Operating and service information

DN332

Graphic equaliser

Audio engineering
like no other
in the world

Introduction

The graphic equaliser is a vital component in any audio system. The entire signal passes through it and so any limitations imposed by the equaliser will compromise the performance of the whole system. For example, an indifferently designed equaliser may introduce severe phase distortion, noise and other anomalies related to centre frequency accuracy, filter shape and attenuation accuracy which may manifest themselves as an overall deterioration in the perceived sound quality of the system. Clearly this is an unacceptable state of affairs, but fortunately your choice to utilise Klark-Teknik graphic equaliser product will eliminate these problems, offering you unprecedented product performance coupled with the highest filter calibration and reliability standards in the industry.

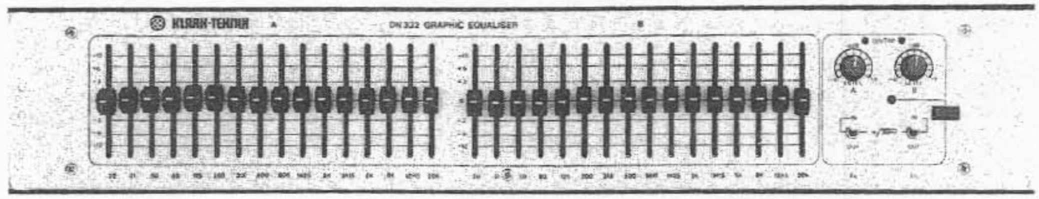
For many years Klark-Teknik has been at the forefront of equaliser design, and have carried out detailed research into optimum filter response characteristics, including their sonic performance.

The Series 300 range of equalisers is a direct result of this research. It should be noted that graphic equalisation cannot always overcome all frequency response related problems. There are applications where the ability to cut or boost the response at a particular frequency, or over a certain bandwidth other than the equaliser specified one, is required to overcome exceptionally difficult response anomalies or narrow band feedback problems. When such an instance is encountered, it may be more appropriate to use the greater range of control provided by a parametric type equaliser, where the centre frequency, bandwidth and amplitude are all controllable.

Reliability is also of paramount importance which is why our filters are designed around a technique commonly used in computer manufacturing - thick film engineering. This technique has enabled Klark-Teknik to build these new filter circuits into self-contained packages which are referred to as **"MELT"**. These micro-electronic circuits are so consistent and reliable that we are able to warrant those solid state devices for 5 years. This type of "fit and forget" technology, already proven all over the world, provides users with products that perform brilliantly year after year.

When using an equaliser remember that the need to use large amounts of boost or cut within the equalisation curve indicates that there may be something fundamentally wrong with the sound system or room acoustics, which should be further investigated and corrected before final equalisation is applied.

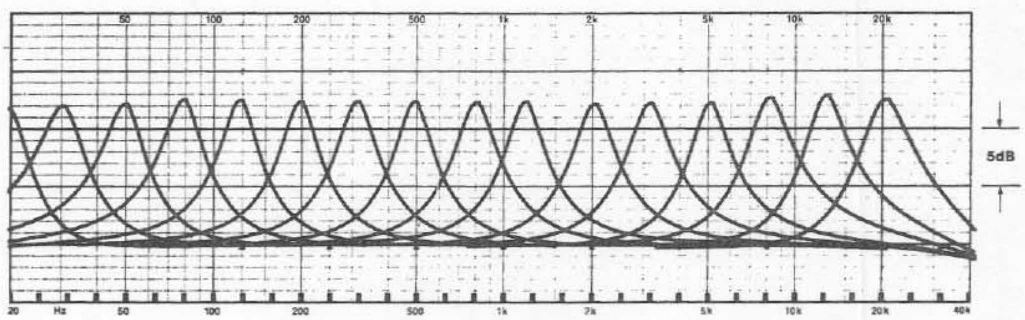
DN332 Graphic Equaliser



The Klark-Teknik DN332 is a dual channel, 16 band equaliser offering 12dB of cut or boost in 2/3 octave steps between the frequencies of 20Hz and 20kHz.

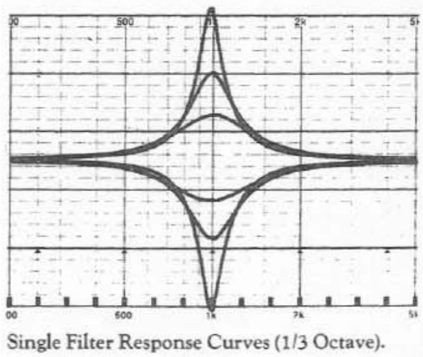
Filter Shape and Combining Action

ISO CENTRE FREQUENCIES (IN HERTZ)															
20	31	50	80	125	200	315	500	800	1.25K	2K	3.15K	5K	8K	12.5K	20K



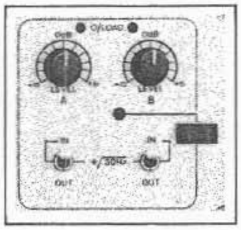
2/3 Octave Band Equaliser Filter Curves.

At the heart of any graphic equaliser is the bank of filters used to shape the signal response, and Klark-Teknik utilise a proprietary filter circuit which replaces the conventional inductor based circuit, at the same time, offering several performance advantages. Inductor based circuits are heavy, expensive to produce and suffer from low frequency distortion and induced hum. Klark-Teknik's proprietary filters on the contrary suffer none of these problems, yet offer unequalled phase response and control accuracy with the additional benefits of low noise and minimal ripple.

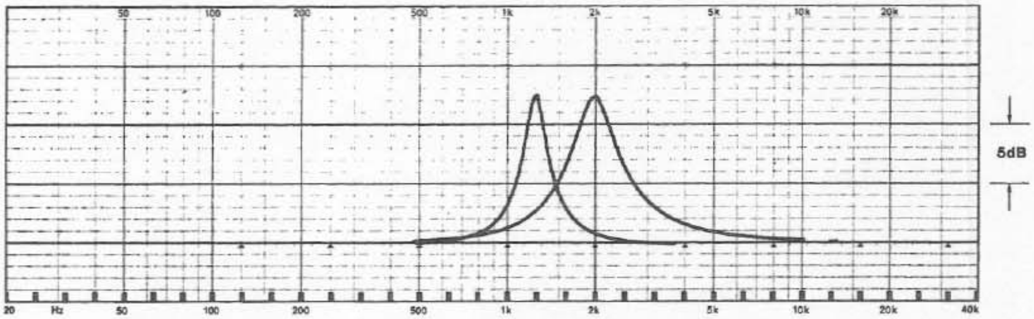


Single Filter Response Curves (1/3 Octave).

