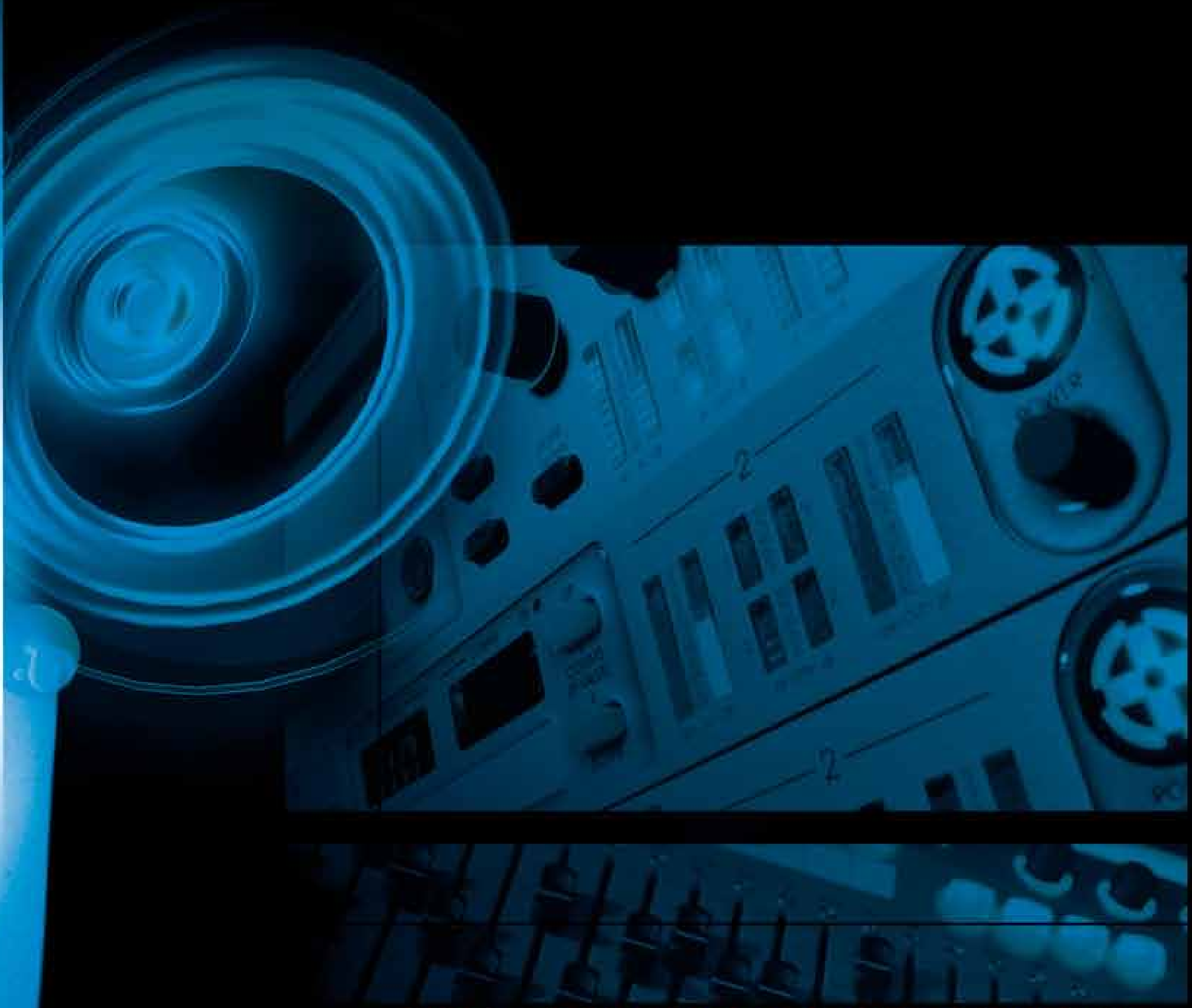

KLARK TEKNIK DN SERIES PRODUCT CATALOGUE 7

Introduction

In 1974, brothers Phil and Terry Clarke founded Klark Teknik Research Ltd. In the years immediately following, their pioneering approach to design and development allowed them to introduce some truly groundbreaking designs. One of 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 resulting in the uniquely capable DN370 and the world famous DN360.

Today Klark Teknik continues to bring innovation in design, engineering and sonic quality in both the analogue and digital realm of signal processing. 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.

