

Before connecting this unit to the mains power, ensure that the operating voltage is correct for your local supply.

It is important that you observe the following instructions if another voltage setting is required.

Do not install this unit in a location subjected to excessive heat, dust or mechanical vibrations.

Voltage Selection and Power Connection

Connection is by means of an IEC standard power socket. The rear panel voltage label, indicates the voltage required for the satisfactory operation of the unit.

Before connecting this unit to the mains supply, ensure the fuse fitted is the correct type and rating as indicated on the rear panel, adjacent to the fuse holder.

To change the mains voltage please refer to "Voltage Change-Over" section on page 10

* Mains voltage change should be carried out by a qualified service technician only.

Safety Warning

This unit is fitted with 3-pin power socket: For safety reasons the earth lead should not be disconnected. If you encounter a problem with earth-loops, use the ground-lift switch on the rear panel to isolate the signal earth from the chassis earth.

To prevent shock or fire hazard, do not expose the unit to rain or moisture.

To avoid electrical shock do not remove covers. Refer servicing to qualified personnel only.

After you have unpacked the Unit

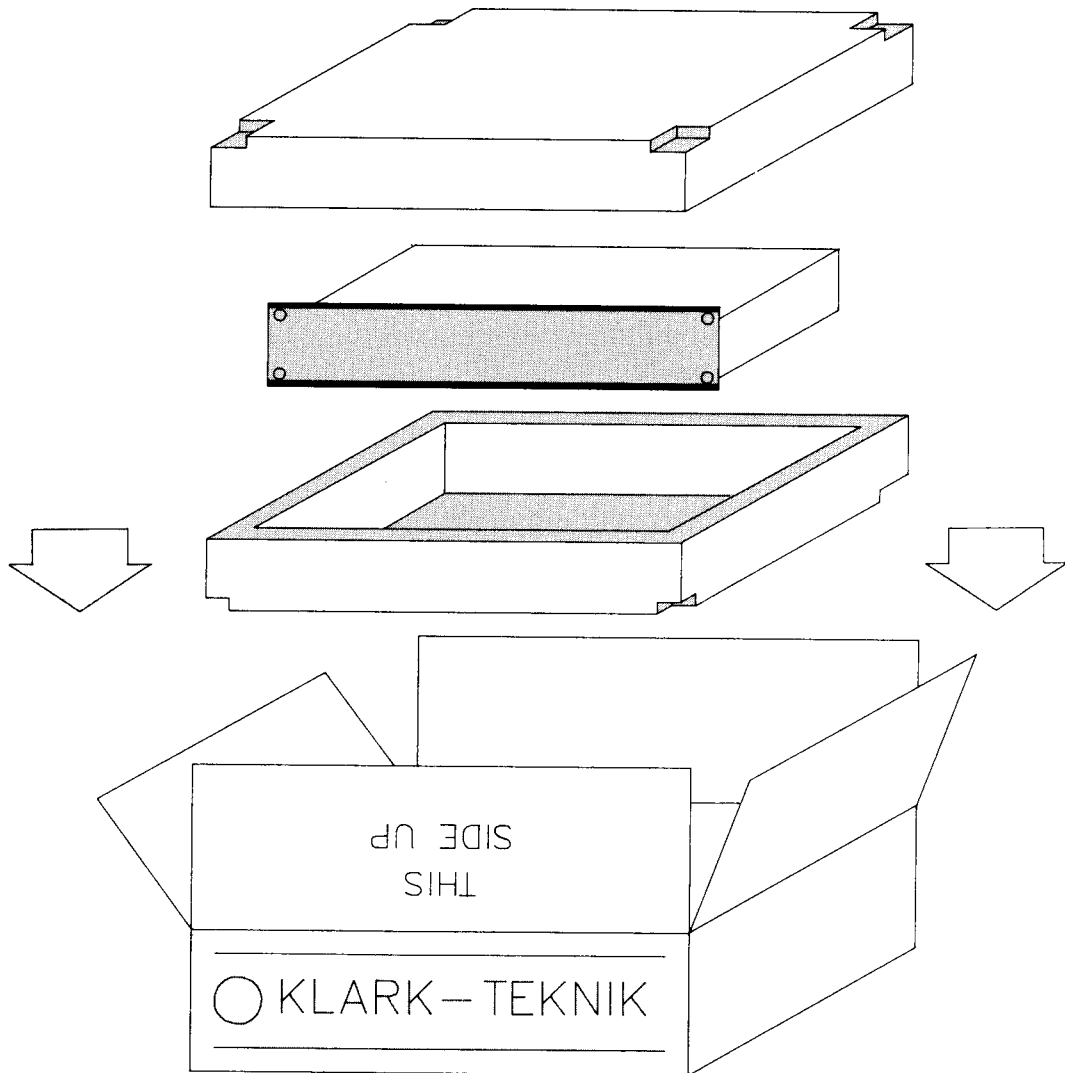
Save all the packing materials - they will prove valuable should it become necessary to transport or ship this product.

Please inspect this unit carefully for any signs of damage incurred during transportation. It has undergone stringent quality control inspection and tests prior to packing and left the factory in perfect operating and mechanical condition.

If, however, the unit shows a sign of damage, notify the transportation company without delay. Only you, the consignee, may institute a claim against the carrier for damage during transportation.

If necessary, contact your supplier or as a last resort, your Klark-Teknik importing agent,* who will fully co-operate under such circumstances.

* See enclosed world-wide importing agents list.



Introduction

Over the past few years, the Digital Delay Line (DDL) has become one of the most widely used items of signal processing equipment to be found both in the studio and sound reinforcement installations.

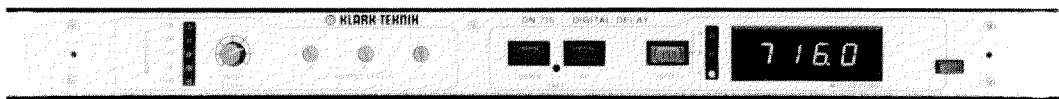
Until comparatively recently the DDL tended to be regarded rather as a laboratory curiosity with its use confined only to the very large and complex sound reinforcement systems or to the larger recording studios. Recent advances in digital technology have not only improved the electronic performance and versatility of digital delay lines but have also brought them within the budget of the smaller studio and sound reinforcement installation. The possible applications of the DDL are correspondingly increasing, ranging from improving the intelligibility, clarity and naturalness of sound reinforcement systems, compensation of varying signal arrival times in multi media/multi-path transmission systems, to creating a wide variety of completely new sounds for the recording and broadcast studio and live event. In addition, the digital delay line can also be used to alter the perceived spatial and reverberant characteristics of a room or auditorium. The wide audio bandwidth, high precision and resolution of modern delay lines also opens up new avenues of use in the field or laboratory to provide high quality delayed audio frequency signals e.g. for acoustic and electro-acoustic measurement purposes or to accurately time align loudspeaker systems and recording channels.

Many of the delay line's applications are perhaps not so immediately apparent as those of other signal processing equipment such as an equaliser for example, for many applications derive from psycho-acoustic origins as well as acoustic and physical effects.

To get the best effect from the delay line or indeed to be able to decide whether one is needed or might prove useful in a particular situation, an understanding of both the way in which the device works as well as an appreciation of some of its applications and their psycho-acoustic basis is an advantage. Because of the wide and diverse range of applications in which digital delay lines may be employed, finding the appropriate information is not always straightforward. The enclosed short form notes aim to overcome this problem by presenting a number of practical applications together with a discussion on the underlying aspects of the physical and psycho-acoustical premise on which they are based. Essentially digital delay line applications may be divided into two distinct groups:-

- i) Where the delay line, as its name implies, delays an audio signal in order for other signals, either acoustic or electronic, to basically 'catch up' or synchronize, and
- ii) Where the delay line delays an audio signal to produce a psycho-acoustic effect or illusion.

DN716 Digital Delay Line



The Klark-Teknik DN716 is a multiple output digital audio delay line offering over 1.3 seconds of signal delay in each of its three outputs.

In-house research and development by Klark-Teknik Engineers placed special emphasis on the design of very high quality converters and the used 16 bit successive approximation A/D convertor and 16 bit D/A converter ensures optimum performance in the DN716.

The input and output filters are proprietary low-pass hybrid types, specially designed to achieve optimum noise performance, low distortion and high reliability.

A high speed CPU gives the unit an increased versatility and lower component count, resulting in better reliability in a compact package.

The delay time on each output can be changed in 20 micro second steps from zero to 1.311 seconds. The display is auto-ranging giving the following delay time resolutions:-

00.00 msec to 99.98 msec	(counting in 0.02 msec steps)
100.0 msec to 999.9 msec	(counting in 0.1 msec steps)
1000 msec to 1311 msec	(counting in 1 msec steps).

