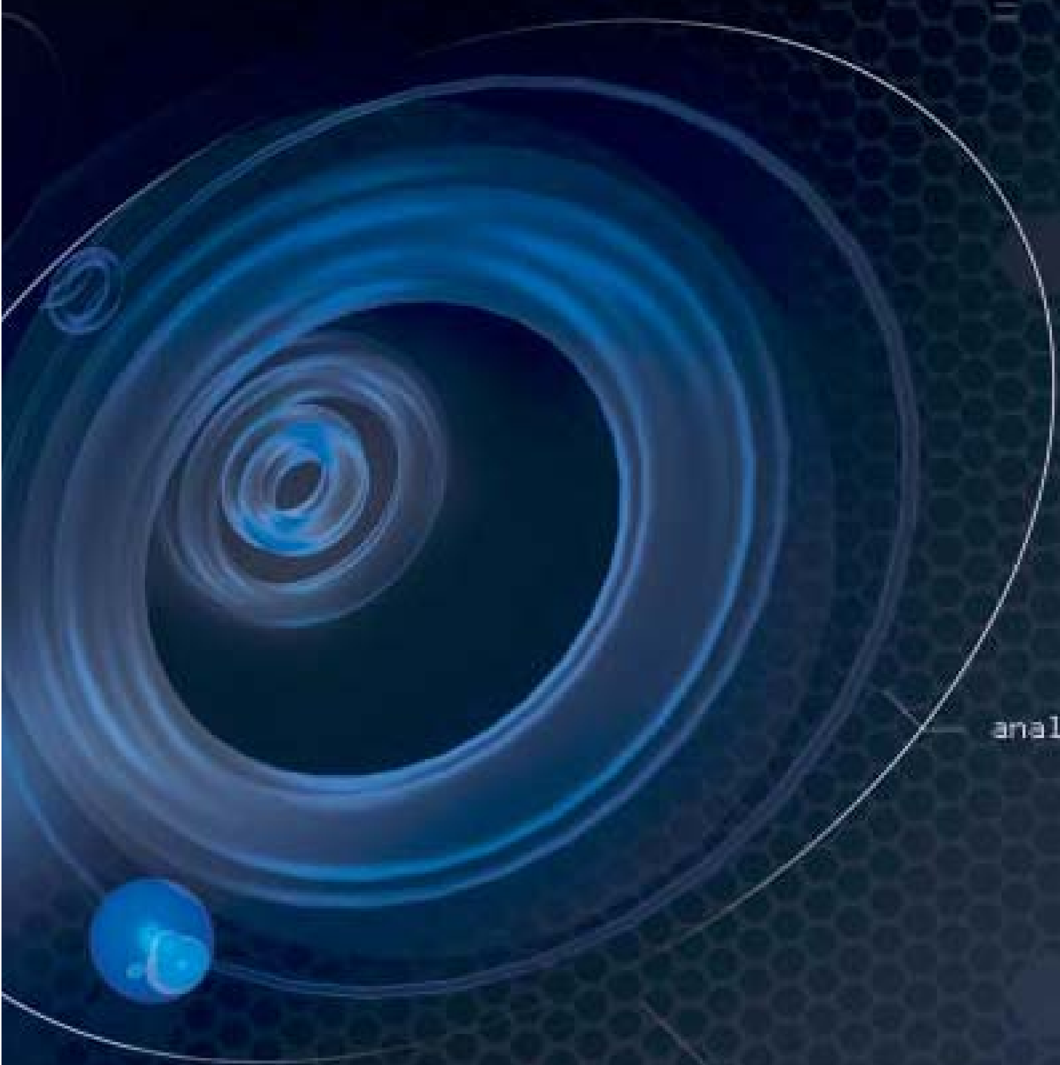
signal processing by definition



analogue excellence

digital innovation

KLARK TEKNIK PRODUCT CATALOGUE

INTRO:

In 1974, brothers Phil and Terry Clarke founded Klark Teknik Research Ltd. In the years immediately following, their innovative approach to design and development allowed them to introduce some truly groundbreaking designs. The world's first digital delay and digital reverb units emanated from their laboratory, and their descendants remain in common usage all over the world to this day. However, it was their concepts for equalisation devices that really changed the world of professional audio. Their earliest designs eventually matured into the world famous DN360 that remains the de facto standard graphic EQ for audio professionals everywhere.

Today, Klark Teknik continues to uphold the original vision of the Clarke brothers: innovation in design, followed by uncompromising dedication to engineering and sonic quality. Most of our units are still made and tested by hand, a time consuming and labour intensive process that remains the only method by which we can maintain the quality that our customers expect. Uniquely in its field, Klark Teknik also provides the customer with an opportunity to invest in leading-edge equipment with an extraordinary working lifespan and unrivalled retained value. Global support for our products is readily available from the factory in Kidderminster, from our international distributor network covering more than 50 countries, and direct from the Klark Teknik website.

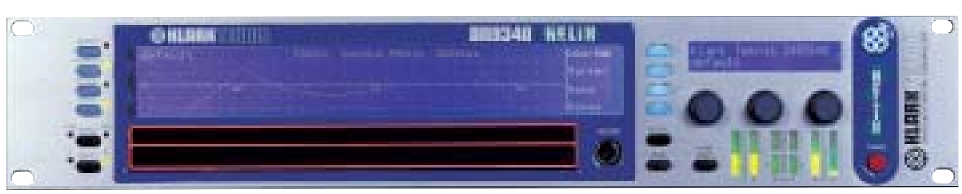
CONTENTS:

DN9340	digital equaliser	page 3
DN9344	digital slave equaliser	page 5
DN9848	loudspeaker processor	page 7
DN1248 *Plus*	mic splitter	page 9
DN360	dual channel 30 band 1/3 octave graphic equaliser	page 11
DN300	single channel 30 band 1/3 octave graphic equaliser	page 13
DN332	dual channel 16 band 2/3 octave graphic equaliser	page 15
DN410	dual channel 5 band parametric equaliser	page 17
DN500 *Plus*	dual compressor/limiter expander	page 19
DN504 *Plus*	quad compressor limiter	page 21
DN514 *Plus*	quad auto gate	page 23
DN7453	user configurable digital audio delay line	page 25
DN7454	user configurable digital audio delay line	page 27
DN6000	audio analyser	page 29
DN800	active crossover	page 31
DN1414	di module	page 33
DN100	active di	page 35
Accessories	factory options	page 37
FAQ's	frequently asked questions	page 38
Line drawings		page 43

The concept of an equalisation device which allows the user to not only select from a menu of various EQ types but also to integrate them with each other in any combination is one that could only have come from Klark Teknik. The DN9340 Helix has, within a year of launch, become the new world standard for digital equalisation, finding applications in live sound, broadcast, film and TV post-production and recording. At the heart of Helix is the fact that it sounds incredible – the most common remark from first-time users is 'it sounds just like a great analogue EQ!' Continuous software and hardware development has added a slew of new features and retro-fittable factory options including auto-EQ linking with DN6000 and now AES/EBU digital inputs and outputs.

For the uninitiated, the DN9340 Helix is a dual channel, 2RU, all-digital equalisation unit that simultaneously offers a five-mode 31-band graphic EQ, twelve bands of full parametric EQ, four configurable filters (HPF, LPF, hi shelf, lo shelf and notch) and two bands of true dynamic EQ (T-DEQ) per channel. Control is provided by both a touch strip and rotary controls, full input and out metering is provided plus dedicated meters for the T-DEQ function. A large, bright LCD display provides visual reference for all functions and the entire menu structure is simple and intuitive. Helix can also be linked to Midas Heritage and Legend consoles to provide an 'auto-solo' function: this allows any channel of a connected Helix system to be edited from the control surface of a single master Helix unit simply by activating the appropriate PFL function on the console.

DN9340: digital equaliser



Architect's & Engineer's Specification

The Digital Equaliser shall provide two audio channels in a standard 2U 19" rack mount chassis.

Each audio channel shall include: Input gain, delay up to one second; up to four filters, two dynamic EQ bands, up to 12 parametric EQ bands and a 31 band graphic EQ.

All delay times shall be set in milliseconds and microseconds, or in distance units (metric and imperial) with a temperature compensation facility.

The filters shall be selectable from notch, high pass, low pass, high shelf and low shelf types. The Low pass and high pass filters shall have selectable slopes of 6, 12, 18, 24, 36 and 48 dB per octave and the high and low shelf filters shall have selectable slopes of 6 and 12 dB per octave and ±12 dB of gain.

The dynamic EQ sections shall have independent high and low level thresholds and gain and be selectable from parametric EQ, or high shelf or low shelf filter types. The parametric EQ shall provide proportional-Q, constant-Q and symmetrical-Q responses. The dynamic EQ sections shall also have independent attack and release times.

The parametric EQ sections shall have up to 12 dB of cut or boost and a Q value variable from 0.4 to 20. The parametric EQ shall provide proportional-Q, constant-Q and symmetrical-Q responses.

The graphic EQ section shall provide 31 bands on standard frequencies defined in BS EN ISO 266:1997. Proportional-Q, constant-Q and symmetrical-Q responses shall be provided as well as emulations of Klark Teknik DN27 and DN360 Graphic Equalisers.

Each Digital Equaliser shall meet or exceed the following performance specifications:

Frequency response: ±0.3 dB (20 Hz to 20 kHz)
Distortion @ +4 dBu: <0.01% (20 Hz to 20 kHz)
Dynamic Range: 115 dB (20 Hz to 20 kHz unweighted)

All audio inputs and outputs shall be electronically balanced and use XLR connectors. A 480 x 64 graphic LCD shall be provided to display a graphical representation of the equaliser section responses. All parameters shall be displayed and adjusted via a 20 x 2 alphanumeric LCD display, three rotary encoders and individual menu buttons for each equaliser section. A dual touchstrip shall be provided for use with the graphic LCD to allow the selection of graphic EQ band and gain, and centre or corner frequency for filters, and dynamic and parametric EQ. The graphic and alphanumeric LCDs and the dual touchstrip shall have LED backlights.

There shall be provision for 32 system memories and 32 factory presets with a security lock-out feature. There shall also be a security lock-out feature that is enabled when the unit is under remote control.

The Digital Equaliser shall be provided with RS-232 ports on the front and rear panels and RS-485 ports on the rear panel. The RS-485 ports and front panel RS-232 port shall be provided for remote control from a master Digital Equaliser or a PC and additionally the front panel RS-232 port shall also provide the facility to download software updates and preset memories into the Digital Equaliser. The rear panel RS-232 port shall be provided for remote control from Midas Heritage and Legend mixing consoles.

The unit shall be capable of operating from a 100 to 240V, 50 to 60 Hz a.c. power source.

The Digital Equaliser shall be the Klark Teknik model DN9340 and no alternative option is available.

Technical Specification

Inputs	**Two**
Type	Electronically balanced (pin 2 hot)
Impedance (Ohm)	20k
Common Mode Rejection	>80dB @ 1 kHz
Outputs	**Two**
Type	Electronically balanced (pin 2 hot)
Maximum Level	+21dBu into >2k
Performance	
Frequency response (20Hz to 20kHz)	0.3 dB with all filters and EQ flat
Distortion (THD+N) @ +4dBu (20Hz to 20 kHz)	<0.01%
Dynamic range (20Hz-20kHz unweighted)	115 dB
Processing (Per Channel)	
Input Gain	+12dB to -40dB in 0.1dB steps plus Off
Delay	0-1 second (342.25 m or 333'10" at 20C in 20.8us steps)
Filters	4 Filters (max)
Types	Low Pass, High Pass, Low Shelf, High Shelf, Notch
Dynamic EQ	2 Bands (max)
Range	±12dB
Responses	Proportional, Constant, Symmetrical
Parametric EQ	12 Bands (max)
Range	±12dB
Responses	Proportional, Constant, Symmetrical
Graphic EQ	31 Bands On ISO standard frequencies
Range	±12dB
Responses	Proportional, Constant, Symmetrical, DN27, DN360
Power Requirements	
Voltage	90 V to 250 V a.c. 50/60 Hz
Consumption	<60W
Terminations	
Audio inputs/outputs	3 pin XLR
RS-485 inputs/outputs	3 pin XLR
RS-232	8 pin Mini-DIN socket (front) 9 pin D-type (rear)
Power	3 pin IEC
Dimensions	
Width	483 mm (19inch)
Height	88 mm (3.5 inch) - (2RU)
Depth	303 mm (12 inch)
Weight	
Net	6kg
Shipping	8kg

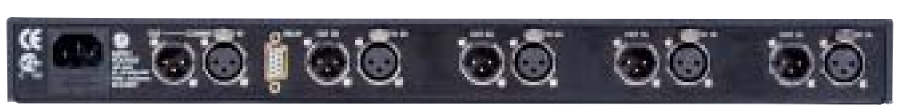
Trade Descriptions Act:
Due to the company policy of continuing improvement, we reserve the right to alter these specifications without prior notice. E&OE.

KLARK TEKNIK
SIGNAL PROCESSING BY DEFINITION

DN9344 is a fantastic example of how clever digital design can make products smaller and lighter without sacrificing functionality. DN9344 Helix slave is actually TWO complete DN9340 Helix units in just a single rackspace device, providing four discrete or two pairs of stereo-linked channels of multi-configurable EQ, that can be controlled singly or as part of a larger system from a single DN9340 Helix Master unit, or by the free proprietary software developed by KT for the purpose. Up to 64 channels can be controlled from one master unit. Perfect for installations, it is also fitted with contact closures to allow for memory recall by a mechanical device.

For full operational and technical details on the extraordinary Klark Teknik Helix range, please visit www.klarkteknik.com and download the full Helix brochure, or contact your nearest Klark Teknik dealer.

DN9344: digital slave equaliser



Architect's & Engineer's Specification

The Digital Slave Equaliser shall provide four audio channels grouped as two linkable pairs in a standard 1U 19" rack mount chassis.

Each audio channel shall include: Input gain, delay up to one second; up to four filters, two dynamic EQ bands, up to 12 parametric EQ bands and a 31 band graphic EQ.

All delay times shall be set in milliseconds and microseconds, or in distance units (metric and imperial) with a temperature compensation facility.

The filters shall be selectable from notch, high pass, low pass, high shelf and low shelf types. The Low pass and high pass filters shall have selectable slopes of 6, 12, 18, 24, 36 and 48 dB per octave and the high and low shelf filters shall have selectable slopes of 6 and 12 dB per octave and ±12 dB of gain.

The dynamic EQ sections shall have independent high and low level thresholds and gain and be selectable from parametric EQ, or high shelf or low shelf filter types. The parametric EQ shall provide proportional-Q, constant-Q and symmetrical-Q responses. The dynamic EQ sections shall also have independent attack and release times.

The parametric EQ sections shall have up to 12 dB of cut or boost and a Q value variable from 0.4 to 20. The parametric EQ shall provide proportional-Q, constant-Q and symmetrical-Q responses.

The graphic EQ section shall provide 31 bands on standard frequencies defined in BS EN ISO 266:1997. Proportional-Q, constant-Q and symmetrical-Q responses shall be provided as well as emulations of Klark Teknik DN27 and DN360 Graphic Equalisers.

Each Digital Slave Equaliser shall meet or exceed the following performance specifications:

Frequency response: ±0.3 dB (20 Hz to 20 kHz)
Distortion @ +4 dBu: <0.01% (20 Hz to 20 kHz)
Dynamic Range: 115 dB (20 Hz to 20 kHz unweighted)

All audio inputs and outputs shall be electronically balanced and use XLR connectors.

There shall be two three-character starburst LED displays per pair of audio channels for displaying recalled memory, communications channel setting and remotely-set user information. There shall also be physical write-on strips for each pair of audio channels plus an additional one for the unit as a whole.

There shall be provision for 32 system memories and 32 factory presets.

The Digital Slave Equaliser shall be provided with an RS-232 port on the front panel and RS-485 ports on the rear panel. The RS-485 ports and RS-232 port shall be provided for remote control from a master Digital Equaliser or a PC and additionally the front panel RS-232 port shall also provide the facility to download software updates and preset memories into the Digital Slave Equaliser. There shall also be a rear panel relay contact closure port to allow the recall of specific preset memories.

The unit shall be capable of operating from a 100 to 240V, 50 to 60 Hz a.c. power source.

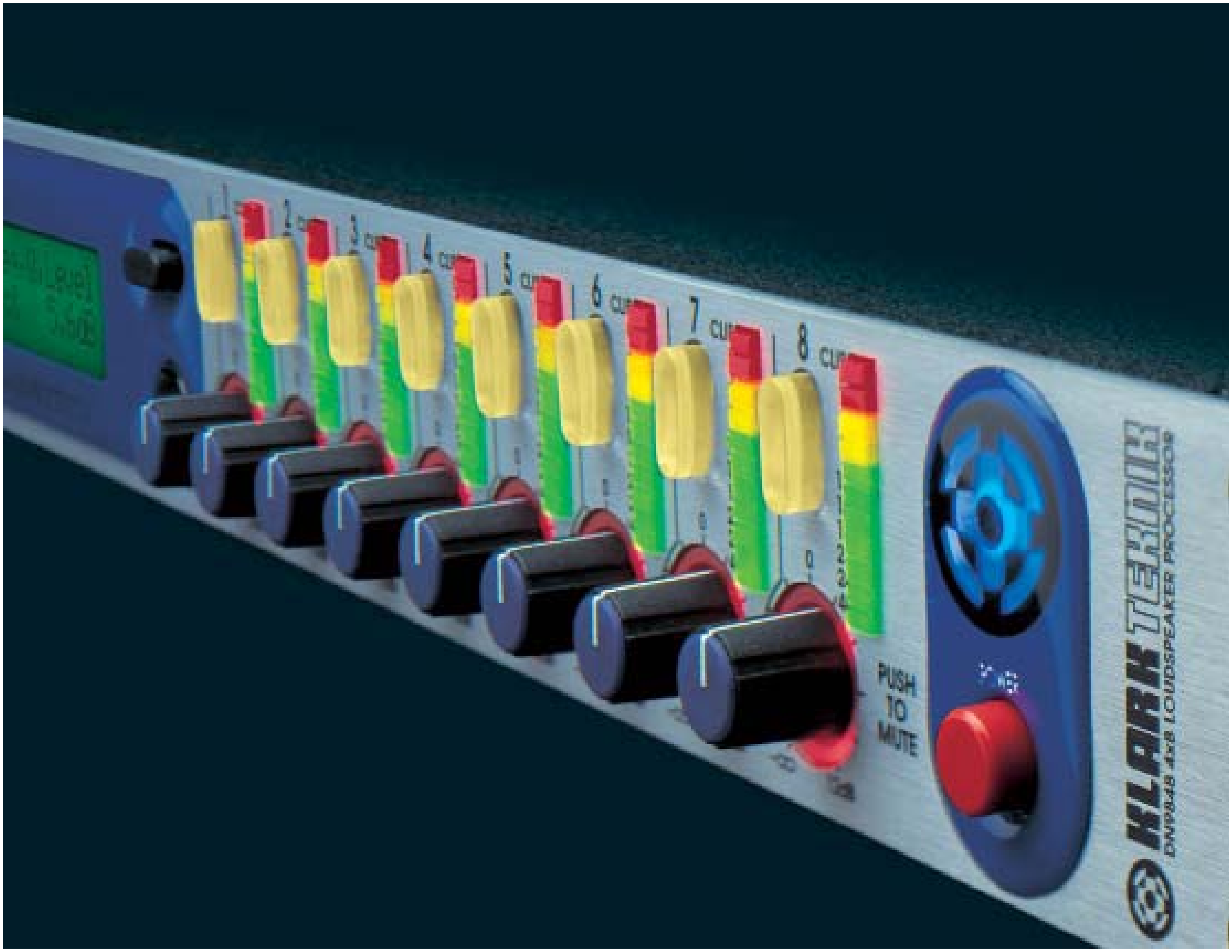
The Digital Slave Equaliser shall be the Klark Teknik model DN9344 and no alternative option is available.

