

QA

**Operating  
and service  
information  
DN504  
Quad Compressor/  
Limiter**

**IMPORTANT:**

**Product registration**

Please check that the product registration card is enclosed with this book. If **NOT**, immediately inform the factory quoting the unit's serial number.

## Introduction

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The Klark Teknik DN504 Quad Compressor/Limiter has been designed to give the studio or PA engineer comprehensive and convenient automatic control over gain and level. Four channels of compression are available in one unit of nineteen inch rack space. Even at this high packing density, the degree of control over each channel of audio is greater than that found on many conventional two channel compressor/limiters.

As a Compressor, the DN504 can achieve almost undetectable dynamic range reduction by the use of sophisticated automatic attack and release circuitry. Or compression can be used as an effect in its own right with full manual control over Threshold, Ratio, Attack, Release and Gain. There is also a switchable 'Knee' control which provides the option of Hard or Soft compression styles.

With the compression ratio set to the maximum value of 50:1, the DN504 is also able to operate as a high quality, fast acting, Limiter.

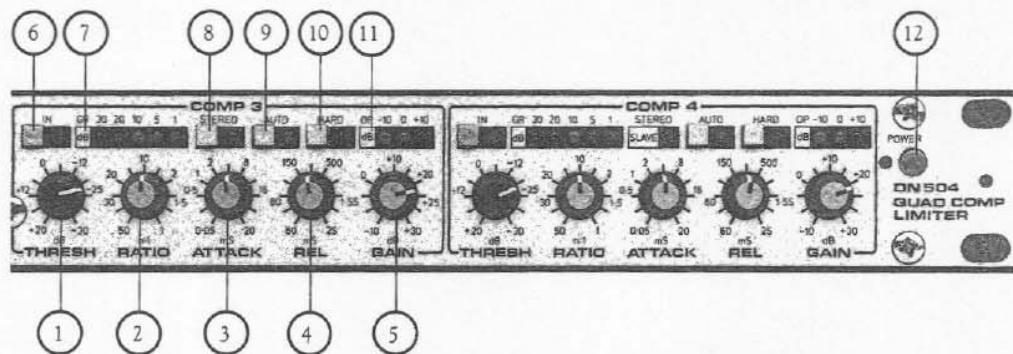
The DN504 has LED indicators for every switched audio function, and also has separate LED bar graph meters for gain reduction and for output level on each channel.

Four side chain inputs are provided. A stereo link function is available to allow channels to be linked in pairs to compress stereo signals while avoiding the possibility of image shift.

The DN504 Quad Compressor/Limiter is built to Klark Teknik's high standards of design and construction. A well thought out aesthetic presentation is also an aid to realising full dynamic control over gain and level in today's complex productions.

## Instrument familiarisation

### Front panel controls

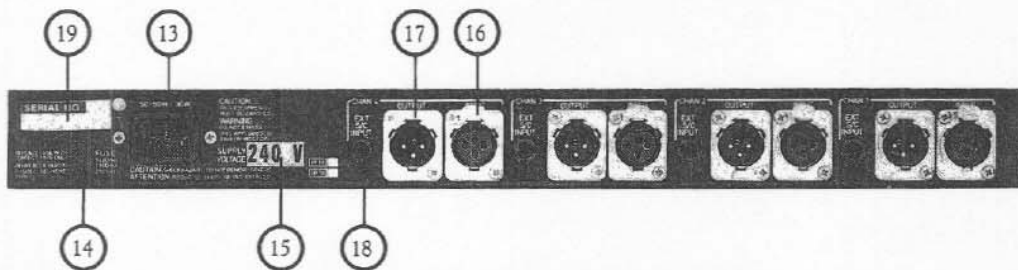


The DN504 Quad Compressor/Limiter consists of four identical channels.

1. **Threshold** determines the level above which gain reduction (compression) will take place, variable from +20dBu down to -30dBu. The Gain Reduction LED bar graph meter indicates in decibels by how much the signal is compressed at any instant.
2. **Ratio** sets the degree of compression applied above the threshold level, variable from 1:1 to 50:1. A ratio of 1:1 corresponds to zero compression: Signal In = Signal Out. A ratio of 10:1 indicates that when the input signal, above the threshold, rises in level by 10dB, the output level will rise 1dB.
3. **Attack** sets the time taken for compression to commence after the threshold is exceeded, from 0.05 milliseconds (50 microseconds) to 20 milliseconds.
4. **Release** determines the time taken for the gain to return to normal after the signal drops below the threshold level, adjustable from 60 milliseconds to 2 seconds.
5. **Gain** is used to restore level lost as a result of the compression process, variable from -10dB to +30dB.
6. The **In** switch selects the channel to be in or out of circuit.
7. The Gain Reduction (**GR**) five-stage LED bar graph meter indicates by how much the signal is compressed at any instant.
8. **Stereo** links together pairs of channels of the DN504 for compressing stereo signals. Channel 1 operates as the master for Channel 2, which becomes the slave. Channels 3 and 4 can be linked in the same way. In Stereo mode, the unit still monitors the levels of both channels of each pair to detect when the signal is above the threshold level (on Channels 2 and 4, the Stereo switch is replaced by a 'Stereo Slave' LED indicator).
9. **Auto** allows the setting of attack and release times to be automatic or manual. On Auto, the attack time of the input signal and the amount by which it exceeds the threshold level are assessed, and suitable Compressor/Limiter attack and release values set to give an almost undetectable reduction in signal dynamic range. On Manual, the rotary Attack and Release controls come into operation.

10. **Knee**, is switchable between a 'Hard' and 'Soft' compression effect. A hard knee means that when the input signal level rises above the threshold, it is immediately compressed at the full ratio as set. A soft knee means that compression will start below the threshold level at a low ratio and gradually attain the full ratio selected only at higher levels.
11. The output level three-stage LED bar graph meter indicates OUTPUT level when the channel is switched in circuit. When the channel is bypassed, the meter indicates INPUT level.
12. Power switches mains power on or off.

#### Rear Panel Functions



13. **Mains** is supplied via an IEC standard 3-pin connector. A compatible power cord is supplied with the unit.
14. The **mains fuse** is located in a fuse holder fitted to the rear panel. Always replace with the correct type and rating as indicated on the unit.
15. **Voltage selector switch**. This unit is switchable between two nominal supply voltages, 110V and 220V, via a slide switch. The switch **MUST** be set before the supply is connected. Any attempt to run the unit from a 220V supply with the switch set to 110V is liable to result in severe damage to the unit.
16. **Signal Input** is made via a female XLR type connector.
17. **Signal Output** is available on a male XLR type connector. For wiring details see page 15 of this manual.
18. **External Side Chain Input** is made via a 1/4 inch type A stereo jack connector.
19. Always quote the **Serial Number** in any correspondence concerning the unit.

## Reliability Control

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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

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Security cover  
Transformer input\*/output balancing

\* input transformer balancing is non-retrofittable and has to be specified with order.

## Options Ordering Information

## Parts Number

Perspex security cover	SCP..... Model No.
Aluminium security cover	SCA..... Model No.
Output balancing transformer *	BU37
Input balancing transformer	BN37

## Using the DN504 Quad Compressor/Limiter

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The natural sounds of life have an extremely wide dynamic range, from the rustle of a falling leaf to the roar of a jet engine on take off. The human ear has an automatic gain control which enables it to accommodate all of these sounds from the threshold of hearing to close to the threshold of pain, a dynamic range of approximately 120 decibels.

Even the most modern audio equipment is incapable of handling the full range that the ear can cope with. Analogue tape without noise reduction can manage almost 70 decibels dynamic range between its noise floor and the 3% distortion point. 16 bit digital audio equipment can achieve over 90dB. Still almost 30dB less than the ear's range.