The nature, shape and way in which individual equaliser filters combine, has a profound effect on the control provided by the equaliser and on the resulting quality of sound. The majority of applications within the sound reinforcement, broadcast and recording fields, require a smooth and continuous equalisation response curve in order to correctly contour the overall response characteristics of a sound system, loudspeaker, recording effect or audio channel. To achieve this, the individual filters must be capable of combining smoothly together to result in a continuous response curve, free from shape discontinuitities in order to avoid unwanted audible peaks or anomalies in the final sound.



Individual channel level controls, overload indicators and bypass switches with incorporated low cut filters are featured on the control panel.



Comparison Between 1/3 Octave and 2/3 Octave Filter Shapes.

Other Features

The bypass switch silently removes the graphic equaliser sections from the signal path. Incorporated in the bypass switch is a subsonic 18dB/octave roll-off filter (−3dB @ 30Hz).

Other features include an overload LED per channel, which warns of impending overload at any point in the equaliser. A signal ground lift switch and an optional security cover to prevent unauthorised personnel from tampering with the control settings.

This product 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 two units of rack space and has electronically balanced inputs and unbalanced outputs. Output balancing transformers are available and retrofittable.

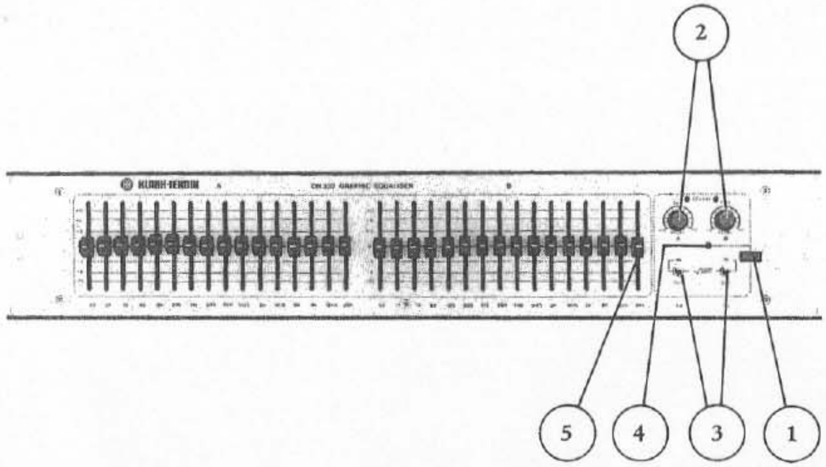
Reliability Control

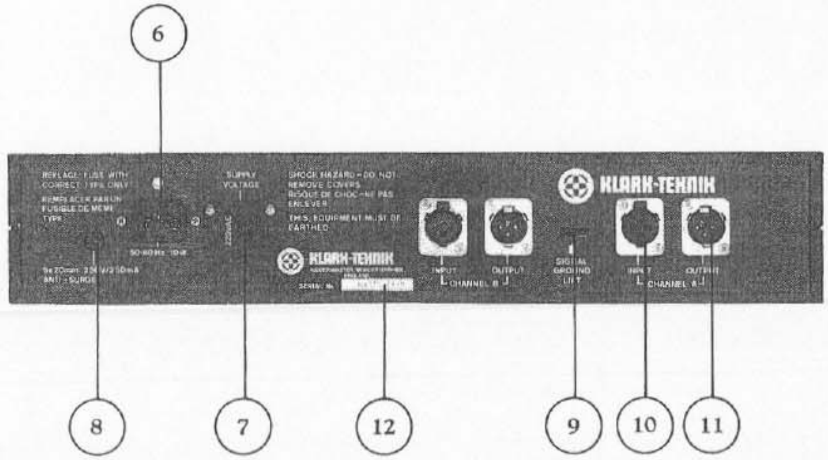
Even with the advanced technology incorporated in this product, each instrument is given the full backing of Klark-Teknik's **"reliability control"** which proves each product against a specification consistent with the highest professional standards. Only top quality components are used, and every unit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

Options

Aluminium security cover
Perspex 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	SC30
Aluminium security cover	SC31
Output balancing transformer	BU37
Input balancing transformer	BN37





Front Panel Functions

1. **The power switch** is a two pole type isolating both the live and neutral conductors. When the power is on, a red status LED lights.
2. **The input level control** allows the system gain to be up to +6dB when in its fully clockwise position, and offers full attenuation in its anti-clockwise position.
3. **The low cut filter** switch enables a 30Hz subsonic filter to be connected in or out of circuit. This switch also performs the **bypass** functions selected as follows:-
   Postions Down: Both the equaliser section and the subsonic filter are removed from the signal path.
   Centre: Both the equaliser section and the subsonic filter are in circuit.
   Up: The subsonic filter is out of circuit.
4. **The Overload LED** The signal level is monitored at several separate points within the circuitry of the unit, and any one of these signals exceeding a threshold, set 3dB below clipping, will cause the LED to light. This threshold is set at +19dB, but it must be remembered that excessive boost of some frequencies combined with a high average input signal, can occassionally cause this level to be exceeded. In this event, the output gain control should be turned down to correct the problem. However, if the input signal itself exceeds +19dBm the input stage will be overloaded. If this problem arises, the signal level from the output of the preceding piece of equipment must be turned down.
5. **The high quality faders** used in this equaliser have an oil-damped action for smooth operation and feature a centre detent allowing accurate "flat" setting.

Rear Panel Functions

6. **Mains** is supplied via an IEC standard 3 pin connector. A compatible power cord is supplied with the unit.
7. **Voltage selector switch** This unit is designed to be switchable between two nominal supply voltages, 110V and 220V. To facilitate this, a slide switch is fitted, accessible from the rear panel. The power supply **MUST** be disconnected before the switch is reset. Note too that any attempt to operate the unit from a 220V supply with the switch set to 110V is liable to result in severe damage to the unit.
8. **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 fuse holder.
9. **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.
10. **Input and output connections** are made via complementary XLR style sockets.
11. For wiring details see page entitled Audio Connections.
12. **The serial number** of this unit should be quoted in any correspondence concerning the unit.