Klark Teknik products are represented by an international network of appointed distributors, all of whom are authorised to sell and provide technical support for our products. Full contact details for all our distributors are available from the website at www.klarkteknik.com, however please contact the factory direct for information if necessary.

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KLARK TEKNIK
SIGNAL PROCESSING BY DEFINITION

Helix DN9331 RAPIDE - Graphic Controller

Unique, immediate and tactile, the DN9331 Helix RAPIDE offers direct access to all of the graphic equalisation functions of Helix digital equalisers.

Instant recall of fader positions is made possible by the use of thirty one console-quality 100 mm long travel high resolution motorised faders, custom manufactured to Klark Teknik's exacting standards, featuring long life conductive plastic tracks and driven by fast acting precision servo control circuits. A generously specified power supply ensures high speed of response, and can supply the peak currents required by simultaneous multiple fader movements, without the lag effects experienced with lower-grade remote fader units.

Integrating a Midas/Klark Teknik STS Solo Tracking System interface, the Helix RAPIDE is ideal for use in stage monitoring systems, when combined with a Midas Heritage, Legend or Siena console, the solo buttons on each aux send can be used to instantly recall the graphic equaliser settings of the connected channel of Helix digital equalisation, offering the monitor engineer unparalleled speed of access in situations demanding an immediate response.

A flexible user interface allows custom remote channel assignments across four banks of 32 channel selection buttons. Four freely assignable group buttons and a global 'all channels' button allow relative adjustment of channels, especially important when the priority is to stop on-stage feedback first, and determine the source second.

The Helix RAPIDE is the networking centre of the Show Command System, an eight external port Ethernet hub is incorporated into the device, allowing the connection of Helix digital equalisers and system processors, with wired or wireless connections to laptop or tablet PCs running the Elgar Helix EQ RCS and System Controller RCS.

The Helix RAPIDE is fully backwards-compatible with the original Helix DN9340 and DN9344 digital equalisers which may be interfaced using Ethernet-to-serial converters.

Architect's & Engineer's Specification

The Digital Graphic Equaliser Remote Controller shall offer control of 31 bands of Klark Teknik Helix graphic equalisation and have 31 motorised 100 mm faders representing frequencies from 20Hz-20kHz on 1/3 octave centres to BS EN ISO 266:1997.

The unit shall be contained in a 6U 19" rackmount enclosure, which shall be so designed as to also allow the unit to perform as a freestanding console.

The unit shall provide remote control of up to 64 channels of Klark Teknik Helix graphic equalisation. The remote control interface shall be via Ethernet and there shall be an Ethernet switch integrated into the unit with eight external ports fitted with Ethercon connectors. There shall also be a rear panel RS-232 port provided for remote control from Midas Heritage, Legend and Siena mixing consoles, which implements the Klark Teknik and Midas Solo Tracking System control.

The user interface shall provide for four banks of 32 channels of user-assignable channel access, implemented as four bank and 32 channel non-latching illuminating pushbutton switches. There shall be four non-latching illuminating pushbutton switches provided for the selection of groups, each of the four groups permitting the relative adjustment of multiple channels of Helix graphic equalisation. There shall also be a non-latching illuminating pushbutton for global access of all addressable channels of Helix graphic equalisation and applying relative adjustment to all channels. A non-latching illuminating pushbutton shall also be provided to bypass the currently selected channel(s) of Helix graphic equalisation.

Eleven-segment LED bargraph meters shall be provided for monitoring the input and output audio signal levels of an individual channel.

A 20 x 2 alphanumeric LCD display shall be provided for the display of parameter information and three rotary encoders shall be provided for parameter adjustment. Momentary pushbutton switches shall be provided for memory store and recall and setup menu access.

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

The digital graphic equaliser remote controller shall be the Klark Teknik DN9331 Helix RAPIDE, and no alternative specification option is available.