Even if a 120dB dynamic range were possible in audio equipment, would it be desirable – and useful? A listener in a domestic setting enjoying the exhilarating effects of a 96dB Sound Pressure Level will almost certainly be causing his neighbour a significant amount of annoyance, if not distress! At the other end of the dynamic scale, a typical ambient noise level of at least 40dB SPL precludes the use of very quiet levels in recorded or broadcast sound media.

Almost always, it is necessary to compress the dynamic range of natural sounds to fit them into a window suitable both for the equipment and for comfortable listening.

The Compressor/Limiter is a valuable tool for the control of dynamic range. Compression and Limiting reduce and control the dynamic range of any instrument or programme source.

Compression and Limiting have their artistic uses too. The sounds of instruments and voices can be altered. A mix of instruments can be compressed to give a 'tighter' dynamic effect. Or individual voices or instruments can be treated to give a musical effect unobtainable by any other means.

The DN504 Quad Compressor/Limiter offers in a compact unit versatile control over dynamic range, for corrective and for artistic purposes.

## Use of Compression

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One of the principal uses of compression is the control of level in vocals. Many singers train for years to achieve the degree of breath control necessary for an even tone and expressive performance. Other vocalists rely on an instinctive voice production technique, which may need help in the studio to maintain a consistent level, and result in a vocal track which 'sits' correctly in the mix.

The level of a vocal may vary widely, and appear like the unprocessed signal in the diagram:

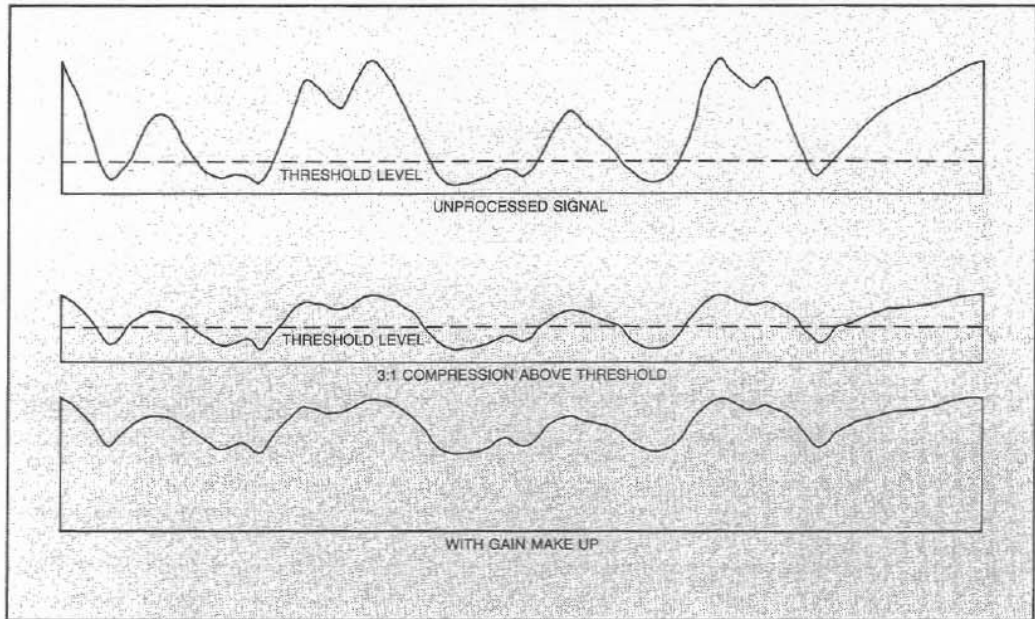


Diagram 1

The unprocessed signal has a large dynamic range between the highest and lowest levels. Applying compression reduces the highest levels, reducing the dynamic range. Because the peak level of the signal is now lower, make-up gain is added to restore the original peak level. The result is a much more controlled and useable sound.

### Interface with the console

The Klark Teknik DN504 Quad Compressor/Limiter is optimised for use at line level, therefore to process the signal from a microphone, the input to the DN504 has to be taken from the console – preferably from the channel insert point send. The output from the DN504 comes back to the channel insert return. By connecting the DN504 at this position in the signal chain, its operation is unaffected by the use of any of the console controls, except Input Gain.

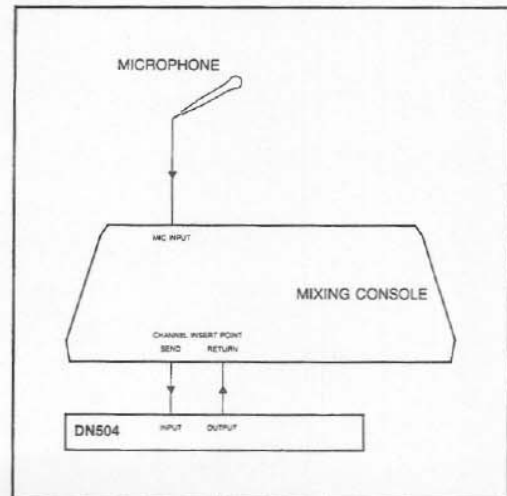


Diagram 2

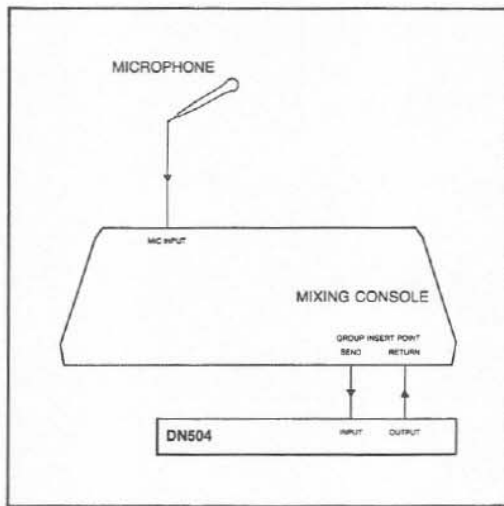


Diagram 3

An alternative is connection to the group insert point of the console:

This connection has two uses: The input from the microphone may be compressed post-EQ, which offers an alternative sound quality which may be desirable in some cases. Alternatively, several instruments may be compressed together in the mix to achieve a 'harder' sound.

### Setting the Controls

**Threshold** sets the level above which compression takes place. Signals below the threshold will remain unaltered. Turning the control clockwise lowers the threshold level and allows more of the signal to be compressed.\*