Technical Specification

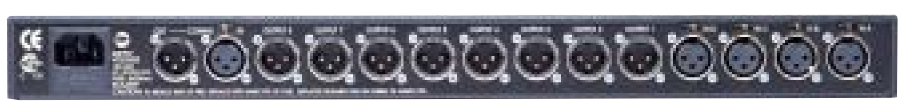
Inputs	**Four**
Type	Electronically balanced (pin 2 hot)
Impedance (Ohm)	20k
Common Mode Rejection	>80dB @ 1 kHz
Outputs	**Four**
Type	Electronically balanced (pin 2 hot)
Maximum Level	+21dBu into >2k
Performance	
Frequency response (20Hz to 20kHz)	0.3 dB with all filters and EQ flat
Distortion (THD+N) @ +4dBu (20Hz to 20 kHz)	<0.01%
Dynamic range (20Hz-20kHz unweighted)	115 dB
Processing (Per Channel)	
Input Gain	+12dB to -40dB in 0.1dB steps plus Off
Delay	0-1 second (342.25 m or 333'10" at 20C in 20.8us steps)
Filters	4 Filters (max)
Types	Low Pass, High Pass, Low Shelf, High Shelf, Notch
Dynamic EQ	2 Bands (max)
Range	±12dB
Responses	Proportional, Constant, Symmetrical
Parametric EQ	12 Bands (max)
Range	±12dB
Responses	Proportional, Constant, Symmetrical
Graphic EQ	31 Bands On ISO standard frequencies
Range	±12dB
Responses	Proportional, Constant, Symmetrical, DN27, DN360
Power Requirements	
Voltage	90 V to 250 V a.c. 50/60 Hz
Consumption	<60W
Terminations	
Audio inputs/outputs	3 pin XLR
RS-485 inputs/outputs	3 pin XLR
RS-232	8 pin Mini-DIN socket (front)
Relay Socket	9 pin D-type (rear)
Power	3 pin IEC
Dimensions	
Width	483 mm (19 inch)
Height	44 mm (1.75 inch) - (1RU)
Depth	287 mm (12 inch)
Weight	
Nett	4kg
Shipping	6kg

Trade Descriptions Act:
Due to the company policy of continuing improvement, we reserve the right to alter these specifications without prior notice. E&OE.

KLARK TEKNIK
SIGNAL PROCESSING BY DEFINITION



DN9848: loudspeaker processor



Digital loudspeaker system control has been one of the fastest developing areas of signal processing in recent years. This is principally because it allows designers to combine a number of key control functions within a single device, thereby lowering overall costs and adding convenience. Unfortunately, in many cases the relevance of the audio performance of the device has been overshadowed by the ‘bells and whistles’ functionality of the unit, ultimately somewhat defeating the object of the exercise. With the Klark Teknik DN9848, no compromise has been made in either the feature set or the audio performance.

Unique in that it is the only such device fitted with XLRs to pack four inputs and eight outputs into a single rackspace, DN9848 brings a new level of flexibility to system control whether for live production or installation use. Since there is no preset routing within the device, it can be easily pre-programmed to perform almost any system-control task. Limiters and compressors on all outputs plus compressors on all inputs provide ultimate speaker control and protection, whilst no less than ninety-six bands of fully parametric EQ allow for both room and system equalisation. Best of all, there’s enough processing power onboard to allow every function to be available all the time, regardless of what is already in use.

The proprietary Preset Editor software (utilising Microsoft Excel) allows simple up-and-down-loading of system parameters into the FLASH memory locations, as well as storage and transmission of system information.

Need a new system configuration to be loaded into a unit on the other side of the world? No problem, put the information into Preset Editor and e:mail it to wherever it needs to go, where it can then be uploaded into the unit in seconds.

Online remote control and offline system creation is provided via the Stardraw Audio system. Whilst it allows intuitive control of every function of an individual unit or units, it also (and uniquely) allows inputs and outputs to be assigned to control groups. These groups can then be made to control any parameter of the unit or the system – muting, delay, EQ, dynamics, speaker or room zones, whatever you need. Simple screens with easy-access controls make for quick adjustments, and entire system set-ups can then be stored as a computer file.

All supporting software programs are available free of charge from the Klark Teknik website (www.klarkteknik.com). A high-quality RS-232 to RS-485 converter (the LBB-485) is also available. This allows computer connection to DN9848s at greater range than RS-232 will support, and is fitted with standard XLR connectors to interface with regular audio cables.

Architect's & Engineer's Specification

The Loudspeaker Processor shall provide four input channels and eight output channels with configurable routing in a standard 1U 19" rack mount chassis.

Each input channel shall include: input gain control, delay up to one second; twelve parametric EQ stages (+6dB boost, -18dB cut); a compressor.

Each output channel shall include: configurable routing; delay up to 300 milliseconds; all-pass and phase correction filters and low and high pass crossover filters with slopes of 6, 12, 18, 24, 36 and 48 dB per octave and options of Linkwitz-Riley, Butterworth and Bessel characteristics where appropriate; six parametric EQ sections with up to 12 dB of cut or boost (optionally two of these stages are configurable as low frequency and high frequency shelf filters respectively); a phase invert function; an output level control a compressor and a limiter.

All delay times shall be set in milliseconds and microseconds, or in distance units (metric and imperial) with a temperature correction facility.

Each Loudspeaker Processor shall meet or exceed the following performance specifications:

Frequency response
+/- 0.3dB (20Hz to 20kHz)

Distortion (THD+N)
<0.02% @ 1kHz, +8dBu

Dynamic Range
>114dB (20Hz to 20kHz unweighted)

All inputs and outputs shall be electronically balanced and use XLR connectors.

All parameters shall be displayed and adjusted via an alphanumeric LCD display, three rotary encoders and individual menu buttons for each input and output channel.

The Loudspeaker Processor shall be provided with an RS-232 and RS-485 ports for remote control and software updates*.

There shall be provision for six user memories and in addition 32 system memories and 99 factory presets with a security lock-out feature.

There shall also be a security lock-out feature that is enabled when the unit is under remote control.

The unit shall be capable of operating from a 90 to 250V, 50 to 60Hz AC power source.

The Loudspeaker Processor shall be the Klark Teknik model DN9848 and no alternative option is available.

** Software updates via RS-232 port only.*

Technical Specification

Inputs	**Four**
Type	Electronically balanced (Pin 2 Hot)
Impedance (□)	
Balanced	20k
Unbalanced	10k
Common Mode Rejection	>80dB @ 1kHz
Maximum level	+ 21dBu
Audio Outputs	**Eight**
Type	Electronically Balanced (Pin 2 Hot)
Minimum load impedance	56□ /20nF
Source impedance	56□
Maximum level	+ 21dBu into > 2k□
Performance	
Frequency response	(20 Hz to 20 kHz) +/- 0.3dB with all filters and EQ flat
Distortion (THD+N)	<0.02% @ 1kHz, +8 dBu
Dynamic range	(20 Hz to 20 kHz unweighted) >114dB
Input Processing (per channel)	
Input gain	+12dB to -40dB in 0.1 dB steps plus Off
Parametric EQ 1-12	
Frequency range:	20Hz to 20kHz in 21 steps per octave Boost/cut: +6/-18dB in 0.1dB steps Q: 3.0 to 0.08
Compressor	
Threshold:	+21dBu to - 10dBu in 0.1dB steps
Attack:	40us to 100ms Insert: On/Off Release: 10ms to 2000ms Ratio: 1:1 to 5:1 Knee: Hard/Soft
Delay	0 to 1 second 342.25 m or 1122' 10" at 20(C) in 20.8us steps
Output Processing (per channel)	
Routing	Route from inputs: A, B, C, D, A+B, C+D, A+B+C+D
Delay	0 to 300ms (102.68 m or 333' 10" at 20(C)in 5.02 us steps)
Phase correction filters	0˚ to 180˚ in 5˚ steps
All pass filter	1st and 2nd order
Low pass filter	frequency range 20Hz to 20kHz in 21 steps per octave. Supported configurations are:- Butterworth (6dB/Oct, 12dB/Oct, 18dB/Oct, 24dB/Oct, 36dB/Oct, 48dB/Oct) Linkwitz-Riley (12dB/Oct, 24dB/Oct) Bessel (12dB/Oct, 18dB/Oct, 24dB/Oct, 36dB/Oct, 48dB/Oct)
High pass filter	frequency range 20Hz to 20kHz in 21steps per octave. Supported configurations are:- 12dB/Oct Peaking 24dB/Oct Peaking Butterworth (6dB/Oct, 12dB/Oct, 18dB/Oct, 24dB/Oct, 36dB/Oct, 48dB/Oct) Linkwitz-Riley (12 dB/Oct, 24dB/Oct) Bessel (12dB/Oct, 18dB/Oct, 24dB/Oct, 36dB/Oct, 48dB/Oct) Peaking Filter Boost: 0dB to +6dB in 0.1dB steps.
Parametric EQ 1/ Low shelf filter	frequency range 20Hz to 20kHz in 21steps per octave. Boost/cut: +12/-12dB in 0.1dB steps Parametric EQ Q: 3.0 to 0.08 Shelf slope: 6dB/Oct and 12dB/Oct
Parametric EQ 2-5	frequency range 20Hz to 20kHz in 21steps per octave. Boost/cut: +12/-12dB in 0.1dB steps Q: 3.0 to 0.08
Parametric EQ 6/ Hi shelf filter	frequency range 20Hz to 20kHz in 21steps per octave. Boost/cut: +12/-12dB in 0.1dB steps Parametric EQ Q: 3.0 to 0.08 Shelf slope: 6 dB/Oct and 12dB/Oct
Polarity invert	Normal/invert
Output gain	+12dB to -40dB in 0.5dB steps plus Off
Look-ahead limiter	Threshold: +21dBu to - 10dBu in 0.5dB steps Release: 10ms to 1000ms Knee: Hard/Soft
Compressor	Threshold: +21dBu to - 10dBu in 0.1dB steps Attack: 40us to 100ms Insert: On/Off Release: 10ms to 2000ms Ratio: 1:1 to 5:1 Knee: Hard/Soft
Mute	On/off
Terminations	
Audio inputs/outputs	3 pin XLR
RS-485 inputs/outputs	3 pin XLR
RS-232	8 pin Mini-DIN socket
Power	3 pin IEC
Power Requirements	
Voltage / Consumption	90 to 250V a.c @ 50/60Hz < 60VA
Dimensions	
Height	44 mm (1.75 inch) - (1U)
Width	483mm (19 inch)
Depth	287mm (12 inch)
Weight	
Nett	4kg
Shipping	6kg

Trade Descriptions Act:
Due to the company policy of continuing improvement, we reserve the right to alter these specifications without prior notice. E&OE.

KLARK TEKNIK
SIGNAL PROCESSING BY DEFINITION

Back in 1999, Klark Teknik responded to market demand by producing exactly what our customers had been requesting for years – a roadworthy and flexible active signal splitter system with the superlative audio performance they'd expect from Klark Teknik. So, DN1248 was born, and also delivered with a host of features simply not found in any comparable unit.

We specified an internal power supply (with a factory option of dual auto-switching PSUs at very low cost), more inputs and outputs per unit than any competitor, a uniquely flexible solo buss system, and a Midas Heritage-series microphone preamp, all made available at a per-channel price appreciably lower than any comparable device. These features have made DN1248 one of our most successful units worldwide, but still some customers were not satisfied. So, once again we have responded to market demand, hence the introduction of the new DN1248 Plus.

This unit takes all the operational and cost advantages of the original, and adds a duplicate set of inputs and outputs to the rear panel. This adds a further dimension of flexibility, and allows users to upgrade their existing systems with the minimum of re-wiring. Add in the regular KT 5-year international factory warranty, and you have a unit that exceeds the expectations of even the most demanding users.

DN1248 *Plus* : mic splitter

Architect's & Engineer's Specification

The Mic Splitter shall provide 12 discrete audio channels in a standard 3U 19" rack mount chassis.

Each channel shall have a microphone preamplifier, two transformer-isolated outputs, and two electronically balanced outputs. Optionally, all outputs may be transformer-isolated.

Each channel shall also provide separate +30dB boost and -15dB pad switches, switchable +48V phantom power, an earth lift function and a soloing facility.

The Mic Splitter shall have a headphone amp to allow the monitoring of soloed audio channels.

The headphone amplifier shall have a headphone jack socket for the headphones, a rotary level control for the headphones output and a seven-segment LED bargraph for monitoring the soloed signal level.

Each Mic Splitter shall meet or exceed the following performance specifications:

Electronically Balanced Outputs
Distortion (THD+N)
< 0.01% @1 kHz, +4 dBu
Frequency response
+0 / -0.5 dB (20 Hz to 20 kHz)
Transformer Balanced Outputs
Distortion (THD+N)
<0.04% @1 kHz , +4 dBu
Frequency response
+0 / -1.0 dB (20 Hz to 20 kHz)

The audio connections for each of the twelve audio channels shall be via 3-pin XLR style connectors -

Inputs :
two parallel-connected female XLR connectors (one on the front panel and the other on the rear panel).
Transformer Outputs:
one male XLR connector on the front panel for each output.
Electronic Outputs:
one output with one front and one rear panel male XLR connectors, one output with one rear panel male XLR connector only.

The rear panel input XLRs and output XLRs shall be mounted on three removable plates, and be grouped as one panel of input connectors and two panels of output connectors

All inputs and outputs shall be made available internally on PCB-mounted terminal strips to enable users to retrofit alternative rear panel connector configurations.

The unit shall be capable of operating from a 90 to 250V, 50 to 60 Hz AC power source. The unit shall have the option of dual redundant power supplies.

The Mic Splitter shall be the Klark Teknik model DN1248 plus and no alternative option is available.

Technical Specification

Inputs

	two parallel-connected female XLR connectors (one on the front panel and the other on the rear panel
Input impedance	> 2kΩ
CMRR	> -100 dB @ 100 Hz to 10 kHz
Equivalent input noise	< - 100 dBm @ unity gain
Connectors	3 pin female XLR (external) 3 way terminal strip (internal)
Signal present level	> - 25dBu
Signal clip level	> + 21dBu

Outputs

Electronically balanced	one output with one front and one rear panel male XLR connectors, one output with one rear panel male XLR connector only
Source impedance	50Ω
Min Load	600Ω
Max level	+ 21dBu @ 1kHz
Connectors	3 pin male XLR (external) 3 way terminal strip (internal)
Transformer balanced & isolated	one male XLR connector on the front panel for each output
Source impedance	70Ω
Min Load	600Ω (-3dB level loss into 200Ω)
Max level	+ 18dBu @ 1kHz
Connectors	3 pin male XLR (external) 3 way terminal strip (internal)

Performance

Electronically balanced outputs	
Frequency response	20Hz to 20kHz + 0 / - 0.5dB
Distortion (THD+N)	< 0.01 % @1kHz, +4dB
Transformer balanced & isolated outputs	
Frequency response	20Hz to 20kHz + 0 / - 1.0dB
Distortion (THD+N)	< 0.04 % @1kHz, +4dB

Terminations

Audio Inputs / Outputs	3 pin XLR
Power	3 pin IEC

Power Requirements

90 to 250V AC, 50/60Hz < 60W

Dimensions

Height	132 mm (5.2 inches) - (3U)
Width	483 mm (19 inches)
Depth	300 mm (12 inches)

Weight

Nett	7.4 kg
Shipping	8.4 kg

Options

*Dual power supply
*All outputs transformer balanced

** All options are non retrofittable and must be specified with order*

Trade Descriptions Act:
Due to the company policy of continuing improvement, we reserve the right to alter these specifications without prior notice. E&OE.

KLARK TEKNIK
SIGNAL PROCESSING BY DEFINITION

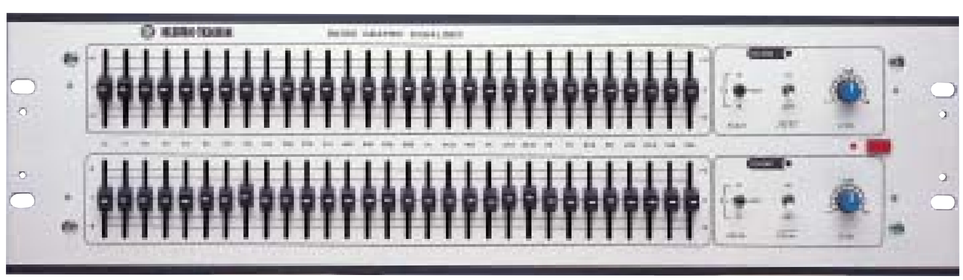
Now approaching some 20 years in continuous production, the DN360 dual graphic EQ has achieved ubiquity in professional audio circles. With nearly 30,000 units in the field worldwide, and the lowest failure rate of any comparable product, the DN360 even today remains the dual graphic EQ of choice in most instances.

Why is it still so popular, especially in this menu-driven digital age? The answer is threefold: instant access, total reliability, and the great sound of the best analogue EQ money can buy. One of the main contributors to DN360s audio performance is its variable 'Q' design, meaning that the 'Q' value of any fader becomes narrower as the fader approaches maximum cut or boost.

So rather than a collection of unconnected cuts and boosts (as provided by a 'constant-Q' device) the DN360 user is rewarded with a flowing, musical response with any overall fader setting. This proprietary design also allows every fader to function correctly regardless of the relative position of its neighbours, another design fault inherent in 'constant-Q' units. Each channel also features an 18dB/octave high-pass filter set at 30Hz to eliminate subsonic 'rumble' if required, plus an EQ in/out switch and an overall 6dB / 12dB fader scale switch for normal or high fader resolution.

A design classic, still made as only KT know how.

DN360: dual channel 30 band 1/3 octave graphic equaliser



Architect's & Engineer's Specification

The equaliser shall provide 30 bands of 12dB* of boost and cut on ISO frequency centres, from 25Hz-20kHz.
*Selectable to 6dB for increased fader resolution.

Each equaliser shall meet or exceed the following performance specifications:

Distortion (THD+N) <0.01% @1kHz, +4dBu
Frequency response ±0.5dB(20Hz-20kHz)
Noise <-90dBu (20Hz-20kHz unweighted)
Maximum Output level into 600¤ +22dBu

Each equaliser shall allow for; subsonic frequency attenuation at 18dB/octave, equalisation section by-pass and shall be fail-safe, that is the unit shall return automatically to the by-pass condition in the event of power supply interruption.

Each equaliser shall use centre detented slide potentiometers arranged to give a graphical display of frequency plotted against level.

A rear panel switch shall be provided to isolate the signal ground connections, quickly and safely, from the chassis ground.

All audio connections shall be via XLR style connectors and a tamperproof front panel cover shall be available to fit the unit.

The unit shall be capable of operating from a 115/230V ± 12% 50/60Hz AC power source.