Technical Specification

Equalisation	**31 Bands**
Centre Frequencies	To BS EN ISO 266:1997 20Hz-20kHz, 1/3 octave
Maximum Boost/Cut	±12dB
Power Requirements	
Voltage	100V – 240V a.c. ± 10% 50/60 Hz
Terminations	
Ethernet Communications	Ethercon
Solo Tracking System	9-pin D-type
Power	3-pin IEC
Dimensions	
Width	483 mm (19 inch)
Height*	264 mm (7 inch) 6RU High
Depth	150 mm (6 inch) Top 80 mm (3 inch) Bottom
Weight	
Net	10kg
Shipping	13kg

- **STS Compatible**
- **Show Command Component**

** Requires an additional 1U space above the unit for ventilation when rack mounted.*

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

Helix DN9340E - Dual EQ

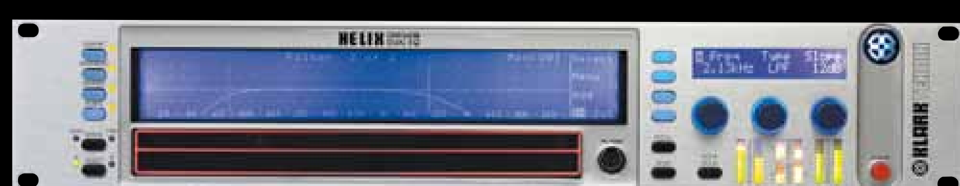
The concept of an equalisation device which allows the user to not only select from a menu of various EQ types but also to integrate them with each other in any combination is one that could only have come from Klark Teknik. At the heart of Helix DN9340E is the fact that it sounds incredible – the most common remark from first-time users is 'it sounds just like a great analogue EQ!'

For the uninitiated, the new Helix DN9340E 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. Unit 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 Siena, Legend and Heritage consoles via the Solo Tracking System (STS): this allows any channel of connected Helix EQ to be edited from the control surface of a single Helix DN9340E unit or PC simply by activating the appropriate PFL function on the console (STS only works on the outputs of the Siena).

By linking a DN9331 RAPIDE to the DN9340E it allows the user to have instant control over the Helix 31 band graphic EQ function thanks to the unique 31 Klark Teknik 100mm motorised faders fitted to this unit.

The Helix DN9340E also features AES/EBU digital inputs and outputs as standard, complete with word clock sync inputs. Whilst the internal sample rate of all DN9340/44E units remain at 48kHz/44.1kHz, these digital connections are all 96 kHz compatible allowing easy interface with any other digital device featuring the higher sample rate.

Secondly the Helix DN9340E unit also features a dual port Ethernet communications interface. This is to facilitate much faster communication, response and metering between units than was previously possible with serial comms. The Ethernet ports allow for control of the units from a PC, either via Ethernet or wirelessly with the Helix EQ Remote Control Software (RCS) an ELGAR Add-In.



Architect's & Engineer's Specification

The Digital Equaliser shall provide two audio channels (analogue and digital, in and out) in a standard 2U 19" rack mount chassis.

Each audio channel shall include: Source select (analogue or digital), 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.

Digital inputs shall run at any sample rate up to 96kHz with internal sample rate conversion. The sample rate converter can be bypassed when the incoming digital signal has a sample rate of 44.1kHz or 48kHz.

A word clock input shall be provided to allow the system to lock to an external clock source. In addition, the word clock input can be used to only clock the digital outputs allowing digital audio to come into the unit at one sample rate and go out at another.

Digital outputs shall also run at any sample rate up to 96kHz. In standard operation, they shall run at the system sample rate (48kHz or 44.1kHz). Using the internal SRC, the digital outputs can also run at the same sample rate as the digital inputs or the word clock input.

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

The high and low pass 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 analogue audio inputs and outputs shall be electronically balanced and use XLR connectors. All digital audio inputs and outputs shall be 110Ω AES/EBU 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 two Ethernet ports on the rear panel. The Ethernet 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 Digital Equaliser shall be controllable from the ELGAR remote control PC software.

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 Te knik model DN9340E and no alternative option is available.

Technical Specification

Digital Inputs	**One 2-Channel Input**
Type	AES / EBU
Impedance	110Ω
Sample Rate	44.1kHz, 48kHz, 88.2kHz*, 96kHz*
Word Length	24-bit or 16-bit

**SRC Input at these sample rates*

Analogue Inputs	**Two**
Type	Electronically balanced (pin 2 hot)
Impedance (Ω)	20k
Common Mode Rejection	>80dB @ 1 kHz
Maximum Level	+21dBu
Word Clock Input	
Type	BNC
Impedance	75Ω
Digital Outputs	**One 2-Channel Output**
Type	AES / EBU
Impedance	110Ω
Sample Rate	44.1kHz, 48kHz, 88.2kHz**, 96kHz**
Word Length	24-bit

***SRC Output - these rates are only available when used in conjunction with a word clock or a digital input running at that frequency.*