Audio Connections

Input

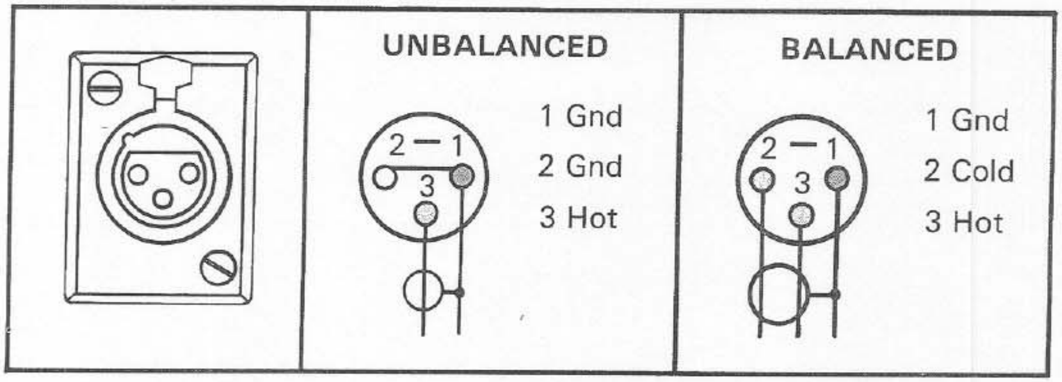
The input circuitry is a transformerless, electronically balanced design which achieves a symmetry of better than −50dB from 20Hz to 10kHz.

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

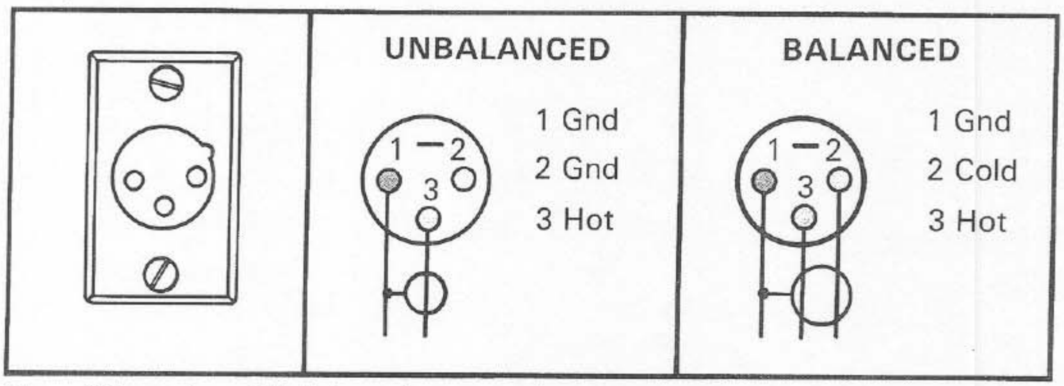
Output

The standard output is unbalanced, but balancing transformers are available and may be retrospectively fitted. The output circuitry is capable of driving a 600 ohm load at a level of +22dBm.

Input



Output



Note: When using a fully balanced system, either pin 2 or pin 3 may be the HOT terminal.

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.

Specifications

Input

Type	Electronically balanced
Impedance (ohm)	
Balanced	20k
Unbalanced	10k

Output

Type	Unbalanced
Min. load impedance	600 ohms
Source impedance	$<$60 ohms
Max. level	+22dBm

Performance

Frequency response	±0.5dB (20Hz - 20kHz)
Distortion (@ +4dBm)	$<$0.01% @ 1kHz
Equivalent input noise	$<-$90dBm (20Hz - 20kHz unweighted)
Channel separation	$>$75dB @ 1kHz
Overload indicator	+19dBu threshold
Level control	+6dB to $-\infty$ (infinity)

Filters

Type	* MELT
Centre frequencies	2 X 16
ISO	20Hz - 20kHz 2/3 octave
Tolerance	±5%
Maximum boost/cut	±12dB
Subsonic filter	18dB/octave — 3dB @ 30Hz

*MELT - Proprietory Microcircuit

Power Requirements

Voltage	110/120/220/240V 50/60Hz
Consumption	$<$15 VA

Weight

Nett	3.5kg
Shipping	6kg

Dimensions

Width	482mm (19 inch)
Depth	205mm (8 inch)
Height	89mm (3½ inch)

Terminations

Input	3 pin XLR
Output	3 pin XLR
Power	3 pin CEE

DN332 Circuit Description

Channel A is described. For channel B, substitute corresponding component numbers.

Input Section

The input signal enters via XLR-type connector SKT2 and may be balanced or unbalanced (see audio connector wiring details). The signal is then connected via first order R.F. filter networks R3/C20 and R8/C21 to the differential input amplifier IC1. Common mode rejection is factory set with P1 to give maximum rejection at 100Hz; typically 85dB. This stage is followed by voltage follower IC2A and the input level control VR33. IC3 is a non-inverting amplifier with a gain of 7dB and is A.C. coupled by C24. D.C. offset of IC3 is adjustable by preset P2.

D.C. Offset Trim

Preset P2 is factory set to give minimum D.C. voltage difference (typically less than 1mV) at test point TP1 when the Eq. switch is switched between 'IN' and 'IN' + 30Hz' positions only.

N.B. not 'out' position.

This avoids the possibility of audible clicks being generated by use of the Eq. switch.

Subsonic Filter

From IC3 the signal passes through an 18dB/octave third order Butterworth high pass filter with −3dB turnover point fixed at 30Hz. Signal from the output of IC2B passes via voltage follower IC2C to the equaliser section.

Equaliser Section

The 16 equaliser filters are connected to IC4 via their respective faders. To demonstrate the working principles of the equaliser section the behaviour of one filter is now examined. The remaining filters perform in a similar manner.