The equaliser shall be the Klark Teknik Dual Channel Model DN360, and no alternative specification option is available

Technical Specification

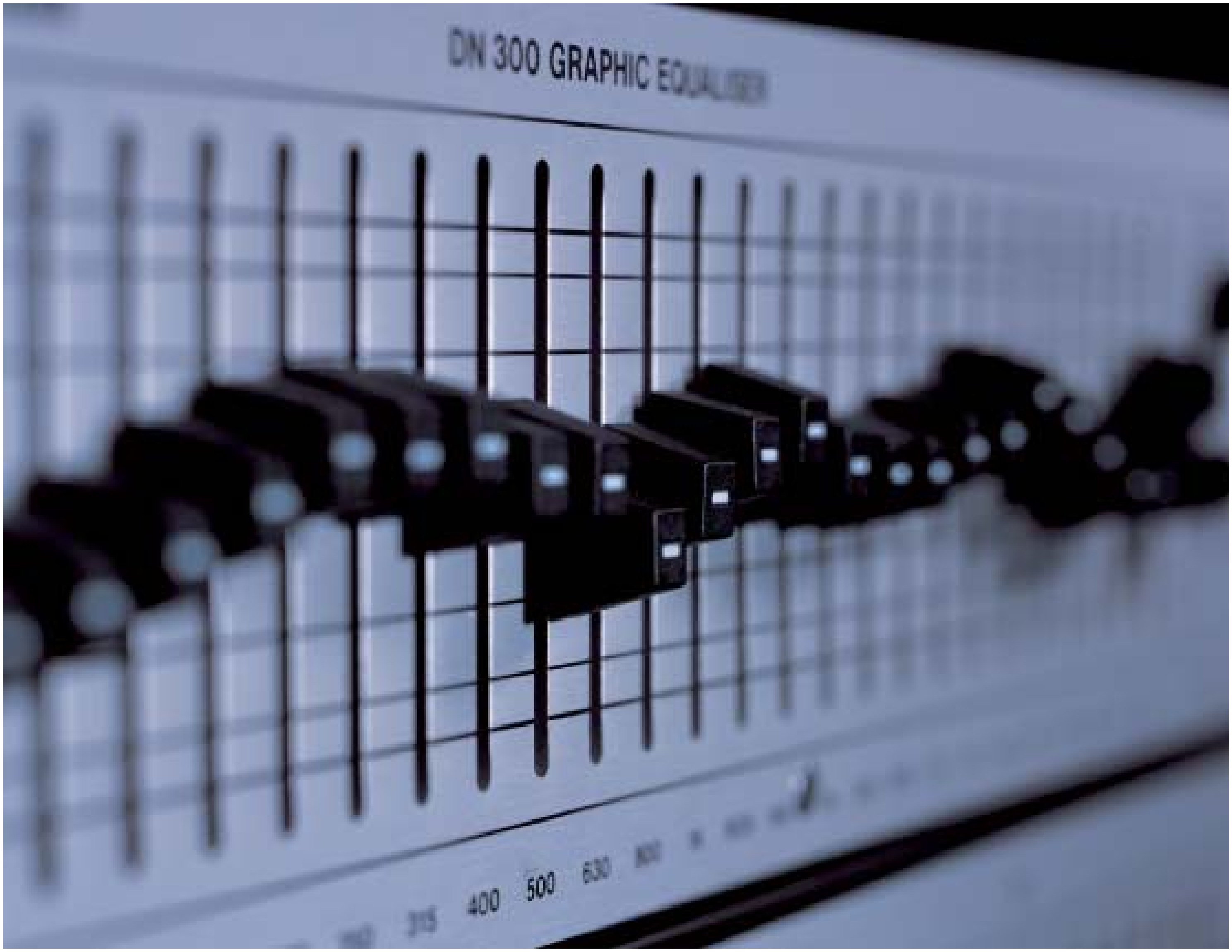
Inputs	**Two**
Type	Electronically balanced (pin 3 hot)
Impedance (¤)	
Balanced	20k
Unbalanced	10k
Outputs	**Two**
Type	Unbalanced (pin 3 hot)
Min. load impedance	600¤
Source impedance	<60¤
Max. level	+22dBu
Performance	
Frequency response (20Hz-20kHz)	
Eq out	±0.5dB
Eq in (flat)	±0.5dB
Distortion (THD+N)	<0.01% @ 1kHz, + 4dBu
Equivalent input noise	(20Hz-20kHz unweighted)
Eq in (flat)	<-90dBu
Channel separation	>75dB @ 1KHz
Overload indicator	+19dBu
Gain	-¤ to +6dB
Filters	
Type	MELT**
Centre frequencies	2x30, to ISO 266:1997 25Hz-20kHz 1/3 octave
Tolerance	±5%
Maximum boost/cut	±6/12dB
Subsonic filter	18dB/octave - 3dB @ 30Hz
Terminations	
Inputs	3 pin XLR
Outputs	3 pin XLR
Power	3 pin IEC
Power Requirements	
Voltage	115/230V 50/60Hz
Consumption	<15VA
Dimensions	
Height	133mm (5.25 inch) - (3U)
Width	482mm (19 inch)
Depth	205mm (8 inch)
Weight	
Nett	5.8kg
Shipping	7kg

Options
Security Cover
Transformer input* /output balancing

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

*** "MELT": Proprietary thick-film circuit.*

Trade Descriptions Act:
Due to the company policy of continuing improvement, we reserve the right to alter these specifications without prior notice. E&OE.



Klark Teknik has always striven to provide our equalisation units in various formats to suit different applications and customers.

DN300 is the single channel version of DN360, the main advantage of which is the use of 45mm (rather than 30mm) faders, thus allowing greater fader resolution.

The unit also features swept high and low pass filters, with switchable 6/12dB per octave resolution on the LPF, a single input level rotary providing a range of +6dB to off, and an EQ in / out switch.

DN300: single channel 30 band 1/3 octave graphic equaliser

Architect's & Engineer's Specification

The equaliser shall provide 12dB of attenuation and accentuation at 30 1/3 octave ISO centre frequencies from 25Hz-20kHz.

The equaliser shall meet or exceed the following performance specifications:

Distortion (THD+N) <0.01%@1kHz, +18dBu
Frequency response ±0.5dB (20Hz-20kHz)
Noise <-94dBu (20Hz-20kHz unweighted)
Maximum output level into 600¤ +22dBu

The equaliser shall have adjustable low & high cut 12dB/octave slope filters ranging from 15Hz-300Hz & 2.5kHz-30kHz and provide for selectable high cut filter slope 6/12dB.

The equaliser shall have an equalisation section by-pass facility and shall be fail-safe, that is the unit shall return automatically to the by-pass condition in the event of power supply interruption.

The equaliser shall use centre detented slide potentiometers arranged to give a graphical display of frequency plotted against level.

A rear panel switch shall be provided to isolate the signal ground connections, quickly and safely, from the chassis ground.

All audio connections shall be via XLR style connectors and a tamperproof front panel cover shall be available to fit the unit. The unit shall be capable of operating from a 115/230V ± 12% 50/60Hz AC power source.

The equaliser shall be the Klark Teknik Model DN300 and no alternative specification option is available.

Technical Specification

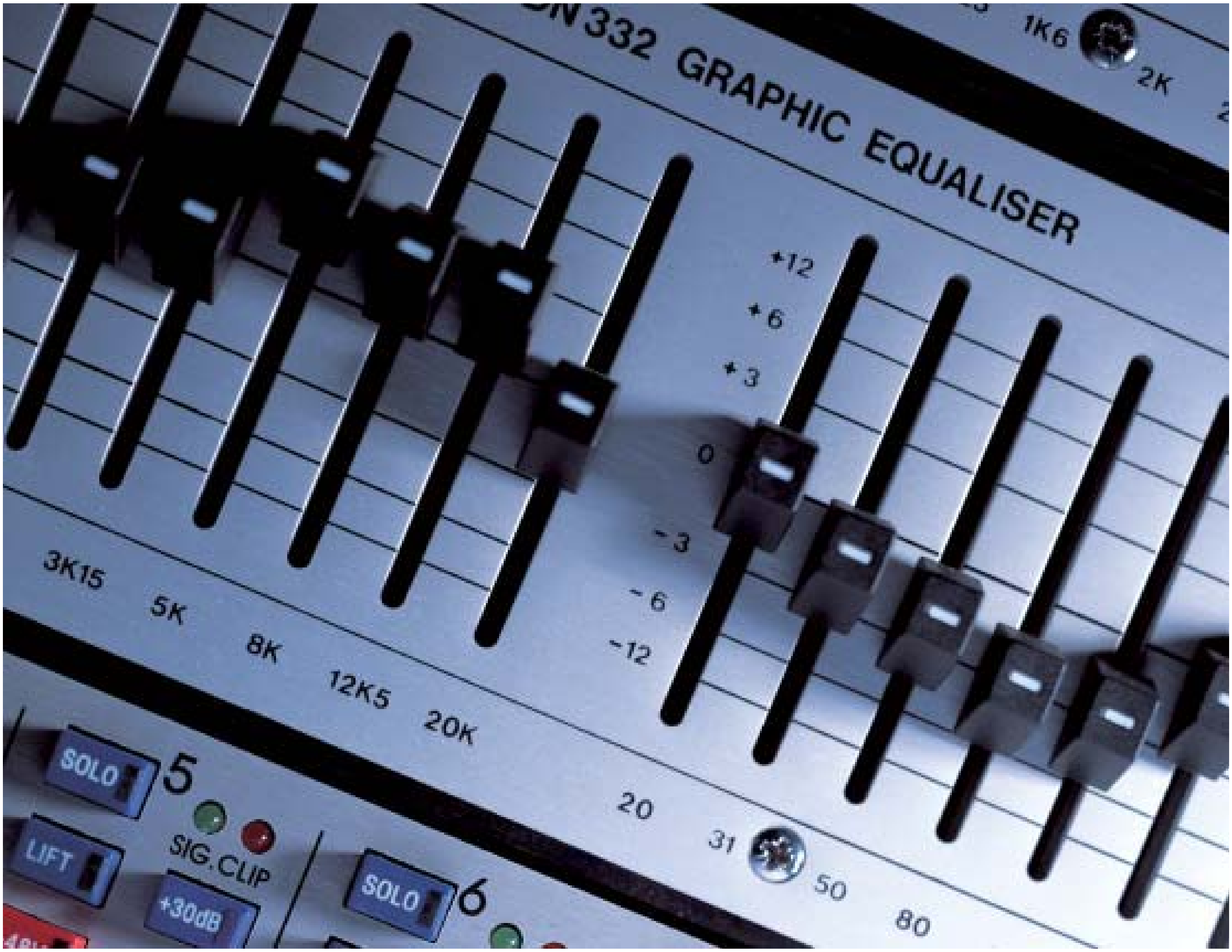
Input	**One**
Type	Electronically balanced (pin 3 hot)
Impedance(¤)	
Balanced	20k
Unbalanced	10k
Output	**One**
Type	Unbalanced (pin 3 hot)
Min. load impedance	600¤
Source impedance	<60¤
Max. level	+22dBu
Performance	
Frequency response (20Hz-20kHz)	
Eq out	±0.5dB
Eq in(flat)	±0.5dB
Distortion (THD+N)	<0.01% @ 1kHz, +18dBu
Equivalent input noise	(20Hz-20kHz unweighted)
Eq in	<-94dBu
Overload indicator	+19dBu
Gain	-¤ to +6dB
Filters	
Type	MELT**
Centre frequencies	30, to ISO 266:1997 25Hz-20kHz 1/3 octave
Tolerance	±5%
Maximum boost/cut	±12dB
High Pass filter slope	15Hz-300Hz 12dB/octave
Low Pass filter slope	2k5Hz-30kHz 6/12dB/octave
Terminations	
Input	3 pin XLR
Output	3 pin XLR
Power	3 pin IEC
Power Requirements	
Voltage	115/230V 50/60Hz
Consumption	<15VA
Weight	
Nett	4kg
Shipping	6kg
Options	
Security Cover	
Transformer input* /output balancing	

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

*** "MELT": Proprietary thick-film circuit.*

Trade Descriptions Act:
Due to the company policy of continuing improvement, we reserve the right to alter these specifications without prior notice. E&OE.

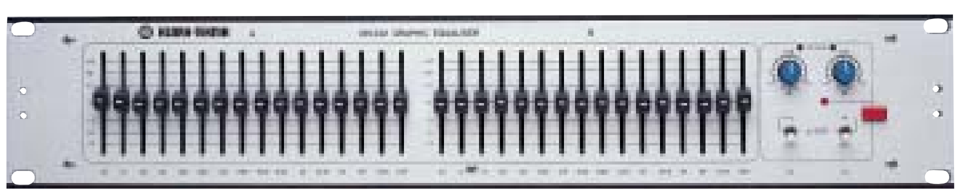
KLARK TEKNIK
SIGNAL PROCESSING BY DEFINITION



DN332 packs genuine Klark Teknik stereo graphic equalisation into a cost and space-effective 2RU package.

Features include 45mm high resolution faders, input level controls providing ranges from +6dB to off, very useful 18dB / octave 30Hz hi-pass subsonic filters (ideal for removing stage rumble), and three-position switches providing EQ in, EQ in with subsonic filter or EQ bypass.

DN332: dual channel 16 band 2/3 octave graphic equaliser



Architect's & Engineer's Specification

The equaliser shall provide 12dB of attenuation and accentuation at 2x16 2/3 octave ISO centre frequencies from 20Hz-20kHz.

Each equaliser shall meet or exceed the following performance specifications:

Distortion (THD+N) <0.01% @ 1kHz, +4dBu
Frequency response ±0.5dB (20Hz-20kHz)
Noise <-90dBu (20Hz-20kHz unweighted)
Maximum output level into 600¤ +22dBu

Each equaliser shall allow for subsonic frequency attenuation at 18dB/octave and have an equalisation section by-pass facility.

Each equaliser shall use centre detented slide potentiometers arranged to give a graphical display of frequency plotted against level.

A rear panel switch shall be provided to isolate the signal ground connections, quickly and safely, from the chassis ground.

All audio connections shall be via XLR style connectors and a tamperproof front panel cover shall be available to fit the unit.

The unit shall be capable of operating from a 115/230V ± 12% 50/60Hz AC power source.

The equaliser shall be the Klark Teknik Dual Channel Model DN332, and no alternative specification option is available.

Technical Specification

Inputs	**Two**
Type	Electronically balanced (pin 3 hot)
Impedance (¤)	
Balanced	20k
Unbalanced	10k
Outputs	**Two**
Type	Unbalanced (pin 3 hot)
Min. load impedance	600¤
Source impedance	<60¤
Max. level	+22dBu
Performance	
Frequency response (20Hz-20kHz)	
Eq out	±0.5dB
Eq in (flat)	±0.5dB
Distortion (THD+N)	<0.01% @ 1kHz, + 4dBu
Equivalent input noise	(20Hz-20kHz unweighted)
Eq in (flat)	<-90dBu
Channel separation	>75dB @ 1kHz
Overload indicator	+19dBu
Gain	-¤ to +6dB
Filters	
Type	MELT**
Centre frequencies	2x16, to ISO 266:1997 20Hz-20kHz 2/3 octave
Tolerance	±5%
Maximum boost/cut	±12dB
Subsonic filter	18dB/octave - 3dB @ 30Hz
Terminations	
Inputs	3 pin XLR
Outputs	3 pin XLR
Power	3 pin IEC
Power Requirements	
Voltage	110/120/220/240V
50/60Hz	
Consumption	<15VA
Dimensions	
Height	89mm (3.5 inch) - (2U)
Width	482mm (19 inch)
Depth	205mm (8 inch)
Weight	
Nett	4kg
Shipping	6kg
Options	
Security Cover	
Transformer input*/ output balancing	

**Input transformer balancing is non retrofittable and has to be specified with order.*
*** "MELT" Proprietary thick-film circuit.*

Trade Descriptions Act:
Due to the company policy of continuing improvement, we reserve the right to alter these specifications without prior notice. E&OE.

Klark Teknik's international reputation is founded on their EQ products, and one of the reasons for this is the DN410. Built to stand years of hard use yet sensitive and accurate, they remain the premier choice of audio professionals who require the very best in great-sounding analogue parametric EQ.

Each EQ channel features five bands of fully parametric EQ, with each band having an active range of 20Hz to 20kHz. This design makes it simple to accurately EQ out problem frequencies by dialling in a narrow-Q notch filter then sweeping it across the frequency range. It also means that EQ bands can be placed very close together or overlapped if required, unlike some competitive units which allocate specific frequency ranges to their units.

Each EQ band has an in / out button, and the unit is also fitted with an overall in / out switch for easy comparison of EQ'd and non-EQ'd responses.

DN410: dual channel 5 band parametric equaliser

Architect's & Engineer's Specification

The dual channel equaliser shall provide five bands of fully parametric filters and separate tuneable high & low cut filters. Each equaliser filter shall provide 25dB of attenuation and 15dB of accentuation at continuously variable frequencies ranging from 20Hz-20kHz and shall allow for bandwidth adjustment from 1/12 to 2 octaves.

The equaliser shall meet or exceed the following performance specifications:

Distortion (THD+N) <0.01% @ 1kHz, 4dBu
Frequency response ±1dB (20Hz-20kHz)
Noise <-94dBu (20Hz-20kHz unweighted)
Maximum output level into 600¤ +22dBu

The equaliser shall have adjustable low & high cut 12dB/octave slope filters ranging from 15Hz-300Hz & 2.5kHz-30kHz.

Stereo and mono operation of the unit shall be possible with all 10 filters available in mono mode.

Separate in/out switches shall be provided for each parametric filter section, and each complete equaliser channel.

The equaliser shall be fail-safe, that is the unit shall return automatically to the bypass condition in the event of power supply interruption.

A rear panel switch shall be provided to isolate the signal ground connections, quickly and safely, from the chassis ground.

All audio connections shall be via XLR style connectors and a tamperproof front panel cover shall be available to fit the unit.

The unit shall be capable of operating from a 115/230V ±12% 50/60Hz AC power source.

The equaliser shall be the Klark Teknik Model DN410 and no alternative specification option is available.

Technical Specification

Inputs	**Two**
Type	Electronically balanced (pin 3 hot)
Impedance (¤)	
Balanced	20k
Unbalanced	10k
Outputs	**Two**
Type	Unbalanced (pin 3 hot)
Min. load impedance	600¤
Source impedance	<60¤
Max. level	+22dBu
Performance	
Frequency response	(20Hz-20kHz)
Eq in (Flat, one band active)	±1dB
Eq out	±1dB
Distortion (THD+N)	<0.01% @ 1kHz, +4dBu
Equivalent input noise	(20Hz-20kHz unweighted) <-94dBu
Channel separation	>75dB @ 1kHz
Gain	-¤ to +6dB
Overload indicator	+19dBu
Filters	
Type	Parametric (2 x 5)
Bandwidth	Variable from 1/12 ~ 2 octaves
Max. boost/cut	+15/-25dB
Frequency ranges	20Hz-200Hz/ 200Hz-2kHz/2kHz-20kHz
High Pass filter	15Hz-300Hz/12dB octave
Lower Pass filter	2k5Hz-30kHz/12dB octave
Terminations	
Input	3 pin XLR
Output	3 pin XLR
Power	3 pin IEC
Power Requirements	
Voltage	115/230V 50/60Hz
Consumption	<15VA
Dimensions	
Height	89mm (3.5 inch) - (2U)
Width	482mm (19 inch)
Depth	235mm (9.25 inch)
Weight	
Nett	5kg
Shipping	6kg

Options

Security cover
Transformer input* / output balancing

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

KLARK TEKNIK
SIGNAL PROCESSING BY DEFINITION

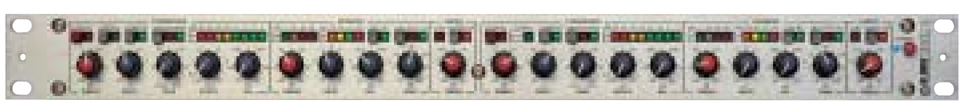
The revised DN500 Plus series utilises a new 'THAT's' chip which features lower distortion, improved dynamic range and faster slew rate. We've updated the cosmetics too, bringing them into the new KT 'family look'.