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 20.8µs 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	100V -240V a.c ±10%
Consumption	<60W
Terminations	
Audio inputs/outputs	3 pin XLR
Ethernet inputs/outputs	Ethercon
RS-232	8 pin Mini-DIN socket (front) 9 pin D-type (rear)
World Clock	BNC
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

STS Compatible

Show Command Component

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

Helix DN9344E - Quad EQ

Helix DN9344E is a fantastic example of how clever digital design can make products smaller and lighter without sacrificing functionality. Helix DN9344E Quad EQ is actually, as the name suggests, TWO complete DN9340E Helix Dual EQ 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 DN9340E Helix Dual EQ, or via Helix EQ Remote Control Software (RCS). Up to 64 channels can be controlled from one master unit or the RCS. Additionally the 31 band graphic function of the unit can be controlled via the unique DN9331 RAPIDE with its 31 motorised 100mm faders. Perfect for installations, it is also fitted with contact closures to allow for memory recall by a mechanical device.

The principal operational advantage of the Helix DN9344E is that it offers all the functionality of several standalone devices in one package, thus saving massively on both cost and rackspace. For instance, enough EQ for a 24-way monitor mix plus two sidefills will fit into just SIX rackspaces (six DN9344E Quad EQs), at a comparative cost to the same number of channels of top-class analogue graphic EQ.

The Helix DN9344E also features AES/EBU digital inputs and outputs as standard, complete with word clock sync inputs. Whilst the internal sample rate of all DN9340/44E units remain at 48kHz/44.1kHz, these digital connections are all 96kHz compatible allowing easy interface with any other digital device featuring the higher sample rate.

The DN9344E unit is equipped with dual port Ethernet communications interface. This is to facilitate much faster communication, response and metering between units than was previously possible with serial comms. The Ethernet ports allow for control of the units from a PC, either via Ethernet or wirelessly with the Helix EQ Remote Control Software (RCS) an ELGAR Add-In.

Architect's & Engineer's Specification

The Digital Slave Equaliser shall provide four audio channels (analogue and digital, in and out) grouped as two linkable pairs in a standard 2U 19" rack mount chassis.

Each audio channel shall include: Source select (analogue or digital), 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.

Digital inputs shall run at any sample rate up to 96kHz with internal sample rate conversion. The sample rate converter can be bypassed when the incoming digital signal has a sample rate of 44.1kHz or 48kHz.

A word clock input shall be provided to allow the system to lock to an external clock source. In addition, the word clock input can be used to only clock the digital outputs allowing digital audio to come into the unit at one sample rate and go out at another.

Digital outputs shall also run at any sample rate up to 96kHz. In standard operation, they shall run at the system sample rate (48kHz or 44.1kHz). Using the internal SRC, the digital outputs can also run at the same sample rate as the digital inputs or the word clock input.

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

The high and low pass 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 k Hz)
Dynamic Range 115 dB (20 Hz to 20 kHz unweighted)

All analogue audio inputs and outputs shall be electronically balanced and use XLR connectors. All digital audio inputs and outputs shall be 110Ω AES/EBU 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 two Ethernet ports on the rear panel. The Ethernet 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 Digital Slave Equaliser shall be controllable from the ELGAR remote control PC software. and have utility software to allow the editing of factory presets using an industry standard PC spreadsheet application.

The unit shall be capable of operating from a 100 to 240V, ±10%.

The Digital Slave Equaliser shall be the Klark Teknik model DN9344E and no alternative option is available.

Technical Specification

Digital Inputs	**Two 2-Channel Inputs**
Type	AES / EBU
Impedance	110Ω
Sample Rate	44.1kHz, 48kHz, 88.2kHz*, 96kHz*
Word Length	24-bit or 16-bit

**SRC Input at these sample rates*

Analogue Inputs	**Two**
Type	Electronically balanced (pin 2 hot)
Impedance (Ω)	20k
Common Mode Rejection	>80dB @ 1kHz
Maximum Level	+21dBu
Word Clock Input	
Type	BNC
Impedance	75Ω
Digital Outputs	**Two 2-Channel Outputs**
Type	AES / EBU
Impedance	110Ω
Sample Rate	44.1kHz, 48kHz, 88.2kHz**, 96kHz**
Word Length	24-bit

***SRC Output - these rates are only available when used in conjunction with a word clock or a digital input running at that frequency.*