The DN716 is built to the same high electrical and mechanical standards as all Klark-Teknik equipment and is both robust and stylish. It occupies a standard one unit of rack space and has an electronically balanced input and unbalanced outputs. Output balancing transformers are available and retrofittable.

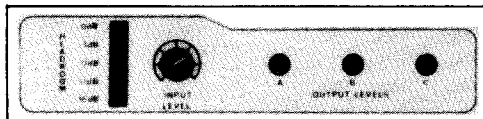
Comprehensive diagnostic test routines for fault diagnosis are included.

Operating Considerations

Audio Signal Levels

Due to the inherent limit of digital audio products it is important that special care be placed on setting the operating levels to achieve the best possible performance.

This can be achieved by the following:



a) Turn down to minimum the three output level controls.

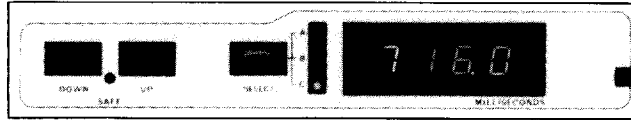
b) Inject normal operating signal and turn up the input level control as high as practical (0db LED = 2dB below internal clipping of the unit.)

c) Adjust the output level controls as required.

Displayed Delay Times

Although the display is capable of displaying the delay to an accuracy of 20 microseconds, the display for clarity reasons does not include a fixed delay of approximately 110 microseconds caused by the sampling process and necessary filter networks. Therefore the actual delay is approximately 110 microseconds greater than that displayed. The fixed delay varies slightly with frequency due to the filter networks.

20Hz - 5kHz	110 micro seconds
8kHz	115 micro seconds
13kHz	120 micro seconds
16kHz	125 micro seconds
18kHz	135 micro seconds
20kHz	145 micro seconds



Furthermore

For safety and interference reasons the supply earth connection should always be used.

The audio signal group (pin 1 XLR's) may be isolated from the chassis ground by means of the 'earth lift' switch located on the rear panel. The shells of the XLR's are always connected to the chassis earth.

Do not rely on the front panel mounting as a means of chassis earthing.

Although the DN716 has been designed for minimum EMF interference it is not advisable to mount the unit near radio sensitive equipment (i.e. radio microphone receivers).

For the best possible performance in sensitive areas it is advisable that the units have transformer balanced inputs and outputs.

Make sure that the unit's voltage setting is in the correct range, incorrect setting can cause intermittent operation or excessive heating of the unit.

If normal operating conditions are adhered to, the DN716 will not require any periodic maintenance other than cosmetic removal of dust from the front panel using a dry soft cloth.

Due to the complexity of the circuitry used in the DN716 it is essential that all servicing is carried out by an authorised Klark-Teknik service technician.

Attempted repairs by non-authorised service personnel will invalidate the warranty.

Diagnostic error codes on Switch-On should be reported to the nearest Klark-Teknik service point (See "Importing Agents" on page 23).

Typical Error Codes:-

1. EE01	Eprom (T-Bug)
2. EE02	C-Mos Memory Failure
3. Repetitive EE03	Data Corruption

Reliability Control

Even with the advanced technology incorporated in this product, each instrument is given the full backing of Klark-Teknik's "**reliability control**" which proves each product against a specification consistent with the highest professional standards. Only top quality components are used, and every unit is bench tested and aligned before a burn-in period and final performance test.

Options

Aluminium security cover

Relay by-pass

Transformer input*/output balancing

* Input transformer balancing is non retrofittable and has to be specified with order.

Options Ordering Information	Parts Number
Aluminum security cover	SC71
Relay by-pass kit	RL70
Output balancing transformer	BU37
Input balancing transformer	BN37

Output Balancing Transformer Retrofit Procedure

(one transformer required per output)

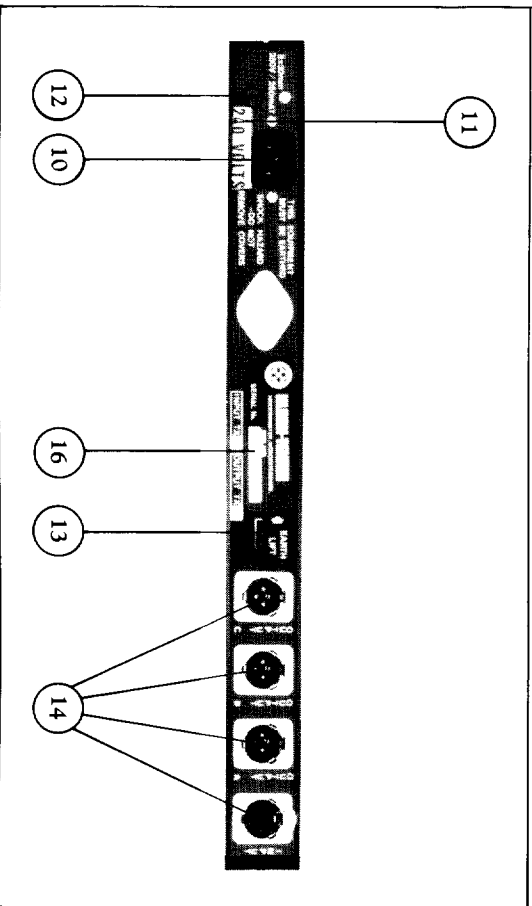
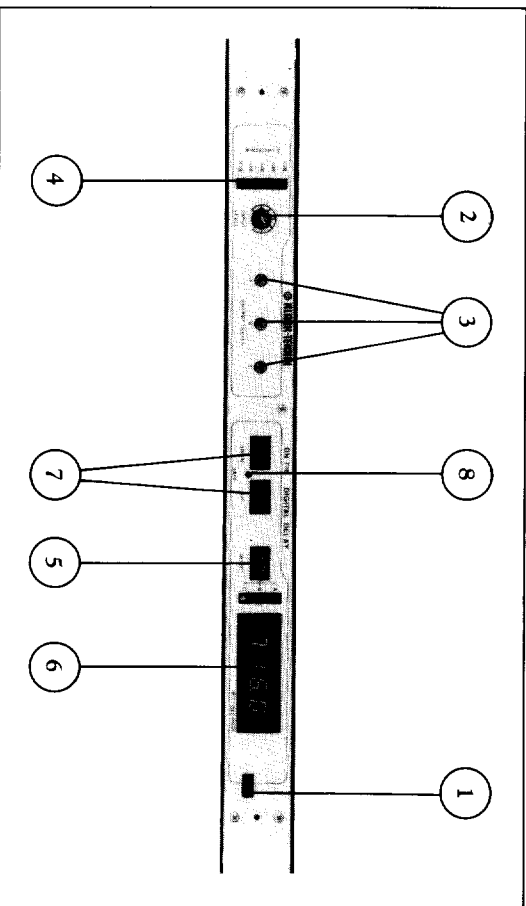
- a) Removes links on connectors ST6, ST7 and ST8.
- b) Mount transformer using screw provided.
- c) Connect transformer plug.
- d) Change link on ST5 to BAL position.

Relay By-Pass Retrofit Procedure

(two relays required - KTR part ES-R4201)

- a) Remove yellow links L1 to L8.
- b) Solder in the relay units.

Instrument Familiarisation



Front Panel Functions

1. **The Power Switch** is a two pole type isolating both the live and neutral conductors. When the power is on the display lights.
2. **The Input Level Control** should be set so that with normal signal the 3dB LED on the headroom indicator lights up during loud passages.
3. **The Output Level** is set at +8dBu for maximum level i.e. the onset of 0dB LED on the headroom indicator. The output level can be adjusted with the output level controls labelled A, B, C on the front panel. The range of adjustment is from -2dBu to +18dBu for maximum level.
4. **The Headroom Indicator** shows the headroom left in the digital section before clipping commences. The top LED (0dB) activates at 2dB below clipping.
5. **Display Select Switch** control the display only and will not affect the delay settings. The LEDs A, B and C adjacent to the switch indicate the relevant output channel delay setting. Pressing the "SELECT" switch will indicate the next output delay setting.
6. **Display** shows the amount of delay (time) on the output channel selected. The display is auto-ranging.
7. **Delay Setting Switches** These are used to increase or decrease the delay time on any of three outputs. To alter delay settings:- First select the channel required by pressing the "SELECT" switch. The display will show the present delay time. Then press the "UP" or "DOWN" switch to alter the delay setting accordingly. The "UP" and "DOWN" switches can be used in the following manner:- Nudge - by single taps on the switch, the delay time can be incremented or decremented in 20 micro second steps. Slow - by permanent depression of the relevant switch. Fast - by permanent depression of one switch followed by the depression of the other switch.
8. **Safe Switch** is an alternate action switch, and is located within the front panel, so that it cannot be pressed unintentionally. Use a small screwdriver to depress it. When pressed, the delay settings are frozen and cannot be altered by the "UP" and "DOWN" Delay Setting Switches. To return the unit to normal operation, depress the switch again.
9. **Display Rotate** - when in the "SAFE" mode, the display will automatically rotate showing the respective delay of each output for approximately one second. This facility can be cancelled by changing the position of the switch on the main circuit board labelled "ROTATE".