The DN500 Plus dual compressor provides two channels of full function compression, expansion, limiting and peak clipping in 1RU. A fully variable knee control allows continuous definition of compression style, and auto / manual modes provide either fast set-up or the necessary control for advanced compression effects.

The DN500 Plus has been a broadcast-industry standard for many years, due mainly to its extremely low noise performance, typically >2dB quieter than any comparable product.

Expansion characteristics are continuously variable between hard gating and gentle expansion thanks to the flexible expander section, and both compressor and expander section are fitted with their own side chain inputs. The channels can be ganged together for stereo operation, and the peak clipper eliminates transient overload whilst tracking the limiter threshold for total protection.

DN500 *Plus* : dual compressor/limiter expander



Architect's & Engineer's Specification

The compressor/limiter shall provide two complete channels of compression, expansion, peak limiting and peak clipping. The compressor section shall provide for adjustment of Threshold, Ratio, Knee, Attack and Release and have push button selection of auto or manual modes. The expander section shall provide for adjustment of Threshold, Ratio and Release and have push button selection of Auto or Fixed attack times. The limiter section shall provide for adjustment of Threshold and have push button selection of a Peak Clipper. An output gain control and level meter shall be provided. Gain reduction meters shall be provided for both compressor and expander sections.

The compressor/limiter shall meet or exceed the following specifications:

Distortion (THD+N) <0.03% @1kHz, +4dBu
Frequency response ±0.5dB (20Hz-20kHz)
Noise <-94dBu (20Hz-20kHz unweighted)
Compressor Attack time 50µs-20ms
Compressor Release time 60ms-2 secs
Maximum output level into 600¤ +21dBu

Push button switches shall be provided to select compressor, expander and channel bypass and to link both channels for stereo operation. Side chain inputs shall be provided for both compressor and expander sections. Channel inputs and outputs shall be via XLR style connectors, external side chain inputs shall be via 1/4" jack. A tamperproof front panel cover shall be available to fit the unit. The compressor/limiter shall be 19" standard rack mountable and 1U high. The unit shall be capable of operating from a 100V, 115V, 220-240V 50/60Hz AC power source.

The compressor/limiter shall be the Klark Teknik Model DN500 Plus and no alternative specification option is available.

Technical Specification

Audio Inputs	**Two**
Type	Electronically balanced (pin 3 hot)
Impedance (¤)	
Balanced	20k
Unbalanced	10k
Side Chain Inputs	**Two (Compressor) + Two (Expander)**
Type	Electronically balanced (tip hot)
Impedance (¤)	
Balanced	20k
Unbalanced	10k
Audio Outputs	**Two**
Type	Unbalanced (pin 3 hot)
Min. Load impedance	600¤
Source impedance	<60¤
Max.level	+21dBu
Performance	
Frequency response	(20Hz-20kHZ) ±0.5dB
Distortion (THD+N)	<0.03% @ 1kHz, +4dBu
Equivalent input noise	(20Hz-20kHz unweighted) <-94dBu
Compressor	
Threshold	-30dBu to +20dBu
Ratio	1:1 to 50:1
Knee	1dB (Hard) to 40dB (soft)
Envelope	Switchable auto (attack and release controls disabled) or manual
Attack (90% capture)	50µs to 20ms
Release (90% recovery)	60ms to 2 secs
Expander	
Threshold	-40dBu to +20dBu
Ratio	1:1 to 25:1
Attack	Switchable auto or fixed (2ms)
Release (90% recovery)	40ms to 2 secs
Output Gain	-10dB to +30dB
Limited/Clipper	
Threshold	0dBu to +20dBu
Terminations	
Audio inputs/outputs	3 pin XLR
Side-Chain inputs	Normalled 1/4 inch stereo jack
Power	3 pin IEC
Power Requirements	
Voltage	100V, 115V, 220-240V 50/60Hz
Consumption	<30VA
Dimensions	
Height	44.5mm (1.75 inch) - (1U)
Width	482mm (19 inch)
Depth	292mm (11.5 inch)
Weight	
Nett	5kg
Shipping	6kg

Options
Security cover
Transformer input* / output balancing

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

Trade Descriptions Act:
Due to the company policy of continuing improvement, we reserve the right to alter these specifications without prior notice. E&OE.

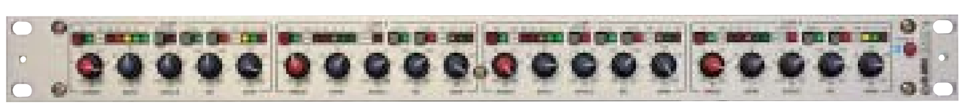
KLARK TEKNIK
SIGNAL PROCESSING BY DEFINITION

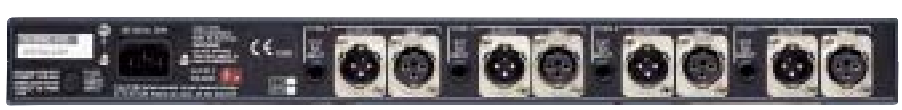
The revised DN500 Plus series utilises a new 'THAT's' chip which features lower distortion, improved dynamic range and faster slew rate. We've updated the cosmetics too, bringing them into the new KT 'family look'.

The DN504 Plus packs four fully featured compressors into just 1RU, and boasts audio performance equal to its super-quiet stable mate the DN500 Plus.

Fitted with hard / soft knee controls, auto and manual attack / release functions, and side chain inputs for each channel, the DN504 Plus is especially suited for in-ear monitoring applications, especially since channels can be linked as stereo pairs if required. Comprehensive gain reduction and output level metering completes this extremely useful and space-saving professional tool.

DN504 *Plus* : quad compressor limiter





Architect's & Engineer's Specification

The compressor/limiter shall provide four complete channels of compression. Each channel shall provide for adjustment of Threshold, Ratio, Attack and Release and have push button selection of auto or manual modes and hard or soft knee. An output gain control and level meter shall be provided. Gain reduction meters shall also be provided for each channel.

The compressor/limiter shall meet or exceed the following specifications:

Distortion (THD+N) <0.03% @1kHz, +4dBu
Frequency response ±0.5dB (20Hz-20kHz)
Noise <-94dBu (20Hz-20kHz unweighted)
Compressor Attack time 50µs-20ms
Compressor Release time 60ms-2 secs
Maximum output level into 600¤ +21dBu

Push button switches shall be provided to select channel bypass and to link adjacent channels for stereo operation. Side chain inputs shall be provided for each compressor section. Channel inputs and outputs shall be via XLR style connectors, external side chain inputs shall be via 1/4" jack. A tamperproof front panel cover shall be available to fit the unit. The compressor/limiter shall be 19"standard rack mountable and 1U high. The unit shall be capable of operating from a 100V, 115V, 220-240V 50/60Hz AC power source.

The compressor/limiter shall be the Klark Teknik Model DN504 Plus and no alternative specification option is available.

Technical Specification

Audio Inputs	**Four**
Type	Electronically balanced (pin 3 hot)
Impedance (¤)	
Balanced	20k
Unbalanced	10k
Side Chain Inputs	**Four**
Type	Electronically balanced (tip hot)
Impedance (¤)	
Balanced	20k
Unbalanced	10k
Audio Outputs	**Four**
Type	Unbalanced (pin 3 hot)
Min. Load impedance	600¤
Source impedance	<60¤
Max. Level	+21dBu
Performance	
Frequency response	(20Hz-20kHz) ±0.5dB
Distortion (THD+N)	<0.03% @ 1kHz, +4dBu
Equivalent input noise	(20Hz-20kHZ unweighted) <-94dBu
Channel separation	>90dB @ 1kHz
Compressor	
Threshold	-30dBu to +20dBu
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)	50µs to 20ms
Release (90% recovery)	60ms to 2 secs
Output gain	-10dB to +30dB
Terminations	
Audio inputs/outputs	3 pin XLR
Side-chain inputs	Normalled 1/4 inch stereo jack
Power	3 pin IEC
Power Requirements	
Voltage	100V, 115V, 220-240V 50/60Hz
Consumption	<30VA
Dimensions	
Height	44.5mm (1.75 inch) - (1U)
Width	482mm (19 inch)
Depth	292mm (11.5 inch)
Weight	
Nett	5kg
Shipping	6kg

Options
Security cover
Transformer input* / output balancing

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

Trade Descriptions Act:
Due to the company policy of continuing improvement, we reserve the right to alter these specifications without prior notice. E&OE.

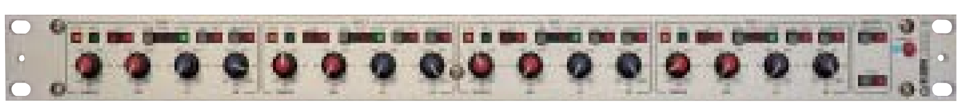
KLARK TEKNIK
SIGNAL PROCESSING BY DEFINITION

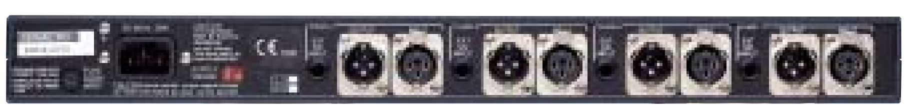
The revised DN500 Plus series utilises a new 'THAT's' chip which features lower distortion, improved dynamic range and faster slew rate. We've updated the cosmetics too, bringing them into the new KT 'family look'.

The DN514 Plus has assumed industry standard status as the multi-channel frequency-conscious gate unit of choice for live and recording applications. Providing the same ultimate audio performance as its DN500 Plus series siblings, the DN514 Plus is extremely comprehensive but easy to set up.

Two semi-automatic attack modes (calibrated for 'Fast' and 'Slow') allied with a hold value that is automatically scaled to the release time, allow each gate to be precisely configured to its application. It is also fitted with the unique 'Sync' function, which locks all four gate release times, allowing easy synchronisation of harmony parts. Each gate also features a side chain input, and an additional key input to allow external triggering if required. LED indicators show gate status, and both Master (unit) and individual channel bypass switches aid set-up.

DN514 Plus : quad auto gate





Architect's & Engineer's Specification

The noise gate shall provide four channels of frequency-conscious gating with each channel having adjustable low and high cut 12dB/octave filters, variable from 20Hz-5kHz and 80Hz-20kHz, switchable into side chain or audio signal path.

The noise gate shall meet or exceed the following specifications:

Distortion (THD+N) <0.03% @1kHz, +4dBu
Frequency response ±0.5dB (20Hz-20kHz)
Noise<-100dBu gate closed (20Hz-20kHz unweighted)
<-94dBu gate open (20Hz- 20kHz unweighted)
Attack time 50µs-2ms
Hold time/Release time 40ms-2 secs
Maximum output level into 600¤ +21dBu

A tamperproof front panel cover shall be available to fit the unit. The noise gate shall be 19" standard rack mountable and 1U high.

The unit shall be capable of operating from a 100V, 115V, 220-240V 50/60Hz AC power source.

The noise gate shall be the Klark Teknik Model DN514 Plus and no alternative specification option is available.

Technical Specification

Audio Inputs	**Four**
Type	Electronically balanced (pin 3 hot)
Impedance(¤)	
Balanced	20k
Unbalanced	10k
Key Inputs	**Four**
Type	Electronically balanced (tip hot)
Impedance (¤)	
Balanced	20k
Unbalanced	10k
Audio Outputs	**Four**
Type	Unbalanced (pin 3 hot)
Min. Load impedance	600¤
Source impedance	<60¤
Max. level	+21dBu
Performance	
Frequency response	(20Hz-20kHz) ±0.5dB
Distortion (THD+N)	<0.03% @ 1kHz, +4dBu
Equivalent input noise	(20Hz-20kHz unweighted)
Gate open	<-94dBu
Gate closed	<-100dBu
Attack programme related, semi-automatic	50µs to 200µs "Fast" 500µs to 2ms "Slow"
Hold/Release	Variable 40ms to 2sec
Threshold	Variable-40dBu to +20dBu
Attenuation	>84dB Gate closed
Key Filters	
High pass filter	20Hz-5kHz/12dB octave
Low pass filter	80Hz-20kHz/12dB octave
Terminations	
Audio inputs/outputs	3 pin XLR
Key inputs	Normalled 1/4 inch stereo jack
Power	3 pin IEC
Power Requirements	
Voltage	100V, 115V, 220-240V 50/60Hz
Consumption	<30VA
Dimensions	
Height	44.5mm (1.75 inch) - (1U)
Width	482mm (19 inch)
Depth	292mm (11.5 inch)
Weight	
Nett	5kg
Shipping	6kg

Options
Security cover
Transformer input* / output balancing

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

Trade Descriptions Act:
Due to the company policy of continuing improvement, we reserve the right to alter these specifications without prior notice. E&OE.

SIGNAL PROCESSING BY DEFINITION

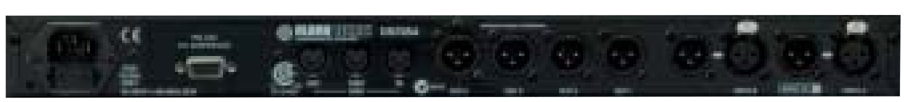
Ostensibly a delay unit with no less than 5.4s of delay per output, the 1 input / 3 output DN7453 is actually a powerful audio 'toolbox' featuring six EQ filters per input and seven per output.

All EQ bands are fully configurable as full PEQ, hi/lo pass or hi / lo shelving filters, and a full-function compressor is available on every input. Every output also features high quality compression and limiting circuitry.

Accurate input and output metering is provided on front panel LED ladders, and all functions are available either via free PC software or the very intuitive front panel user interface. MIDI in/out and thru is provided as is RS-232 connection via 9-pin 'D' type connector. Like all DSP-driven Klark Teknik devices, there is more than enough processing power available for all functions to be accessible at all times.

Applications include time alignment for post-production, for delay towers in line production, and for broadcast satellite signals.

DN7453: user configurable digital audio delay line



Architect's & Engineer's Specification

The delay line shall provide for one input and three outputs, housed in a standard 1U 19" rack mount chassis. It shall have a maximum total delay time of 5400ms at a full bandwidth of 20kHz. Delay times shall be displayed in units of time and distance and shall be adjustable to a resolution of 21 microseconds.

When displaying distance, a temperature compensation facility will allow the delay time to be automatically recalculated for a specified temperature. The unit shall incorporate a master delay time on the input and individual delay times on each of the outputs.

Each input shall include seven bands of full parametric EQ which can be individually configured to be any of the following:- Low shelf, High shelf, Low pass, High pass, Parametric EQ and can be individually bypassed. In addition, each output shall include six bands of full parametric EQ which can also be individually configured to be any of the following:- Low shelf, High shelf, Low pass, High pass, Parametric EQ and can be individually bypassed.

Each output shall have individually controllable compressor and limiter functions.

The delay line shall meet, or exceed, the following specifications:

Frequency response +0/-0.5dB (20Hz to 20kHz)
Distortion (THD+N) <0.01% @1kHz, +8 dBu
Dynamic Range >112dB (20Hz to 20kHz unweighted)

Options for the various delay and equalisation parameters shall be presented on a liquid crystal display and shall be selectable by six front panel control buttons and shall be altered by a continuous rotary controller.

User memories shall be provided for setup storage. A security lock out system shall be available, including a user defined code number.

Each input shall have a gain control and meter and each output shall have an attenuator control and meter, for system matching. Output levels can also be individually adjusted from within the software and levels recalled as part of the user memories.

A MIDI interface shall be provided as standard. The delay line shall also be capable of being controlled remotely by a PC via an RS-232 port.

All audio connections shall be via XLR style connectors. Inputs and outputs shall be electronically balanced and there shall be an option for input transformer isolation. The unit shall be capable of operating from a 90V to 250V a.c., 50/60Hz, power source.

The delay line shall be the Klark Teknik DN7453 and no alternative option is available.