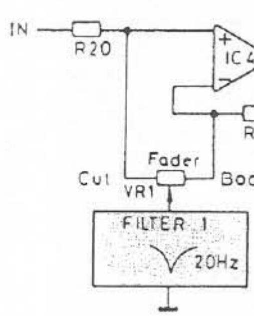


Figure A
Block Diagram

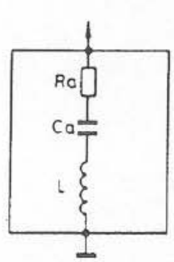


Figure B
Equivalent passive
circuit of filter

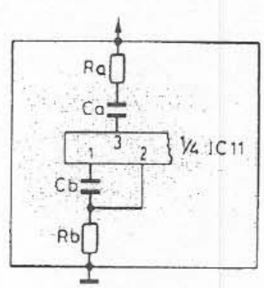
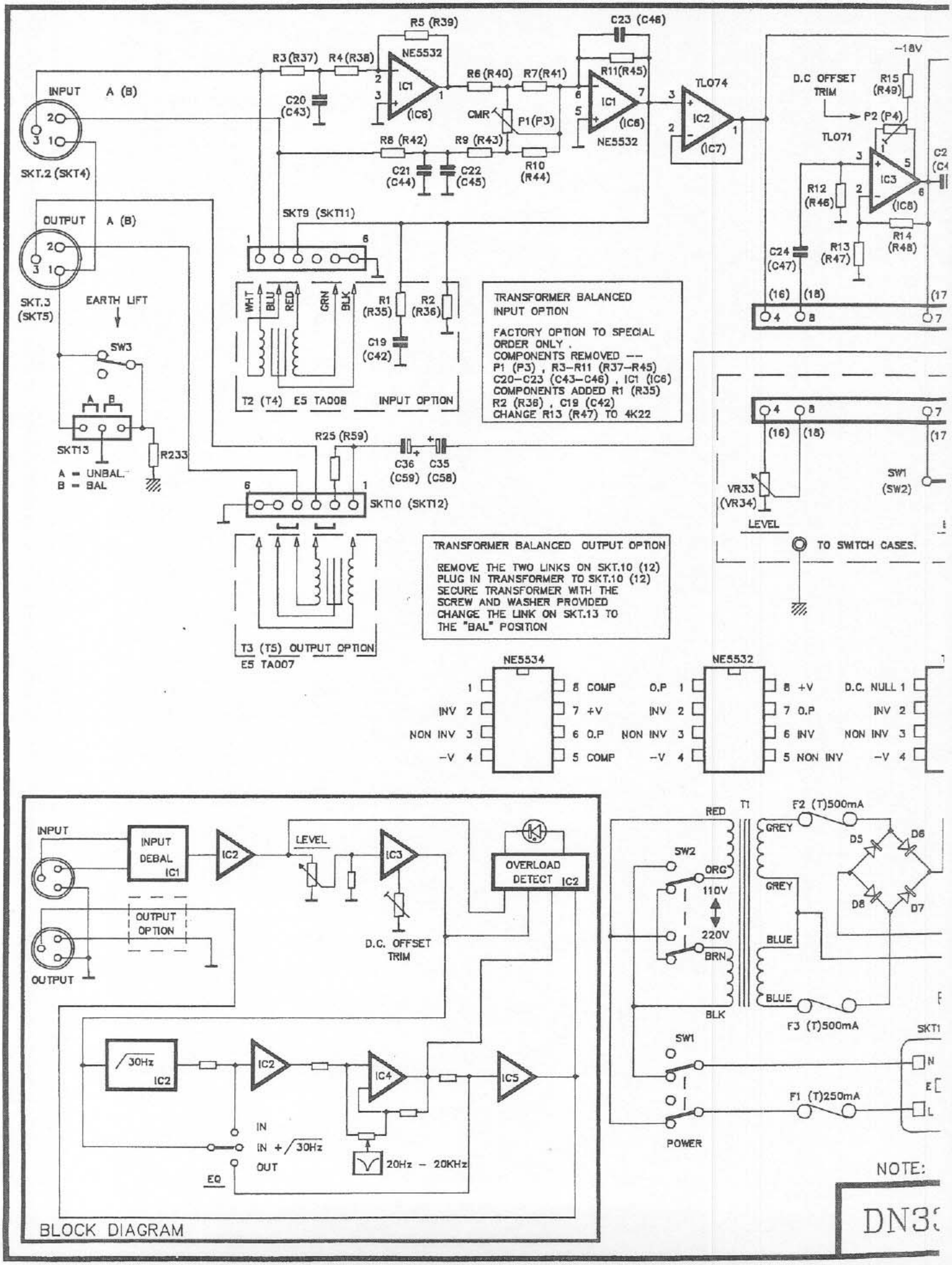
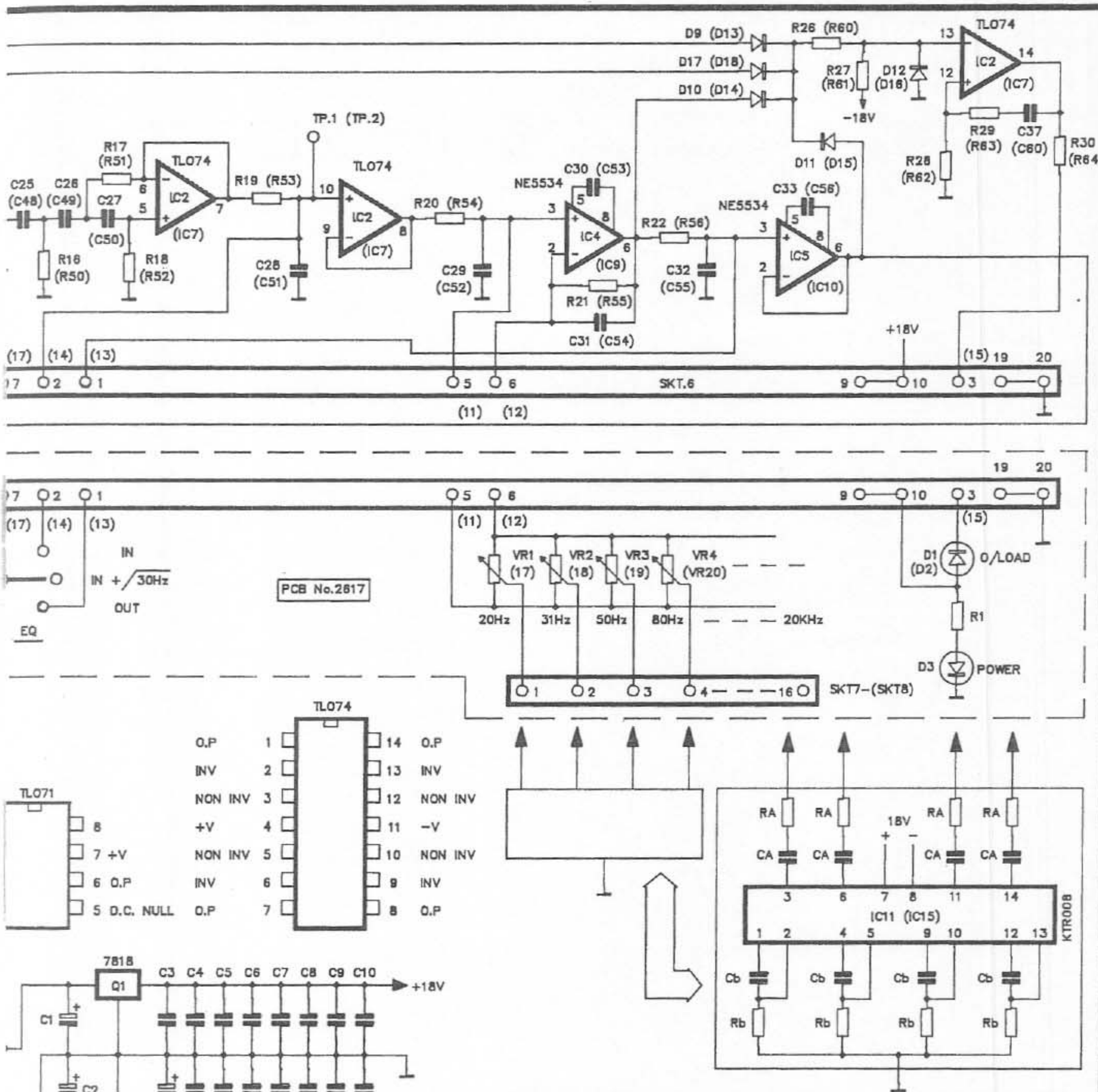
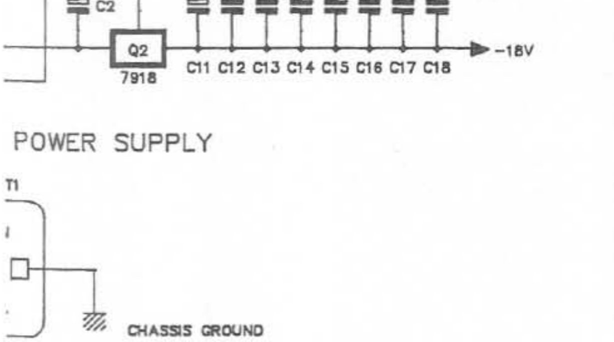