Rear Panel Functions

10. **Mains** is supplied via an IEC standard 3 Pin connector. A compatible power cord is supplied with the unit.
11. **Operating Voltage** - this is clearly marked on an adhesive label. See section INSTALLATION for voltage change-over instructions.
12. **The Mains Fuse** is located in a fuse holder fitted to the rear panel. Always replace with the correct type and rating of fuse, as indicated adjacent to the fuseholder.
13. **Earth Lift Switch** situated on the rear panel, this switch disconnects the signal ground from the mains and chassis earth. This should be used if hum attributable to earth-loops is experienced and will generally solve the problem. It is also safe, unlike the practice of disconnecting the mains earth from the power cord.
14. **Input and Output Connections** are made via complementary XLR style sockets.
15. For wiring details see page entitled Audio Connections.
16. **The Serial Number** of this unit should be quoted in any correspondence concerning the unit.

Audio Connections

Input

The input circuitry is a transformerless, electronically balanced design which achieves a symmetry of better than -50dB from 20Hz to 10kHz .

If transformer balancing of the input is required, this must be stipulated at the time of order; it is not retrofittable.

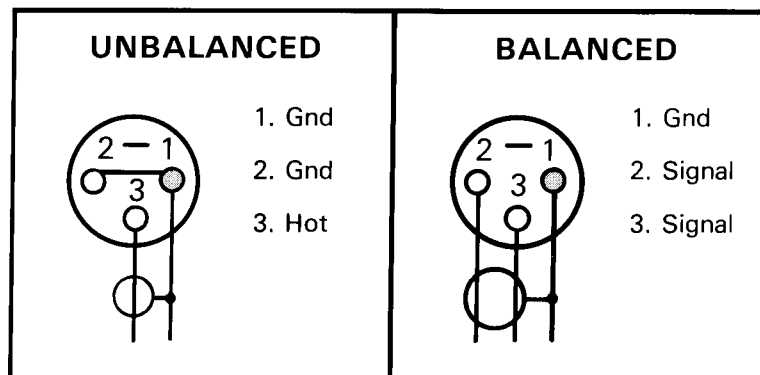
Output

The standard output is unbalanced, but balancing transformers are available and may be retrospectively fitted. The output circuitry is capable of driving a $600\ \text{ohm}$ load at a level of $+18\text{dBm}$.

The unbalanced output XLR connections are: Pin 1 circuit ground; Pin 2 circuit ground; Pin 3 signal.

This can be changed internally to make Pin 2 the signal as follows:

- Remove jumper links which short out Pins 2 and 3, Pins 4 and 5 on ST6, ST7, ST8.
- Join Pins 2 and 4, Pins 3 and 5 on ST6, ST7, ST8. (6 way Molex Plug).



Balanced Circuits

Transformer or electronically balanced connections have the benefit of "common mode rejection" which eliminates externally induced interference such as mains hum etc. Balancing is especially useful when long cable runs are used between pieces of equipment.

Transformer balanced circuits have the added advantage of being, "fully floating" with the ground (earth) or screen being totally isolated from the signal. In installations where a difference in earth potential is likely to occur this isolation prevents grounding problems which can, in some cases, damage the equipment.

Installation

Mounting

The DN716 is designed for standard 19 inch rack mounting and is 44mm (1.75 inch) high and 300mm (11.75 inch) deep, excluding connectors.

Observe the following when mounting:-

- Use a well ventilated rack.
- Do not mount on top of heat producing equipment (i.e. power amplifiers etc.)
- Use fan assisted ventilation if rack exceeds 40°C.

Power Requirements

The factory set nominal operating voltage is clearly indicated on the rear panel. Power connection is made via the standard 3 pin CEE Power Cable provided. The chassis to mains earth connection made by this cable must not be removed. Maximum power consumption of the unit is 20 VA. For power cables using flying leads, the wires in the cable are colour coded as follows:-

BROWN:	LIVE
BLUE:	NEUTRAL
GREEN/YELLOW:	EARTH

Voltage Change-Over

* NOTE: Noisy supply or insufficient mains voltage will cause the DN716 to operate in an intermittent manner.

The unit can be set to operate on 110, 120, 220 or 240 (nominal) VAC @ 50/60Hz, by changing the position of the two voltage selector switches inside the unit, adjacent to the Mains Power Transformer.

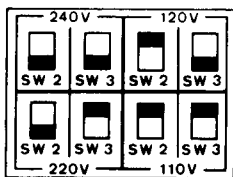
The range of supply voltages suitable for each switch position are as follows:

Switch Position	Range
240V	215-250V @ 50Hz
220V	200-230V @ 50Hz
120V	105-130V @ 60Hz
110V	95-115V @ 60Hz

Voltage Switch Settings

To change the operating voltage, a qualified service technician or an approved Klark-Teknik dealer must carry out the following procedure:-

OPERATING VOLTAGE



Voltage
Switch Settings

1. Remove the mains cable and connector from the socket.
2. Remove the four screws retaining the top cover.
3. Remove the top cover.
4. Set the voltage selector switches inside the unit to the appropriate positions as shown.

5. Re-install the top cover and refit and tighten all screws.
6. Affix a label on the rear panel with the new voltage details.

Mains Fuse

The correct value is T 250mA for all operating voltages.

Specifications

Input

Type (one)	Electronically balanced
Impedance (ohm)	
Balanced	20k
Unbalanced	10k
Level	0dBu to +22dBu

Outputs

Type (three)	Unbalanced
Min. Load Impedance	600 ohms
Source Impedance	<60 ohms
Level	+8dBm nominal for maximum level, adjustable from -2 to +18dBm via the output level controls on front panel. (maximum level = 2dB below clipping level).

Performance

Frequency response	± 1 dB (20Hz - 20kHz) <i>(Any Output at any level at all delay settings.)</i>
Total Harmonic Distortion	<0.01% @ 1kHz <0.03% @ 20Hz - 20kHz <i>(Distortion plus noise at any delay setting.)</i>
Dynamic Range	>90dB (20Hz - 20kHz unweighted) <i>(At any delay setting.)</i>

Features

Delay Range	Adjustable from 0 to 1.311 seconds in 20 micro second steps
Memory Retention	One year minimum (unit powered down)

Power Requirements

Voltage	110/120/220/240V 50/60Hz AC
Consumption	<20VA

Weight

Nett	2.8kg
Shipping	3.5kg

Dimensions

Width	482mm (19 inch)
Depth	300mm (11.75 inch)
Height	44mm (1.75 inch)

Terminations

Input	3 Pin XLR
Outputs	3 Pin XLR
Power	3 Pin CEE

Circuit Description

The system uses three printed circuit boards (PCB's)

- 1) The Main PCB (No. 2647).
- 2) The Front panel Analogue PCB (No. 2646).
- 3) The Front Panel Digital PCB (No. 2645).

The circuits can be split into four sections:-

- 1) Power Supply.
- 2) Analogue Input and Output.
- 3) A/D and D/A Conversion.
- 4) Digital Section.

1. Power Supply (Circuit Diagrams Sheet 4)

The power supply uses a toroidal mains transformer and covers the range of 95V AC at 60Hz to 250V AC at 50Hz, switched via SW3 and SW4 on the main PCB.

The three regulated DC voltages (+15V, -15V and +5V) required by the DN716 are derived from three separately fused secondaries all using bridge rectifiers and linear regulators.

The power supply can be completely isolated from the system by unplugging the connector ST17 and removing link L10.

2. Analogue Input and Outputs (Circuit Diagram Sheet 1)

2.1. Input Section

The standard unit is supplied with an electronically balanced input stage. IC1a, 1b are configured to provide both current and voltage balancing. C8 and C9 provide RF suppression whilst in order to achieve maximum CMR (Common Mode Rejection) at high frequencies and compensate for component tolerances, C7 is selected on test. P1 adjusts low frequency CMR.

Relay by-pass and input isolation transformers can be provided as options. Fitting instructions are on the circuit diagram.