**Ratio** is the 'strength' of compression above the threshold level. The higher the ratio, the greater the effect.

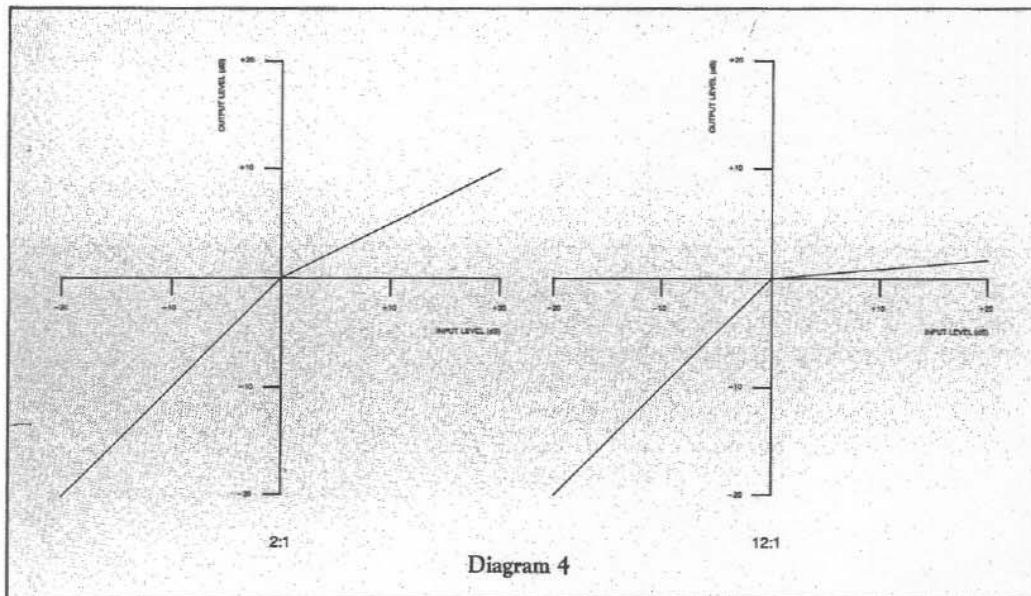
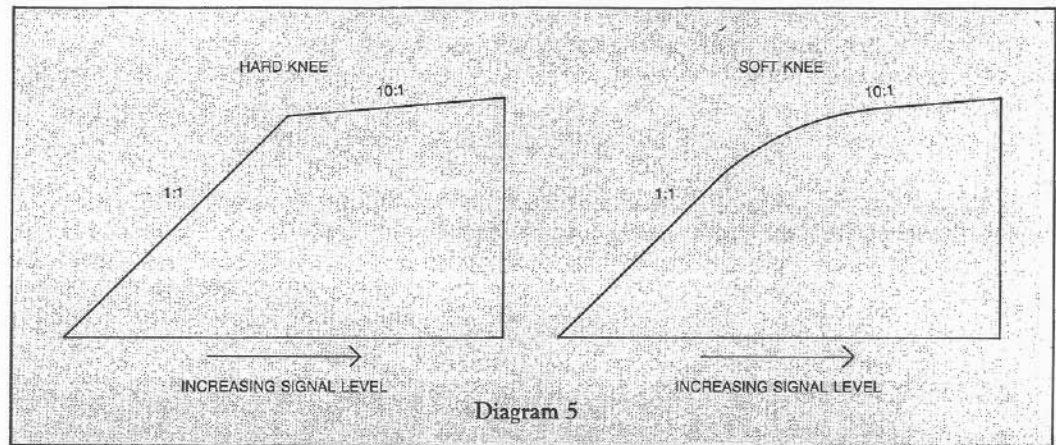


Diagram 4

At a compression ratio of 2:1, the effect is mild and suitable for the subtle compression of vocals or of a complete mix. At 12:1, compression is becoming stronger and more noticeable. Ratios between 3:1 and 15:1 are suitable for the 'compressor' sound, used as an effect in its own right. Higher ratios are used for the control of extremely peaky signals.



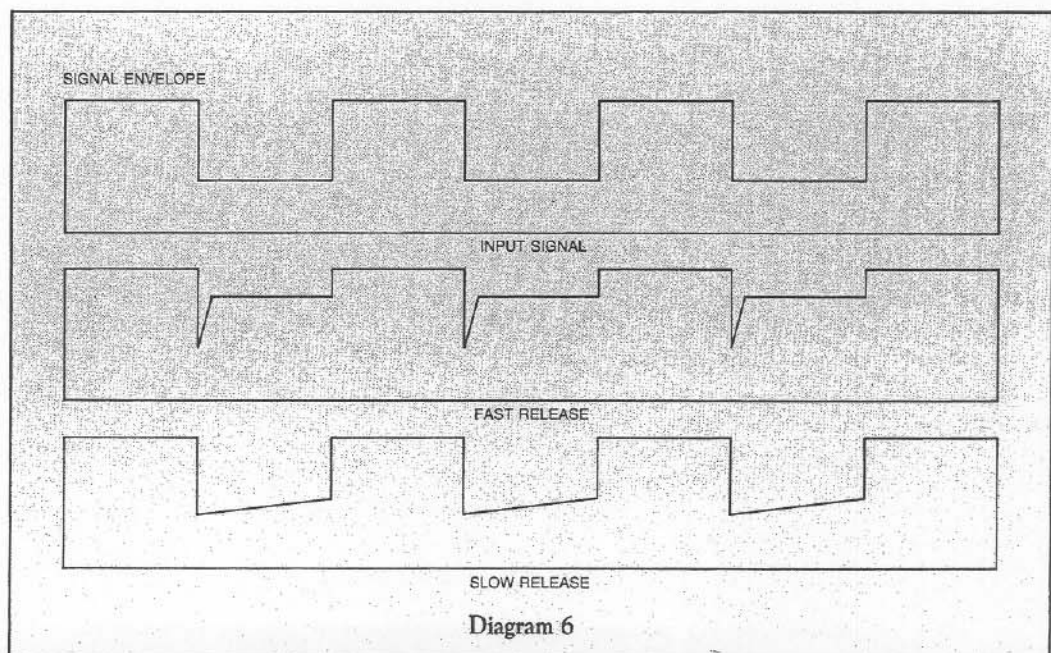
The point where the slope of the Compressor curve changes is known as the **Knee**. The DN504 has an adjustable knee, switchable between 'Hard' and 'Soft':



With a soft knee, signals which only just exceed the threshold level are compressed at a low ratio, the ratio increasing the higher the signal level.

**Attack** sets the time the Compressor takes to respond once the threshold has been exceeded. Attack may be set so that the initial transient of the instrument passes through unaltered, or set to a faster value so that the very start of the sound is compressed. Particularly with drum sounds, careful adjustment of attack time can make the sound more 'punchy' and 'driving'.

**Release** time plays a very important role in compression. During periods of high signal level, gain is reduced. When the signal level falls below the threshold, the gain will increase at a rate determined by the Release control. If the release time is short, the gain will rise quickly. A long release time will mean that the gain will stay at its reduced level, only recovering gradually:



The setting of the correct release time is a compromise. If the release time is too short, background noise can cause effects often known as 'breathing' and 'pumping'. If the release time is too long, the signal will not be compressed, but simply reduced in level. For effective compression, the release time must be set to as short a value as possible before modulation of the background noise becomes too noticeable. The gain reduction bar graph meter will show how much actual compression is going on. If it stays steady, there is little active compression, just a steady-state reduction in level. The faster the bar graph moves up and down, the harder the Compressor is working.

For a natural unnoticeable compression, attack and release times may be set to Auto. The signal characteristics are continuously monitored for optimum values.

## Limiting

By switching any channel of the DN504 to Auto Attack/Release, and setting the compression ratio to a high value – greater than 20:1 – the DN504 acts as a high quality Quad Limiter.

A Limiter acts as a last check on signal level. If the level goes over the threshold, fast-acting high-ratio compression is applied to bring it back within bounds.