Figure C
Actual circuit of
filter using KTR008

In Figure A the equaliser filter 'block' represents a minimal impedance to ground at its centre frequency, in this case 20Hz. Moving the fader towards 'boost' increases gain in IC4 at the filter frequency only. Moving the fader towards 'cut' causes attenuation through R20 at the filter frequency only. With the fader set to 'flat' (50%), the filter has no effect and since R20 and R21 are equal in value, all frequencies are passed at unity gain.


INPUT A (B)
SKT.2 (SKT4)
OUTPUT A (B)
SKT.3 (SKT5)
EARTH LIFT
SW3
SKT13
A = UNBAL.
B = BAL
R233
R3 (R37)
R4 (R38)
R5 (R39)
NE5532
IC1
(IC6)
C20 (C43)
R6 (R40)
R7 (R41)
CMR
P1 (P3)
R8 (R42)
R9 (R43)
R10 (R44)
C21 (C44)
C22 (C45)
C23 (C46)
R11 (R45)
TL074
IC2
(IC7)
D.C OFFSET TRIM
-18V
R15 (R49)
P2 (P4)
TL071
IC3
(IC8)
R12 (R46)
R13 (R47)
R14 (R48)
C24 (C47)
SKT9 (SKT11)
WHT BLU RED GRN BLK
R1 (R35)
R2 (R36)
C19 (C42)
T2 (T4) E5 TA008
INPUT OPTION
TRANSFORMER BALANCED INPUT OPTION
FACTORY OPTION TO SPECIAL ORDER ONLY .
COMPONENTS REMOVED --
P1 (P3) , R3-R11 (R37-R45)
C20-C23 (C43-C46) , IC1 (IC6)
COMPONENTS ADDED R1 (R35)
R2 (R36) , C19 (C42)
CHANGE R13 (R47) TO 4K22
R25 (R59)
C36 (C59)
C35 (C58)
SKT10 (SKT12)
T3 (T5) OUTPUT OPTION
E5 TA007
TRANSFORMER BALANCED OUTPUT OPTION
REMOVE THE TWO LINKS ON SKT.10 (12)
PLUG IN TRANSFORMER TO SKT.10 (12)
SECURE TRANSFORMER WITH THE SCREW AND WASHER PROVIDED
CHANGE THE LINK ON SKT.13 TO THE "BAL" POSITION
VR33 (VR34)
LEVEL
SW1 (SW2)
TO SWITCH CASES.
NE5534
1 / INV 2 / NON INV 3 / -V 4
8 COMP / 7 +V / 6 O.P / 5 COMP
NE5532
O.P 1 / INV 2 / NON INV 3 / -V 4
8 +V / 7 O.P / 6 INV / 5 NON INV
D.C. NULL 1 / INV 2 / NON INV 3 / -V 4
INPUT
INPUT DEBAL IC1
IC2
LEVEL
IC3
OVERLOAD DETECT IC2
OUTPUT OPTION
OUTPUT
D.C. OFFSET TRIM
30Hz IC2
IC4
IC5
IN
IN + / 30Hz
OUT
EQ
20Hz - 20KHz
BLOCK DIAGRAM
RED
T1
SW2
ORG
110V
220V
BRN
BLK
GREY
BLUE
F2 (T)500mA
F3 (T)500mA
D5 D6 D7 D8
SW1
POWER
F1 (T)250mA
N
E
L
NOTE:
DN3





FREQUENCY	IC No.	FILTER No.	A RA	(B)	A RB	(B)	A CA	(B)	A CB	(B)
20Hz	11 (15)	1	R101	R201	R102	R202	C101	C201	C102	C202
31Hz		2	R103	R203	R104	R204	C103	C203	C104	C204
50Hz		3	R105	R205	R106	R206	C105	C205	C106	C206
80Hz		4	R107	R207	R108	R208	C107	C207	C108	C208
125Hz	12 (16)	5	R109	R209	R110	R210	C109	C209	C110	C210
200Hz		6	R111	R211	R112	R212	C111	C211	C112	C212
315Hz		7	R113	R213	R114	R214	C113	C213	C114	C214
500Hz		8	R115	R215	R116	R216	C115	C215	C116	C216
800Hz	13 (17)	9	R117	R217	R118	R218	C117	C217	C118	C218
1K25		10	R119	R219	R120	R220	C119	C219	C120	C220
2K		11	R121	R221	R122	R222	C121	C221	C122	C222
3K15		12	R123	R223	R124	R224	C123	C232	C124	C224
5K	14 (18)	13	R125	R225	R126	R226	C125	C225	C126	C226
8K		14	R127	R227	R128	R228	C127	C227	C128	C228
12K5		15	R129	R229	R130	R230	C129	C229	C130	C230
20K		16	R131	R231	R132	R232	C131	C231	C132	C232

ONLY CHANNEL "A" IS SHOWN. CHANNEL "B" COMPONENTS ARE SHOWN IN (BRACKETS)

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Each filter circuit is functionally similar to the passive LCR network shown in figure B. The inductor, L, is however replaced by a better performing active circuit consisting of 1/4 of a proprietary KTR008 device plus Cb and Rb and, as in figure B, Ra sets the maximum Boost/cut figure for the filter at its centre frequency (nominally 12.5dB). Ca, as in figure B, is the frequency determining capacitor.