The signal from the input stage (IC1 pin 7) goes to the input level control on the front panel analogue PCB (2646) via the cable at ST13 and returns to the buffer amplifier IC5. This IC also provides an approximate 3dB boost at 20kHz for the SIN X/X compensation (R32, R33, C25).

The output from IC5 returns to the front panel analogue PCB to drive the LED indicator circuit (IC1). It also goes to the input of the hybrid thick film anti-aliasing filter (IC6). This is a 7 pole elliptical low pass filter providing the bandwidth limiting required by the A/D converter.

2.2 Output Section

The three output sections A, B, and C are identical. Taking A:- The signal from the sample and hold amplifier (IC12) goes to the reconstruction filter (IC7) via the mute circuit (R47, IC10a). IC10a is a CMOS analogue switch which shorts the signal to ground during the power-on routine, preventing any "garbage" data in the memory from reaching the outputs. The reconstruction filter (IC7) is a thick film hybrid 7 pole elliptical filter that converts the digitally sampled signal into a smooth analogue signal. The output from the filter goes to the output line drive amplifier (IC2). The gain of IC2 is controlled by the output level control on the front panel analogue PCB.

The standard output configuration is unbalanced, with the signal going via R8 to pin 3 on the XLR socket. Relay by-pass and output balancing transformers are available as an option, fitting instructions are on the circuit diagram.

2.3 Terminations

The input circuitry is designed to operate either balanced or unbalanced. If used with an unbalanced input it is advisable to short the unused pin to ground at the connector, especially if long cables are used. This will prevent cable capacitance affecting the high frequency performance. As both inputs (pin 2, pin 3) are identical, either one can be used as the HOT terminal. If pin 3 is used, then the outputs will be in phase with the inputs at zero delay.

The outputs are unbalanced as standard and use pin 3 of the XLR as the live signal.

The XLR configuration on both inputs and outputs can be changed by cross linking the output links L1 to L8, or on the outputs only by changing the configuration of the Jumper links on ST6, ST7 and ST8.

3. A/D and D/A Conversion (Circuit Diagram Sheet 1, 2 and 3)

The conversion system used is P.C.M. (Pulse Code Modulation) with 16 bit linear coding. This gives optimum performance with present technology. The A/D converter is of the Successive Approximation type which enables a high sampling rate (50kHz) to be used.

The A/D and D/A conversion is accomplished using the same 16 bit DAC (IC19).

3.1 A/D Converter

During the A/D conversion, the DAC (IC19) is routed to the input section via the analogue multiplexer (IC15a, 15d). The A/D converter consists of the Track and Hold amplifier (IC15b, 15c, 16), the comparator (IC17, 18), the DAC (IC19) and the successive Approximation Registers (IC24, 25).

The output from the track and hold amplifier (IC16) is compared with the output from the DAC by the comparator. IC18 acts as a high speed low noise buffer amplifier for the comparator IC17.

In order to optimise the noise performance, it is essential that the DAC is operated around its most linear point. The bias preset P3 adjusts the offset voltages in the system and should be set for the lowest noise figure.

3.2 D/A Conversion

During D/A conversion the analogue multiplexer (IC15a, 15d) routes the DAC output to the current to voltage converter IC14. This voltage then goes to the inputs of the three output sample and hold amplifiers (IC10d, 12) (IC11c, 13a) (IC11d, 13b). Each output is sampled when its signal is valid, the sequence can be seen from the timing diagram at the end of the section. The sampled outputs go to the reconstruction filters via the mute circuit (described in 2.2).

3.3 A/D and D/A Timing and Control (Circuit Diagram Sheet 2)

All timing signals for the converters are generated from the PROM (IC62) and counter (IC63). IC61 is an 8 bit register which synchronizes all the signals. The RWR signal from the microprocessor synchronizes the counter with the microprocessor. Using this system eliminates all component errors and temperature drift as all the signals are locked into the Master Clock (6MHz).

4. Digital Section (Circuit Diagrams Sheet 2 and 3)

The entire system is controlled by the Z80 microprocessor (IC48) and its associated circuitry. The system clock is 6MHz. This is derived from the 12MHz oscillator (IC57) and divider (IC58).

The control programme is stored in the Eprom (IC49), as this cannot run at the speed required by the system, the processor copies the programme into fast RAM (IC50) as part of the start-up routine. IC59 generates the "wait" states required by the Eprom.

The front panel display is driven by IC1 on the front panel digital board. All front panel functions are controlled by the processor via the interface IC's (IC60, 64, 65, 68, 69, 70) on the main PCB.

The CMOS RAM (IC50) stores the delay settings and on power down is powered by the lithium battery (BT1). This should have a minimum life of 5 years.

The power supply monitor (IC26) senses any drop out on the 5V DC supply and resets the processor, it also switches the CMOS RAM (IC50) into the battery powered standby mode.

The processor also controls the Analogue Data Memory. The addresses, read and write signals, and refresh control are all generated by the processor. The timing of these is locked into the converter timing by the sync. pulse RWR.

2.2 Output Section

The three output sections A, B, and C are identical. Taking A:- The signal from the sample and hold amplifier (IC12) goes to the reconstruction filter (IC7) via the mute circuit (R47, IC10a). IC10a is a CMOS analogue switch which shorts the signal to ground during the power-on routine, preventing any "garbage" data in the memory from reaching the outputs. The reconstruction filter (IC7) is a thick film hybrid 7 pole elliptical filter that converts the digitally sampled signal into a smooth analogue signal. The output from the filter goes to the output line drive amplifier (IC2). The gain of IC2 is controlled by the output level control on the front panel analogue PCB.

The standard output configuration is unbalanced, with the signal going via R8 to pin 3 on the XLR socket. Relay by-pass and output balancing transformers are available as an option, fitting instructions are on the circuit diagram.

2.3 Terminations

The input circuitry is designed to operate either balanced or unbalanced. If used with an unbalanced input it is advisable to short the unused pin to ground at the connector, especially if long cables are used. This will prevent cable capacitance affecting the high frequency performance. As both inputs (pin 2, pin 3) are identical, either one can be used as the HOT terminal. If pin 3 is used, then the outputs will be in phase with the inputs at zero delay.

The outputs are unbalanced as standard and use pin 3 of the XLR as the live signal.

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The processor also controls the Analogue Data Memory. The addresses, read and write signals, and refresh control are all generated by the processor. The timing of these is locked into the converter timing by the sync. pulse RWR.

Acoustic and Psycho-Acoustic Basis of Operation

Let us begin by taking speech as our signal, as this will be the most common source we shall probably be dealing with.

In Figure 1 we can see the effect that sounds travelling different distances can have upon speech intelligibility. In this illustration we can see that each of the three loudspeakers emits exactly the same sound, but due to the different distances between the loudspeakers and the reception point, the sounds do not arrive together, but instead arrive in a ragged fashion after a series of slight delays. The result is that the signal from the first loudspeaker does not quite synchronise with signals from the others, resulting in a 'blurring' of the sound and a loss of speech clarity. With longer delays, a complete loss of speech intelligibility may well occur!

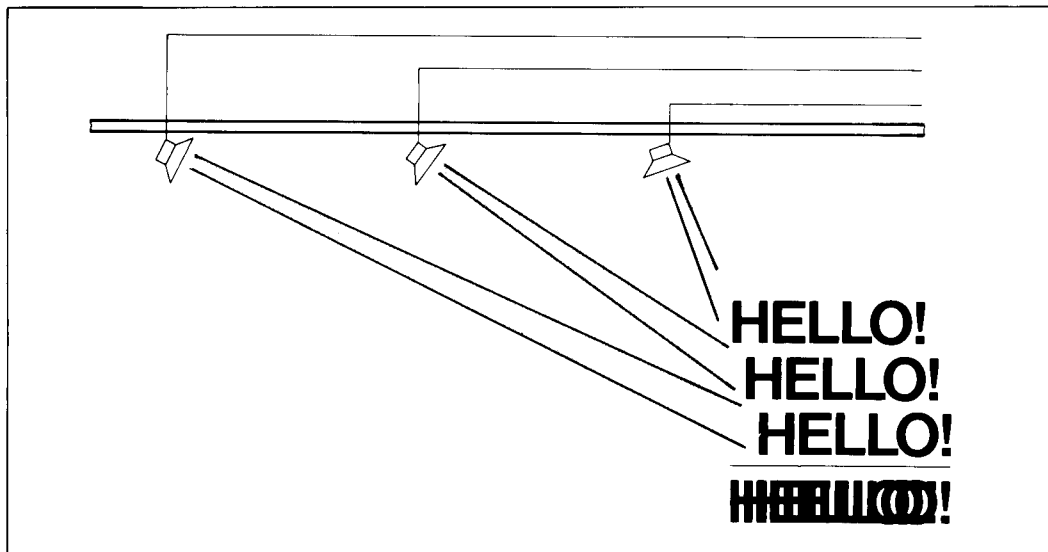


Fig 1 Illustration of how time delay is generated and how speech intelligibility can be affected.