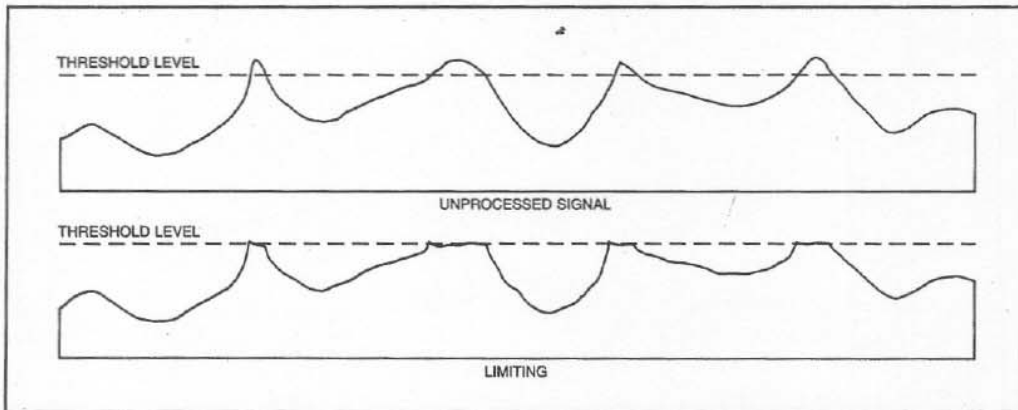
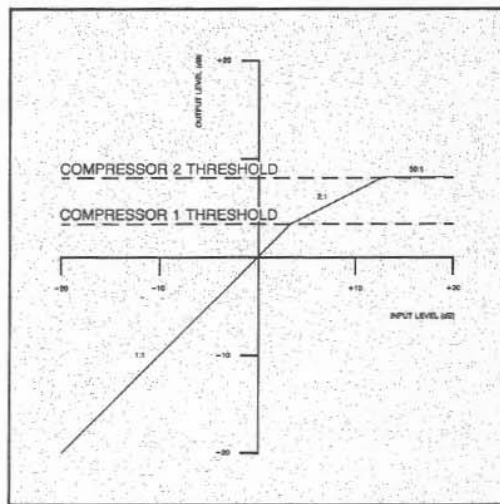


Diagram 7



If a compression effect is required together with limiting of high level transients, two channels of the DN504 may be cascaded. The output of Channel 1 – compression – is fed to the input of Channel 2 – limiting. This gives powerful two stage control over gain:

Diagram 8

Below the compression threshold, the signal is unaltered. In this example, above the first threshold, it is compressed at a mild 2:1 ratio. Above the second threshold it is firmly limited at a ratio of 50:1.

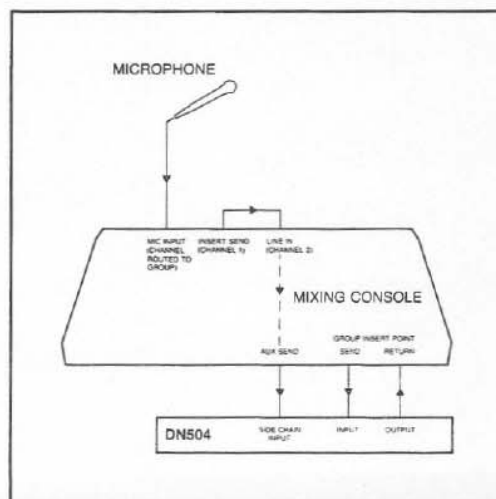
### Applications Using the Side Chain Inputs

Each channel of the DN504 has a **Side Chain** input, in addition to the normal audio signal input.

In normal use, the amount of compression or limiting is related to the dynamics of the input signal. The Side Chain allows the signal passing through the unit to be controlled by the dynamics of another separate signal.

Connection to the Side Chain input is made via the rear panel jack sockets. See 'Audio Connections', page 16.

### De-Essing



De-essing is an important compression technique using the Side Chain. Many singers have high level sibilants – 'sss' sounds – which detract from their performance. Equalising the signal will reduce the sibilants, but also make the overall vocal sound dull.

The sibilants can be selectively removed by compressing only when there is an excessive level of high frequencies. Here is one way to do it:

Diagram 9

The microphone channel is routed to a group with the Compressor patched into the group insert points. The microphone channel is also paralleled into another channel via the line input. The signal in the second channel is equalised so that high frequencies in the sibilant range are boosted. This channel is fed via an auxiliary output to the Side Chain input.

Now, the Compressor will react whenever there is a sibilant, reducing the gain for the duration of the sibilant and cleaning up the vocal sound.

This technique can also be used to compensate for a 'boomy' bass, or other situations where a band of frequencies is occasionally obtrusive.

## Ducking

If speech is to be mixed with music from a stereo source, it is best if the music is dropped in level during the speech, and raised during pauses. This can be done automatically by passing the music through Channels 1 and 2 of the DN504 and patching the microphone into the Side Chain input of Channel 1:

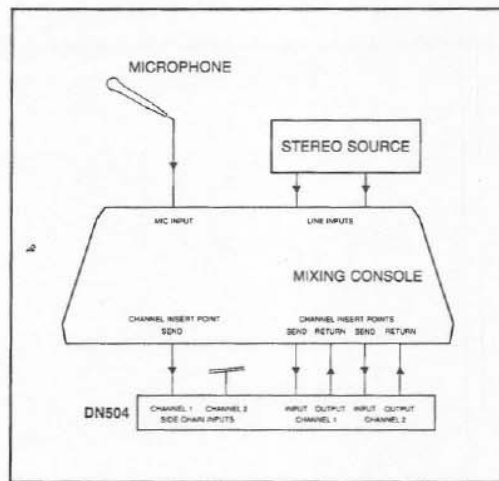


Diagram 10

Since stereo music is to be controlled by an input into the Side Chain of just one channel of the DN504, the Stereo link button must be pressed. To prevent the music input to Channel 2 affecting the degree of compression, an open circuit jack plug must be inserted in the Side Chain input of Channel 2 to break away the jack socket's switch connection.

Now whenever the voice-over artist speaks, the music will be reduced in level by the compressor. When he is silent, the music will rise back in level to take up the full available dynamic range.

Ducking can be accomplished with even greater flexibility using another Klark Teknik product, the DN510 Advanced Dual Gate.

The Klark Teknik DN504 Quad Compressor/Limiter packs a lot of processing power into a small rack space. Engineers will find it convenient to use and will also enjoy its creative potential.

## Audio Connections

### Input

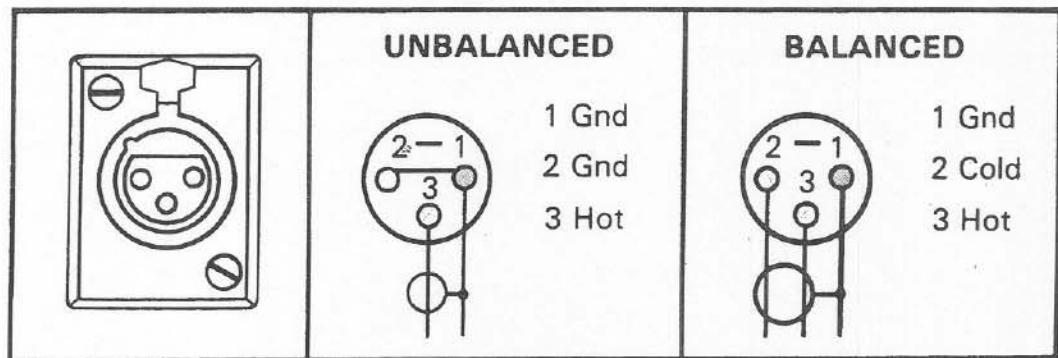
The input circuit is a transformerless, electronically balanced design which achieves a symmetry of better than  $-50\text{dB}$  from  $20\text{Hz}$  to  $10\text{kHz}$ .

If transformer balancing of the input is required, this must be specified at the time of order. It is not retrofittable.