Technical Specification

Audio Inputs — One

Type	Electronically Balanced (Pin 2 Hot)
Impedance (◘)	
Balanced	20 k
Unbalanced	10 k
Maximum Level	+21dBu

Audio Outputs — Three

Type	Electronically Balanced (Pin 2 Hot)
Source impedance	>100◘
Maximum Level	+21 dBu into > 2k◘

Performance

Frequency response	(20Hz to 20kHz) +0/-0.5 dB with all filters and EQ flat
Distortion (THD+N)	<0.01% @ 1kHz, +8 dBu
Dynamic Range	(20Hz to 20kHz unweighted) >112 dB

Input Processing

Input Gain	+6dB to - ◘ under front panel control
Master EQ 1-7	Parametric EQ Mode frequency range 20Hz to 20kHz in 21 steps per octave Boost/cut: (12 dB in 1 dB steps Q: 0.4 to 20 Hi-Shelf/Lo Shelf Filter Modes Boost/cut: (12 dB in 1 dB steps Slope: -6dB/Oct, -12 dB/Oct Hi-Pass/Lo-Pass Filter Modes Q: 0.4 to 2.0 (-12dB/Oct only) Slope: -6dB/Oct, -12 dB/Oct
Delay	0 to 4500 milliseconds in 21 us steps

Output Processing (per channel)

Delay	0 to 900 milliseconds in 21 us steps
Channel EQ 1-6	Parametric EQ Mode frequency range 20Hz to 20kHz in 21 steps per octave Boost/cut: (12 dB in 1 dB steps Q: 0.4 to 20 Hi-Shelf/Lo Shelf Filter Modes Boost/cut: (12 dB in 1 dB steps Slope: -6dB/Oct, -12 dB/Oct Hi-Pass/Lo-Pass Filter Modes Q: 0.4 to 2.0 (-12dB/Oct only) Slope: -6dB/Oct, -12dB/Oct
Output gain control	0 dB to -◘ under front panel
Compressor	Threshold: +21dBu to -9dBu in 1.0dB steps Ratio: 1:1, 1.4:1, 2:1, 4:1, 8:1 Attack: 0ms to 99 ms Release: 50ms to 999ms
Limiter	Threshold: +21dBu to -9dBu in 1.0dB steps Release: 50ms to 999ms

Terminations

Audio inputs/outputs	3 pin XLR
MIDI	5 pin DIN
RS-232	9 pin D-Type socket
Power	3 pin IEC

Power Requirements

Voltage / Consumption	90 to 250V a.c @ 50/60Hz / 20watts

Dimensions

Height	44mm (1.75 inch)- (1U)
Width	483mm (19 inch)
Depth	374mm (14.72 inch)

Weight

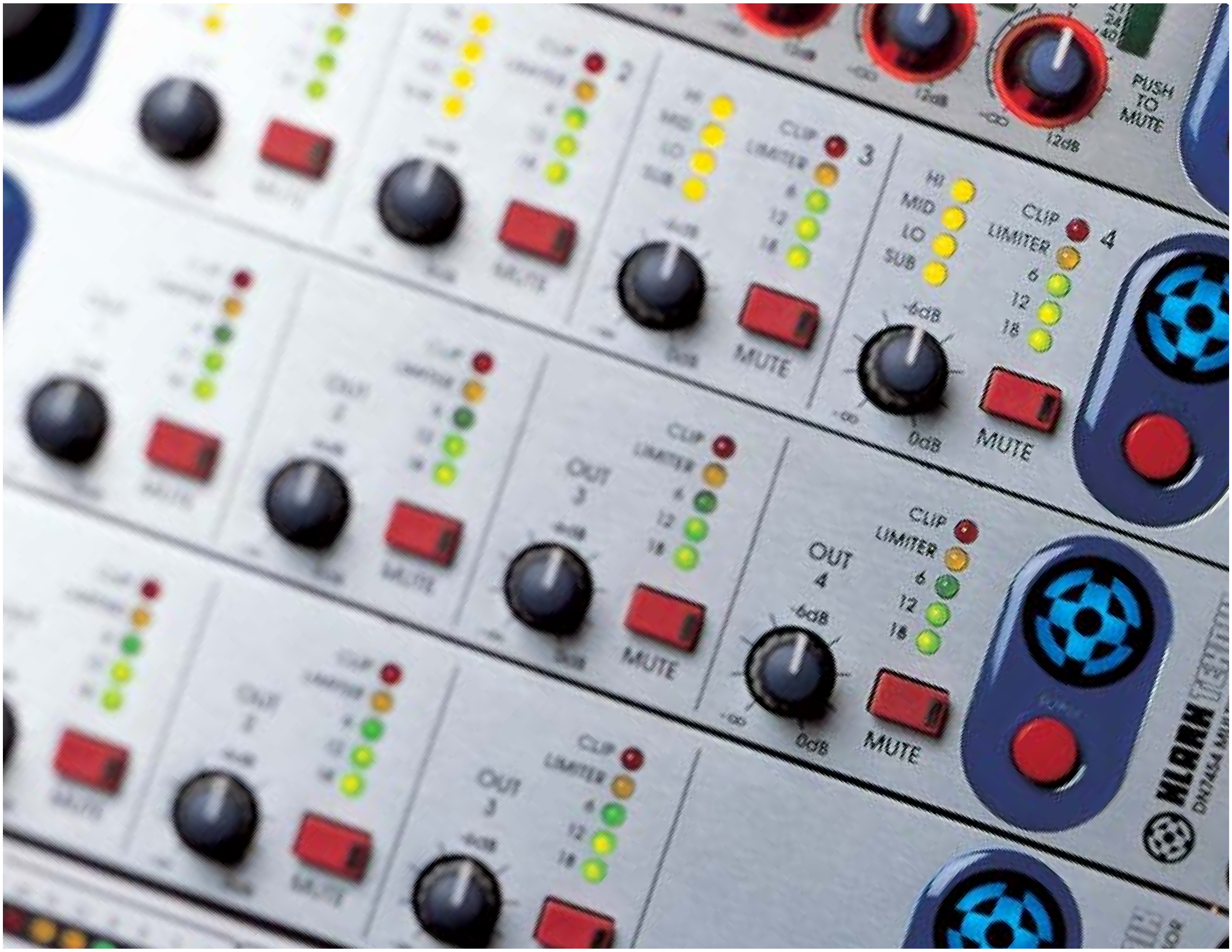
Nett	5kg
Shipping	7kg

Options

Transformer input balancing (must be specified with order).

Trade Descriptions Act:
Due to the company policy of continuing improvement, we reserve the right to alter these specifications without prior notice. E&OE.

KLARK TEKNIK
SIGNAL PROCESSING BY DEFINITION



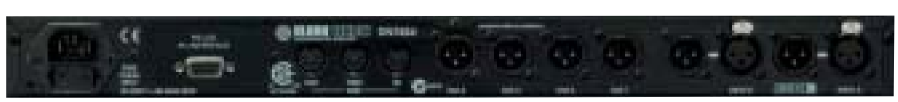
The DN7000 series of multipurpose 'audio toolkits' continues Klark Teknik's longstanding principle of offering flexible, integrated solutions to a wide range of sonic demands.

With no less than 5.4s of delay per output (2.7s in stereo mode) and user-definable routing, DN7545 is also a powerful EQ tool featuring six EQ filters per input and seven per output.

All EQ bands are fully configurable as full PEQ, hi/lo pass or hi / lo shelving filters, and a full-function compressor is available on every input. Every output also features high quality compression and limiting circuitry. Accurate input and output metering is provided on front panel LED ladders, and all functions are available either via free PC software or the very intuitive front panel user interface. MIDI in/out and thru is provided as is RS-232 connection via 9-pin 'D' type connector. Like all DSP-driven Klark Teknik devices, there is more than enough processing power available for all functions to be accessible at all times.

Applications include time alignment for post-production, for delay towers in line production and for broadcast satellite signals.

DN7454: user configurable digital audio delay line



Architect's & Engineer's Specification

The delay line shall provide for two inputs and four outputs, housed in a standard 1U 19" rack mount chassis. It shall have a maximum total delay time of 5400ms at a full bandwidth of 20kHz. Delay times shall be displayed in units of time and distance and shall be adjustable to a resolution of 21 microseconds.

When displaying distance, a temperature compensation facility will allow the delay time to be automatically recalculated for a specified temperature. The unit shall incorporate a master delay time on the input and individual delay times on each of the outputs.

Each input shall include seven bands of full parametric EQ which can be individually configured to be any of the following:- Low shelf, High shelf, Low pass, High pass, Parametric EQ and can be individually bypassed. In addition, each output shall include six bands of full parametric EQ which can also be individually configured to be any of the following:- Low shelf, High shelf, Low pass, High pass, Parametric EQ and can be individually bypassed.

Each output shall have individually controllable compressor and limiter functions.

The delay line shall meet, or exceed, the following specifications:

Frequency response +0/-0.5dB (20Hz to 20kHz)
Distortion (THD+N) <0.01% @1kHz, +8 dBu
Dynamic Range >112 dB (20Hz to 20kHz unweighted)

Options for the various delay and equalisation parameters shall be presented on a liquid crystal display and shall be selectable by six front panel control buttons and shall be altered by a continuous rotary controller.

User memories shall be provided for setup storage. A security lock out system shall be available, including a user defined code number.

Each input shall have a gain control and meter and each output shall have an attenuator control and meter, for system matching. Output levels can also be individually adjusted from within the software and levels recalled as part of the user memories.

A MIDI interface shall be provided as standard. The delay line shall also be capable of being controlled remotely by a PC via an RS-232 port.

All audio connections shall be via XLR style connectors. Inputs and outputs shall be electronically balanced and there shall be an option for input transformer isolation. The unit shall be capable of operating from a 90V to 250 V a.c., 50/60 Hz, power source.

The delay line shall be the Klark Teknik DN7454 and no alternative option is available.

Technical Specification

Audio Inputs	**Two**
Type	Electronically Balanced (Pin 2 Hot)
Impedance (¤)	
Balanced	20 k
Unbalanced	10 k
Maximum Level	+21dBu
Audio Outputs	**Four**
Type	Electronically Balanced (Pin 2 Hot)
Source impedance	>100¤
Maximum Level	+21 dBu into > 2k¤
Performance	
Frequency response	(20Hz to 20kHz) +0/-0.5 dB with all filters and EQ flat
Distortion (THD+N)	<0.01% @ 1kHz, +8 dBu
Dynamic Range	(20Hz to 20kHz unweighted) >112 dB
Input Processing (per channel)	
Input Gain control	+6dB to - ¤ ,under front panel
Master EQ 1-7	Parametric EQ Mode frequency range 20Hz to 20kHz in 21 steps per octave Boost/cut: (12dB in 1dB steps Q: 0.4 to 20 Hi-Shelf/Lo Shelf Filter Modes Boost/cut: (12dB in 1dB steps Slope: -6dB/Oct, -12dB/Oct Hi-Pass/Lo-Pass Filter Modes Q: 0.4 to 2.0 (-12dB/Oct only) Slope: -6dB/Oct, -12 dB/Oct
Delay	0 to 4500 milliseconds in 21 us steps
Output Processing (per channel)	
Delay	0 to 900 milliseconds in 21 us steps
Channel EQ 1-6	Parametric EQ Mode frequency range 20Hz to 20kHz in 21 steps per octave Boost/cut: (12 dB in 1 dB steps Q: 0.4 to 20 Hi-Shelf/Lo Shelf Filter Modes Boost/cut: (12 dB in 1 dB steps Slope: -6dB/Oct, -12 dB/Oct Hi-Pass/Lo-Pass Filter Modes Q: 0.4 to 2.0 (-12dB/Oct only) Slope: -6dB/Oct, -12 dB/Oct
Output gain control	0 dB to -¤ under front panel
Compressor	Threshold: +21dBu to -9dBu in 1.0dB steps Ratio: 1:1, 1.4:1, 2:1, 4:1, 8:1 Attack: 0ms to 99 ms Release: 50ms to 999ms
Limiter	Threshold: +21dBu to -9dBu in 1.0dB steps Release: 50ms to 999ms
Terminations	
Audio inputs/outputs	3 pin XLR
MIDI	5 pin DIN
RS-232	9 pin D-Type socket
Power	3 pin IEC
Power Requirements	
Voltage / Consumption	90 to 250V a.c @ 50/60Hz / 20watts
Dimensions	
Height	44mm (1.75 inch)- (1U)
Width	483mm (19 inch)
Depth	374mm (14.72 inch)

Weight	
Nett	5kg
Shipping	7kg

Options

Transformer input balancing (must be specified with order).

Trade Descriptions Act:
Due to the company policy of continuing improvement, we reserve the right to alter these specifications without prior notice. E&OE.

KLARK TEKNIK
SIGNAL PROCESSING BY DEFINITION

DN6000: audio analyser

The ability to monitor and analyse audio systems, regardless of their application, remains one of the most important issues facing engineers. The DN6000 Audio Analyser continues Klark Teknik's long tradition of providing audio professionals with the tools they need in the field, with a unique feature set that provides every important analysis function in one roadworthy device.

Current SPL is constantly shown on a large red LED display, and measurements may be taken unweighted or with A or C weighting, in 1/3rd or 1/6th octave analysis modes, and shown in either peak or average responses. A high-quality measurement microphone is supplied with every (individually calibrated) DN6000, for which a front panel microphone input is fitted. The unit is also provided with dual line level inputs for stereo 1/3 octave analysis, with sum and differences display options, to allow it to run 'in-line' with a system if required.

An internal signal generator with sine wave and band limited pink noise test signals is also supplied, with the additional capability to provide test signal burst and frequency sweeps with automatic data capture, thus providing numerous system-test options. A proprietary ribbon-cable interface allows connection to a Klark Teknik Helix system for auto-EQ functions, allowing real-time room or system EQ adjustments as acoustic characteristics change.

Additional functions include RT60 (reverb decay time) measurement, as well as Leq (average SPL over a given period) and Let (SPL at a specific point). Test measurements can be taken over any period configurable between 25ms and 180hrs, and two separate banks of data storage are available, 32 locations for frequency analysis and 16 for time-related measurements. The ability to store, recall, compare and print this data plus any subsequent measurements taken thereon makes the DN6000 an invaluable tool for numerous applications.

Architect’s & Engineer’s Specification

The analyser shall conform to the Type 1 requirements of IEC 804: 1985 - Standard Specification for Integrating-averaging sound level meters. It shall be a standard 2U, 19” rack mounted unit, capable of frequency domain and time domain analysis of a single mic level or twin line level signals introduced via a front panel XLR microphone input socket equipped with 48 volt phantom power, or via twin rear panel XLR line input sockets respectively. The unit shall feature a large, backlit LCD graphic display area, multiple function switches and an LED numerical display that can be read from a distance. It shall be equipped with switchable A and C weighting filters.

The analyser shall have an integral signal generator, capable of sine wave, swept sine wave and gated, band limited pink noise generation via a rear panel XLR output.

In frequency analysis mode, the unit shall perform 1/3 octave and 1/6 octave real time spectrum analysis. The 1/3 octave bands shall range from 12.5 Hz to 31.5 kHz at intervals defined by ISO 266:1997. The user shall have control over display range, reference level and response time, and over a cursor to pick out any frequency band or the over all signal level for numerical read out on the LED display.

In time analysis mode, the unit shall be capable of RT60 (reverberation time) analysis at any 1/3 octave or 1 octave band; of up to 180 sequential Leq measurements of durations ranging from 1 second to 1 hour, of Let measurements over durations of 1 minute to 180 hours.

The unit shall be able to freeze the real time analysis and store to any of 48 non volatile memory locations - 32 for frequency analysis and 16 for time analysis. The analyser shall be equipped with a parallel printer port for creation of hard copies of any measurement both graphically and in tabular form. An RS-232 serial port shall also be provided to allow printing via an external computer. The analyser shall also be equipped with a data port for direct connection to Klark Teknik DN9340 for automatic equalisation purposes.

The unit shall be the Klark Teknik Model DN6000 and no alternative specification option is available.

Technical Specification

Frequency Response	**5Hz to 40kHz**
Microphone Input	**One, Differential** Compatible with microphone sensitivities from 0.25mV/uBar to 1mV/uBar
Sensitivity	140dBSPL to 50dBSPL (with optional 6051 microphone)
Powering (nominal)	48V DC phantom power
Connector	XLR on front panel
Line Input	**Two, Differential** balanced or unbalanced
Sensitivity	+40dBu to -50dBu
Impedance	47k¤
Connector	XLRs on rear panel
Pink Noise output	
Type	Digital pseudo-random white noise generator with pink noise filter
Frequency distribution	-3dB/Octave 20Hz to 20kHz ±0.2dB
Level	+4dBu, -10dBu, -30dBu
Impedance	50¤ balanced
Connector	XLR on rear panel
Filters	
Attenuation accuracy	±0.1dB
‘A’-weighting	Selectable to IEC 651 type 1 requirement
‘C’-weighting	Selectable to IEC 651 type1 requirement
Interfaces	DN9340, parallel printer, RS-232 serial port via external computer
Terminations	
Audio inputs/outputs	3 pin XLR
Mic	3 pin XLR
Printer Port	25 way D socket
Data Output	16 way IDC Latching Header (Male)
RS-232	9 way D socket
Power	3 pin IEC
Power requirements	
Voltage	100 to 240V, 50 to 60Hz
Consumption	< 40VA
Dimensions	
Height	89mm (3.5 inch) - (2U)
Width	482mm (19 inch)
Depth	302mm (11.8 inch)
Weight	
Nett	5.5kg
Shipping	9.5kg
6051 Microphone	
Frequency Response	Flat to 15kHz
Sensitivity	0.5mV per uBar nominal @ 1kHz
Dynamic range	20 to 130dBSPL
Capsule	0.25 inch electret condenser
Type	Pressure - omnidirectional
Power required	14V minimum phantom power (compatible with 48V)

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

Trade Descriptions Act:
Due to the company policy of continuing improvement, we reserve the right to alter these specifications without prior notice. E&OE.

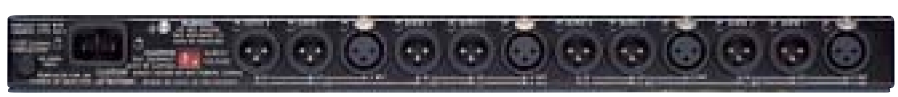
KLARK TEKNIK
SIGNAL PROCESSING BY DEFINITION

DN800: active crossover

Whilst DN9848 was the first digital system controller to fit eight outputs and four inputs into one rackspace, the DN800 remains the only analogue device to offer this format.

You might wonder where, in this digital world, an old-fashioned analogue crossover fits in? The fact remains that DN800 still offers some features that make it uniquely useful, and now at a price that makes it very attractive.

It can be configured in a number of ways, but its most common application is to provide four mixes for bi-amped monitor systems. Its combination of rugged engineering, unmatched audio performance and ease of use makes it the perfect choice for permanently configured systems that do not require the myriad features provided by our digital controllers.



Architect's & Engineer's Specification

The DN800 electronic active crossover shall provide up to 6 crossover points/8 bands in one rack unit.

The crossover shall be configurable as 4-way stereo, 3-way stereo or 2-way 4 channel.

The crossover shall be able to provide any frequency, slope and response by the use of plug-in cards. Each frequency band shall have controls for mute, gain, phase invert and band-edge phase adjust.

The crossover shall meet or exceed the following performance specifications:

Distortion (THD+N) <0.01% @1kHz, +4dBu
Equivalent input noise < -95dBu (any output)
(20Hz to 20kHz unweighted)

High quality VCA limiters shall be available on plug-in cards. These shall have threshold controls available on the front panel.

Fixed equalisation shall be available on plug-in cards to suit constant directivity horns etc.

The Unit shall incorporate a fixed 18dB/Oct Subsonic filter at 30Hz.

The crossover shall feature front panel LEDs to indicate signal present, limit and +6dB over-limit. Front panel controls apart from gain and mute shall be recessed and covered after initial setup by security plates. An overall tamper-proof cover shall be available.

All audio connections shall be via XLR style connectors. All inputs and outputs shall be electronically balanced. Input balancing transformers should be available as an option and must be internally fitted.

The unit shall be capable of operating from a 115/230V ±10% 50/60Hz AC power source.

The crossover shall be the Klark Teknik model DN800 and no alternative option is available.

Technical Specification

Inputs	**Four**
Type	Electronically balanced (pin 2 hot)
Impedance (¤)	
Balanced	20k
Unbalanced	10k
Outputs	**Eight**
Type	Electronically balanced (pin 2 hot)
Min. load impedance	600¤
Source impedance	<60¤
Max. Level	>+21dBu
Performance	
Distortion (THD+N)	<0.01% @ 1kHz, +4dB
Equivalent input Noise	(20Hz to 20kHz unweighted) <-95dBu (any output)
Nominal gain	0dB
Adjustable gain	± 6dB on front panel control additional +12dB or -6dB on internal preset
Limiter threshold	-12dBu to +12dBu
Phase relationship	Continuously adjustable 0˚ to 180˚ between bands. Polarity switch provides additional 180˚
Frequency division filters	Butterworth, Bessel, or Linkwitz-Riley 12, 18 or 24dB/Oct
Subsonic filter	18dB/Oct 30Hz
Power requirements	
Voltage	110/120/220/240V 50/60Hz AC
Consumption	<30VA
Terminations	
Audio inputs/Outputs	3 pin XLR
Power	3 pin IEC
Dimensions	
Height	44mm (1.75 inch) - (1U)
Width	482mm (19 inch)
Depth	285mm (11.2 inch)
Weight	
Nett	4kg
Shipping	5kg
Options	
Overall security cover	
System equalisation	
Transformer input balancing*	

**Input balance transformers must be specified with order*

Trade Descriptions Act:
Due to the company policy of continuing improvement, we reserve the right to alter these specifications without prior notice. E&OE.

A good DI (direct-injection) device is essential in almost any system. Given that its primary function is to replace a microphone, the audio performance is critical. They also need to be extremely rugged, and also capable of providing flexible operation. The Klark Teknik DN1414 DI module both meets these criteria, and more.

DN1414: di module

The DN1414 multiple DI module brings all the advantages of the LBB100 to a rackmount format, packing no less than 14 discrete DI boxes into a single 3RU package.