However, by electronically delaying the signals to the nearer loudspeakers, to compensate for the additional distance travelled and hence transit time from the more distant loudspeakers, the sound arrivals can be synchronised, and speech intelligibility and clarity restored as shown in Figure 2.

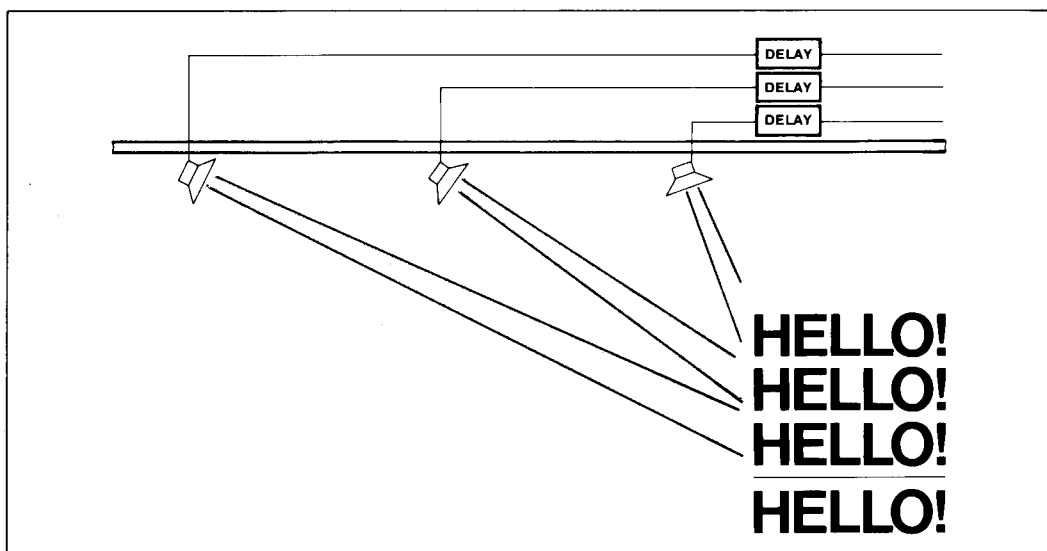


Fig 2 Use of digital delay lines to compensate for sound path/time arrival differences resulting in improved speech clarity and intelligibility.

Another common example of a situation where a digital delay unit can overcome a time delay problem is shown in Figure 3. Here, under balcony loudspeakers are being used to supplement the main sound system which cannot adequately penetrate under the balcony overhang.

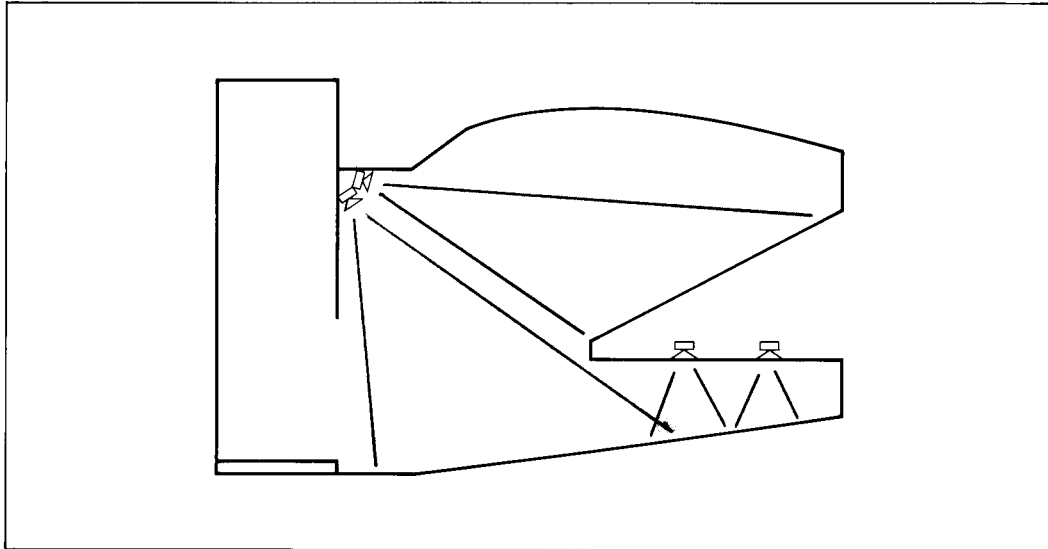


Fig 3 Theatre sound system with underbalcony loudspeakers.

As Figure 3 shows, a region of overlap occurs between the two loudspeaker systems, which due to the difference in arrival times of the two sets of signals will give rise to a loss of the intelligibility and clarity. Furthermore, under the balcony, the sound will be perceived to be coming from the overhead loudspeaker rather than from the stage. The introduction of a signal delay line into the system can overcome both of these problems.

Firstly, by synchronising the arrival times of the sounds emitted from the two sources, blurring and confusion can be eliminated.

Secondly, by taking advantage of the 'Haas' or precedence effect, the correct localisation can be maintained i.e. the sound will still appear to come from the direction of the stage, although it is being reinforced from a local loudspeaker overhead.

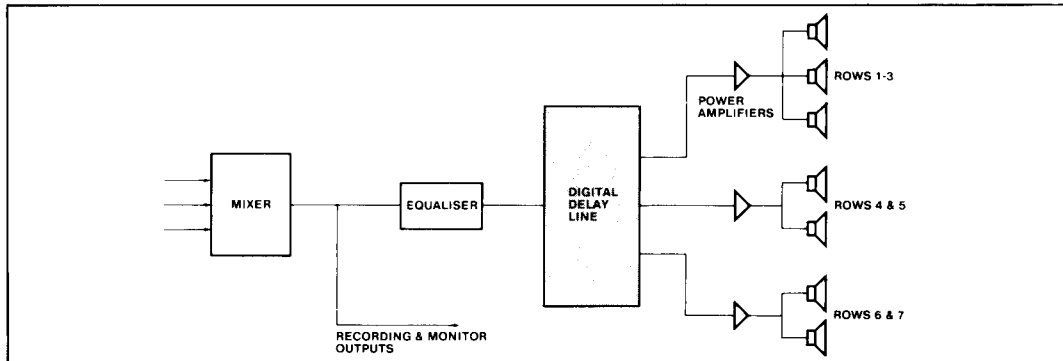
To optimise these effects, an understanding of some basic acoustic and psycho-acoustic factors and characteristics of our hearing mechanism is required.

If you would like to know more about this subject you can as a Klark-Teknik product user obtain the company's "Sound System Design and Engineering" publication, simply by returning the enclosed request reply card.

Applications

Conference and Distributed Loudspeaker Systems

Conference systems employing a central cluster or column loudspeaker on either side of the stage, either with or without supplementary coverage loudspeakers can be treated in exactly the same way as theatre systems. In flat floor venues, with relatively low ceiling heights, a distributed loudspeaker system using ceiling mounted loudspeakers will provide the best sound coverage - the difference in sound level between the front and rear areas of the room becoming too great for the successful operation of a cluster or column loudspeaker type arrangement.



Simplified schematic diagram of conference room sound system.

Ceiling Mounted Distributed Systems

By careful loudspeaker distribution, a very even coverage throughout the room can be achieved. Furthermore, a microphone should be able to be used anywhere within the room without feedback. But will the system benefit from a delay unit? This very much depends on the room acoustics and finishes. For example, if the room has a highly absorbent acoustic tiled ceiling and carpeted floor, then much of the live speech sound will be quickly absorbed. Listeners towards the centre and rear of the room will therefore only hear the sound of the nearest loudspeaker and a delay line will not really benefit the system. But if the room has been built with natural acoustics in mind or if the ceiling is hard and sound reflecting, then the natural voice will be heard to some extent at least throughout the room and this may then be reinforced with a delay unit providing synchronisation between the natural voice or direct sound and the reinforcement system.

Figure 14 shows a section through a typical small conference room, the distance from the stage to the rear of the room being 20 metres whilst the height of the room is 5 metres. A total of 7 rows of loudspeakers are being used to provide an even coverage throughout the room.