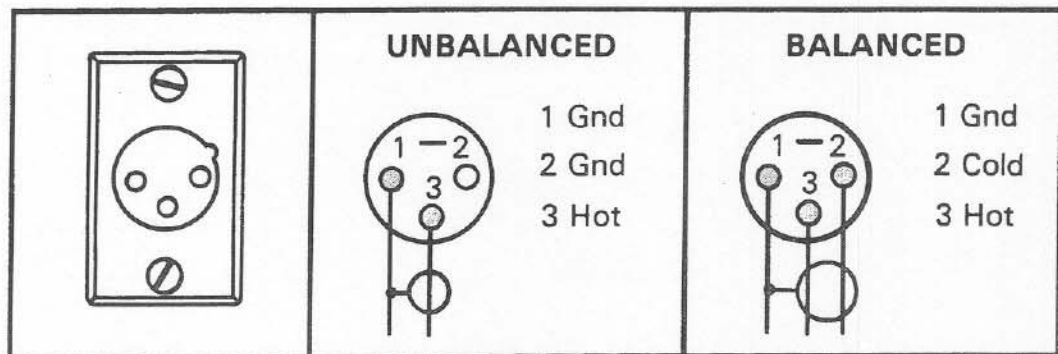
### Output

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  $+22\text{dBu}$ .

### Input



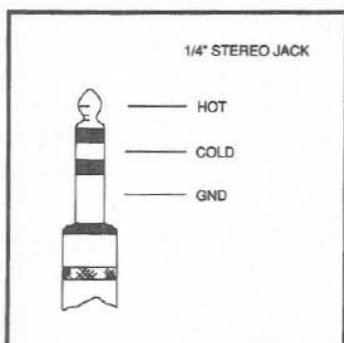
### Output



### Pin 2/Pin 3 Hot Operation

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The unit can be quickly re-configured to accommodate either XLR wiring standard by removing the top cover and changing the orientation of 4 plug-in links per connector, as shown on the adjacent PCB legend. When using a fully balanced system, either pin 2 or pin 3 may be the HOT terminal.



### Side Chain Inputs

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The electronically balanced Side Chain inputs are on stereo quarter inch type A jack sockets, configured TIP = HOT, RING = COLD, SLEEVE = Earth. Inserting a mono jack plug will automatically unbalance the input.

The sockets are internally normalled so that signal continuity is maintained when there is no jack plug inserted. If the unit is wired to a patchbay then normalling must be carried out at the patchbay.

### Balanced Circuits

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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 earth (ground) or screen being totally isolated from the signal. In installations where a difference in earth potential is likely to occur, this isolation prevents earthing problems which can, in some cases, damage the equipment.

## DN504 Technical Specifications

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<b>Audio inputs</b>	<b>Four</b>
Type	Electronically balanced
Impedance (ohms)	
balanced	20k
unbalanced	10k
<b>Side chain inputs</b>	<b>Four</b>
Type	Electronically balanced
Impedance (ohms)	
balanced	20k
unbalanced	10k
<b>Audio outputs</b>	<b>Four</b>
Type	Unbalanced
Minimum load impedance	600 ohms
Source impedance	<60 ohms
Maximum level	+21dBu
<b>Performance</b>	
Frequency response (20Hz – 20kHz)	±0.5dB
Distortion (at +4dBu)	<0.03% at 1kHz
Equivalent input noise (20Hz – 20kHz unweighted)	<-94dBu
Channel separation	>90dB at 1kHz
<b>Compressor</b>	
Threshold	-30dB to +20dB
Ratio	1:1 to 50:1
Knee	Switchable, 1dB (hard)/40dB (soft)
Envelope	Switchable auto (attack and release controls disabled) or manual
Attack (90% capture)	0.05mS to 20mS
Release (90% recovery)	60mS to 2S
Output gain	-10dB to +30dB
<b>Power requirements</b>	
Voltage	110/120/220/240V, 50/60Hz
Consumption	<30VA
<b>Weight</b>	
Nett	4kg
Shipping	6kg
<b>Dimensions</b>	
Width	482mm (19 inch)
Depth	292mm (11½ inch)
Height	44.5mm (1¾ inch)

**Terminations**

Audio inputs/outputs

Side chain inputs

Power

3 pin XLR

Normalised ¼ inch stereo jack

3 pin IEC

**Options**

Security cover

Transformer input/output balancing\*

\* Input transformer balancing is non-retrofitable and has to be specified with order.



## DN504 Circuit Description

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(Note: only Channel 1 is described, other channels are identical except where stated.)

### Audio Circuits

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#### Input Section

The input signal enters via XLR type connector XL1 and may be balanced or unbalanced (see audio connector wiring details).

The input is then fed through the Hot-Pin jumper block which allows for either Pin 2 or 3 being selected as the in phase or active pin on the input.

The signal is then connected via the first order filter network R2/C1 and R7/C2 to the differential input amplifier circuit consisting of A1a and A1b.

Common mode rejection is factory set with P1 to give maximum rejection at 100Hz; typically 85dB.

#### VCA Section

The output from A1b passes through C10 and the IN switch SW1c to the first order LF and HF filter networks C19/R29 and C21/R28 and then to the VCA input buffer A4b. From there it is AC coupled to the DBX VCA A5 via C22 and to the Side Chain input jack JK1 via C20.

Preset P2 is factory set to give minimum distortion at 1kHz and +10dBm, (typically 0.003%), this adjustment also reduces control feed-through to a minimum.

#### Output Section

The VCA output is fed to the current to voltage converter A4a which has a high current drive capability. High frequency stability is optimised with capacitor C41.

The output of A4a is AC coupled by C24 to the IN switch SW1d.

From there the signal is connected via C11 to the output transformer socket TC1.

On the standard unit two plug-in links on TC1 connect signal and ground to pins 2 and 3 of the output XLR connector XL2 via the Hot-Pin jumper block.

On units with transformer balanced output the two links are removed and the transformer is connected to TC1.

#### Side Chain Input

The Side Chain input signal enters via stereo jack connector JK1 and may be balanced or unbalanced (see audio connector wiring details).

The signals then pass to the differential input amplifier A2 which has first order R.F. filter networks formed by R17/C6 and R18/C7.

From A2 output the signal is AC coupled to the IN switch element SW1b by C8.

When no jack is inserted into JK1 the feed from the VCA input buffer A4b is fed via the break contact on JK1 to the Side Chain input circuit.

#### Side Chain Mixers

On channels 1 and 3 the signal from the IN switch element SW1b is fed through AC coupling capacitor C15 to the inverting mixer circuit of A3.

This mixer circuit has a loss of 6dB and high frequency stability is ensured by C17.

The output of the mixer circuit is fed via C18 to the Compressor THRESHold control V1.

On channels 2 and 4 this signal passes through the STEREO LINK switch elements SW2b instead of a mixer circuit.

This arrangement causes channel 1 and 2, or channel 3 and 4, audio signals to be mixed together and fed to channel 1 or 3 control circuits when the sections are Linked in Stereo.

### Side Chain Control Circuits

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The Compressor side chain is a feed forward control circuit allowing much more versatility of control and eliminating the overshoot problems associated with feedback circuits.

Inserting effects units into the side chain is also more straightforward with this system, although the control circuits themselves are necessarily more complicated as a result.

### Threshold Control Amplifier

The output from the wiper of the Threshold control V1 is AC coupled via C25 to the positive amplifier A6b.

The gain of A6b is changed by the AUTO switch SW3b, being 10dB in the manual position and 13dB in AUTO.

These gains are for channel 2 and 4 circuits, channels 1 and 3 having 6dB more gain in both positions to allow for the loss in the Side Chain Mixer circuit on those channels. This stage allows continuously variable selection of threshold level down to -30dBm.

### AVA and RMS section

The output of A6b is fed via C27 to the Absolute Value Amplifier and Root Mean Square circuit formed by A6a, A7a and A7b, A8a and A8b, and transistor array Q1.

This circuit achieves its RMS action using the log relationship between a transistor's base emitter voltage and its collector current.

The transistors in the array Q1 along with their associated amplifiers form log and antilog generators which are arranged to square the input voltage, average it and then the output is fed back into the calculation to perform the square root function.