10 channels are configured as per the LBB100, and the two remaining channels are arranged in pairs, featuring simple jack in / XLR out connection for use as single DI units or as stereo pairs.

Customers can specify a factory-fitted dual power supply option if required, and the unit is also fitted as standard with a multipin retrofit kit. This allows a user to fit the multipin connector of their choice to a blank panel on the rear and then hard wire the outputs direct to it.

All this makes the DN1414 a very flexible device which suits a number of applications in live production, in the studio and in broadcast.

Architect's & Engineer's Specification

The Multiple DI Module shall provide 14 discrete audio channels in a standard 3U 19" rack mount chassis, each channel providing galvanic isolation and impedance matching for a variety of input signals.

Each channel shall also provide separate -30 dB pad and -15 dB attenuation switches, and an earth lift function.

Each Multiple DI Module shall meet or exceed the following performance specifications:

Distortion (THD+N) < 0.01% @1kHz, +4dB
Frequency response +0 / -1.0dB (20Hz to 20kHz)

The DI Module shall have ten single audio channels and two dual audio channels. All channels shall have a 1/4" TRS jack input which is capable of accepting balanced or unbalanced inputs. The ten single audio channels shall have a female 3-pin XLR connector in parallel with the jack socket. In use the XLR input shall present a 20k ohm input impedance and the 1/4" jack socket a nominal 1M ohm input impedance.

The ten single channels shall also have an unbalanced link output on a 1/4" TS jack socket.

All outputs shall be transformer isolated and shall use 3-pin male XLR connectors.

The unit shall be capable of operating from a 90 to 250V, 50 to 60Hz AC power source.The unit should have the option of dual redundant power supplies.

The DI Module shall be the Klark Teknik model DN1414 and no alternative option is available.

Technical Specification

Audio Inputs	**Two per mono channel** **One per stereo channel**
Type	Electronically balanced
Impedance	
TRS jack input	1MΩ
XLR input	20kΩ
Max level	+ 21dBu with no input attenuation
Attenuation	- 15dB
Pad	- 30dB
Audio Outputs	**Two per mono channel** **One per stereo channel**
Type	Transformer isolated
Source impedance	50Ω
Min Load	600Ω (-3dB level loss into 200Ω)
Max level	> + 21dBu @ 1kHz with load > 1kΩ
Link Output (Channels 1-10)	
Source impedance	50Ω
Min Load	600Ohm (-3dB level loss into 200Ohm)
Max level	> + 21dBu @ 1kHz with load > 1kΩ
Performance	
Noise	-100dBu between 20Hz and 20kHz unweighted
Frequency response	20Hz to 20kHz +/- 0.5dB
Distortion (THD+N)	<0.01% @ 1kHz, +4dBu output
Terminations	
Audio Inputs	3 pin XLR & 1/4" TRS jacks
Audio Outputs	3 pin XLR
Power	3 pin IEC
Power Requirements	90 to 250V a.c @ 50/60Hz @ < 60 VA
Dimensions	
Height	132 mm (5.2 inches) - (3U)
Width	483 mm (19 inches)
Depth	300 mm (12 inches)
Weight	
Nett	8kg
Shipping	9kg
Options	
*Dual power supply	

All options are non retrofittable and must be specified with order.

Trade Descriptions Act:
Due to the company policy of continuing improvement, we reserve the right to alter these specifications without prior notice. E&OE.

KLARK TEKNIK
SIGNAL PROCESSING BY DEFINITION

The new Klark Teknik DN100 Direct Injection Box is the natural successor to the long-established LBB100.

A ground-up redesign provides an extended dynamic range, lower noise floor and all the world-class audio performance you'd expect from Klark Teknik. DN100 is also designed to handle the rigours of life on the road: a thick aluminium shell protects the electronics, and this in turn is protected by a tough silicone rubber casing, which is replaceable and available as a spare part.

We've also fitted a Kensington security slot in one of the end panels to allow the unit to be made secure using a Kensington MicroSaver security cable.

Attention to detail – it's what makes a good unit into a great one.

DN100: active di

Architect's & Engineer's Specification

The Direct injection module shall provide the functions of transformer isolation, impedance matching and attenuation into a low impedance active balanced input. The module shall be able to accept a maximum input level of at least 30dBu provide switchable attenuation from 0 to 20dB and output the signal into a balanced 600 ohm load.

Input connectors shall include two quarter inch jack sockets and one 3-pin XLR socket, all linked. Input impedance shall be 1M ohm (jacks sockets), 20K ohm (XLR only).

The output shall be transformer balanced and isolated, with a source impedance of 150 ohms, capable of driving a 10dBu signal into a 2k_ load. The output connector shall be a 3-pin XLR socket.

An earth lift switch shall be provided to disconnect input and output grounds when required.

The unit shall obtain power from a 48V phantom supply.

The unit shall achieve or exceed the following specifications:

Output noise -100dBu. 20Hz to 20kHz unweighted, with input terminated by 10k ohm resistor.

Distortion (THD+N) < 0.01% @ 1 kHz, +4dBu.

Frequency response +0.5/-1dB 20Hz to 20kHz.

Power consumption <10mA

The Direct Injection Module shall be the Klark Teknik model DN100 and no alternative option is available.

Technical Specification

Input

Type	active electronic, balanced or unbalanced
Impedance	1M ohms nominal, balanced or unbalanced (jack connectors) 20K ohms (XLR input only)
Connectors	2 quarter inch jacks and 3-pin XLR linked in parallel
Max. Level	30dBu
Attenuator	20dB, switchable

Output

Type	Transformer Isolated, balanced
Impedance	300 ohms
Connector	3 pin XLR
Max. Level	10dBu with load >2k ohms
Min. load	600 ohms

Performance

Noise	-100dBu, 20Hz to 20kHz unweighted, with input terminated by 10k resistor
Frequency response	+0.5/-1dB 20Hz to 20kHz
Distortion (THD+N)	<0.01% @ 1kHz, +4dBu output

Power Requirement

Voltage	+48V Phantom *
Current consumption	<10mA

Weight

<1kg

Dimensions

Length	142mm (5.6 inch)
Width	106mm (4.2 inch)
Height	60mm (2.35 inch)

** The DN100 has been designed to allow use at phantom voltages less than +48V. The unit will function down to +20V (when used with 6k8 dropping resistors) but with reduced headroom and dynamic range. All the specifications above are quoted using standard +48V Phantom power.*

Trade Descriptions Act:
Due to the company policy of continuing improvement, we reserve the right to alter these specifications without prior notice. E&OE.

KLARK TEKNIK
SIGNAL PROCESSING BY DEFINITION

ACCESSORIES: all the extras

The Klark Teknik LBB485 is a dedicated, high-quality RS-232 to RS-485 converter. It allows connection of a computer to DN9848s and DN8000s to facilitate remote control using the Stardraw control software, and (in the case of DN9848s), manipulation of program data using the Excel-based Preset Editor.

Security Covers: for installation purposes, applicable KT units can be supplied with either smoked perspex or brushed aluminium security covers of the appropriate size.

Balancing Transformers: most KT units can be supplied with input and / or output balancing transformers if required.

Dual Power Supplies: the DN1248 Plus active splitter system and the DN1414 multiple DI module can be factory-fitted with dual power supplies if required. The suffix 'DP' is applied when this option is specified.

All-Transformer Balancing: the DN1248 Plus can also be factory-fitted with all outputs transformer balanced if required. The suffix 'AT' is applied if this option is specified. This unit can also be fitted with both the Dual PSU option and the All-Transformer option if required, in this case the suffix 'FM' is applied.

FAQ: frequently asked questions

The DN9848 displays its filter steepness as "bandwidth" in octaves – what are the corresponding values expressed as "Q"?

DN9848 PEQ Bandwidth	Equivalent Q setting
0.08 Oct	18.03
0.1 Oct	14.42
0.2 Oct	7.21
0.3 Oct	4.80
0.4 Oct	3.60
0.5 Oct	2.87
0.6 Oct	2.39
0.7 Oct	2.04
0.8 Oct	1.78
0.9 Oct	1.58
1.0 Oct	1.41
1.2 Oct	1.17
1.5 Oct	0.92
2.0 Oct	0.67
2.5 Oct	0.511
3.0 Oct	0.40

How do I connect my computer to the RS-485 port on my KT loudspeaker processor ?

Connection options for Klark Teknik Products when controlled from external computers

The following information applies to recent KT products – the DN9848, DN9340 and DN9344. For older products (DN8000, DN3600 family) please contact the factory.

There are two separate ports on the units – a front panel (RS-232) port and a rear panel (RS-485) port. The RS-485 port should be used for long cable runs as it is a balanced interface. Note however, that for safety reasons software upgrades can only be applied using the front panel (RS-232) port. To accommodate all the various possibilities on current computers, we support both serial and USB interfaces. Since these interfaces differ in internal operation, there are three possible modes selectable in the software – "Echo On", "Echo Off" and "KT LBB-485". It is essential to set this correctly for the interface in use.

1. Direct RS-232
Cable connection from an RS-232 serial port (D9 connector), using the grey cable supplied with the device to the front panel port.
Use mode "Echo On" in the software program.

2. Klark Teknik LBB-485
This unit is the ideal solution if you have a PS/2 mouse port (for power) and a D9 serial port on your computer. It has the advantage of using XLR connections, so the cabling is very easy and "roadworthy".
Use mode "KT LBB-485" in the software program.

3. KK systems USB232
This provides an RS-232 port from a USB laptop. This can then be combined with the grey D9 to MINI-DIN lead supplied with the unit to connect to the front port.
Use mode "Echo On" in the software program.

4. KK systems K3-ADE plus power supply
This is a 232-485 converter - combine this with the KK systems USB232 described above to provide USB to 485 conversion. The external power supply (also from KK) is necessary to ensure reliable operation on long cable runs.

Pin connections are:

DN9848 XLR Pin 1 = KK D9 pin 9
DN9848 XLR Pin 2 = KK D9 pin 3
DN9848 XLR Pin 3 = KK D9 pin 8

Switch positions should be:

1 = ON (2 WIRE)
2 = OFF (RXEN)
3 = OFF (TXEN)
4 = ON (baud rate)
5 = ON (baud rate)
6 = ON (baud rate)

Use mode "Echo Off" in the software program.

KK systems info at http://www.kksystems.co.uk/

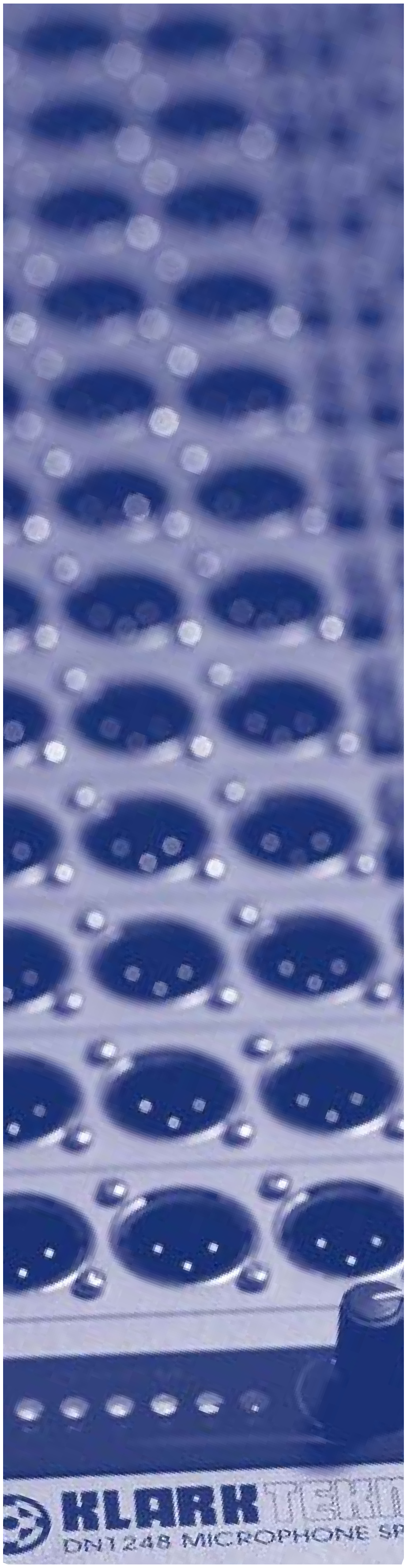
4. B+B Electronics USOTL4
This is a USB-to-485 box, with barrier strip terminals inside the box. The only disadvantage is that you can't use it to reprogram units as it is RS-485 only. However, it is in some ways a more convenient and "roadworthy" package for remote control once the units are re-programmed (either with a different PC or using the KK USB232). .

Pin connections are:
DN9848 XLR Pin 1 = GND
DN9848 XLR Pin 2 = RDB (+)
DN9848 XLR Pin 3 = RDA (-)

Switch positions should be:

1 = ON (TD485)
2 = OFF (Echo on)
3 = ON (2-wire)
4 = ON (2-wire)
Use mode "Echo On" in the software program. B+B info at http://www.bb-elec.com/

The solutions outlined above have been tested with current Klark Teknik software on Windows 98, 2000, and XP. Other interfaces may well also work, but these are the ones we have tested. If you wish to use a different interface, or are a manufacturer who would like to supply a unit for evaluation, please contact Klark Teknik.



FAQ: frequently asked questions

How should I set the gain on my Active Splitter system?

The use of an active microphone splitter system in place of the traditional passive transformer-based splitter provides a number of clear advantages. These include easier control over microphone powering, headphone monitoring facilities, and metering, in addition to the fundamental advantage of improved line drive capability. The combination of low output impedance and higher signal level mean that an active splitter is potentially capable of quieter performance and better noise immunity than a passive one, as well as minimising high-frequency losses due to cable capacitance. However, if these benefits are to be realised in practice, it is necessary to set up the complete system (including both the splitter and the console) with the correct gain structure. Failure to do this may result in the system actually performing worse when compared with a simple passive splitter, so it is well worth spending a few minutes to get familiar with the concepts involved. The reason that it matters at all is that amplifiers are not perfect. All active electronics add a small amount of noise to the signal - for example a typical well-designed amplifier will have residual noise at around -100dBu on its output, irrespective of any input signal.

Figure 1

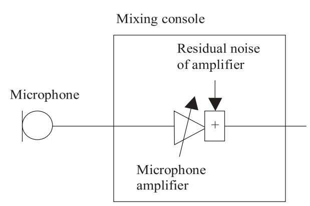


Figure 1 shows a microphone connected directly to a console. In this simple case we bring the microphone signal into the console, and immediately amplify it in the first active stage - the microphone amplifier. To take a practical example, a common dynamic vocal microphone subjected to an SPL of 110dB will produce an output of approximately -33dBu. In order to bring this up to a usable level in the console, we will set the microphone amplifier to +33 dB of gain, resulting in a 0dBu signal leaving the amplifier. To this will be added the noise of the amplifier, but since this is at around -100dBu on the amplifier output, we still have a signal-to-noise ratio of around 100dB.

Figure 2

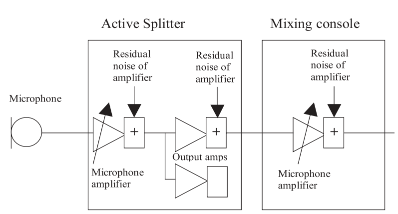


Figure 2 shows the same signal connected using an active splitter system. The splitter contains a variable-gain microphone amplifier, which then feeds a number of independent output amplifiers. One of these is then connected to the console input, which itself has a variable-gain microphone amplifier. The crux of the matter is how best to set the gain of the two microphone amplifiers. It is tempting to simply set the splitter to unity gain, and insert it in the signal path expecting nothing to change - after all, this is what we would do with a passive splitter. However, we can immediately see a problem with this approach. We bring our microphone signal at the same level of -33dBu into the splitter, but now instead of it immediately hitting an amplifier with gain, it is simply passed at the same level through the splitter. The splitter's microphone amplifier and the line driver will each add noise at about -100dBu to this signal, just as the console's microphone amplifier did. Note that because the signal is still at -33dBu, the signal-to-noise ratio at point A is now only 67dB. This signal arrives at the microphone amplifier in the console, and we boost the whole thing by +33dB. This restores the signal to 0dBu as desired, but also brings up the splitter's output noise by 33dB - so we still have a signal-to-noise of only 67dB. The additional noise at -100 from the console's microphone amplifier is of no real consequence in this case.

In order to restore the performance of our system and to actually benefit from the improved line driving ability of the splitter, what we should have done is to use the microphone amplifier on the splitter. If we set the splitter's microphone amplifier to +33dB of gain, then the noise contribution of that amplifier (at -100dBu) will now be added to a signal with a level of 0dBu, instead of -33dBu. This will preserve our 100dB signal-to-noise ratio in the splitter, instead of reducing it to 67dB. The console input section is now set to 0dB of gain, so there is no increase in the splitter's noise contribution as a result, and we merely add the console noise at -100dBu to our signal.

So, the conclusion that we reach is:
When using any active splitter system, as much gain as possible should be added using the splitter's microphone amplifier, and as little as possible using the console.
Obviously the limit of this approach will be the point at which the splitter's output will clip on loud sounds. It is worth noting, however, that with the popular dynamic vocal microphone used for this example, and +30dB of gain on the splitter's microphone amplifier, that it would require an SPL of 133.3dB to produce an output of +20dBu from the splitter - still within the output capability of most professional equipment.

FAQ: frequently asked questions

What is the sampling rate and wordlength of the DN9848 ?

We are often asked questions such as "why don't you quote the number of bits for your analogue-to-digital converters (ADCs)?" by people wishing to compare our equipment with products from other manufacturers. This has been a deliberate policy, because of the danger of making "over simple" comparisons between competing units based on numbers of bits or sample rates. In many cases the actual performance may differ substantially from the "apparent quality" based on the numbers in the specification. So, in response to these questions, here is a summary of the DN9848 architecture, with some background on how this can be sensibly compared with competitor products.

DSP sample rate is 48 kHz. This allows us a theoretical 24kHz audio bandwidth, although we only specify 20 Hz to 20 kHz, and we deliberately roll off above 20kHz. In our opinion, bandwidths wider than this are in general undesirable for live sound, as they merely increase the likelihood of HF driver failure without any sonic advantage. Many people over the years have conducted subjective listening tests comparing 96kHz sampled systems to 48kHz systems and found that they sound different. However, this usually involves different analogue stages, different ADCs and DACs, different phase responses and so on, so it is no surprise that they sound different. On the other hand, if a 96 kHz sampled system is built, and then a 20 kHz digital filter is introduced inside the system, we remain convinced that the result is inaudible. This assumes, of course, that the filter is linear phase and has low ripple in the passband (not always the case !). 96kHz sampling also causes problems with the noise performance of low frequency EQ stages (because the differences between adjacent samples are smaller), so a 96 kHz system typically requires a longer wordlength to achieve the same noise performance as a 48 kHz one. The one advantage of a 96kHz system in live sound is that it is possible to reduce the latency (delay) through the system a little. Note also when comparing 96kHz and 48kHz systems that many 96kHz systems specify audio bandwidths of 30kHz or even 40kHz, and then only specify the noise performance up to 20kHz. Clearly if the system is flat to 30kHz, then all the noise up to 30kHz will be arriving at the power amplifiers and should be included in the noise measurement. This is particularly true when oversampling ADCs are used, which have a noise profile that typically rises with frequency.