Outputs	**Two**
Type	Electronically balanced (pin 2 hot)
Maximum Level	+21dBu into >2k
Performance	
Frequency response (20Hz to 20kHz)	±0.3dB with all filters and EQ flat
Distortion (THD+N) @ +4dBu (20Hz to 20kHz)	<0.01%
Dynamic range (20Hz-20kHz unweighted)	115dB
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 20.8µs 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	100V to 240V a.c ±10%
Consumption	<60W
Terminations	
Audio inputs/outputs	3 pin XLR
Ethernet inputs/outputs	Ethercon
RS-232	8 pin Mini-DIN socket (front) 9 pin D-type (rear)
World Clock	BNC
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

STS Compatible

Show Command Component

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

Helix DN9848E - System Controller

The digital system controller 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 new Klark Teknik Helix DN9848E system controller, no compromise has been made in either the feature set or the audio performance.

The Helix DN9848E 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 preprogrammed 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 DN9848E System Controller features AES/EBU digital inputs as standard. Whilst the internal sample rate of the DN9848E unit remains at 48kHz, these digital connections are all 96 kHz compatible allowing easy interface with any other digital device featuring the higher sample rate. The unit now features a dual port Ethernet communications interface. This is to facilitate much faster communication, response and metering when controlling multiple units, than was previously possible with serial comms.

The DN9848E incorporates some customer-requested operational additions. The user-configurability of the unit includes full matrix mixing capability between inputs and outputs, providing an unequalled level of flexibility. Whilst programming, inputs or outputs can now be 'ganged' so that the user can enter program data into one input or output menu and all connected inputs or outputs will be simultaneously updated. Input and output parameters can also be copied from one to another. The internal memory structure has also been revised such that it is now possible to back up the RAM-based system memories into non-volatile flash memory.

The proprietary ELGAR software coupled with Helix System Controller Remote Control Software (RCS) Add-In allows simple up-and-down-loading of system parameters into the FLASH memory locations, as well as storage and transmission of system information.

Should for example you need a new system configuration to be loaded into a unit on the other side of the world? No problem, simply email the ELGAR file to wherever it needs to go, it can then be uploaded into the unit in seconds.

Architect's & Engineer's Specification

The Loudspeaker Processor shall provide four analogue and four digital input channels and eight output channels with fully featured matrix mixing in a standard 1U 19" rack mount chassis.

Each input channel shall include: input name, input gain control, input source select (analogue or digital), delay up to one second, eight parametric EQ stages (+6 dB boost, -18 dB cut) and a compressor.

Digital inputs shall run at any sample rate up to 96kHz with internal sample rate conversion. The sample rate converter can be bypassed when the incoming digital signal has a sample rate of 48kHz.

Each output channel shall include: output name; configurable routing; delay up to 300 milliseconds; two cascaded all-pass phase correction filters, 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; and 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 analogue inputs and ou tputs shall be electronically balanced and use XLR connectors. All digital inputs shall be 110Ω AES/EBU 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.

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 Loudspeaker Processor shall be provided with an RS-232 and Ethernet ports for remote control and software updates. The Loudspeaker Processor shall be controllable from the ELGAR remote control PC software

The unit shall be capable of operating from a 100V to 240V, ±10%.

The Loudspeaker Processor shall be the Klark Teknik model DN9848E and no alternative option is available.

Technical Specification

Digital Inputs	**Two 2-Channel Inputs**
Type	AES / EBU
Impedance	110Ω
Sample Rate	96kHz*, 88.2kHz*, 48kHz, 44.1kHz*
Word Length	24-bit or 16-bit
**SRC Input at these sample rates*	
Analogue Inputs	**Four**
Type	Electronically balanced (Pin 2 Hot)
Impedance (Ω)	
Balanced	20k
Unbalanced	10k
Common Mode Rejection	>80dB @ 1kHz
Maximum level	+ 21dBu
Audio Outputs	**Eight**
Type	Electronically Balanced (Pin 2 Hot)
Minimum load impedance	56Ω/20nF
Source impedance	56Ω
Maximum level	+ 21dBu into > 2kΩ
Performance	
Frequency response	(20 Hz to 20 kHz) +/- 0.3dB with all filters and EQ flat
Distortion (THD+N)	<0.02% @ 1kHz, +8dBu
Dynamic range	(20Hz to 20kHz 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:	40µs 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 20.8µs steps
Output Processing (per channel)	
Routing	Full featured matrix mixing: any combination of inputs can be routed to any output in .1dB steps from 0dB to - 40dB and OFF.
Delay	0 to 300ms (102.68m or 333' 10" at 20(C)in 5.02µs 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/	frequency range 20Hz to 20kHz in 21steps per octave.
Low shelf filter	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/	frequency range 20Hz to 20kHz in 21steps per octave.
Hi shelf filter	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: 40µs 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
Ethernet inputs/outputs	Ethercon
RS-232	8 pin Mini-DIN socket
Power	3 pin IEC
Power Requirements	
Voltage / Consumption	100V to 240V a.c ±10% 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.