This circuit is changed between its two modes of operation by switching C31, the RMS averaging capacitor, either in or out of circuit with one pole of the AUTO switch SW3c. Capacitors C28, C29, C30 and resistor R49 are included to ensure stability, eliminating the need for any adjustments.

### Ideal Diode section

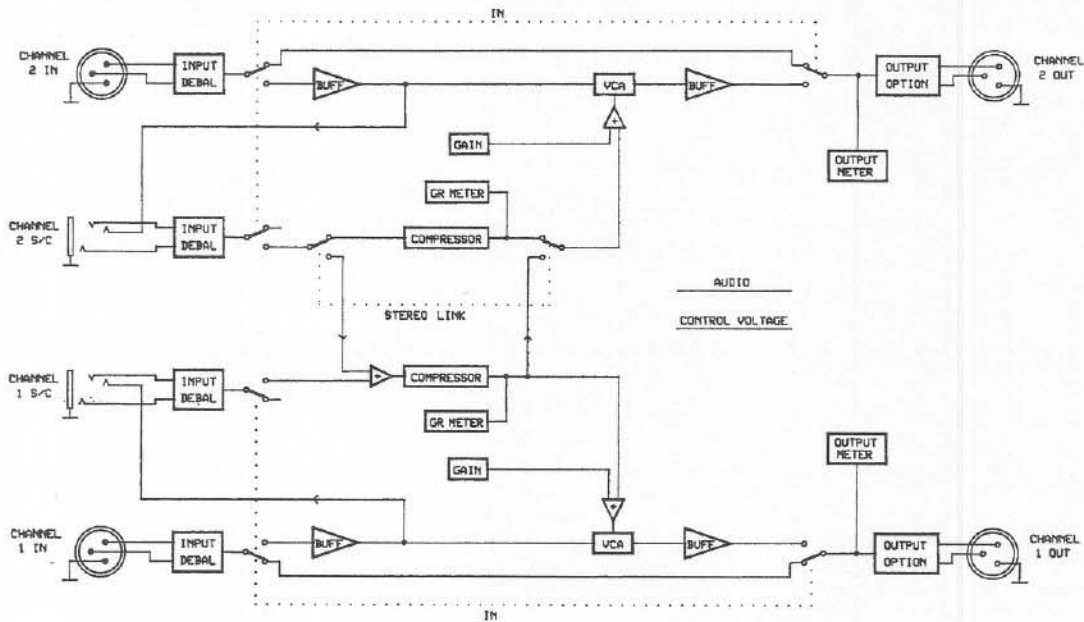
The output of A8b is directly connected to A14 which is arranged with diode D6 to form an ideal diode circuit to pass the output voltage from the RMS section accurately to the following stages.

The output from A14 is fed via a filter/storage network formed by R54 and C32 to limit the high frequencies and voltage swing reaching the next stage.

### Logger section

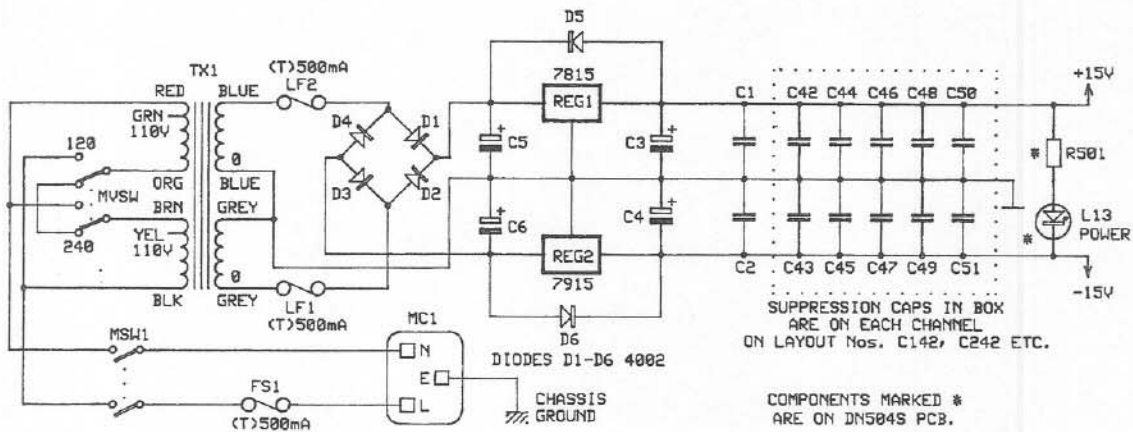
The output of C32 is connected to the logger circuit formed by A10a, A10b and transistor array Q2.

The operation of the logger circuit uses the same principles as the RMS section already described to produce an output voltage which is the log of the input voltage.

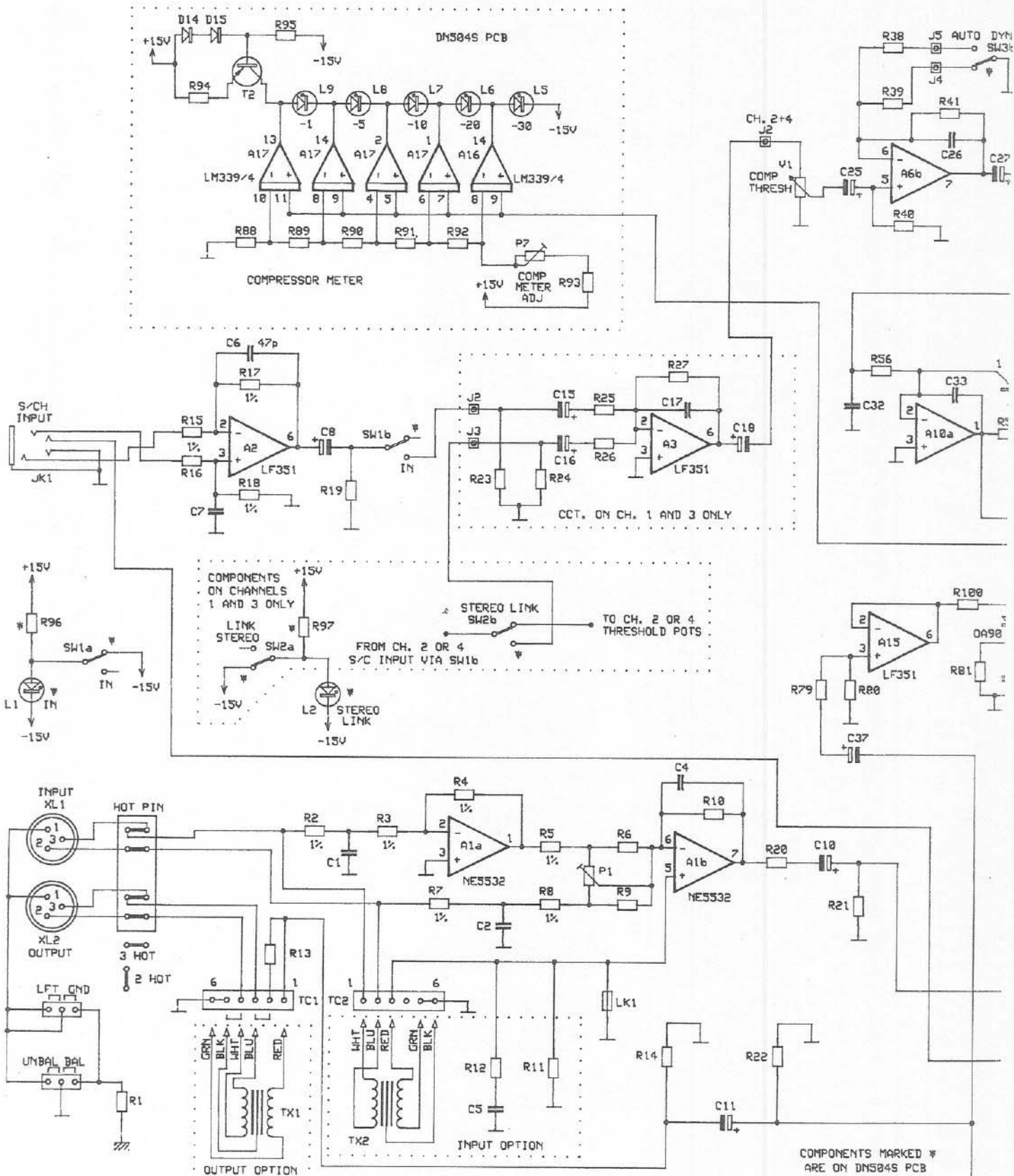


CHANNELS 1 AND 2 ONLY SHOWN.  
CHANNELS 3 AND 4 IDENTICAL.