DSP wordlength is 24-bit, fixed-point (optionally 48-bit fixed-point where necessary for the algorithms). This gives us a theoretical internal dynamic range of 144 dB, so this is comfortably better than the converters that are currently available. Fixed-point versus floating-point is a big discussion, but in general a 24-bit fixed-point system is harder to design than a 24-bit floating-point system but sounds better. This is because when there is a typical loud-ish signal level passing through the unit, the "step size" available between samples is smaller on the fixed-point system. In addition, the step size is fixed, whereas a floating-point system has a variable step size depending on the instantaneous signal level. In other words in a floating-point system the quality of the quiet hi-hat cymbal will be modulated by the signal level of the bass guitar - not generally a good thing... Obviously the floating-point system has a theoretical noise advantage at very low signal levels, but by the time the level is low enough for this to be significant, the ADC and DAC noise will be dominating, not the DSP noise.

The ADC and DAC parts that we use are both "nominal 24-bit" items, but this is essentially meaningless. If a manufacturer claims that they have a "24-bit converter" in their product, then the next question to ask is how you should measure the unit to confirm the 144 dB dynamic range that this implies. In practice no-one is achieving even 20-bit noise performance (=120 dB dynamic range) from a digital system of this kind at the present time. The DN9848 achieves >114 dB dynamic range or "19 bits" overall from input to output. Note that this is an unweighted figure (i.e. flat frequency response). Some manufacturers quote "A-weighted" figures which flatter the unit's performance significantly by applying a psycho-acoustic curve to the measurement. Measurements which specify the dynamic range of the ADC or DAC in isolation should also be treated with caution, since these are often "data sheet" numbers supplied by the IC manufacturer which are rarely if ever achieved in practice. The ultimate safety net is to say "could I verify this measurement myself with an example of the unit and a test set ?" – if you can, then the manufacturer is unlikely to be exaggerating – the potential for embarrassment is too great ! If the figures can only be verified by calculation or internal connections to the circuitry, then the figures may be less useful…

The other key performance issue even for digital products is the analogue audio stages - in particular the difference between bench measurements and real-world performance. KT units are designed to perform not only when connected to test equipment on a bench, but also when driving long cables, unbalanced loads, and in the presence of external electrical and magnetic fields. Issues such as common-mode rejection (especially at high frequencies) and impedance balancing of outputs can have a dramatic effect on the actual performance obtained, as opposed to the "brochure specification".

In the end, the one-sentence summary is "don't worry too much about the bits and sample rates - trust the same real-world performance measurements of noise and distortion that you would apply to analogue".

And after that, there are always your ears…

FAQ: frequently asked questions

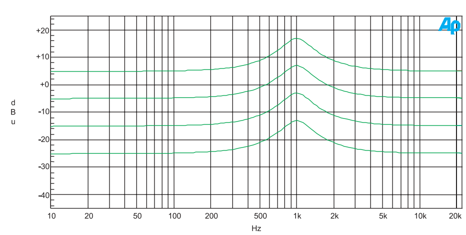
What is Dynamic Equalisation (T-DEQ)?

Over the years a number of professional audio products have provided dynamic equalisation functions of various types. What all these systems have in common is that the frequency response of the device varies depending on the signal level. Many units are based on compressor / expander technology with frequency selection, and the controls often resemble those of a dynamics processor.

The system developed by the Klark Teknik research and development team for the Helix series is rather different. It draws on KT's unrivalled experience in equalisation, and uses the signal level to directly control parametric equalisers. This purely EQ-based solution allows simple controls that directly relate to the signal levels. As a result, it is very easy to set the point at which the dynamic EQ starts to operate, and also to set precisely its maximum effect. We refer to this technique as "Threshold Dependent Equalisation".

In order to understand the operation, let us first consider a conventional parametric EQ section (Figure 1). The three controls available to us are frequency, Q (or bandwidth), and the amount of cut or boost.

Figure 1 – standard parametric EQ

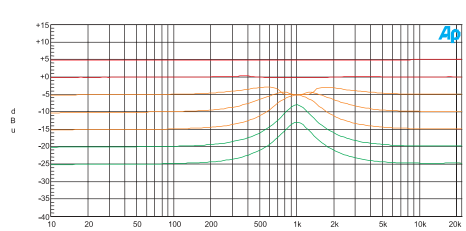


This shows a series of responses for the parametric EQ with different input levels. As expected, there is no change in the shape of the curve with different input levels. If the input is 10dB louder, the output is 10dB louder at every frequency.

If we now replace the parametric with a Helix equaliser and select the dynamic EQ, we have some additional controls. Frequency and Q controls are as before, but now we have two pairs of controls replacing the single cut and boost control; these are [low threshold] / [low level], and [high threshold] / [high level]. If we set the frequency and Q controls to the area that we wish to control, then the processor will monitor the signal level in that frequency range. If the signal level in this part of the spectrum is below the [low threshold] setting, then the unit considers this a 'quiet' signal. The EQ applied to the signal will be controlled by the [low level] control. If the signal level is above the [high threshold] level, then the unit considers this a 'loud' signal, and will apply the amount of EQ set by the [high level] control. If the signal level is between the two thresholds, then the equaliser will seamlessly morph between the two equaliser settings in real time. Manual control over attack and release times is available to set the speed of response to suit the application.

As an example, consider Figure 2, which shows the Helix applying a boost at low signal levels which is automatically 'wound out' at high level.

Figure 2 – Helix with boost at low signal level



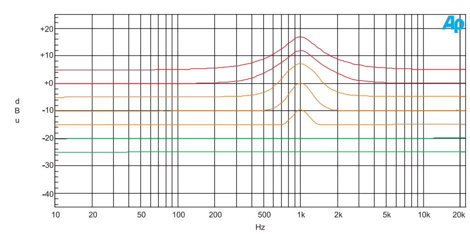
Klark Teknik Proportional Q

In this example, [low threshold] is -20dBu, [low level] is +12dB, [high threshold] is set to -5dBu, and [high level] is 0dB. Thus the lowest trace shows an input at -25dBu with a standard parametric boost of +12dB at 1kHz. The -20dBu trace shows an identical response, as expected. However, once above this level, the filter gradually fades out with increasing signal, until at all levels above 0dBu, the response is flat.

The shape of the curves for -5dBu and -10dBu require some explanation. These appear as they do because of the nature of the frequency sweep measurement. The Helix equaliser uses a copy of the actual filter in use for its level calculation, so that depending on the Q of the filter, our input signals are 'ignored' as we move away from the centre frequency by the correct amount. Thus as the sweep measurement moves across the centre frequency (1kHz in this case), the dynamic EQ is ramping smoothly in and out again, leading to the curves in Figure 2. Note that if the level is outside the range specified by the two thresholds, the unit behaves like a fixed parametric EQ. This means that we do not have to guess how much EQ will eventually be applied - it is explicitly set in advance.

Without changing modes or making any other selections, we can make the unit operate 'the other way up' just by selecting suitable values for the two thresholds and levels see Figure 3.

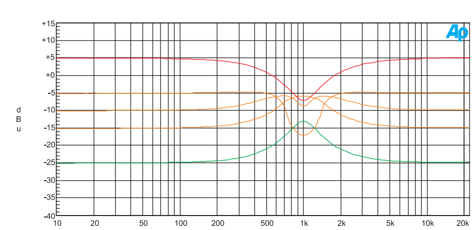
Figure 3 – Helix with boost at high signal level

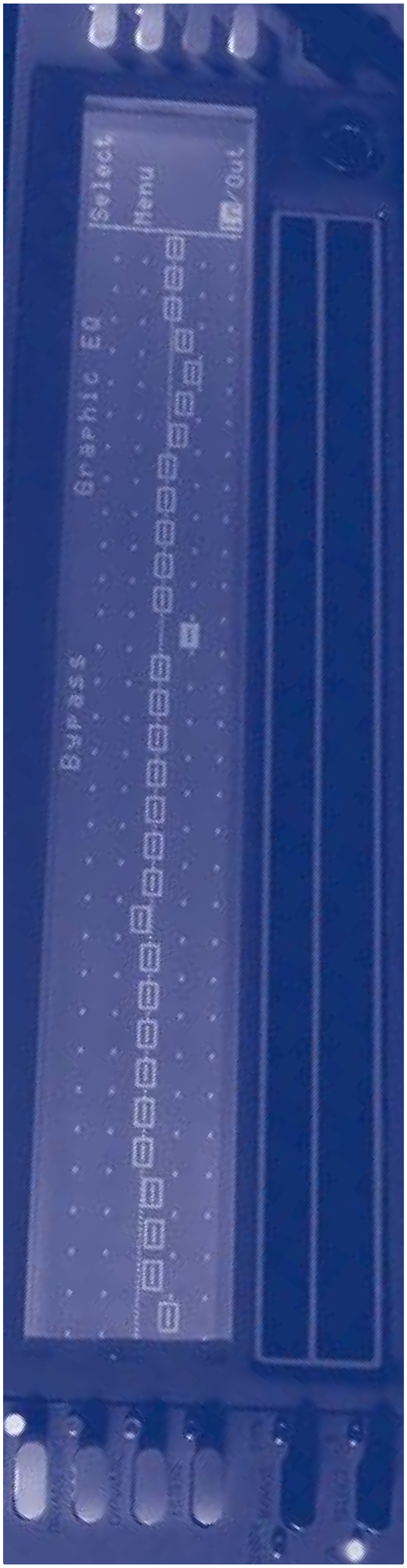


In this case, [low threshold] is -20dBu, [low level] is 0dB, [high threshold] is -5dBu, and [high level] is +12dB, so that instead of cutting this frequency range as the level increases, we are now boosting it. Again, we have precise control over the maximum amount of boost that will be applied, and the level at which this will occur. Note the shape of the curve for -5dBu, which has 'expected values' outside the filter range and at the centre frequency, but intermediate values that show the EQ ramping in and out either side of the centre frequency.

Needless to say, there is no requirement for one of the levels to be 0dB. Figure 4 shows the transition from a +12dB boost at low level to a -12dB cut at high levels. Again, the intermediate curves show the effect of the sweep signal moving in and out of the 'area of interest' of the level detector as the curve is formed.

Helix with boost at low level and cut at high level

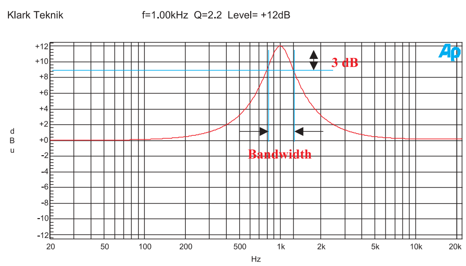




FAQ: frequently asked questions

What is the difference between the various Q types on the DN9340 Helix Equaliser?

The “Q” of an audio equaliser describes the steepness of the filter - the degree to which it will affect signals either side of its nominal or “centre” frequency. In general, the Q of a peaking filter is defined mathematically as , centre frequency / bandwidth where the bandwidth (in Hz) is the range of frequencies affected by the filter. Because the frequency response of such a filter is a smooth curve (not a sharp “brick wall” filter like the ones in an analogue-to-digital converter) we have to decide how we choose to define the bandwidth, and the established convention is that we use the bandwidth to the “-3dB” points on either side of the centre frequency, where the gain is 3dB less than the maximum gain.



In the example above, the filter is centred on 1 kHz, the lower 3dB point is at approximately 800 Hz, and the upper one is at approximately 1.25 kHz. This filter therefore has a Q of 1000 / (1250-800)=2.2 In a typical parametric equaliser (and in the case of the Helix system the graphic and dynamic sections too) we have a manual control for the Q of the filter, and this allows us to set any Q that we require. In general high-Q, narrow filters are used for notching out problem frequencies without affecting the programme material too much, while gentler low-Q filters are useful for adjusting the tonal balance. In the case of graphic equalisers there is another issue - that of interaction between adjacent bands. In general, lower-Q filters will blend together more smoothly, but higher-Q filters provide more selective control of problems - at the expense of more frequency response ripple.

So far so simple - but why the different types? This is due to the way in which the Q of the filter varies (or not) when the gain control is adjusted. There are three modes available in the Helix system, which we term Proportional, Constant, and Symmetrical Q.

Proportional Q

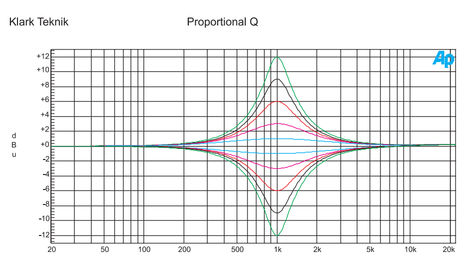
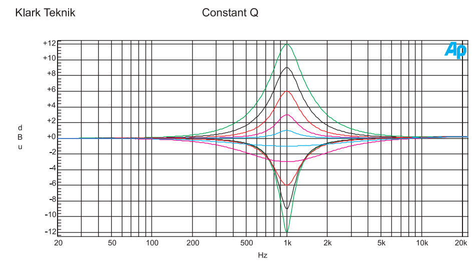


Figure 3 – Helix with boost at high signal level

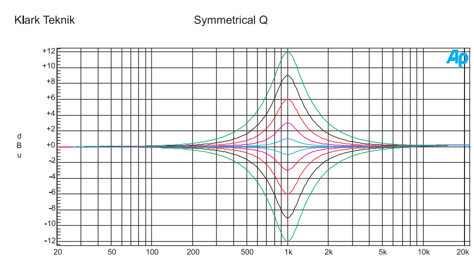
Proportional Q is the mode of operation familiar to users of the Klark Teknik analogue graphic equalisers such as the DN360. As the amount of cut or boost is increased, the Q also increases. This has the effect of making the equaliser “focus” more tightly as the amount of EQ is increased. This allows a fairly low-Q filter at small cut and boost settings, providing gentle control of tonal balance and low ripple. At high gain settings, a proportional-Q equaliser “automatically” increases Q for more dramatic problem solving such as suppression of feedback or unwanted resonances. In the interests of clarity, the Q setting shown on the display is the Q at full cut or boost - the Q at lower gain settings will be lower than that shown on the panel.

Constant Q



A constant Q equaliser has the same Q at all cut and boost settings. In other words, the bandwidth between the 3dB points does not change at all as the gain is adjusted. The really important thing to notice about this is that the resulting frequency response is NOT symmetrical in cut and boost. This is because of the definition of Q which is based on the 3dB points relative to maximum gain. The maximum gain of the filter when in cut is, of course, 0dB, and the bandwidth is determined by the -3dB points relative to 0dB and NOT relative to the minimum gain (at the centre frequency). This makes a lot of sense musically too - if you listen to a music signal and apply a notch filter, and then change the shape of the curve around the minimum gain (centre) point, it will make little difference to the sound (since that area is already attenuated a lot). However, if you change the curve around the 3dB points, this will affect the sound much more, as more or less of the signal “falls into” the notch. It is this bandwidth that the constant-Q filter is keeping constant. Note that many equalisers that are described as “Constant Q” by their manufacturers do NOT fall into this category, and are what we would term symmetrical-Q designs.

Symmetrical Q

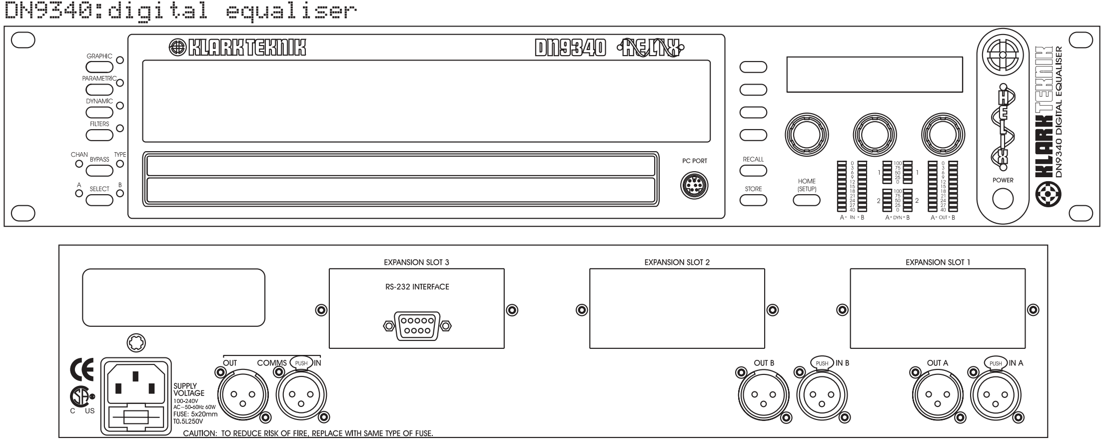


This class of equaliser has the same curves in boost as the constant-Q type, but then has cut responses that are symmetrical with the boost ones. In other words, the bandwidth in cut is defined not according to our usual definition of Q (see constant-Q above) but as “the point were the signal is cut by 3dB less than the maximum cut”. Most equalisers described by their manufacturers as “Constant Q” in fact produce symmetrical responses.