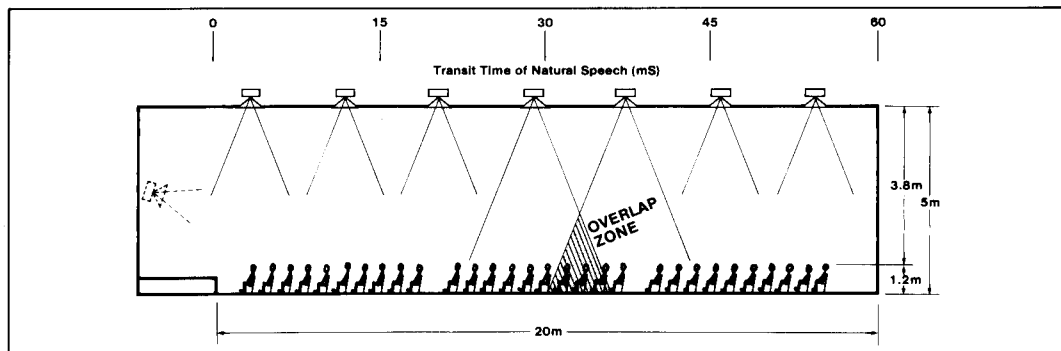
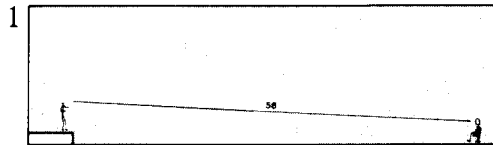


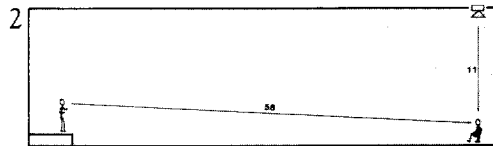
Fig 14 Typical conference room sound system. Time delay zones must be no more than 30 mS steps in order to ensure full speech integration occurs within the overlap zone.

Step by Step Calculation of a Zoned Time Delay System

Rear of Room



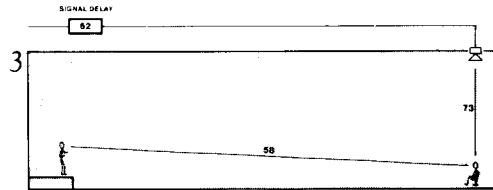
Direct sound arrives after 58 milliseconds



Direct sound arrives after 58mS
 Sound from overhead loudspeaker arrives after 11 mS
 Sound is heard from the overhead loudspeaker
 $58 - 11 = 47\text{mS}$ before the direct sound

A time delay of 47mS is well outside the 30mS rule
 The overhead loudspeaker should therefore be delayed to improve both intelligibility and localisation

Time delay required = $47 + 15$ (Haas) = 62mS

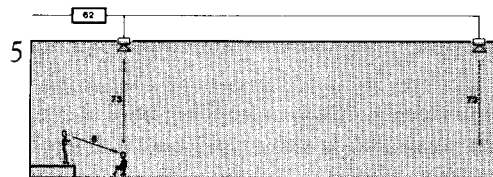


Insertion of signal delay improves intelligibility and localisation for listener at rear of room

Front of Room



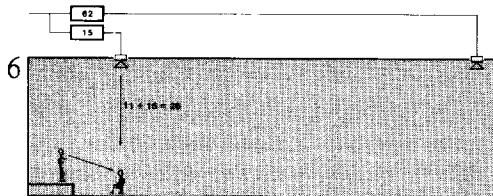
Listener at front of room hears direct sound after 6mS



Listener at front of room hears direct sound after 6mS
 but will hear overhead loudspeaker with a delay of $73 - 6 = 67\text{mS}$ if this is connected to signal delay

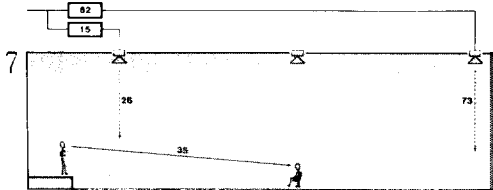
THIS WILL RESULT IN A DISTINCT ECHO AND VERY POOR INTELLIGIBILITY!

The loudspeakers covering the front of the room must therefore be fed from a second signal delay eg. 15mS



The listener at the front of the room now hears:-
 The direct sound after 6mS
 and the sound from the overhead loudspeaker after 26mS
 The direct sound is therefore heard 20mS before the overhead loudspeaker
 Localisation is therefore maintained and speech intelligibility enhanced

Centre of Room



A listener near the centre of the room (eg. 12 metres from the stage) will hear:-
 The direct sound after 35mS

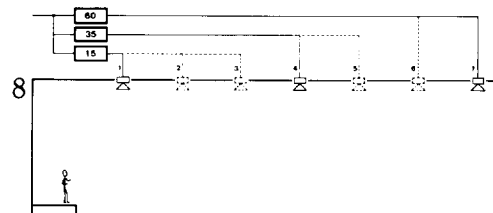
If his overhead loudspeaker is connected to delay 1 (62mS) he will receive sound arrivals at 35 and 73mS (delay = 38mS)

or if the overhead loudspeaker is connected to delay 2 (15mS) he will receive sound arrivals at 35 and 26mS

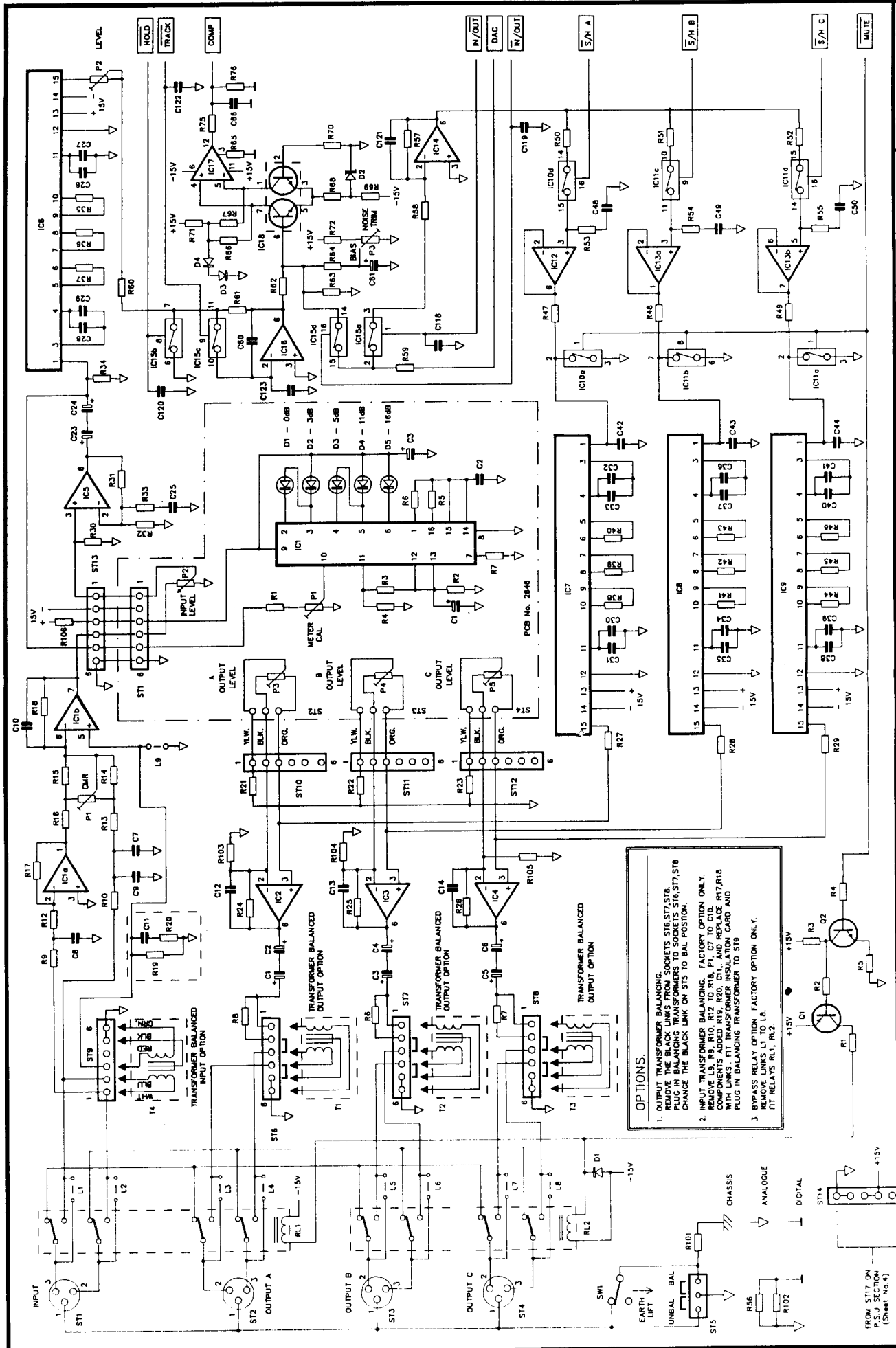
Although the delay is only 9mS — the overhead loudspeaker will be heard first and regarded as the sound source