DN504	BLOCK DIAGRAM	31.12.88
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DN504	POWER SUPPLY SECTION	BRD: DN504M 31.12.88
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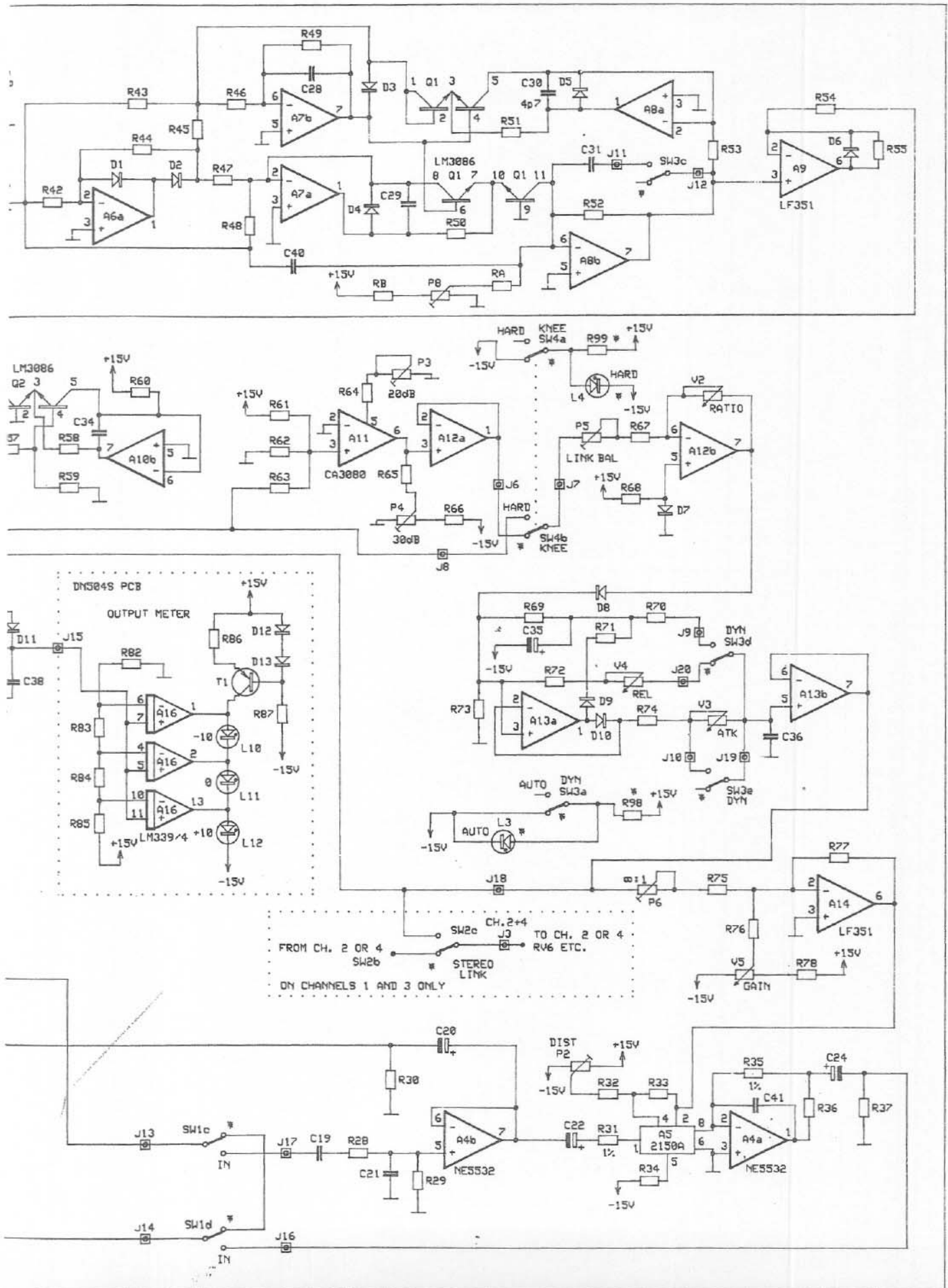
UNLESS OTHERWISE MARKED  
ALL TRANSISTORS BC327  
ALL DIODES 1N4148  
ALL ICs. LF353N

TRANSFORMER BALANCED INPUT OPTION	TRANSFORMER BALANCED OUTPUT OPTION
FACTORY OPTION TO SPECIAL ORDER ONLY COMPONENTS REMOVED: LK1, Rv1, R2 - R10, C1, C2 REPLACED WITH LINKS: R4, R10 COMPONENTS ADDED: R11, R12, C5	REMOVE THE TWO LINKS ON TC1 PLUG TRANSFORMER INTO TC1 SECURE TRANSFORMER WITH THE SCREW AND WASHER PROVIDED CHANGE THE LINK ON SELECTOR TO 'BAL' POSITION

COMPONENTS MARKED #  
ARE ON DNS04S PCB

CIRCUIT FOR ONE CHANNEL ONLY  
OTHERS SAME EXCEPT WHERE MARKED

COMPONENT IDENTS ON LAYOUT  
ARE NUMBERED  
FROM 101 FOR CHANNEL 1,  
AND 201 FOR CHANNEL 2, ETC.  
EXCEPT CONNECTORS, TRANSFORMERS,  
SWITCHES AND LEDS WHICH  
ARE PREFIXED BY CHANNEL No.



DN504

COMPLETE CHANNEL CCT.

BRD: DN504M

28.2.89

This is necessary to ensure that the VCA sees a voltage which corresponds correctly to the input signal since the VCA works on a dBs per volt, or log, law. Capacitors C33 and C34 are included for stability.

#### **Knee section**

The Knee circuit consists of a transconductance amplifier A11 and its current to voltage converter A12a, with presets P3 (20dB) and P4 (30dB) allowing for differences between particular ICs.

The circuit uses the well defined input compression characteristics of the transconductance amp to produce the desired voltage curve for the soft knee.

The HARD switch SW4b selects either the input voltage to the knee section or the output of the knee circuit.

#### **Ratio section**

The RATIO control V2 acts as a feedback element around A12b creating a variable gain circuit with a maximum gain of unity, adjusted accurately by P5 (Link Bal), and a minimum gain of around -70dB.

This circuit determines the relationship between the number of dBs change in input level compared to the number of dBs change in output level, i.e. the Ratio.

#### **Dynamics section**

The output from A12b feeds via D8 to A13a, an ideal diode circuit driving the ATTACK control V3 and the Automatic Release Circuit.

One or the other of these, dependent on the position of the AUTO switch, charge the main storage capacitor C36.

The ATTACK and RELEase controls V3 and V4 are switched in or out of circuit by the AUTO switch elements SW3e and SW3d.