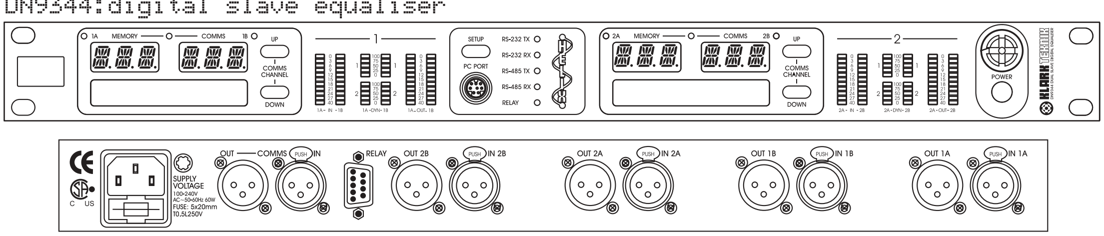
LINE DRAWINGS

These diagrams are for pictorial reference only

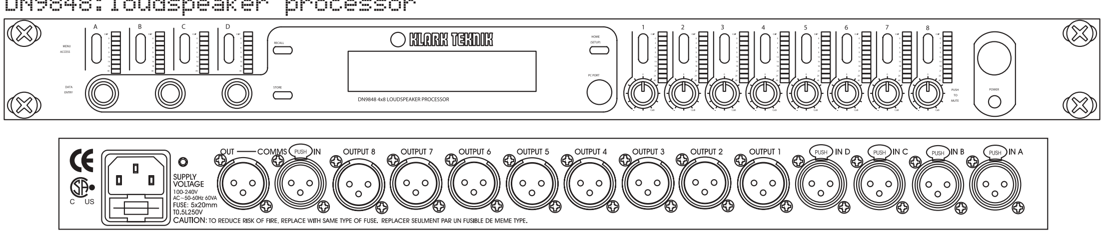
DN9340:digital equaliser



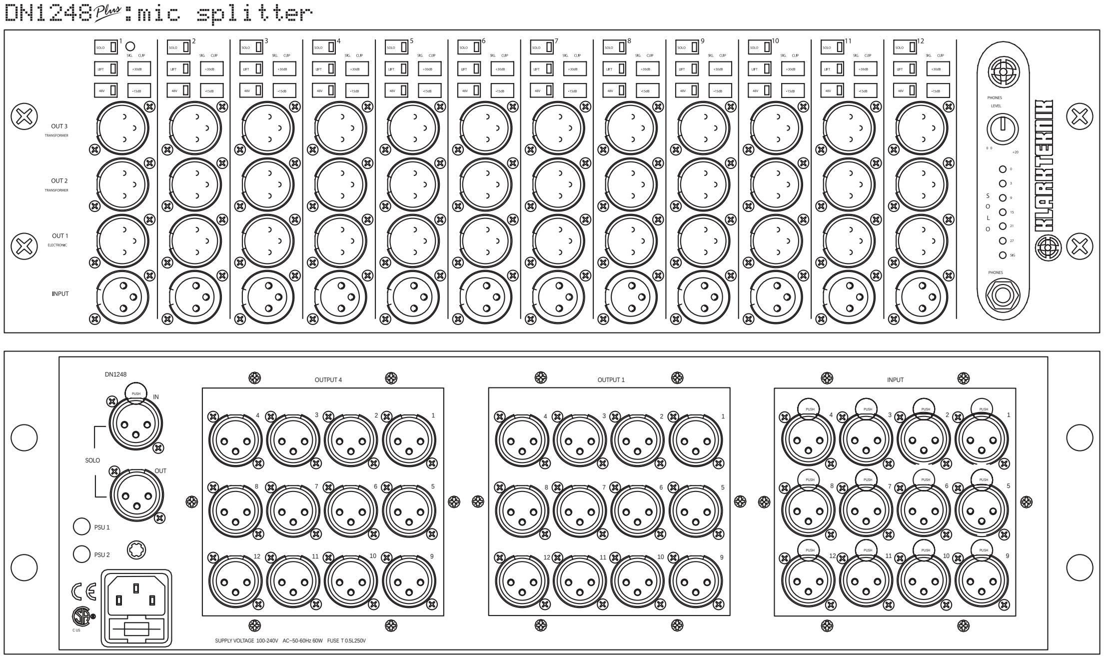
DN9344:digital slave equaliser



DN9848:loudspeaker processor



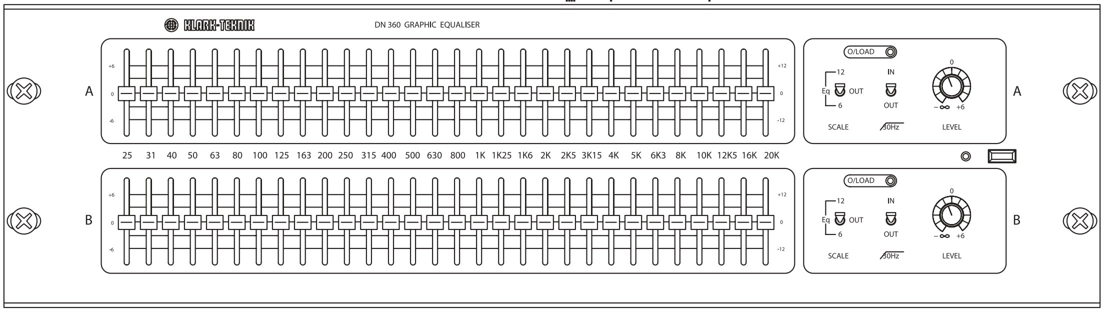
DN1248*Plus*:mic splitter

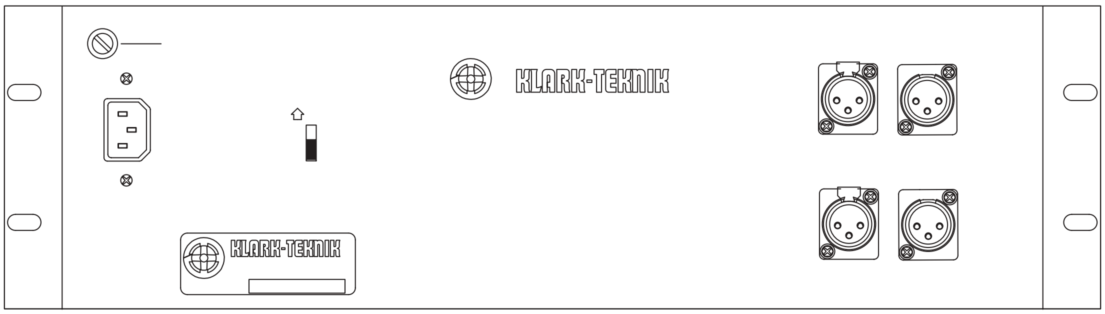


LINE DRAWINGS

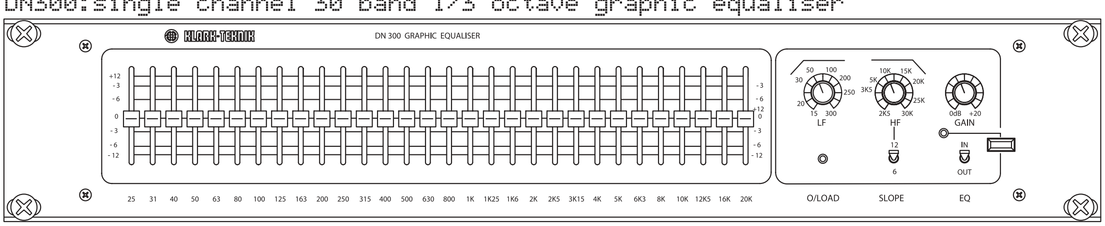
These diagrams are for pictorial reference only

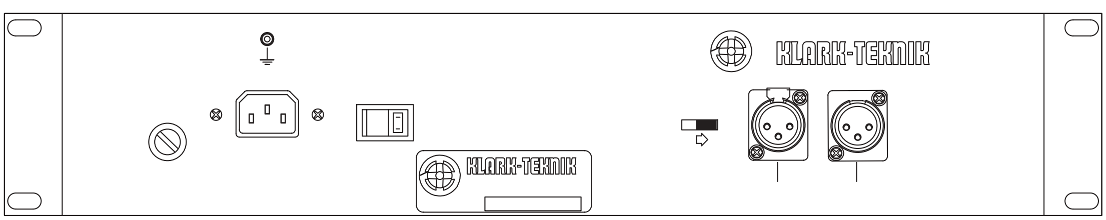
DN360:dual channel 30 band 1/3 octave graphic equaliser



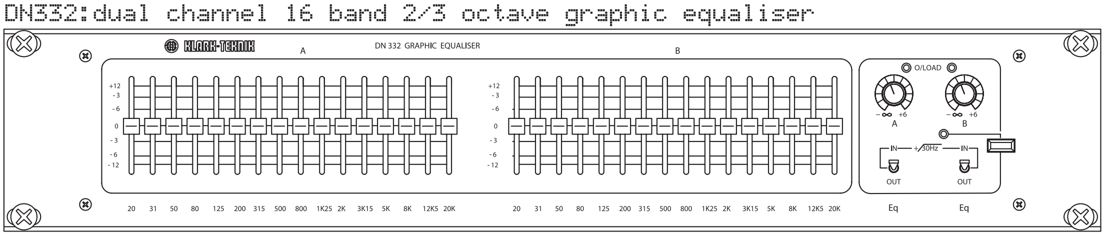


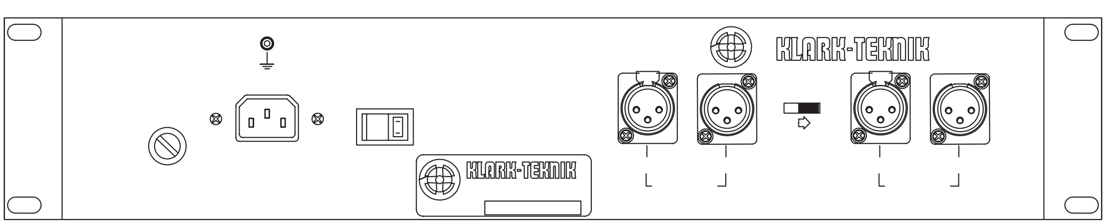
DN300:single channel 30 band 1/3 octave graphic equaliser





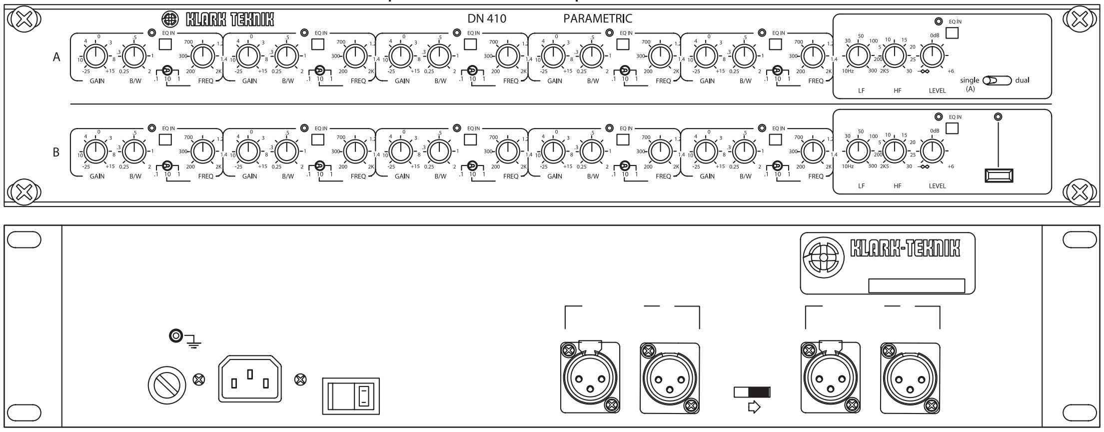
DN332:dual channel 16 band 2/3 octave graphic equaliser



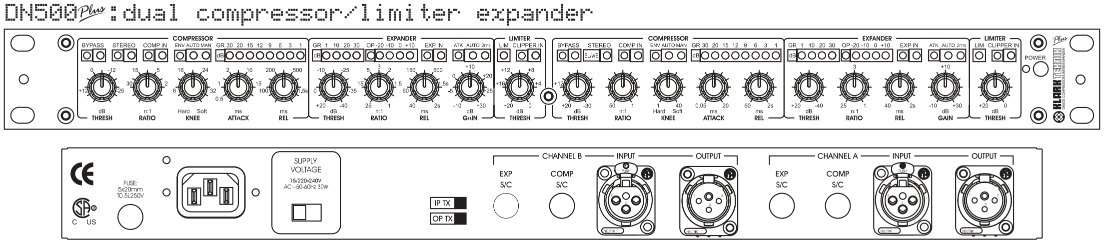


These diagrams are for pictorial reference only

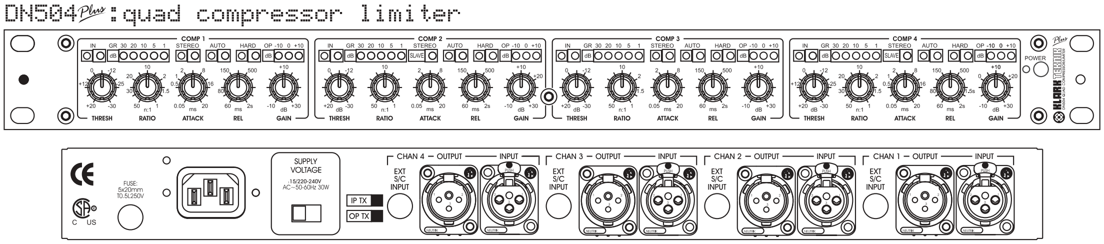
DN410:dual channel 5 band parametric equaliser



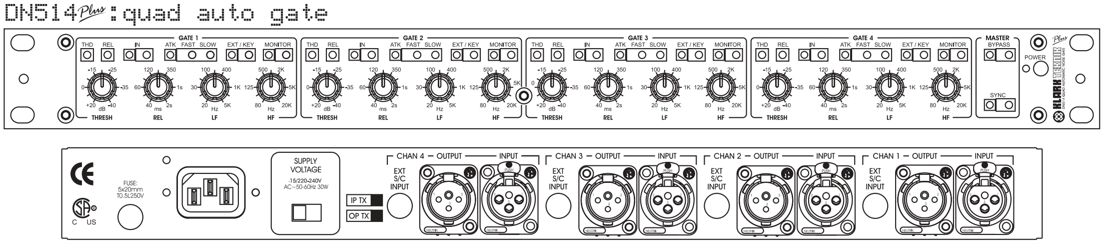
DN500Plus:dual compressor/limiter expander



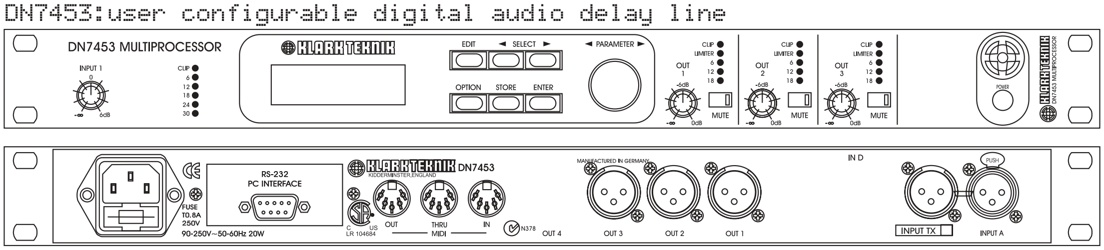
DN504Plus:quad compressor limiter



DN514Plus:quad auto gate



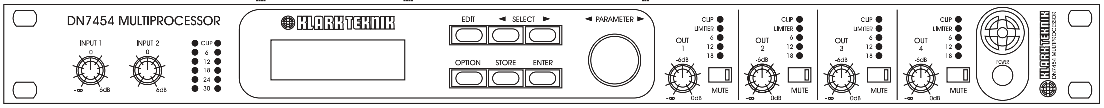
DN7453:user configurable digital audio delay line

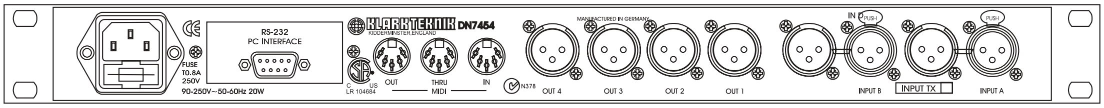


LINE DRAWINGS

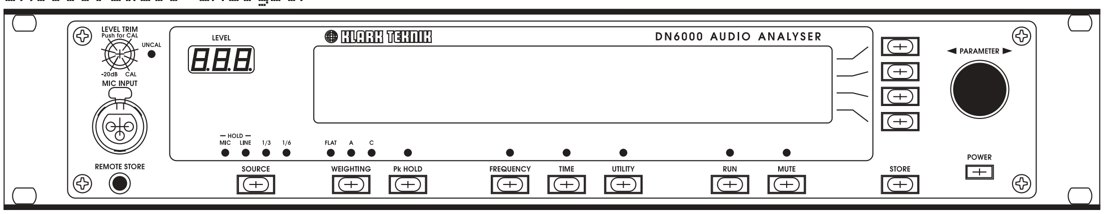
These diagrams are for pictorial reference only

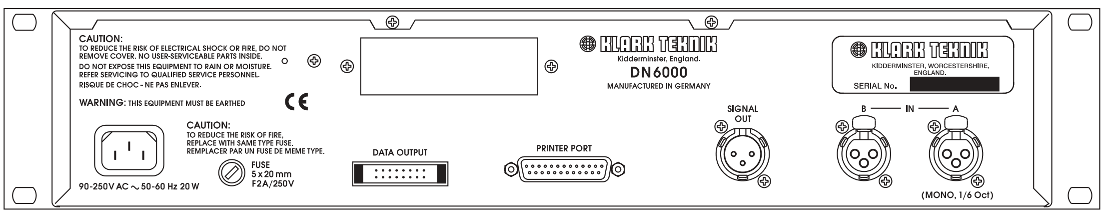
DN7454:user configurable digital audio delay line



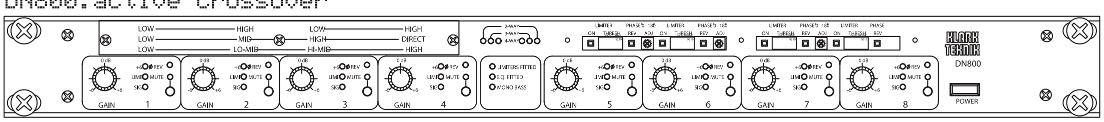


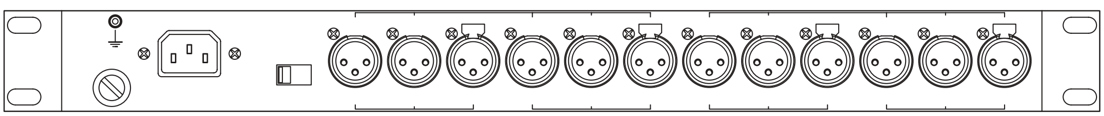
DN6000:audio analyser



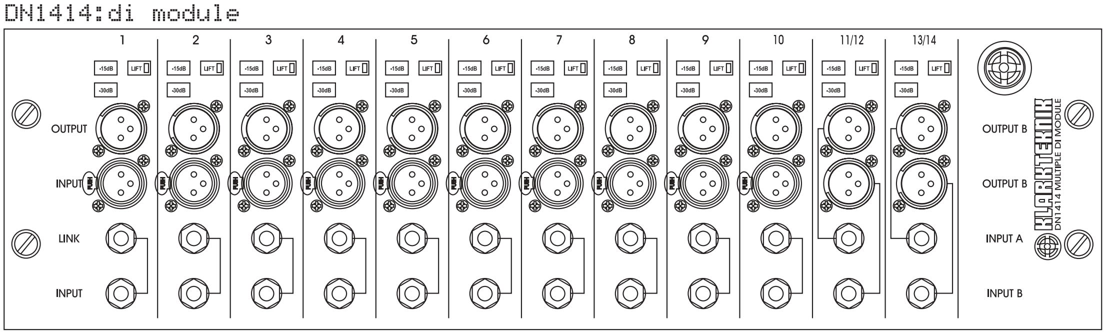


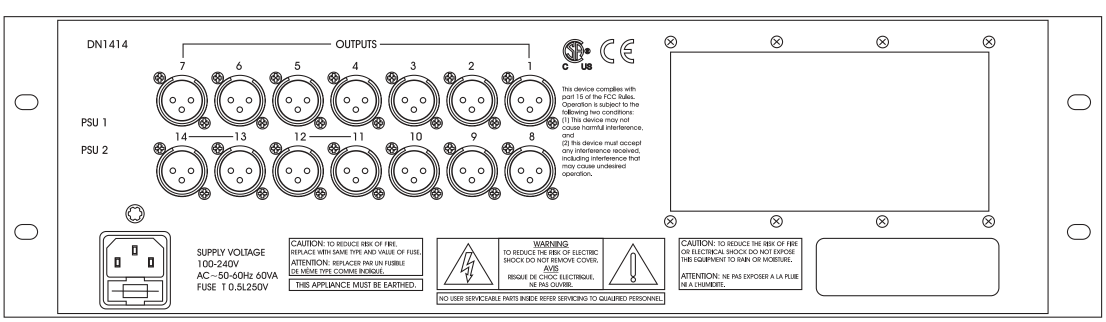
DN800:active crossover



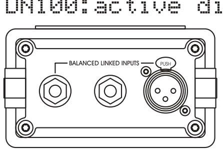


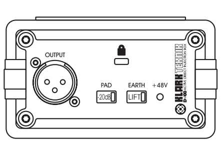
DN1414:di module





DN100:active di





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