Ideally a third delay circuit is required

Final Configuration of Delayed Sound System



Delay overlaps loudspeakers.3 and 4 $35 - 15 = 20\text{mS}$
 loudspeakers 5 and 6 $60 - 35 = 25\text{mS}$

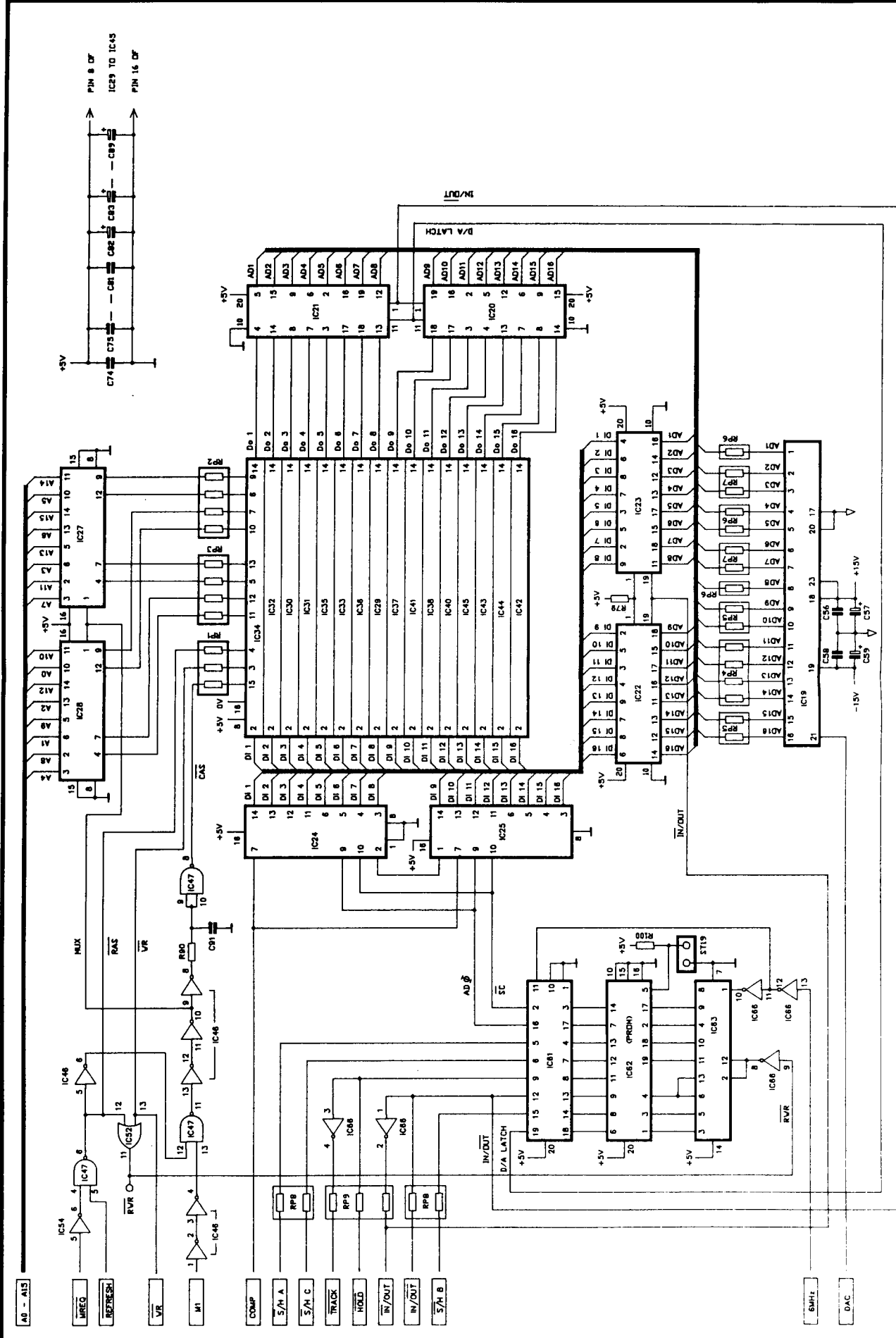


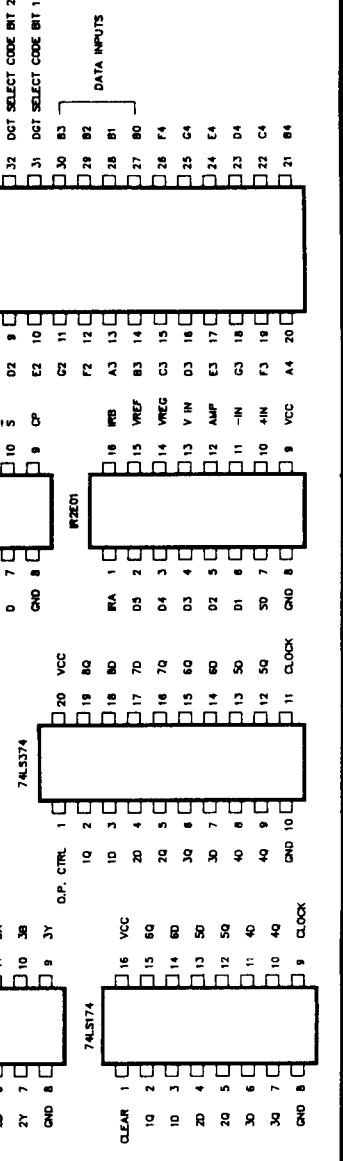
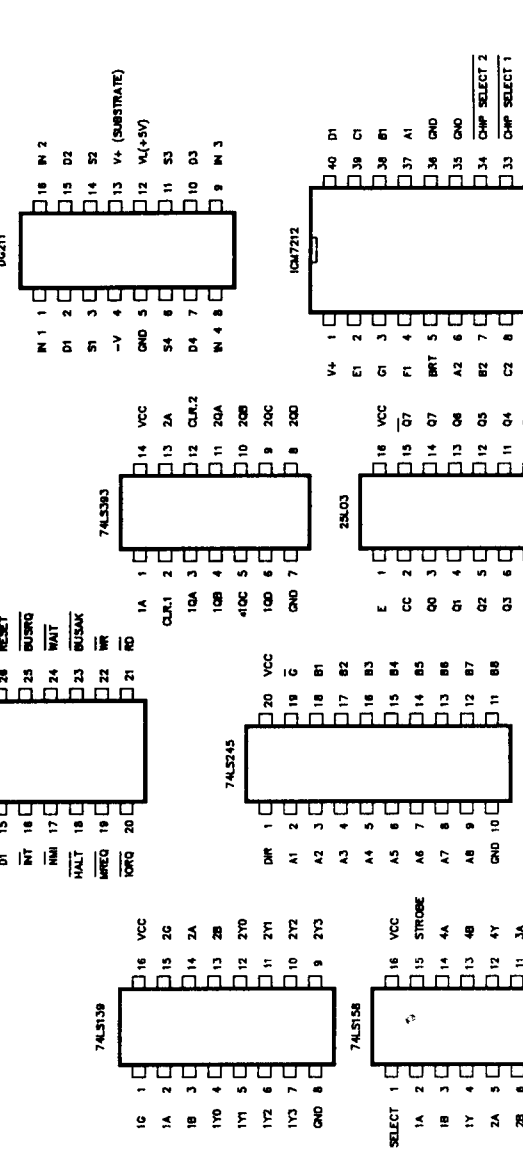
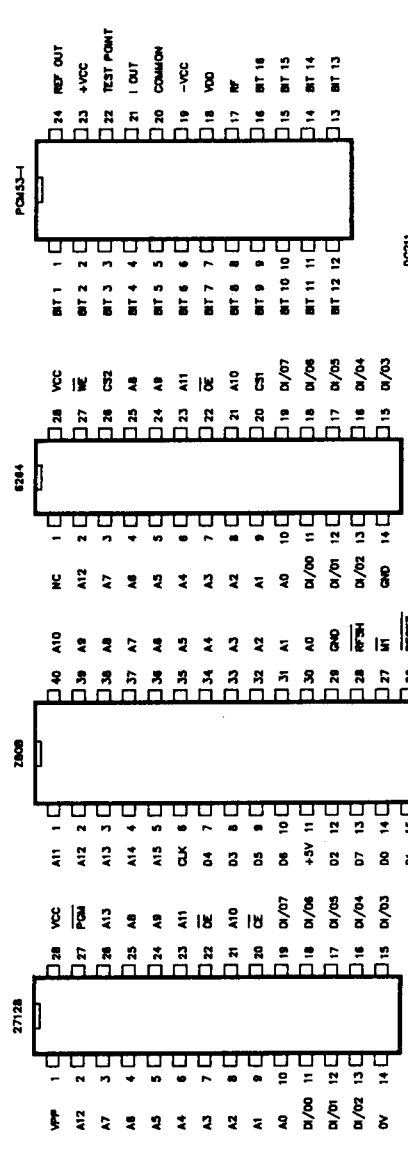
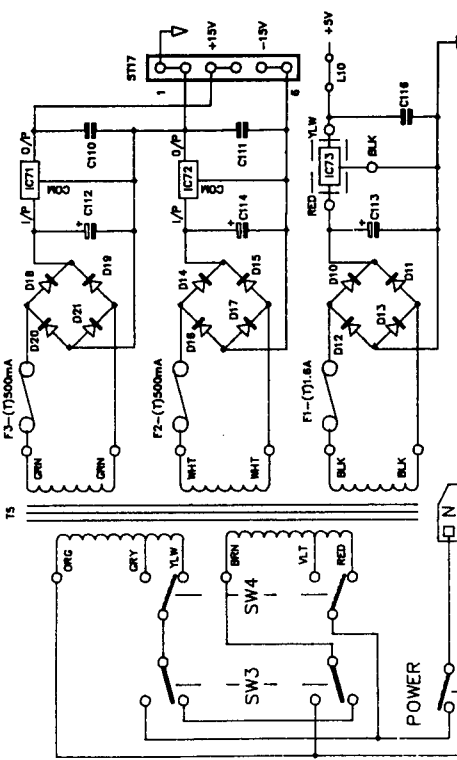
OPTIONS.