The ATTACK control is shorted out when in AUTO mode to stop the RMS attack time constant being interfered with, the fast attack limit in this mode is set by R74.

The RELEase control is completely switched out when in AUTO mode and the Automatic Release Circuit consisting of C35, D9, R69, R70 and R71 is switched in.

The ARC varies the release time in relation to signal burst length, increasing the release time with longer burst lengths, to achieve an almost inaudible release effect due to the masking effects produced by the ear.

#### **Voltage Buffer**

The voltage on the main storage capacitor C36 is buffered by the high impedance voltage follower A13b, and the output of this stage is fed to the VCA driver A14.

#### **VCA Control Voltage circuit**

DC amplifier A14 serves to mix the output of the Compressor control circuit with the feed from the GAIN control V5 into a single voltage and reduce the output voltages of these sections down to the fraction of a volt swing required by the VCA for correct audio attenuation or gain.

Preset P6 (Infinity : 1) is used to accurately match the Compressor maximum Ratio setting to the particular VCA used, and is adjusted for constant output level at Limit.

## Meter circuits

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Both Compressor and Output Meter sections are almost identical and use the same basic constant current LED drive system and comparator ICs to switch the relevant LEDs on and off.

Only the Output Meter section is described below along with any differences in the Compressor meter.

Transistor T1, diodes D12, D13 and resistors R86, R87 form a constant current generator producing approximately 6 milliamps of current flow into the LED diode chain which is wired to the negative supply rail.

This stops the meter circuit generating any interference on the power supply rails when the LEDs are switched on or off.

The top of each LED in the chain is connected to one output of a quad comparator A16, the positive inputs of which are all joined to the input voltage on the storage capacitor C38.

The negative inputs of the comparators are wired to the outputs of a voltage attenuator chain consisting of R82 to R85 which is connected between +15 volts and 0 volts.

This chain sets the voltages at which each comparator will switch off, the open collector output allowing its associated LED to illuminate.

The only difference in the Compressor circuit is the inclusion of preset P7 (COMP METER ADJ) which adjusts the attenuator voltages in order to calibrate the section to match the front panel markings.

The Output Meter section also has an AC to DC converter input stage which is AC coupled by C37 from the IN switch element SW1d.

This stage consists of a unity gain buffer A15 driving germanium diode D11 which in turn charges a storage capacitor C38.

Resistors R81 and R100 give this stage Attack and Release time constants similar to a PPM meter.

The Compressor Meter circuit is driven directly from the output of A13b, the Voltage Buffer.

## Power Supply

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The power supply is a  $\pm 15V$  design using a low noise toroidal transformer with split primary and secondary windings.

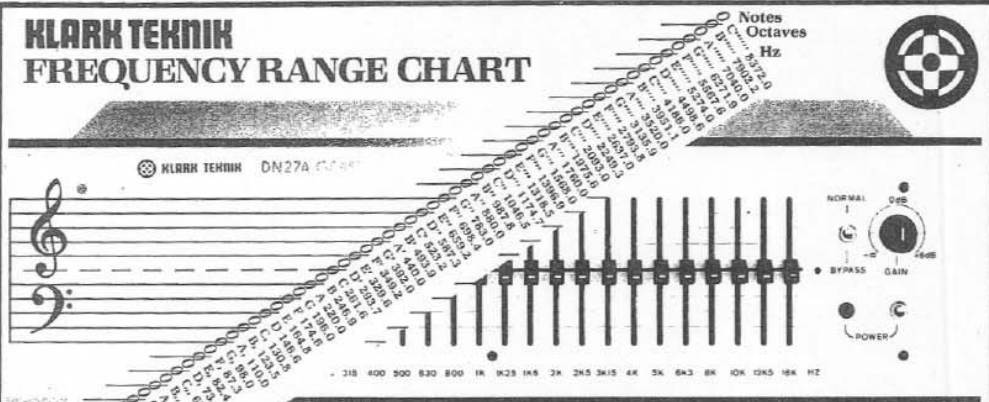
The two primaries are connected in series or in parallel by MVSW to give 220 or 110 Volt nominal operation.

The secondaries drive a full wave bridge rectifier, D1 to D4, two smoothing capacitors, C5 and C6, and integrated circuit positive and negative regulators REG1 and REG2.

Decoupling capacitors C1 and C2 at the power supply and C42 to C51 in each channel ensure low noise supply rails.

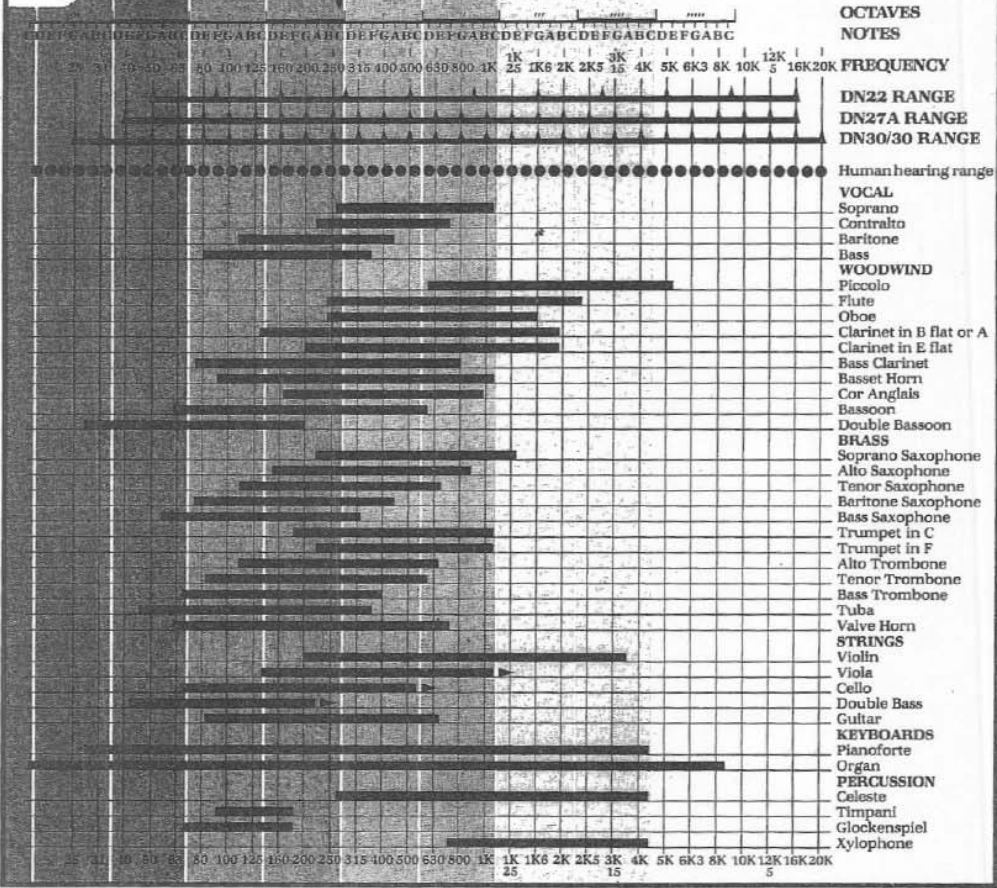
Power 'on' is indicated by LED L13 which is connected between the +15V and -15V rails via R501.

# KLARK TEKNIK FREQUENCY RANGE CHART



### NOTES RELATED TO FREQUENCIES

This chart shows (above) the musical notes and octaves related to their actual frequencies. The details are cross referenced (below) to the frequency range of our Graphic Equalisers and the range capabilities of common musical instruments.



Frequency Range Chart