High frequency stability of the equaliser section is assured by capacitors C29, 30, 31.

Eq. Switch

This 3-position switch combines the selection of equaliser section, 30Hz subsonic filter and bypass function as follows:

1) Switch set to 'Eq. +30Hz'
Signal passes through entire system i.e. 30Hz filter then equaliser.

2) Switch set to 'Eq.'
R19 is in effect 'shorted' by the low impedance output of IC3 and since only signal from IC3 is now passed, the 30Hz filter is bypassed. Signal then passes through the equaliser section.

3) Switch set to 'out'
R22 is effectively 'shorted' by the low impedance output of IC3 so bypassing 30Hz filter and equaliser section.

Output Section

Output amplifier IC5 is a voltage follower with a high current drive capability. High frequency stability is optimised with capacitors C32, C33. The output of IC5 is A.C. coupled by C35, 36 and is connected to output transformer socket SKT10. On the standard (unbalanced output) unit, 2 plug-in links on SKT10 connect signal to pin 3, ground to pin 2, of the output XLR connector. On units with transformer balanced output, the 2 links are removed and the transformer is connected to SKT10.

Overload Detect Circuit

Negative bias on the inverting input of IC2D gives +18V at the output of IC2D, thus overload LED D1 is normally 'off'. Signal levels at 4 points within the equaliser are rectified through diodes D9, 10, 11, 17. When a high level signal (greater than +19dB) gives a voltage on the inverting input of IC2D that exceeds the OV threshold, the output of IC2D switches to −18V, so turning on LED D1.

R29 and C37 extend the 'on' time of the overload circuit to ensure signal transients are adequately displayed.

Power Supply

The power supply is a +/−18V design using a low noise toroidal transformer with split primary and secondary windings. The two primaries are connected in series or parallel by SW2 to give 220 or 110 Volt nominal operation. The secondaries drive a full wave bridge rectifier, 2 smoothing capacitors and integrated circuit positive and negative regulators. Decoupling capacitors C3 to C14 ensure low noise supply rails. Power "on" is indicated by LED D2 which is connected between +18V and ground via R35.

The use of Graphic Equalisers

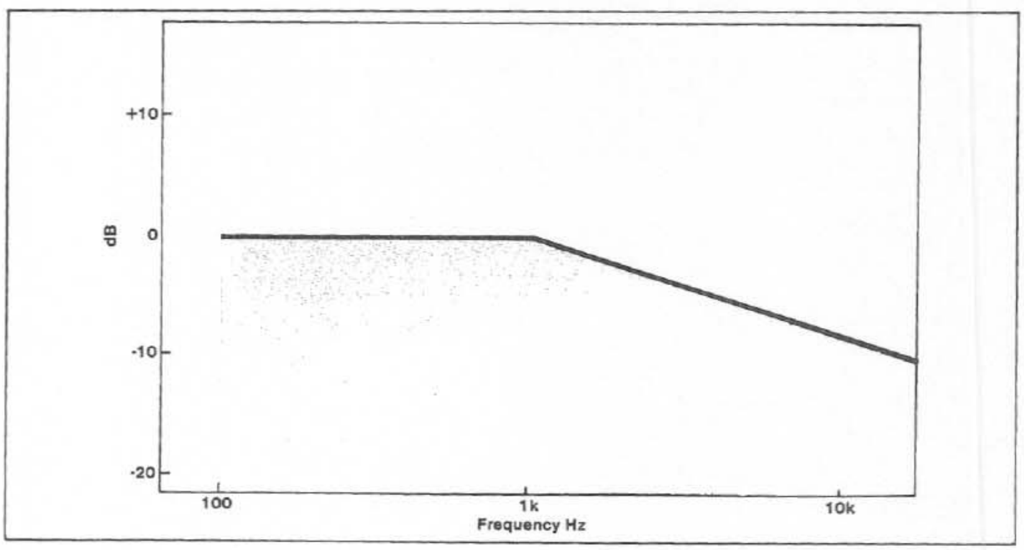
Equalisers may be used for corrective or creative purposes and the Klark-Teknik DN332 is applicable in both live sound and studio applications.

For studio use, a 2/3 octave equaliser might typically be used to compensate for deficiencies in the control room acoustics·if the greater resolution allowed by 30 1/3 octave bands is not required. Because it is almost impossible to set up an equaliser accurately without first analysing the room response, the centre frequencies of the filters have been chosen to correspond with those of the Klark-Teknik spectrum analyser, Model number DN60. In this way, the readings can be transferred directly from the analyser to the equaliser.

It must be stressed however that even a good equaliser doesn't offer a complete solution where the room has severe, inherent acoustic problems. For example; standing waves and resonances cannot be made to disappear simply by using equalisation. True their effects can be reduced, but in a critical listening, environment such as a studio control room or a concert hall, efforts must be made to minimize these problems at source before equalisation is employed. Also, equalisation cannot overcome the lack of sound clarity caused by rooms with unduly long reverberation times though they may be able to effect some improvement in the intelligibility.

On the other hand, the sound company who may well have to set up in different venues night after night have little or no control over the acoustics of the building and so have to use equalisers to arrive at a compromise solution. Depending on the room, some compromises will be more successful than others. Again, effective use of the equaliser means employing the services of a spectrum analyser. It is however not always desirable to achieve a dead flat room response. For example, applying substantial amounts of bass boost to try and restore a weak bottom end is going to use up large amounts of amplifier power and the extra loudspeaker cone excursions so caused will rob the system of headroom and may cause distortion. The harmonics produced by an amplifier driven into clipping may also damage the high frequency drivers and will at any rate sound unpleasant.

On the other hand, cutting the bass may produce real advantages by way of improved intelligibility and subjective naturalness and this is particularly true of buildings made



Typical House Curve.

from concrete or stone where much of the bass is reflected rather than absorbed. Equally, rolling off the high frequency end above 5kHz may also contribute to a more natural sound. The resulting house curve then is far from flat but may well be the ideal compromise. Depending on the individual sound system and the environment, the shape of the optimum house curve will vary and a degree of experience is needed in order to achieve the best results. It should also be borne in mind that the ideal house curves for pure speech and music will not be the same.

In live sound applications, graphic equalisation is almost always applied separately to the stage monitor or foldback system to reduce the level of those frequencies that would otherwise cause feedback problems. These problems come about due to peaks in the frequency response curves of the monitor speaker systems, monitor positioning, and sound reflected from the stage walls. An analyser is probably best employed to do this effectively but many experienced engineers rely on their ears.

In addition to compensating for room acoustics, equalisation can also be used to counteract some of the problems caused by microphone characteristics and positioning or to tailor the response to improve speech intelligibility. Also, many speaker systems have a far from flat response, particularly mobile systems that have to be positioned in physically convenient places rather than the acoustically ideal ones. When equalising the room, these deficiencies are also catered for to a large extent.

Whatever the application, it is generally better to try to attenuate peaks rather than to attempt to boost the surrounding frequencies to the same level. Furthermore, all peaks can be reduced by attenuating their respective bands but some response dips simply cannot be corrected. An example is crossover cancellation where very deep notches may appear covering two or three bands. Attempting to level the response by excessive boosting will simply eat up system power and achieve no useful result. Ultimately a dip in the response is not so audibly objectionable as a peak and so it may be as well to leave these dips alone or to try and solve the problem at source by checking your crossover systems and horn alignment.

In broadcast studios, graphic equalisers are often used during phone-in shows to help compensate for the restricted bandwidth of telephone lines. No equaliser can completely correct the signal in this way as it is impossible to boost frequencies that don't exist and telephone lines have a very restricted bandwidth. Nevertheless, the improvement in subjective terms can be dramatic.

Creative uses may include studio work, live or recorded drama and film soundtrack recording. Voices may be harshly filtered to simulate telephone conversation or the tonal characteristics of an instrument may be modified to fit in with a particular mix.

Though other types of equaliser can often do this job just as well, the graphic equaliser is still the easiest to set up and the controls give an instant visual representation of the response curve. In the commercial studio where time is often of the essence, this attribute should not be overlooked.

Table 1: Effects of Equalisation on Voice Reproduction

1/3 Octave centre frequency (Hz)	Effect on voice
40, 50, 63, 80, 100, 125	Sense of power in some outstanding bass singers.
160, 200, 250	Voice fundamentals.
315, 400, 500	Important for voice quality.
630, 800 1k	Important for voice naturalness. Too much boost in the 315 to 1k range produces a telephone-like quality.
1.25 to 4k	Vocal fricatives - accentuation of vocals. Important to speech intelligibility. Too much boost between 2 and 4kHz can mask certain speech sounds e.g. 'm', 'b' and 'v' can become indistinguishable. Too much boost anywhere between 1 and 4kHz .can produce 'listening fatigue'. Vocals can be highlighted by slightly boosting the vocal at 3kHz and at the same time slightly dipping the instruments at the same frequency.
5, 6.3, 8k	Accentuation of voice.
	The range from 1.25 to 8k governs the clarity of voice.
10, 12.5, 16k	Too much boost causes sibilance.

Table 2: Effects of Equalisation on Music Reproduction

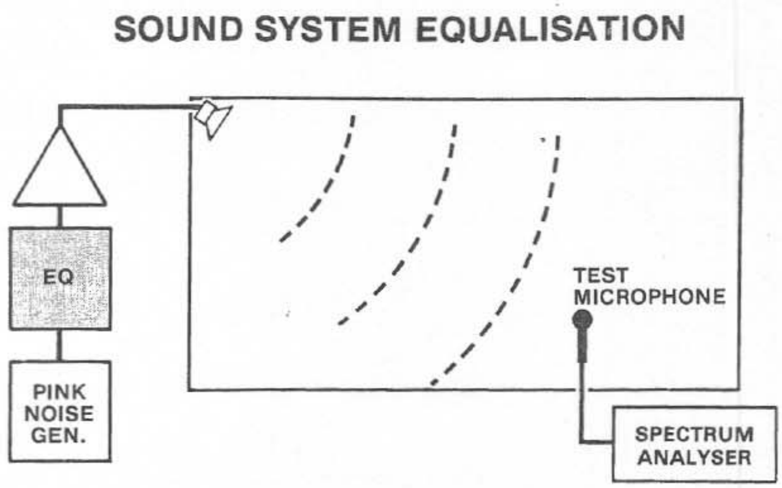
1/3 Octave centre frequency (Hz)	Effect on Music
31, 40, 50, 63	Fundamentals of bass drum, tuba, double bass and organ. These frequencies give music a sense of power. If over-emphasised they make the music 'muddy'. 50 or 60Hz band also used to reject ac. mains hum.
80, 100, 125	Fundamentals of lower tympani. Too much boost produces excessive 'boom'. 100 or 125Hz also used for hum rejection.
160, 200, 250	Drum and lower bass. Too much boost produces excessive 'boom'. Also useful for 3rd harmonic mains hum rejection.
315, 400, 500	Fundamentals of strings and percussion.
630, 800, 1k	Fundamentals and harmonics of strings, keyboards and percussion. Boosting the 600 - 1kHz range can make instruments sound horn-like.
1.25 to 4k	Drums, guitar, accentuation of vocals, strings and brass. Too much boost in the 1 to 2kHz range can make instruments sound tinny. Too much boost anywhere between 1 to 4kHz can produce 'listening fatigue'.
5, 6.3, 8k	Accentuation of percussion, cymbals and snare drum. Reduction at 5kHz makes overall sound more distant and transparent. Reduction of tape hiss and system noise. The 1.25 to 8k governs clarity and definition.
10, 12.5, 16k	Cymbals and overall brightness. Too much boost causes sibilance. Reduction of tape hiss and system noise.

Equalising a Sound System

When equalising a sound system you should always remember just what it is that you are trying to achieve. Two fundamental reasons for equalisation are:-

1. **To increase the potential gain or power output of the system before feedback.**
2. **To improve the naturalness or intelligibility of the sound system.**

In a space with poor acoustics or high levels of background noise, the most natural sound may well not be the most intelligible - a compromise must therefore by reached between these two qualities depending on the particular application in question - but at the end of the day it doesn't matter how natural the system sounds if no one can understand the sound it puts out!



Sound System Equalisation.

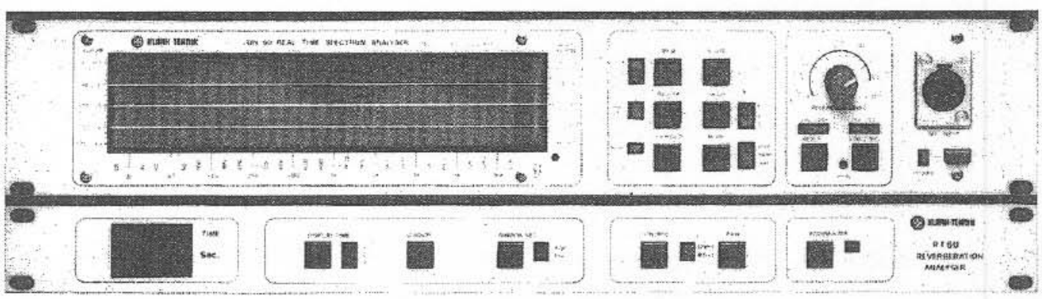
Before beginning to equalise a system, it is a good practice to listen to the "raw" system with speech or music programme. If such signals are distorted then stop and rectify them before attempting to equalise. Another good pre-equalisation test is to use a slow sine-sweep. This can expose a number of problems such as rattles or distortion or poorly controlled room modes and resonances - which pink noise RTA cannot discover. Finally, before equalisation, check the coverage of the system over the 2 to 4kHz bank. (If necessary, use the equaliser as a band pass filter to produce the desired range). If coverage is poor to begin with then no amount of equalisation will overcome this. Again adjustments to the system itself are required. Equalisation is the final tuning stage. Generally, a gradual transition between adjacent bands should be aimed for, particularly in studio monitoring situations where the maximum difference between bands should only be 3dB or so. A warning bell should be ringing if you are using much more than this! This does not mean however that more drastic adjustments should not be used - this is very often necessary with sound systems operating in poor or severe acoustic environments, but the reason why such a particularly large fader excursion is being used at a given frequency should always be carefully considered.

Once satisfied with the basic system, performance equalisation can begin. If using a real time analyser ensure that the microphone is in a sensible position i.e. within the coverage area of the system and not in an area where strong local acoustic effects might be expected such as within 1 metre of a rear or side wall or in a balcony opening.

A good idea is to rotate the measuring microphone in a wide arc or circle round the measuring position and to see if any strong interactions occur causing large deviations in response. If necessary, move to another position. Also ensure that the ambient noise level is at least 6dB (preferably 10dB) below the signal level you are using.

Having set up the desired house curve as smoothly as possible, move round and check the response throughout the listening area. Good equalisation requires time and patience. Do not forget that some interaction will occur between a particular filter and its adjacent bands. A better sound may be produced by adjusting several bands rather than by strongly cutting just the centre one. Do not forget to pause to talk or play music through the system as you go, so that you keep in touch with what the resultant sound quality is like.

If the Real Time Analyser you are using is the Klark-Teknik DN60/RT60 you have a powerful averaging capability which means that averaging the response throughout the coverage area becomes very much easier. The response displayed by the analyser should also become smoother as local fluctuations are averaged out whilst persistent peaks and dips clearly stand out and show where adjustment is truly required.

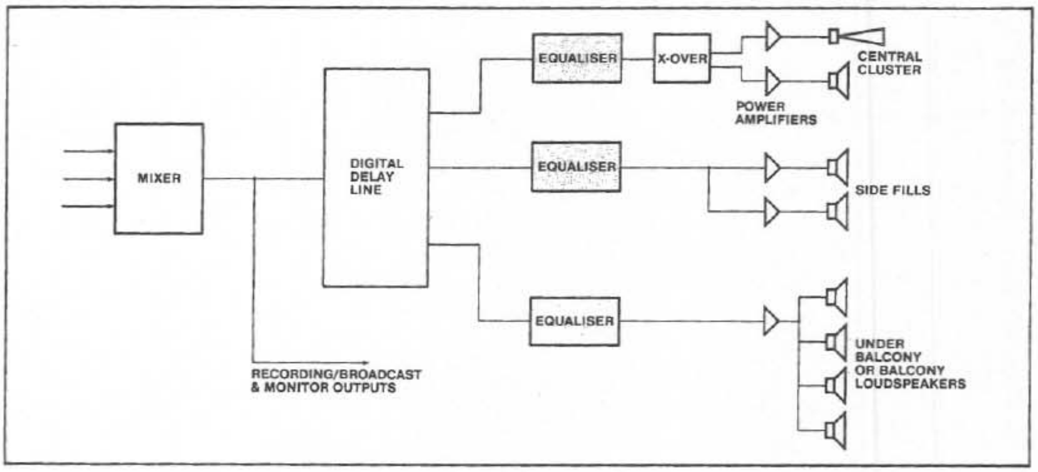


Real Time Spectrum Analyser with Averaging Model DN60/RT60.

Connecting a stage or house microphone into the analyser can be a most instructive exercise - showing up any local reflection or acoustic resonances or loudspeaker sidelobes. Generally repositioning the microphone or adjusting individual microphone channel equalisation will be needed, rather than adjustment to the overall house curve. This technique is particularly useful when investigating acoustic feedback after the initial house curve has been set.

Inserting the Equaliser in the Signal Chain

The exact point of insertion of an equaliser into the signal chain will very much depend on the task in hand e.g. mixer channel/line input, group insert point, group output, auxiliary send or between another signal processing device and the mixer or power amplifier, etc. When using delayed out signals for example, i.e. where a digital delay line is being used to synchronise sound arrivals in order to maintain intelligibility or source directionality, the option may exist to insert the equaliser either before or after the delay line e.g. in a conference venue employing similar loudspeaker types throughout the system, but connected to different delay outputs, the equaliser can be inserted before the delay line. In a more complex system where several loudspeaker types are employed, or where the local acoustic environment differs within the same system e.g. a theatre system with a central loudspeaker cluster and delayed side fills or underbalcony speakers, each delay channel will need its own separate equaliser in order to satisfactorily equalise out either the different loudspeaker responses or the effects of the different local acoustic environment.



Simplified schematic diagram for a typical theatre sound system.

Equaliser Limitations

The equaliser is not the answer to poor sound system design - but instead it should be considered as a final tuning measure - such final tuning can often bring about quite remarkable improvements to the overall intelligibility and perceived sound quality of a system.

After an equaliser, a security cover is probably the most useful accessory a sound system could have. Equalisers, when used competently, can do wonders for your system - but when used badly.......