1. OUTPUT TRANSFORMER BALANCING.
REMOVE THE BLACK LINKS FROM SOCKETS ST6,ST7,ST8.
PLUG IN BALANCING TRANSFORMERS TO SOCKETS ST6,ST7,ST8
CHANGE THE BLACK LINK ON ST5 TO BAL POSITION.
2. INPUT TRANSFORMER BALANCING. FACTORY OPTION ONLY.
REMOVE L9, R9, R10, R12 TO R16, P1, C7 TO C10.
COMPONENTS ADD TO THE CUSTOMER'S EVALUATION CARD AND
PLUG IN BALANCING TRANSFORMER TO ST9
3. BYPASS RELAY OPTION FACTORY OPTION ONLY.
REMOVE LINKS L1 TO L6.
FIT RELAYS RL1, RL2.



FROM ST17 ON P.S.U SECTION (Sheet No 4)





DN716 | PSU SECTION(sheet 4 of 4) BRD:2647/3 23.JUN.86

COPYMASTER DO NOT REMOVE

NOTE! This sheet is your Copymaster. Please duplicate on photocopier when needed.

Service Request Form

Please complete this form and send it to Klark-Teknik before returning the unit. Attach duplicate to the returned unit.

NAME TELEPHONE

ADDRESS
.....
.....
.....

MODEL No. SERIAL No.

PURCHASED FROM DATE

Please tick appropriate box

REPLACEMENT PACKING REQUIRED YES NO

- 1) Describe symptoms of malfunction.
- 2) Which channel(s) exhibit(s) the problem?
- 3) Under what conditions does the problem occur?
 - a) All the time
 - b) After a while
 - c) At high signal levels
 - d) At high temperatures
 - e) Other (please explain)

Is the fault: Permanent Intermittent

- 4) What did you do to isolate the problem to this unit?
- 5) Further comments.

Important Note:

It is often useful to call your dealer or the factory explaining the nature of the problem with the unit. In many instances the problem can be solved without returning the unit to the factory. If the unit has to be returned to the factory, use original packing only. If you do not have one, we will provide a replacement.

Factory authorised service facilities are located throughout the world. Call your dealer or the factory for the location of the service facility nearest you.

Klark Teknik Plc.
Klark Industrial Park
Walter Nash Road
Kidderminster
Worcs. DY11 7HJ
England.

Tel: (0562) 741515
Fax: (0562) 745371

Warranty

This product is manufactured by Klark-Teknik and is warranted to be free from defects in components and factory workmanship under normal use and service for a period of one year from the date of purchase.

During the warranty period Klark-Teknik will undertake to repair or at its option, replace this product at no charge to its owner when failing to perform as specified, provided the unit is returned shipping pre-paid, to the factory or authorised service facility.*

No other warranty is expressed or implied.**

This warranty shall not be applicable and be void when this product is subjected to:-

- a) Repair work or alteration by persons other than those authorised by Klark-Teknik in such a manner as to injure, in the sole judgement of Klark-Teknik, the performance, stability, reliability or safety of this product.
- b) Misuse, negligence, accident, act of God, war or civil insurrection.
- c) Connection, installation, adjustment or use otherwise than in accordance with the instructions provided by Klark-Teknik.

* See enclosed service request form.

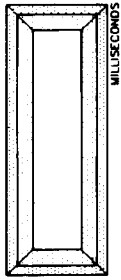
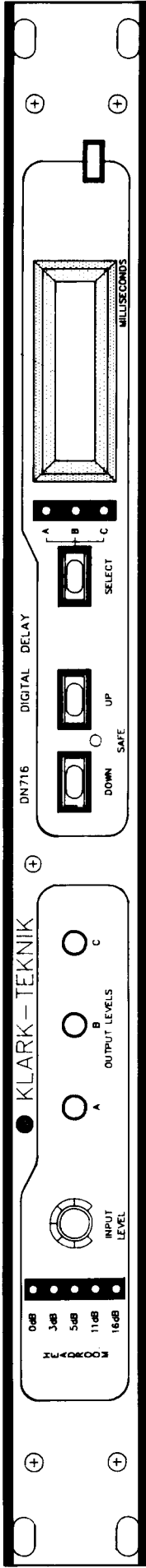
** "MELT" filter circuits used in the Series 300 Graphic Equalisers are warranted for 5 years.

Klark-Teknik reserves the right to alter specifications without notice. This warranty does not affect the statutory rights of the UK customer.



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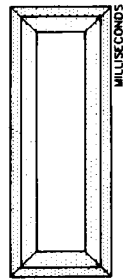
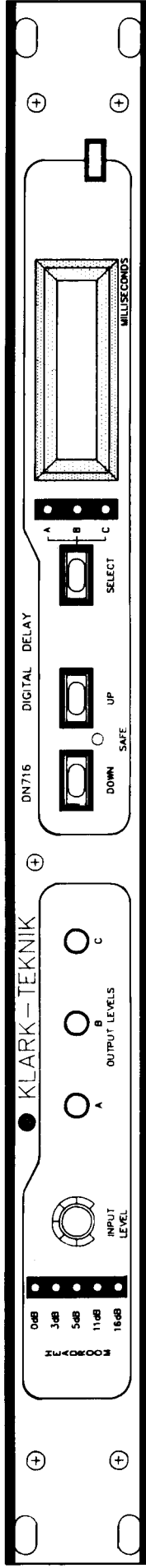
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DELAY LINE No.

SERIAL No.

LOCATION:

DATE:



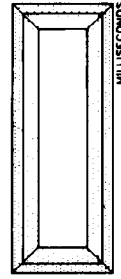
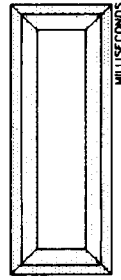
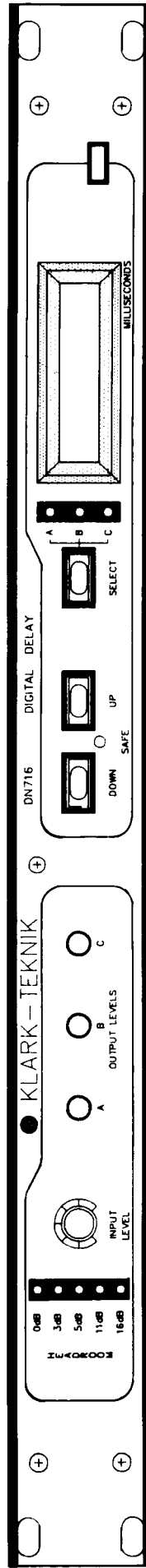
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SERIAL No.

LOCATION:

DATE: