

**Operating
and service
information
DN410
parametric equaliser**

IMPORTANT:

Product Registration

Please check that the enclosed product registration card is part of this book. If **NOT**, immediately inform the factory quoting this units serial number.

Introduction

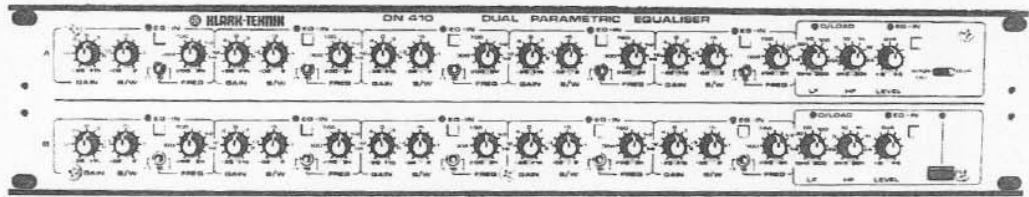
Parametric equalisers have been used for many years for creative equalisation in recording and broadcast studios. Their suitability for corrective equalisation in sound reinforcement systems has been limited by both a lack of accurate spectrum analysis and certain limitations in operational flexibility. Today, the widespread use of improved analysis equipment means that even narrow-band room resonances can be found and the results of corrective equalisation measured accurately.

Series 400 Parametrics can be used with great effect in both creative and corrective applications and offer unsurpassed flexibility as an equalisation device. They are uniquely suited to sound reinforcement applications. Each of five parametric filters can be positioned at any centre frequency between 20Hz and 20kHz with a wide choice of bandwidths from broad to notch. The DN410 offers even greater flexibility with an expanded 10 band single channel mode. Separate high and low frequency cut filters are featured on all models.

Where creative equalisation is required, such as in audio for video post production, broadcast and obviously recording studio's, both the DN405 and DN410 will find instant popularity whenever powerful equalisation is found to be needed.

Like all Klark-Teknik equaliser products the Series 400 Parametrics are carefully designed for excellent noise, distortion and sonic performance. 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.

DN410 Parametric Equaliser

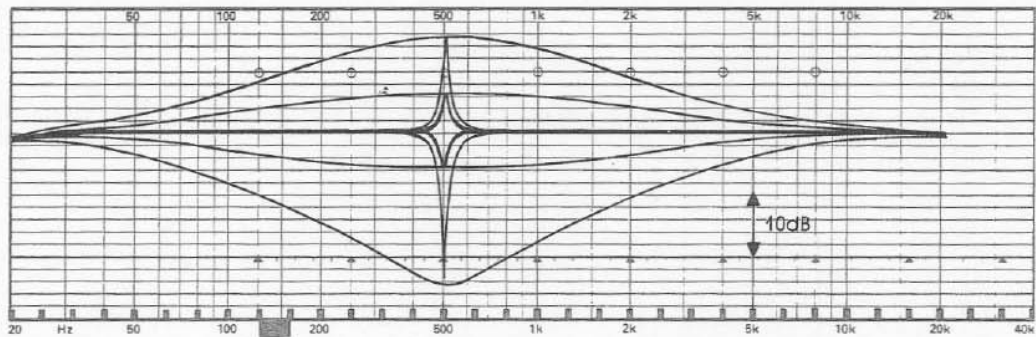


The Klark-Teknik DN410 is a dual channel, 2 x 5 band parametric equaliser offering 15dB of boost or 25dB of cut at any frequency between 20Hz-20kHz, and variable filter bandwidths from less than 1/12 octave to in excess of 2 octaves. In the expanded 'single' mode, all 10 parametric filters are placed into channel A.

Filter Performance

The following curves illustrate the variety of filter responses possible when using the DN410 parametric. The five parametric sections are designed to be free from interaction between each other and so using more than one filter gives a response curve that is simply the sum of the various individual responses.

Parametric



Shows the range of bandwidths and amplitudes possible using a single parametric filter. The centre frequency could, of course, be anywhere between 20Hz and 20kHz.

Notching

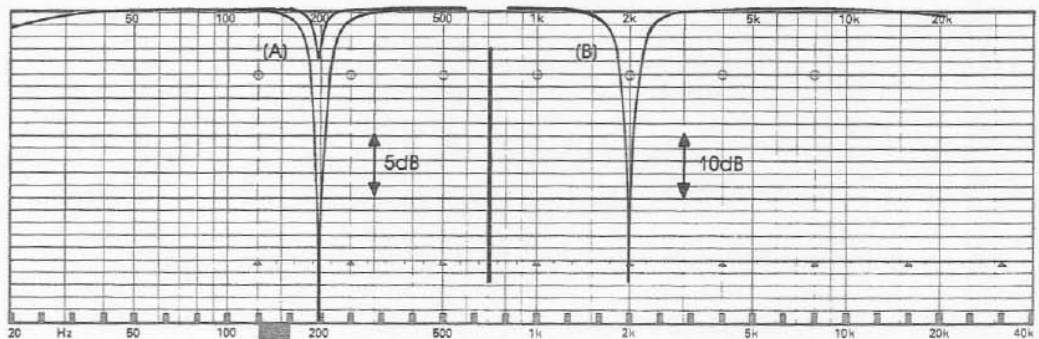
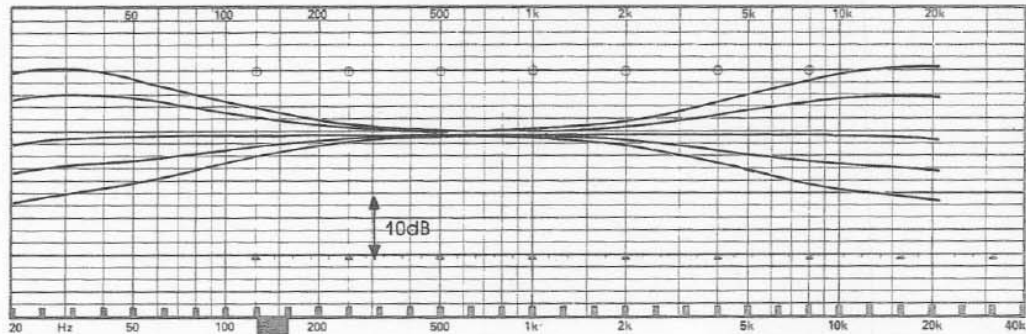


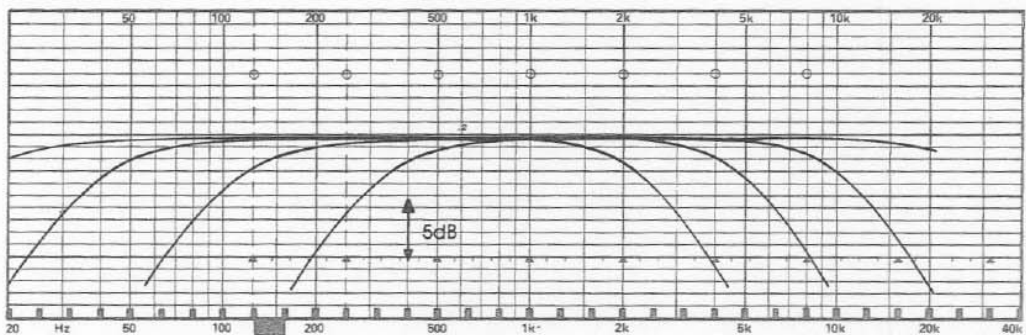
Figure A demonstrates typical notch performance, with filter bandwidths set at minimum. Figure B shows the effect of tuning two notch filters to the same centre frequency for increased rejection.

Shelving



Shows the use of a parametric filter to obtain LF and HF shelving responses. The filter is shown set at minimum and maximum frequencies in turn, with bandwidth at maximum.

LF/HF Cut



Demonstrates the performance of LF and HF cut filters, with controls set for minimum, maximum and intermediate frequencies. Slope for both filters is 12dB octave.

Other Features

The channel eq. in/out switch removes both the equaliser section and the high/low-pass filters from the signal path.

Other features include an overload LED, 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. A failsafe relay is incorporated into the design, which automatically and silently bypasses the equaliser in the event of power supply failure. This relay also acts at switch-on to isolate the equaliser until the power rails have settled, thus precluding the possibility of a potentially damaging switch-on thump.

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 one standard unit of rack space and has an electronically balanced input and unbalanced output. 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

Perspex security cover

Aluminium security cover

Output balancing transformer

Input balancing transformer

Parts Number

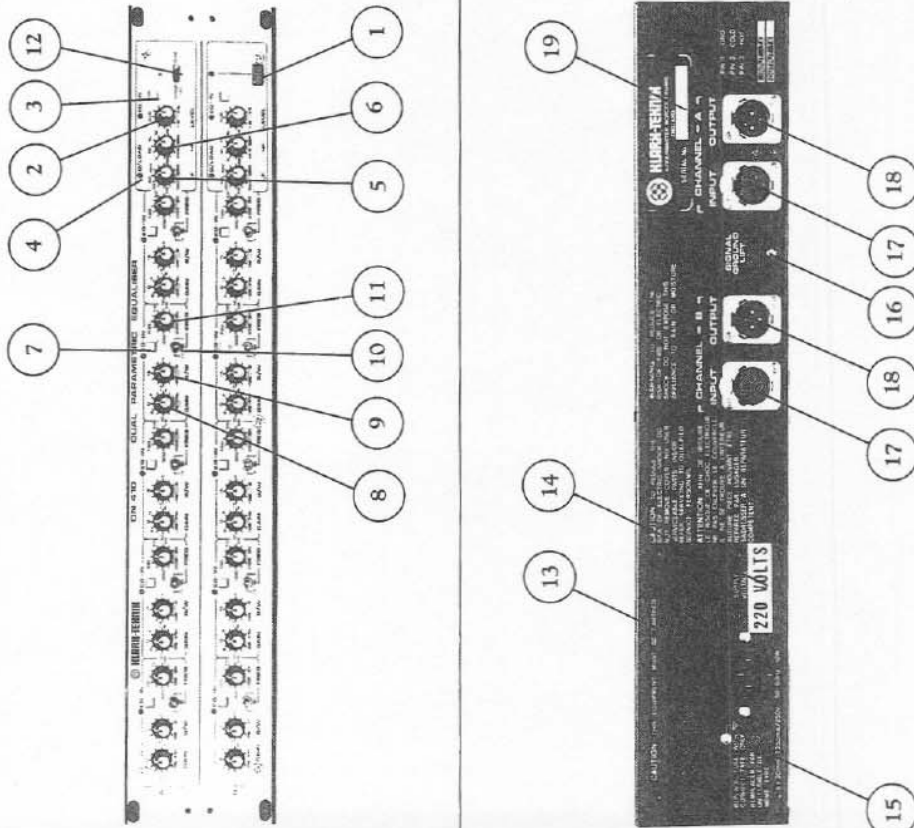
SCP Model No.

SCA Model No.

BU37

BN37

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, a red status LED lights.
2. The input level control allows system gain up to +6dB when in its fully clockwise position, and offers full attenuation in its anti-clockwise position.
3. The channel eq. in/out switch removes both the high/low-pass filters and the complete equaliser section from the signal path. Eq. 'in' status is indicated by LED.
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 occasionally cause this level to be exceeded. In this event, the input level control should be turned down to correct the problem. However, if the problem arises, the signal level from the output of the preceding piece of equipment must be turned down.
5. The low frequency filter (high pass) has a continuously variable turnover frequency from 15Hz to 300Hz.
6. The high frequency filter (low pass) has a continuously variable turnover frequency from 2.5kHz to 30kHz.
7. Eq. in/out switch removes the individual parametric filter from the signal path. Eq. 'in' status is indicated by LED except when cancelled by switching channel eq. to 'out'.
8. The gain control sets gain between +15 and -25dB for the parametric filter. A centre detent indicates the 'flat' position.
9. The bandwidth control sets filter bandwidth between 2 octaves when fully clockwise to less than 1/12 octave when anti-clockwise.
10. The range switch selects the frequency band required as follows:
 - 1/200Hz-200Hz
 - 1/200Hz-2kHz
 - 1/10/2kHz-20kHz
11. The frequency control sets the filter centre frequency within the limits selected by the range switch. A nominal 10:1 sweep range allows precise tuning of notch centre frequencies.
12. The Single/Dual mode switch places all 10 parametric filters into channel A in 'single' and configures the unit for normal 2 x 5 band operation in 'dual'. Note that only channel A level control, sweep filters and eq. in/out switch function in 'single' mode.

Rear Panel Functions

13. Mains is supplied via an IEC standard 3 pin connector. A compatible power cord is supplied with the unit.
 14. Voltage selecto. 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.
 15. 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.
 16. 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.
 17. Input and output connections are made via complementary
 18. XLR style sockets.
 19. For wiring details see page entitled Audio Connections.
- The serial number of this unit should be quoted in any correspondence concerning the unit.

Audio Connections

Input

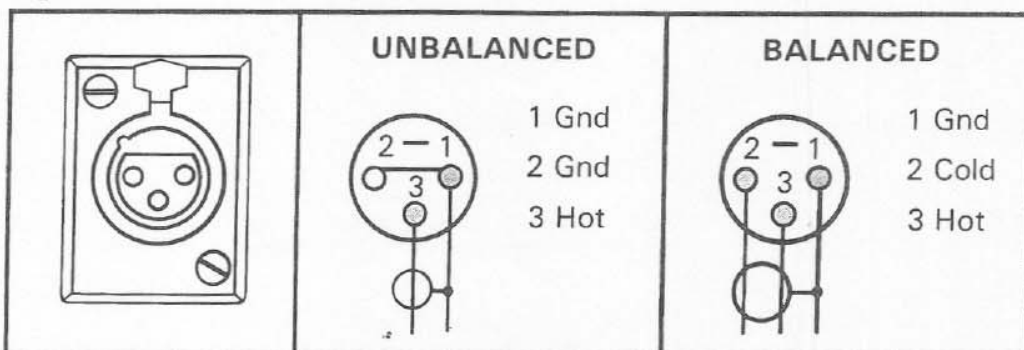
The input circuitry is a transformerless, electronically balanced design which achieves a symmetry of better than -50dB from 20Hz to 10kHz .

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

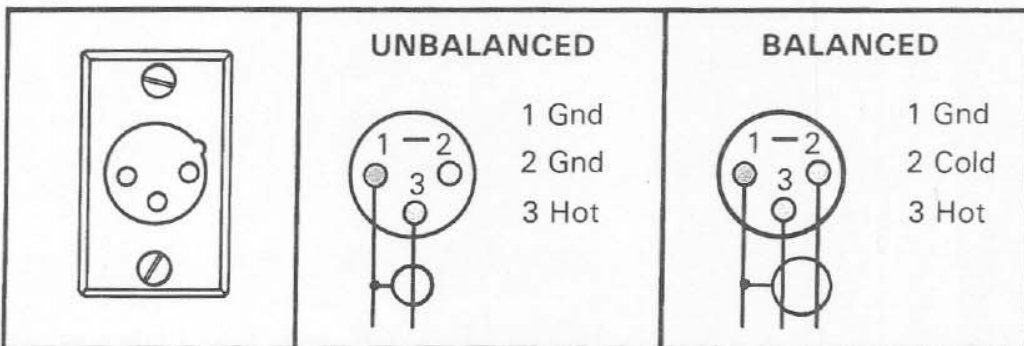
Output

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

Input



Output



Pin 2/Pin 3 Hot operation

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. In "Single" mode use channel A connectors only.

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.

DN410 Technical Specifications

Inputs

Type
Impedance (ohm)
Balanced
Unbalanced

Two

Balanced (electronically)
20K
10K

Outputs

Type
Min. Load impedance
Source
Max. level

Two

Unbalanced
600 ohm
<60 ohm
+22dBm

Performance

Frequency response (20Hz-20kHz)
Distortion (@ +4dBm)
Equivalent input noise
(20Hz-20kHz unweighted)
Channel separation
Gain
Overload indicator

± 1.5 dB
<0.01% @ 1kHz
<-90dBm
>80dB @ 1kHz
+6dB
+19dBu

Filters

Type
Bandwidth
Max. boost/cut
Frequency ranges
High pass filter
Low pass filter

Parametric (2 x 5)
Variable from 1/12 - 2 octaves
+15/-25dB
20Hz-200Hz/200Hz-2kHz/2kHz-20kHz
15Hz-300Hz/12dB octave
2k5Hz-30kHz/12dB octave

Power Requirements

Voltage
Consumption

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

Weight

Nett
Shipping

4.5kg
7kg

Dimensions

Width
Depth
Height

482mm (19 inch)
235mm (9¼ inch)
89mm (3½ inch)

Terminations

Inputs
Outputs
Power

3 pin XLR
3 pin XLR
3 pin CEE

DN410 Circuit Description (Note - only Channel A is described. Channel B is identical).

Input Section

The input signal enters via XLR-type connector SK1 and may be balanced or unbalanced (see audio connector wiring details). The signal is then connected via relay RLA and first order R.F. filter network R2/C1 and R7/C2 to the differential input amplifier IC1. Common mode rejection is factory set with RV1 to give maximum rejection at 100Hz; typically 85dB. This stage is followed by the input level control and non-inverting amplifier IC3B, which has a gain of 3dB.

H.F. and L.F. Cut Filters

From IC3B, the signal passes through the L.F. and H.F. cut filters. These are 12dB/octave second order Butterworth filters arranged around IC4B, and IC4A respectively. The turnover frequency for each filter is continuously variable; between 15Hz and 300Hz for L.F. cut and 2k5Hz to 30kHz for H.F. cut. Resistors R25 and R26 set L.F. cut filter characteristics and result in a gain of 4dB for IC4B. The potential divider R33, R34 attenuates by 4dB and is buffered from the equaliser section by voltage follower IC3A.

Equaliser Section

The five parametric filters are identical high performance state-variable designs connected in cascade. Only filter "A" is described here.

The Output of IC3A is connected via ST5 and ST3A to unity gain inverter IC1B. 'Gain' control VR1 passes signal that is in phase with the input when turned clockwise or out of phase when turned anti-clockwise. Maximum boost is limited to +16dB by series resistor R1. R2 restores unity gain with VR1 set 'flat'. VR1 is connected to Q determining stage IC1A, via R6. Minimum bandwidth is set by R7, maximum by R8. Output from IC1A passes to integrators IC2A, B. Centre frequency of bandpass output IC2A pin 1 is set by VR3, R13 and R15 and switched in 3 ranges, by SW1 and C3-C14.

The output of IC2A is AC coupled by C2, R5 and passes via ST3A and eq. in/out switch SW2A to the non-inverting input of IC1B. The output of IC1B passes to the input of filter B etc. The output of filter E (IC1B) is connected via inverter IC5B to the "Single/Dual" mode relay RLA.

Single/Dual Mode Switch

1. **"Dual" Mode** — With mode switch set to "Dual" RLA is switched off and IC5B output (Channel A) is connected via RLA to Channel A eq. in/out switch SW3. IC5B output (Channel B) is connected to Channel B eq. in/out switch SW3. The output of IC3A (Channel B) passes via ST5, ST6 to RLA and via ST3A to the input of filter A. +18V for Channel B constant current source Q1 is supplied via ST6 and "Single/Dual" switch.
2. **"Single" Mode** — With mode switch set to "Single", RLA is switched on with +18V from ST6 pin 1. IC5B output (Channel A) is connected via RLA and pin 4 of ST6 to ST3A and hence to the input of filter A of channel B. IC5B output (Channel B) is connected via ST6 pin 6 and RLA to Channel A eq. in/out switch. +18V previously supplied via pin 2 of ST6 to constant current source Q1 of Channel B is disconnected via the mode switch and is now supplied via pin 3 of ST6 from Channel A eq. in/out switch.

Channel Eq. In/Out Switch

With eq. switch SW3 to 'out' the non-inverting input of IC5A is connected via R38 to the output of IC3B, bypassing low and high cut filters and equalisation section. With eq. switch to 'in', resistor R38 is in effect 'shorted' by the low impedance output of IC5B and only signal from the equaliser section, via IC5B, is now passed. This method gives an uninterrupted transition from eq. 'out' to 'in' switch positions. SW3 also disconnects +18V from the constant current source Q1 when set to 'out', switching off all 6 eq. 'in' LEDs. (In "Single" mode Channel A eq. in/out switch also supplies Channel B constant current source - see previous section).

Output Section

Output amplifier IC5A is a non-inverting 4dB gain stage with a high current drive capability. High frequency stability is optimised with capacitors C25, C26. The output of IC5A is A.C. coupled by C6, 7 and is connected to output transformer socket STK5. On the standard unbalanced output unit, 2 plug-in links on STK5 connect signal to pin 3, ground to pin 2, of the output XLR connector via bypass relay RLA. On units with transformer balanced output, the 2 links are removed and the transformer is connected to SKT5.

Bypass Relay

With power to the unit removed, input and output XLR pins 2 and 3 are connected together via relay RLA. When power is connected, the delay circuit waits 2 seconds before switching in RLA, so avoiding output 'clicks'. On power 'off' the relay is switched off instantly, again avoiding output 'clicks'.

Relay Delay Circuit

When power is turned on, C25 is charged via R26 until the voltage on C25 exceeds -3V. At this time (2 seconds from turn on), D8 conducts. When the voltage across R28 exceeds 0.6V, Q1 switches on applying -18V to R29, so switching the relay RLA 'on' (and RLB in DN410). R29, 30 limit the voltage across RLA to 20V in DN405. When power is turned off, the +18V supply starts to fall and D7 discharges C25, immediately switching off Q1 and RLA (and RLB for DN410).

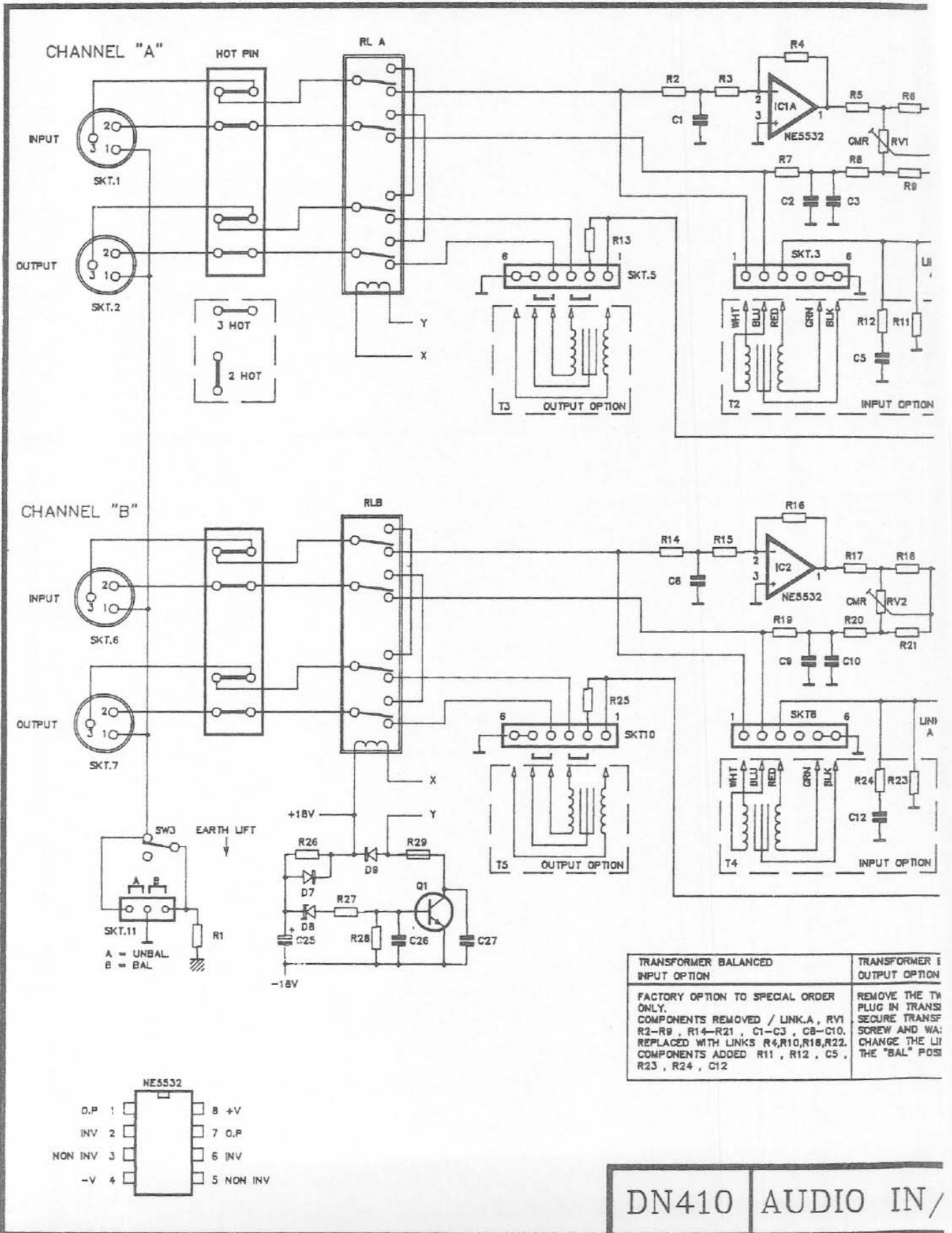
Overload Detect Circuit

Negative bias on the inverting input of IC6A gives +18V at the output of IC6A, thus overload LED D14 is normally 'off'. Signal levels at 19 points within the equaliser are rectified through diodes D1, 2, 3 etc. When a high level signal (greater than +19dB) gives a voltage on the inverting input of IC6A that exceeds the OV threshold, the output of IC6A switches to -18V, so turning on LED D14.

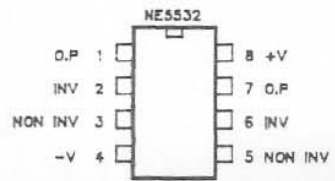
R43 and C31 extend the 'on' time of the overload circuit to ensure signal transients are adequately displayed.

Power Supply

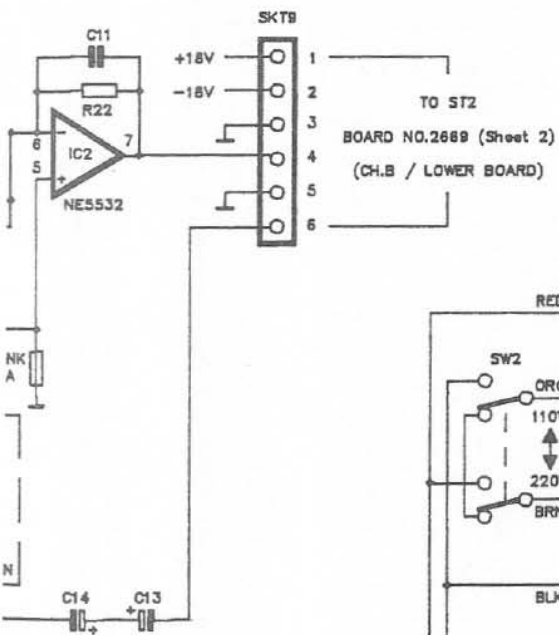
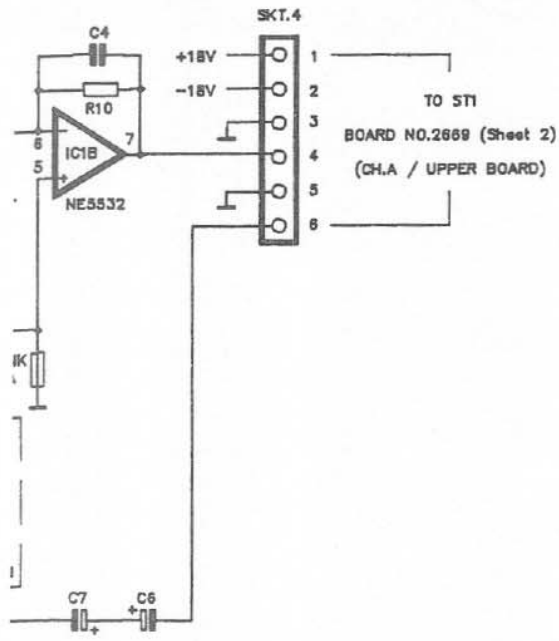
The power supply is a +/- 18V design using a low noise toroidal transformer with split primary and secondary windings. The two primaries are connected in series or parallel by SW2 to give 220 or 110 Volt nominal operation. The secondaries drive a full wave bridge rectifier, 2 smoothing capacitors and integrated circuit positive and negative regulators. Decoupling capacitors C17 to C24 ensure low noise supply rails. Power 'on' is indicated by LED D13 which is connected between +18V and -18V via R50.



TRANSFORMER BALANCED INPUT OPTION	TRANSFORMER E OUTPUT OPTION
FACTORY OPTION TO SPECIAL ORDER ONLY. COMPONENTS REMOVED / LINK.A, R1 R2-R9, R14-R21, C1-C3, C8-C10. REPLACED WITH LINKS R4,R10,R18,R22. COMPONENTS ADDED R11, R12, C5, R23, R24, C12	REMOVE THE T4 PLUG IN TRANSI SECURE TRANSF SCREW AND WA: CHANGE THE LI THE "BAL" POSI

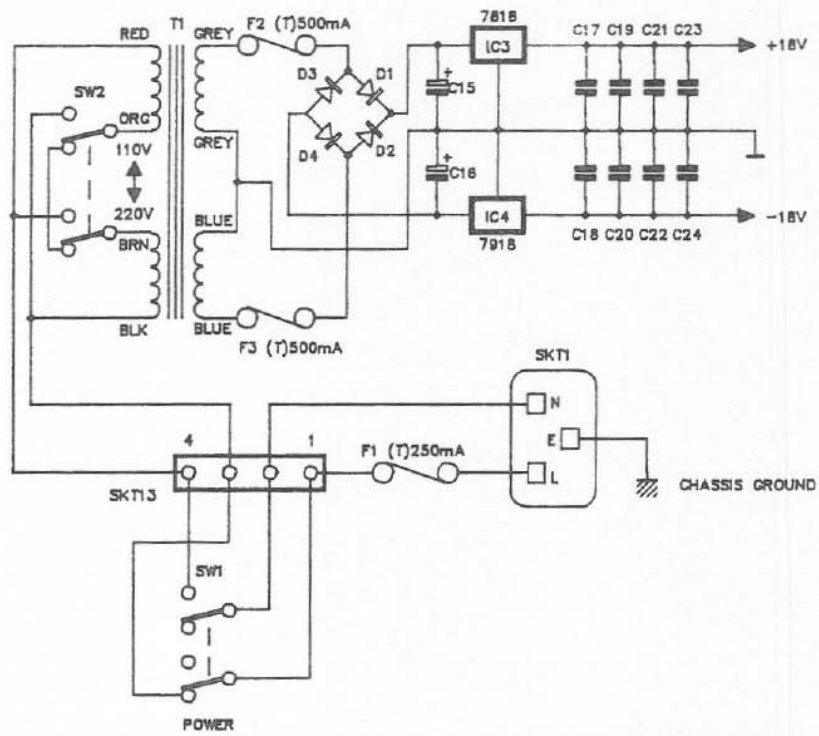


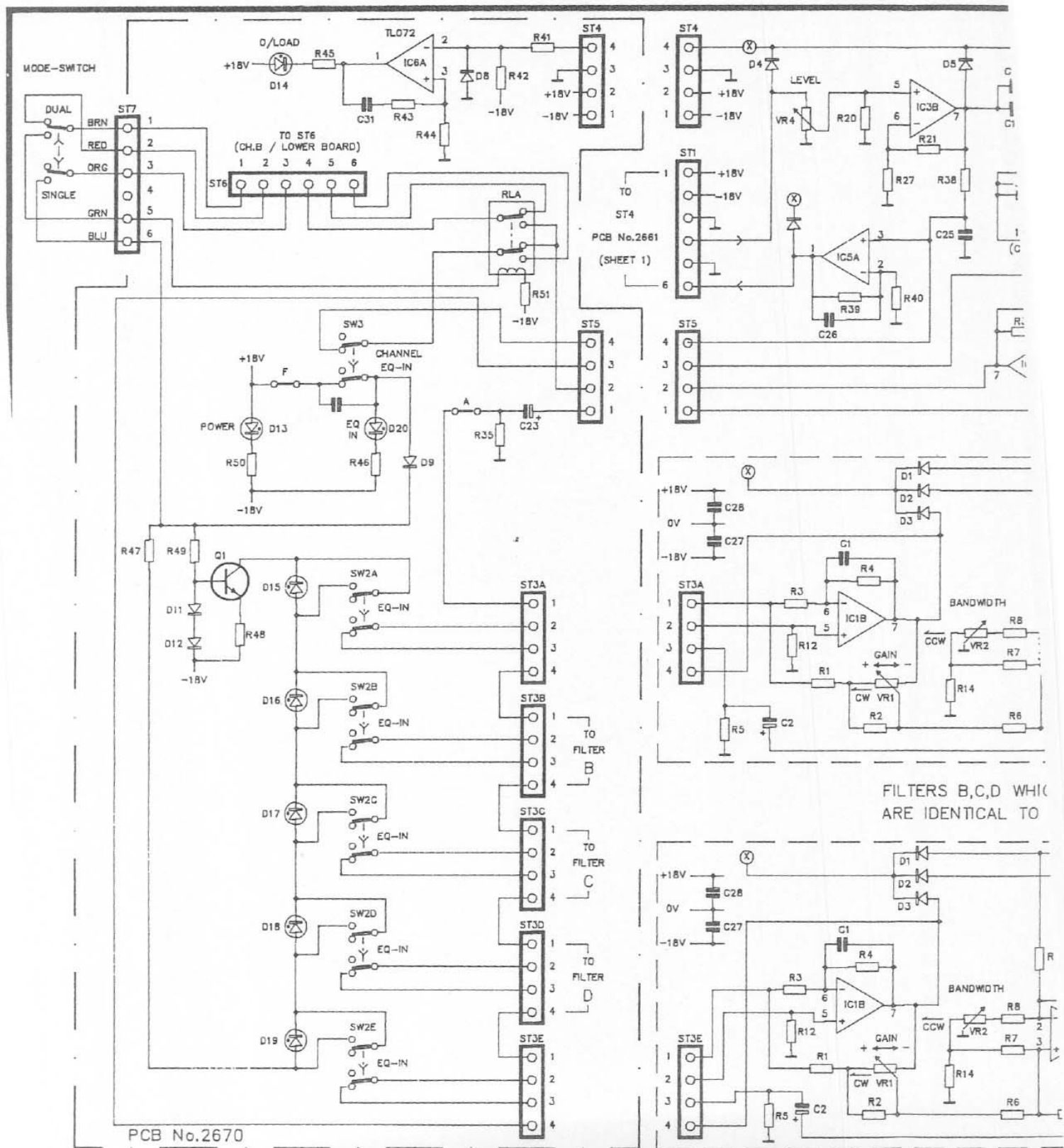
DN410 AUDIO IN/



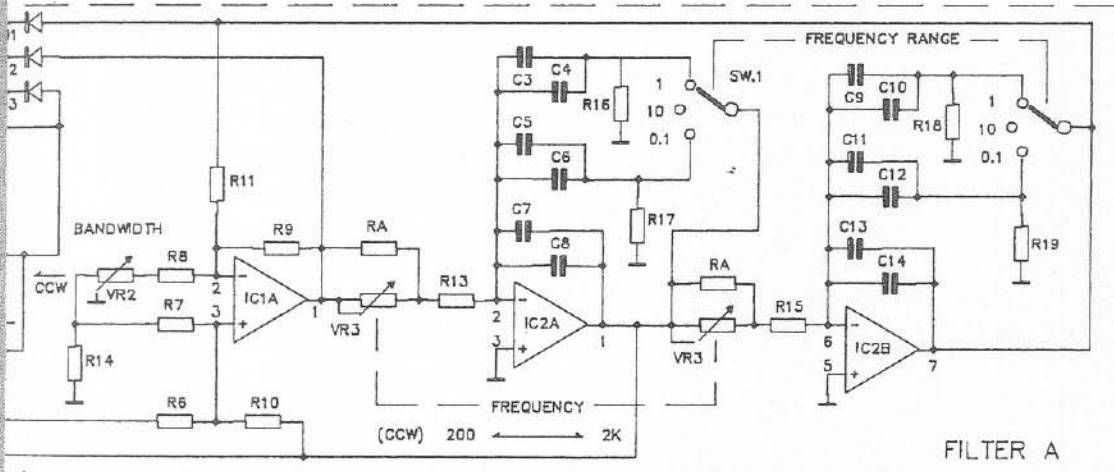
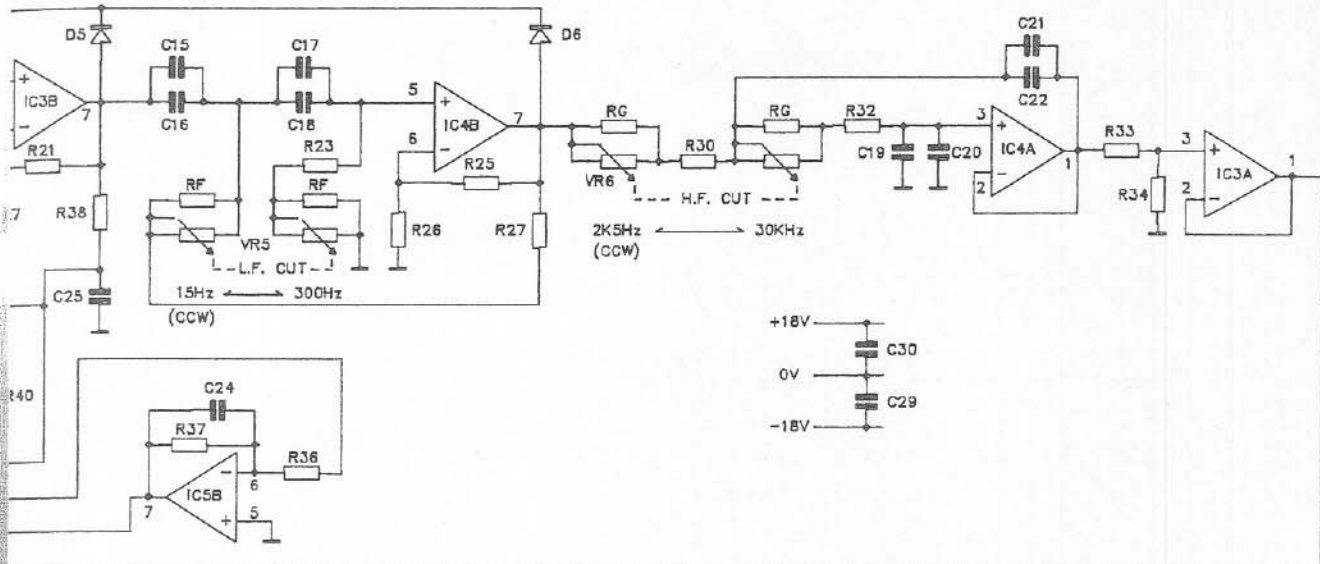
FOR BALANCED
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TWO LINKS ON SKT.5
UNFORMER TO SKT.5
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LINK ON SKT.11 TO
POSITION



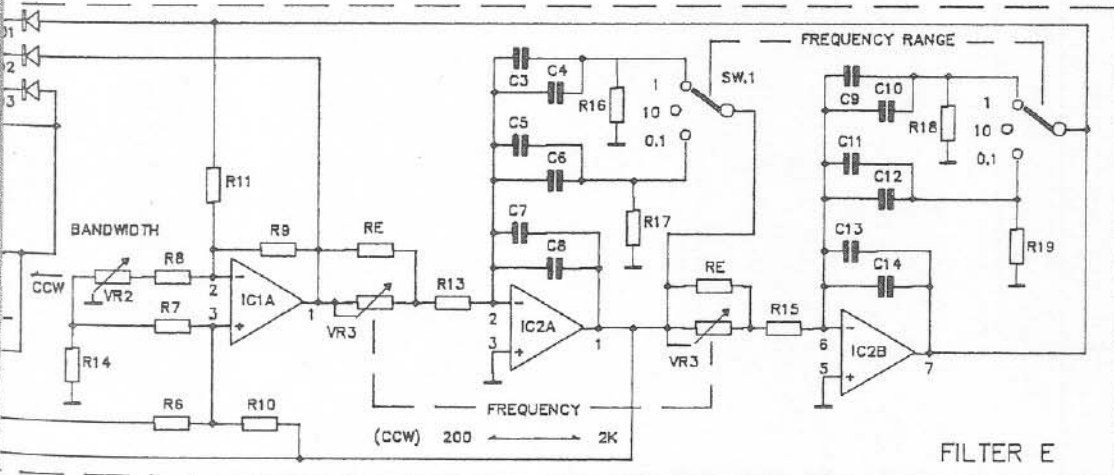


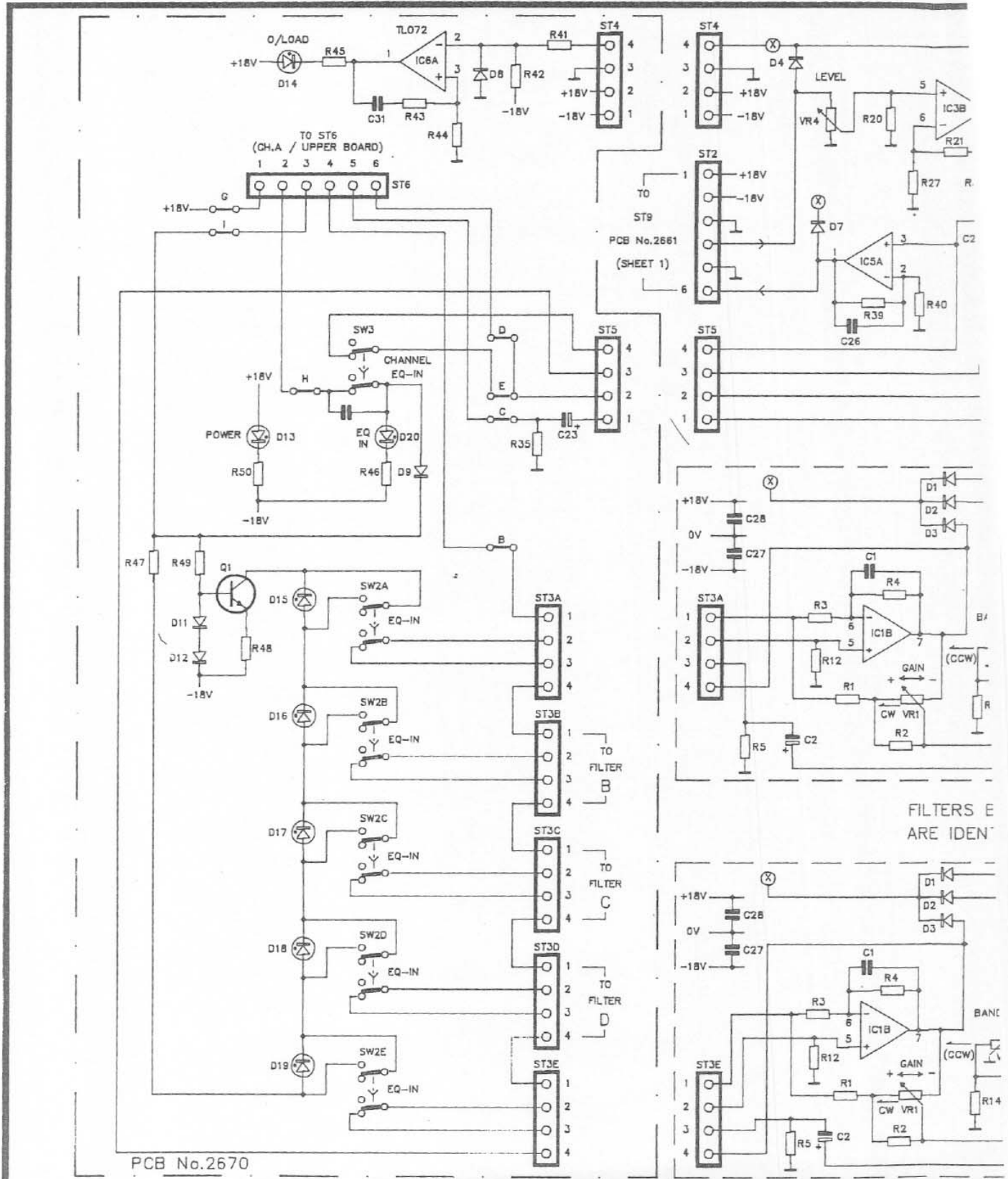
DN410 CHANNEL A FILTER/SWITCH SECTION(SI



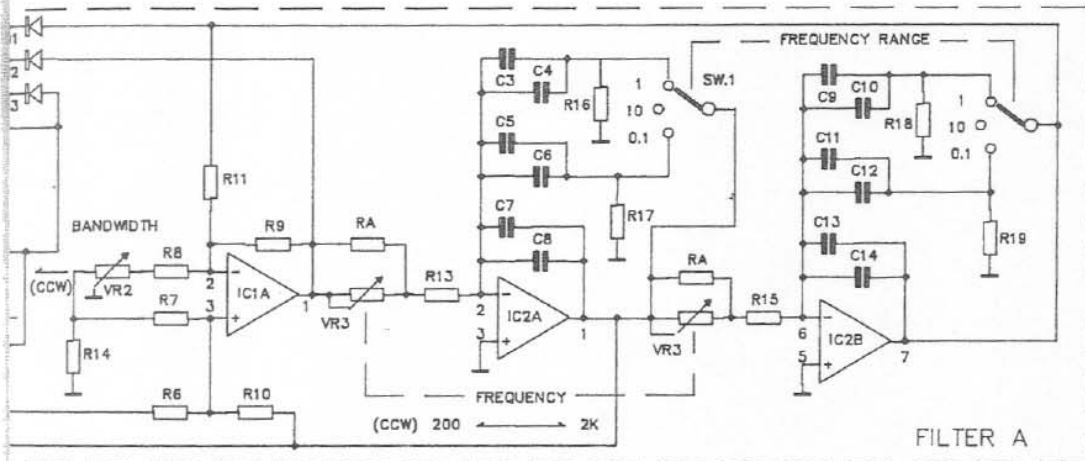
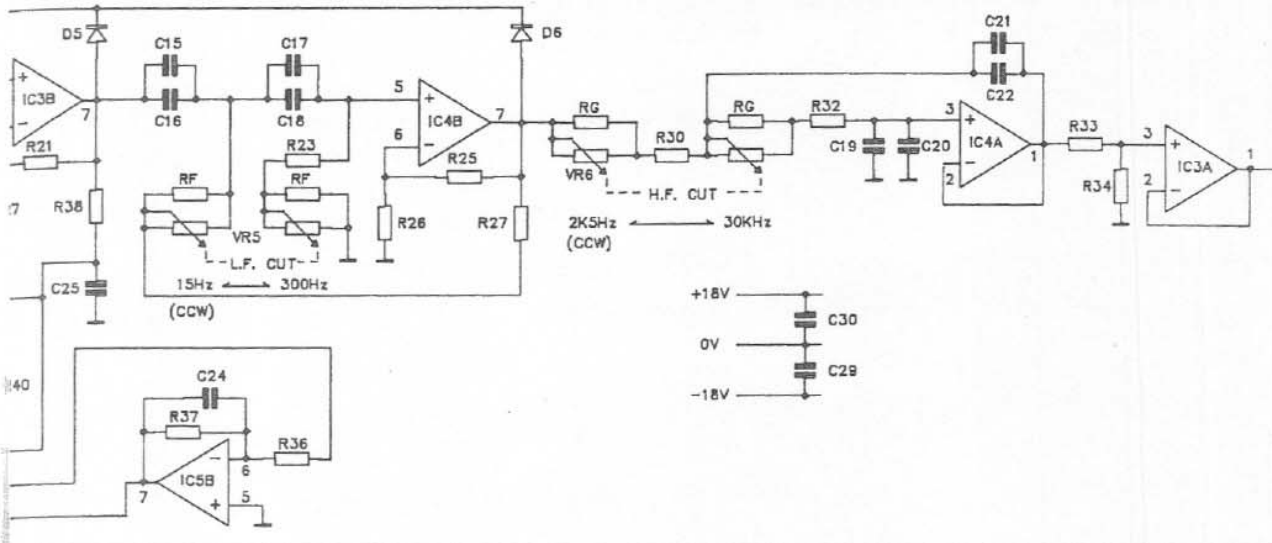
FILTERS B,C,D WHICH ARE NOT SHOWN
ARE IDENTICAL TO FILTERS A,E.

ALL IC'S NE5532 UNLESS STATED



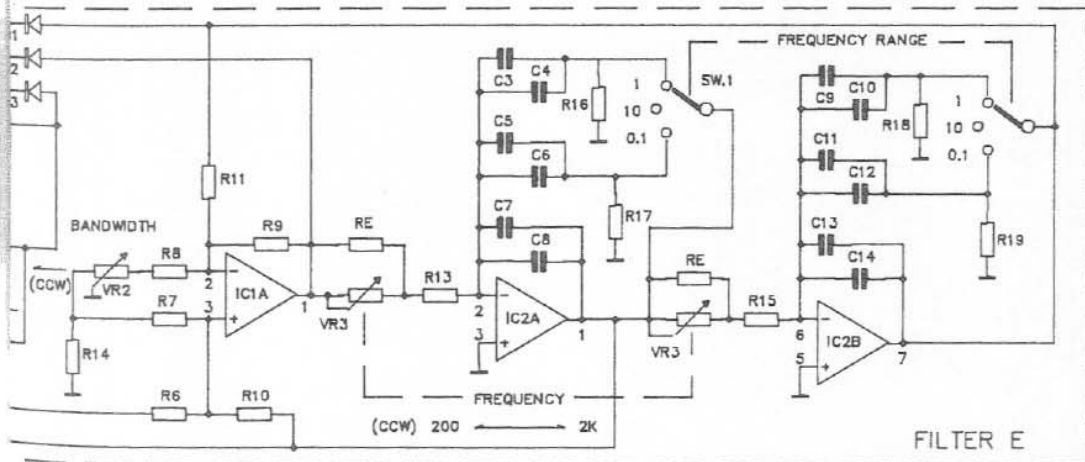


DN410 CHANNEL B FILTER/SWITCH SECTION

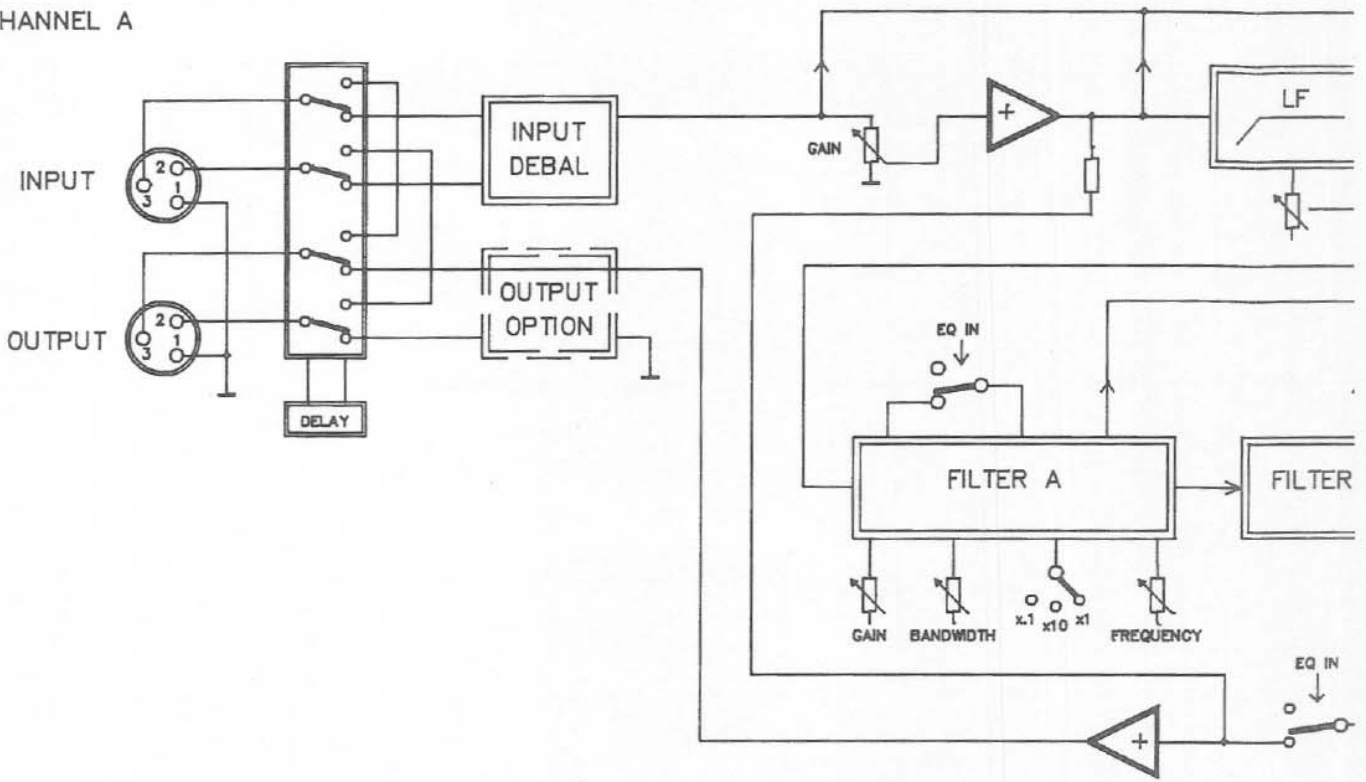


FILTERS B,C,D WHICH ARE NOT SHOWN
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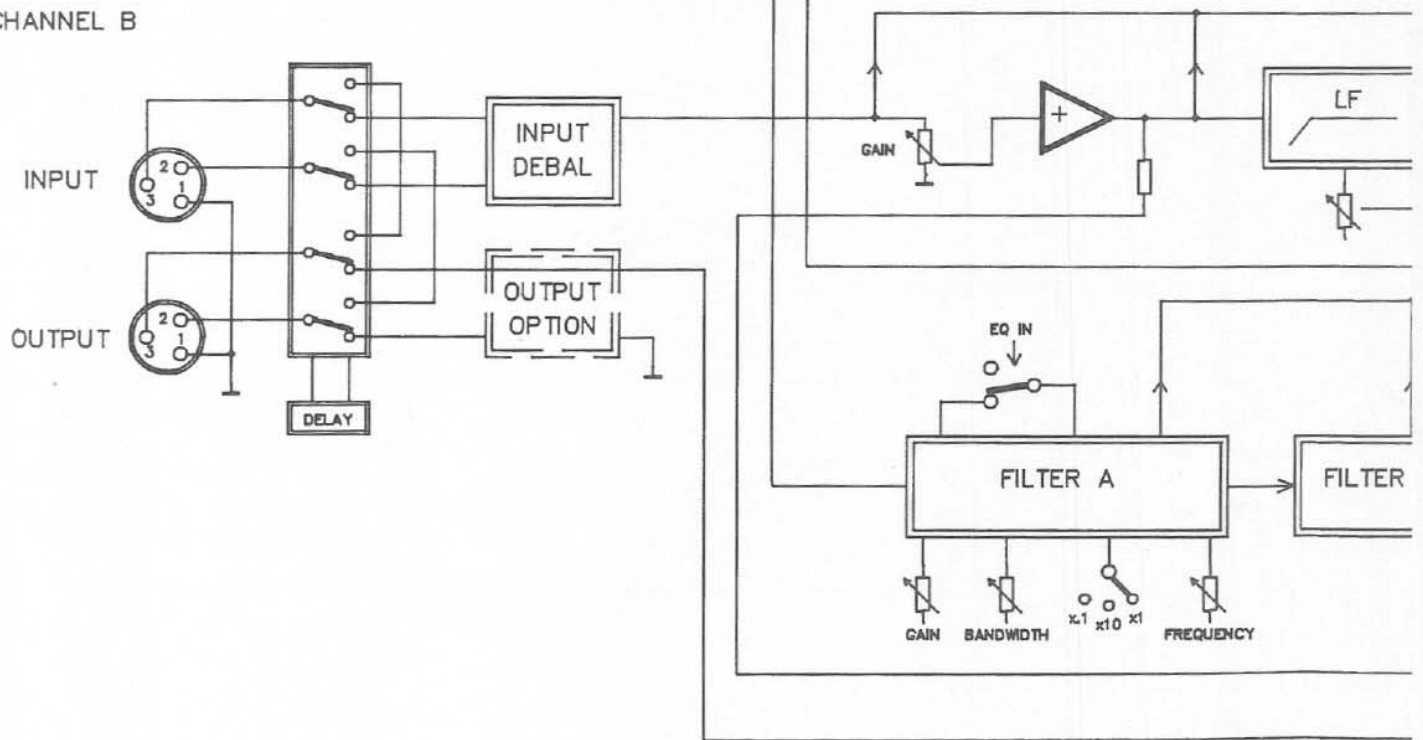
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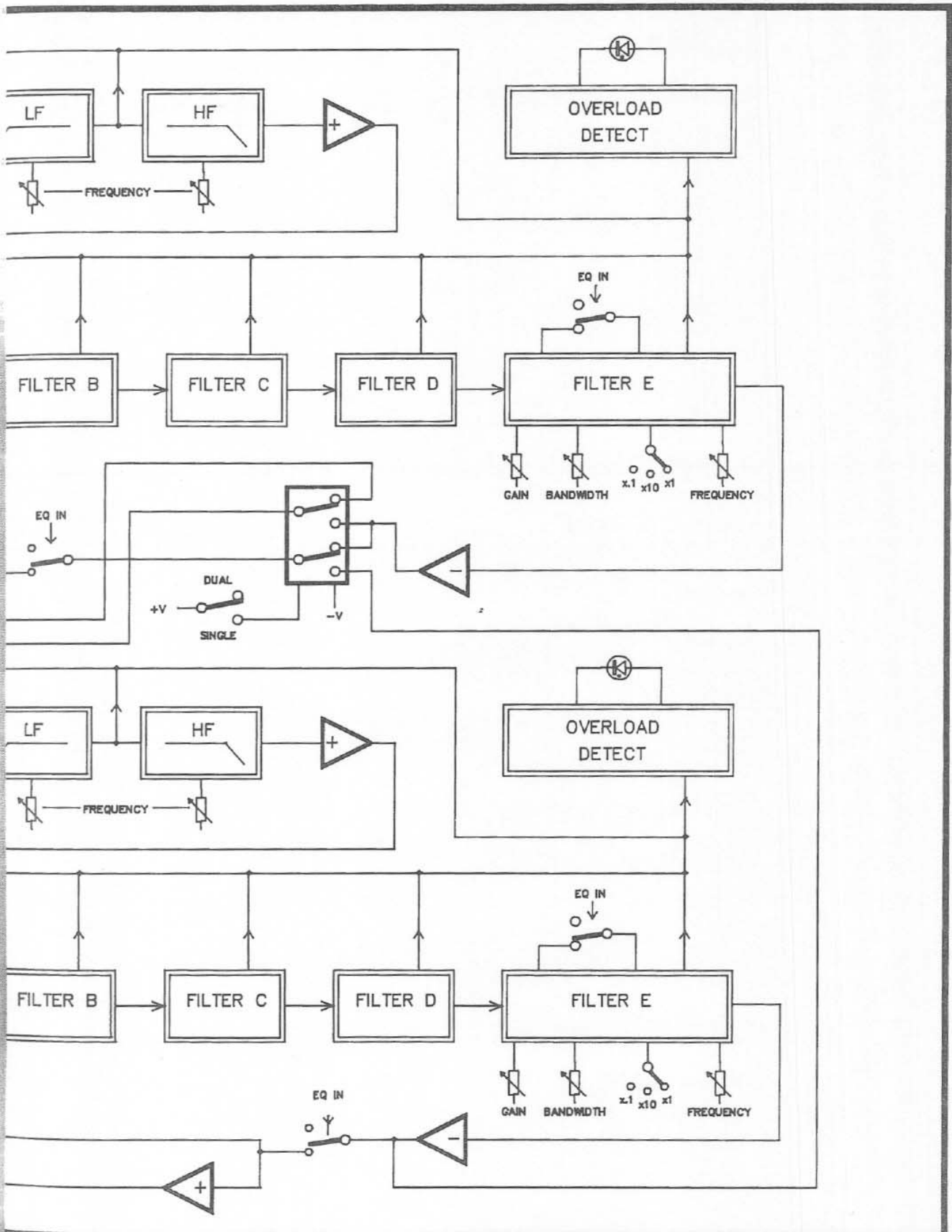


CHANNEL A



CHANNEL B





The Uses of Parametric Equalisers

Parametric equalisers allow the user complete control over the main filter parameters, that is, centre frequency, bandwidth and amplitude or gain (boost or cut). (See Figure 1).

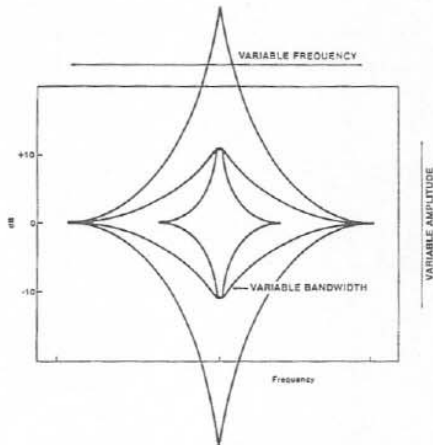


Figure 1 Parametric Filter

The ability to control both the bandwidth and centre frequency of a bank of filters as well as the degree of boost or cut opens up a new dimension in equalisation allowing filter correction and creative filtering capabilities of unparalleled accuracy and precision.

Because Series 400 parametric equalisers allow any filter to be set to any desired frequency over the entire 20Hz to 20kHz audio range the accuracy and versatility of the equalisers is further enhanced and surpasses the performance of other less powerful designs.

Figure 2 shows some examples of the filter characteristics, types of equalisation and general frequency response shaping curves which can be created with the Series 400 equalisers.

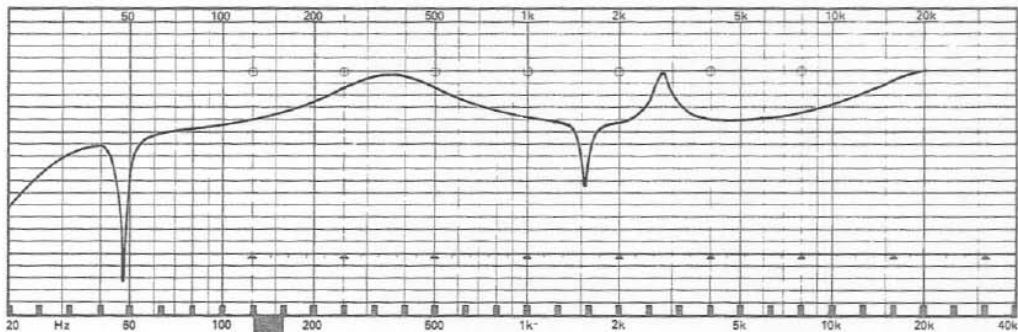


Figure 2 Various Filter Responses

Parametric equalisers may either be used as individual signal processing devices in their own right, or they may be used in conjunction with conventional 1/3 octave equalisers. For example, when equalising a sound system, the 1/3 octave graphic equaliser would be used to provide the overall response shaping, whilst the parametric equaliser would be used to provide detailed narrow band correction or to provide precision notch filtering of narrow band resonances or feedback frequencies.

The ability to accurately 'notch-out' a troublesome frequency or frequency component enables the parametric equaliser to be used to great effect in the recording studio e.g. to 'clean up' sound tracks or remove hum, or other discrete frequency components as well as in 'live' sound reinforcement or Public Address systems e.g. to control feedback without adversely affecting other areas of the frequency spectrum.

Because Series 400 parametric filter stages are cascaded with one another and any filter can be set to any desired frequency, they can be combined to create more than their individually available +15 or -25dB of boost and cut i.e. setting two filters to the same frequency and with the same settings for bandwidth and gain will result in double the amplitude of cut or boost. Figure 3 illustrates this effect by showing the response plot of (a) a filter set to 0.5 octave bandwidth and -25dB of cut and (b) two filters set identically. The increased filtration effect is clearly seen.

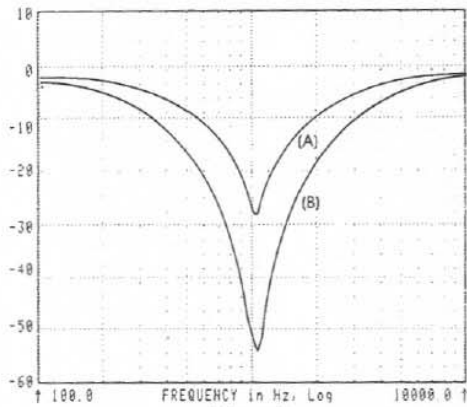


Figure 3
Combination of two identical filters (Q = 0,5)

It is good practice to set all the gain controls to their centre (detent) positions and put each section into by-pass before commencing to create an equalisation curve. This prevents any high gain signals being immediately transmitted into the signal chain thereby preventing corresponding overload of subsequent stages or severe feedback if in a live sound reinforcement situation. It is also good practice to set any unused filter to a medium bandwidth e.g. 0.5. This reduces the possibility of introducing extraneous noise into the signal chain thus raising the noise

floor and reducing the available dynamic range. Whilst in most applications, this is probably of little consequence, in certain critical studio or recording situations, it may be important.

After setting an appropriate equalisation curve, it may be necessary to adjust the overall gain of the equaliser using the gain control facility in order to either avoid overloading the following system stages when considerable gain has been used, or conversely to increase the level of the signal where a substantial degree of cut has been introduced.

Individual parametric filters can also be used to enhance the effect of the incorporated low and high frequency shelf filters. (See Figure 4). Note how the 12dB/octave roll-off filter characteristic has been increased to over 20dB/octave at low frequencies and to over 30dB/octave at high frequency.

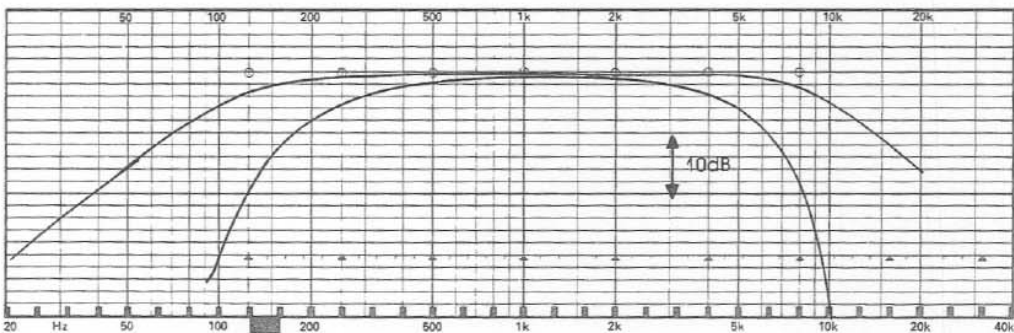


Figure 4
Shelving Filter Characteristics

Filter Q and Bandwidth

The 'Q' of the Series 400 filters is designed to change slightly with the degree of boost or cut applied. This does not compromise the narrowness of a notch filter, but allows the filters to produce a smoother curve when used in combination. (See Figure 5).

The range of control offered by a conventional 1/3 octave combining filter compared with the full range of bandwidth available from a parametric filter is shown in Figure 6 and Figure 7.

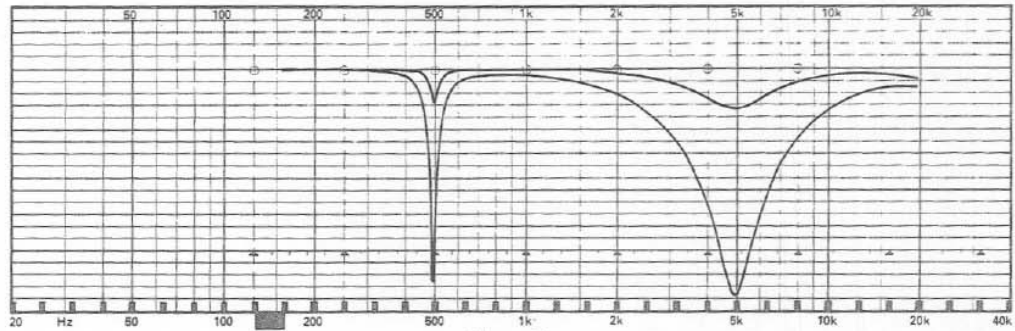


Figure 5

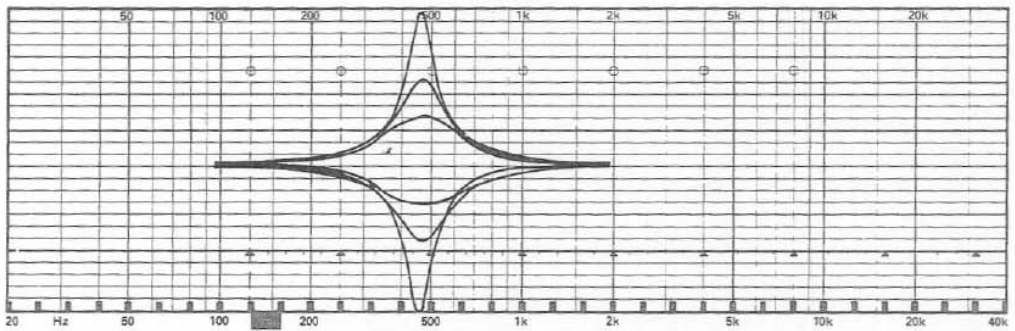


Figure 6 Typical 1/3 Octave Combining Filter

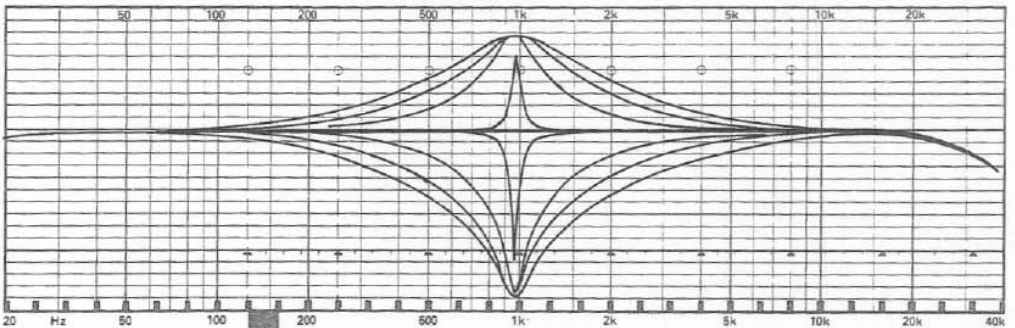


Figure 7 Parametric Filter

Filter Phase and Group Delay Considerations

The fact that any conventional analogue filter, be it a parametric or graphic equaliser filter, will introduce some degree of phase shift should not be neglected when using such devices.

Apart from producing a rapid and substantial change in signal phase, high Q filters also produce a corresponding signal time delay effect. The delay involved can become quite significant at high gain/high Q filter settings. The effect can be important to your particular application and may need to be taken into account. In a similar manner high Q/high gain filters have a natural tendency to ring. Great care should therefore be taken to introduce the minimum amount of boost or cut in any given situation.

Using Analysis

Because the minimum bandwidth of the Series 400 parametric equaliser filters is so narrow ($1/12$ or 0.08 octave) a frequency analyser with equally narrow bandwidth resolution is ideally required to optimally set up the filters. However, a conventional $1/3$ octave Real Time Analyser such as the Klark Teknik DN60 may still be used for many applications. For example, when using the parametric filters to improve the gain before feedback margin in a sound system the $1/3$ octave analyser can be used to determine the frequency of feedback to the nearest $1/3$ octave. The parametric filter section is then tuned to this frequency and set to its narrowest bandwidth with perhaps 6 to 10dB of cut applied. The filter is then switched in and manually swept around the frequency of interest until the feedback component subsides. The depth of the notch filter should then be gradually reduced so as to minimise the effect on adjacent frequencies whilst still maintaining appropriate control over the feedback frequency.

If a narrow band analyser e.g. $1/12$ octave or narrow band FFT or TEF type equipment is available, the feedback frequency may be more accurately resolved and the notch filter set with even greater precision.

In general, acoustic feedback frequencies tend to be very narrow band - making a parametric equaliser an ideal tool to control them. However, the depth of the notch required to provide appropriate control often is not as deep as might be expected - particularly when the prominence of the feedback component is considered. The reasons for this are highly complex, involving the considerations of the Q of the resonance, its phase characteristic and the local acoustic conditions prevailing.

It should be remembered that introducing any degree of cut or boost at a particular frequency or over a relatively narrow range of frequencies will also introduce a corresponding phase shift into the signal path. As feedback is dependent upon both the amplitude and the phase of the signal loop, changing either will reduce the feedback, changing both will result in a corresponding greater potential reduction.

Apart from providing precise narrow band control, the Series 400 equalisers can also create wide band effects - simulating both high and low frequency tone controls, shelving filters or filter shapes suitable for adjusting the low and high frequency response characteristics of certain loudspeaker and compression driver types, where formerly dedicated equalisation filters have been required.

In practice many sound systems require broad band response shaping - e.g. to control excessive low frequency build up or high frequency losses. Broad band control capability enables to carry out these functions, using just two or three filter sections in conjunction with the independent shelf filters leaving 2 or more narrow band filters available to control feedback or other troublesome resonances.

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.

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

The following table of frequency versus effect highlights some further potential uses.

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.

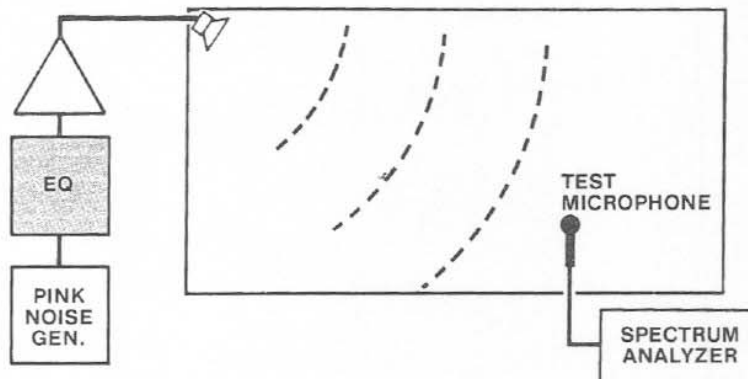
Equalising a Sound System

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



Sound System Equalisation.

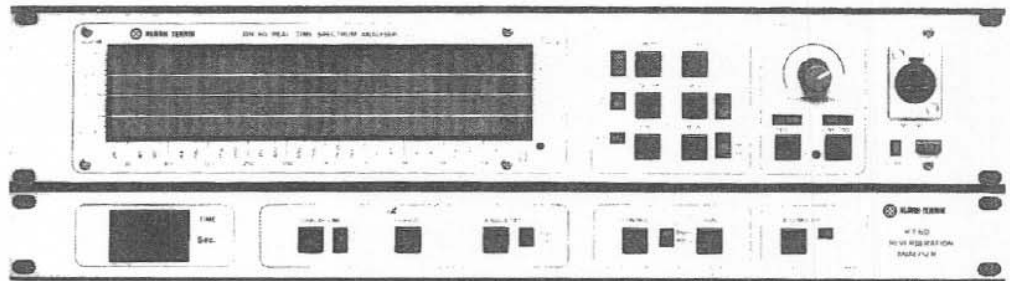
Before beginning to equalise a system, it is a good practice to listen to the "raw" system with speech or music programme. If such signals are distorted then stop and rectify them before attempting to equalise. Another good pre-equalisation test is to use a slow sine-sweep. This can expose a number of problems such as rattles or distortion or poorly controlled room modes or resonances - which pink noise RTA cannot discover. Finally, before equalisation, check the coverage of the system over the 2 to 4kHz band. (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. A warning bell should be ringing if you are using excessive gain settings! 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 a particularly large correction is being applied 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 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.

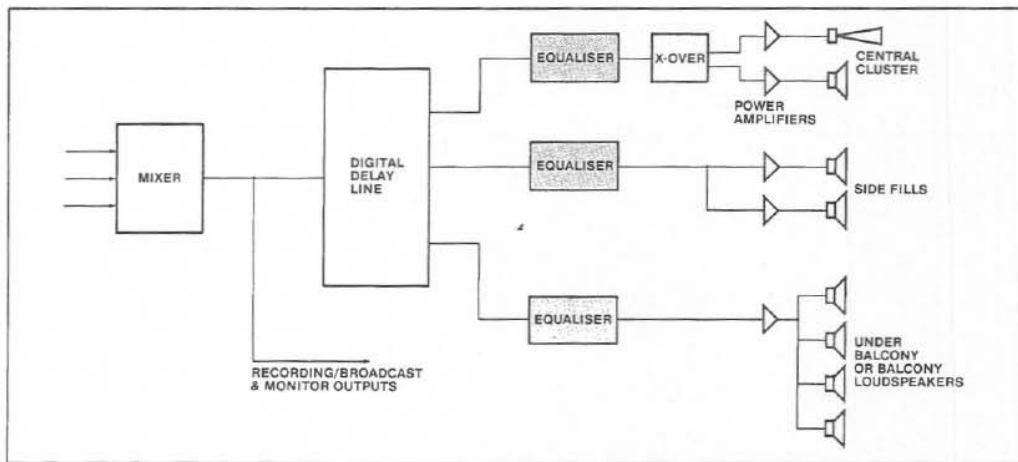


Real Time Spectrum Analyser with Averaging Model DN60/RT60.

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 when 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.....

Attention!

Cables:

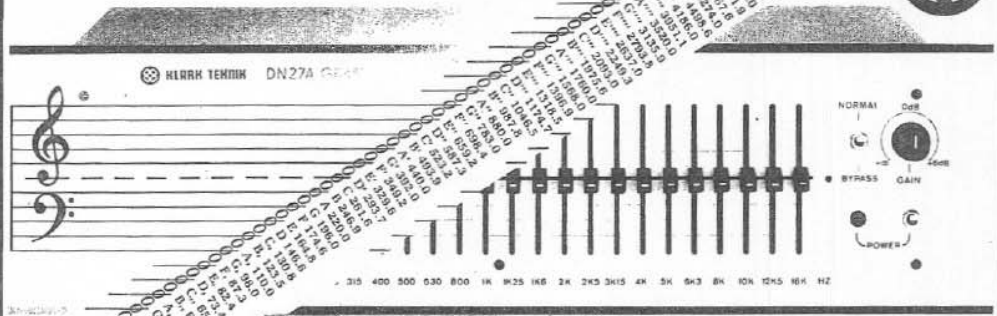
This product should only be used with high quality, screened twisted pair audio cables, terminated with metal bodied 3-pin XLR connectors. The cable shield should be connected to pin 1. Any other cable type or configuration for the audio signals may result in degraded performance due to electromagnetic interference.

Electric Fields:

Should this product be used in an electromagnetic field that is amplitude modulated by an audio frequency signal (20Hz to 20KHz), the signal to noise ratio may be degraded. Degradation of up to 60dB at a frequency corresponding to the modulation signal may be experienced under extreme conditions (3V/m, 90% modulation).

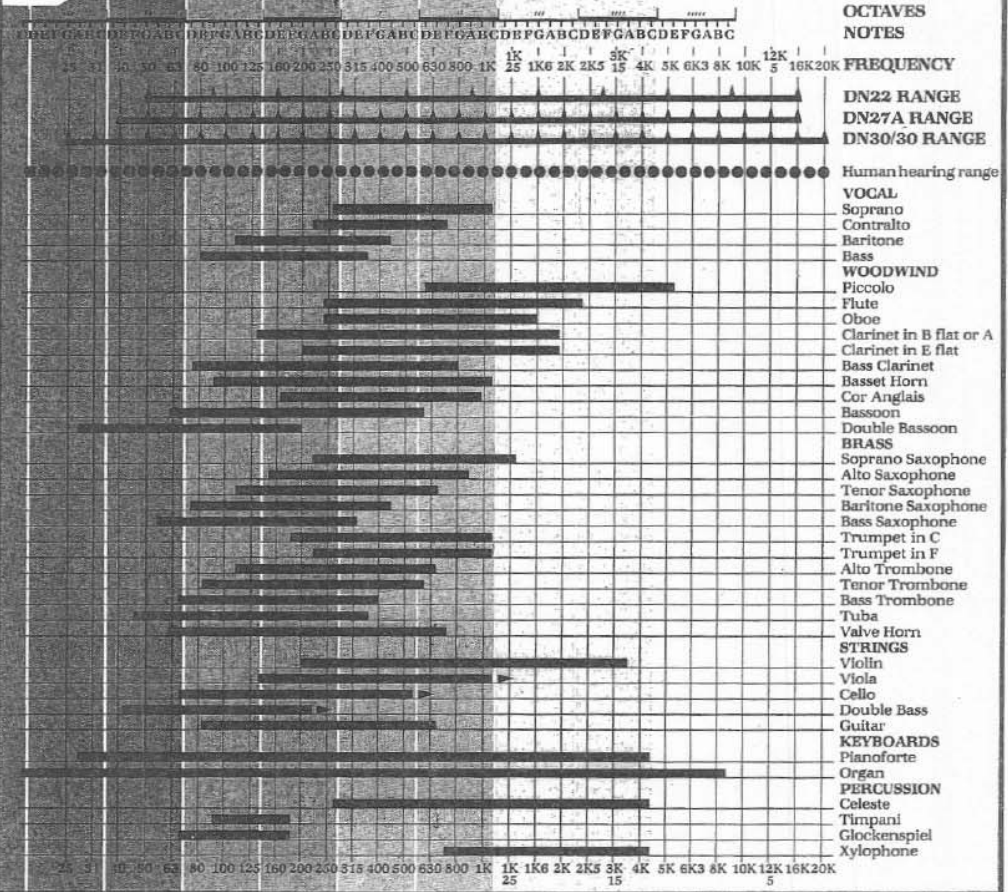
No permanent damage or degradation of performance will be caused by these conditions.

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