Show Command Component

KLARK TEKNIK
SIGNAL PROCESSING BY DEFINITION

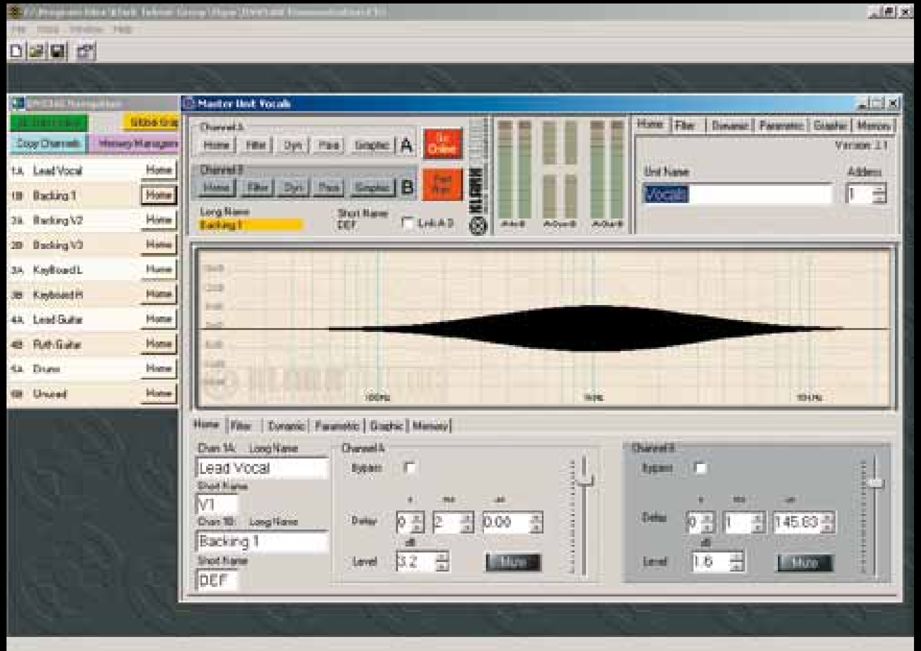
Midas and Klark Teknik ELGAR Framework

ELGAR is a software shell for a PC that allows Midas and Klark Teknik product control software, called Add-Ins, to operate.

ELGAR allows data from individual units, for example a number of Helix units and a Midas Heritage 1000, via the relevant Add-In to be stored within one show file on your PC. You can therefore have your entire show with you on your laptop, allowing you to fine tune settings in your hotel room and then just upload it later at the venue. ELGAR will also ensure that the correct Add-In will only communicate with the correct piece of hardware – in other words it will make certain that a Helix Remote Control Add-In will only talk to the Helix unit and not the Heritage 1000.

Show Command Component

Helix DN9848E Remote Control Software operating under Elgar



Helix EQ Remote Control Software operating under Elgar

Midas Heritage 1000 Library Manager operating under Elgar

Helix EQ Remote Control Software Add-In

To further increase the functionality and control of the Helix EQ is the Add-In for the Midas and Klark Teknik ELGAR control shell – the Helix EQ RCS.

This allows remote PC access to all the functions of Helix EQ, including overall system store and recall. An extremely intuitive Graphical User Interface (GUI) allows simple navigation between function pages, the overall number of which has been kept to a minimum. The system is designed to work with all PCs including the latest handheld PC tablets for ease of wireless connection and portability. Real-time indication of unit online / offline status is visible on all pages and the individual function pages have familiar and easy to use controls whether using a mouse, keypad or stylus.

The make or break of this type of system is always navigation, and this is one of the Helix EQ RCS's real strengths, using our proprietary 'FastNav' page. This is a control panel that is always active, and shows every function of every channel. Thus it is possible to move between, for instance, the graphic EQ for channel 10 and the T-DEQ controls for channel 37 with a single click.

Available now as a free download from www.klarkteknik.com

Show Command Component

DN9848E Remote Control Software Add-In

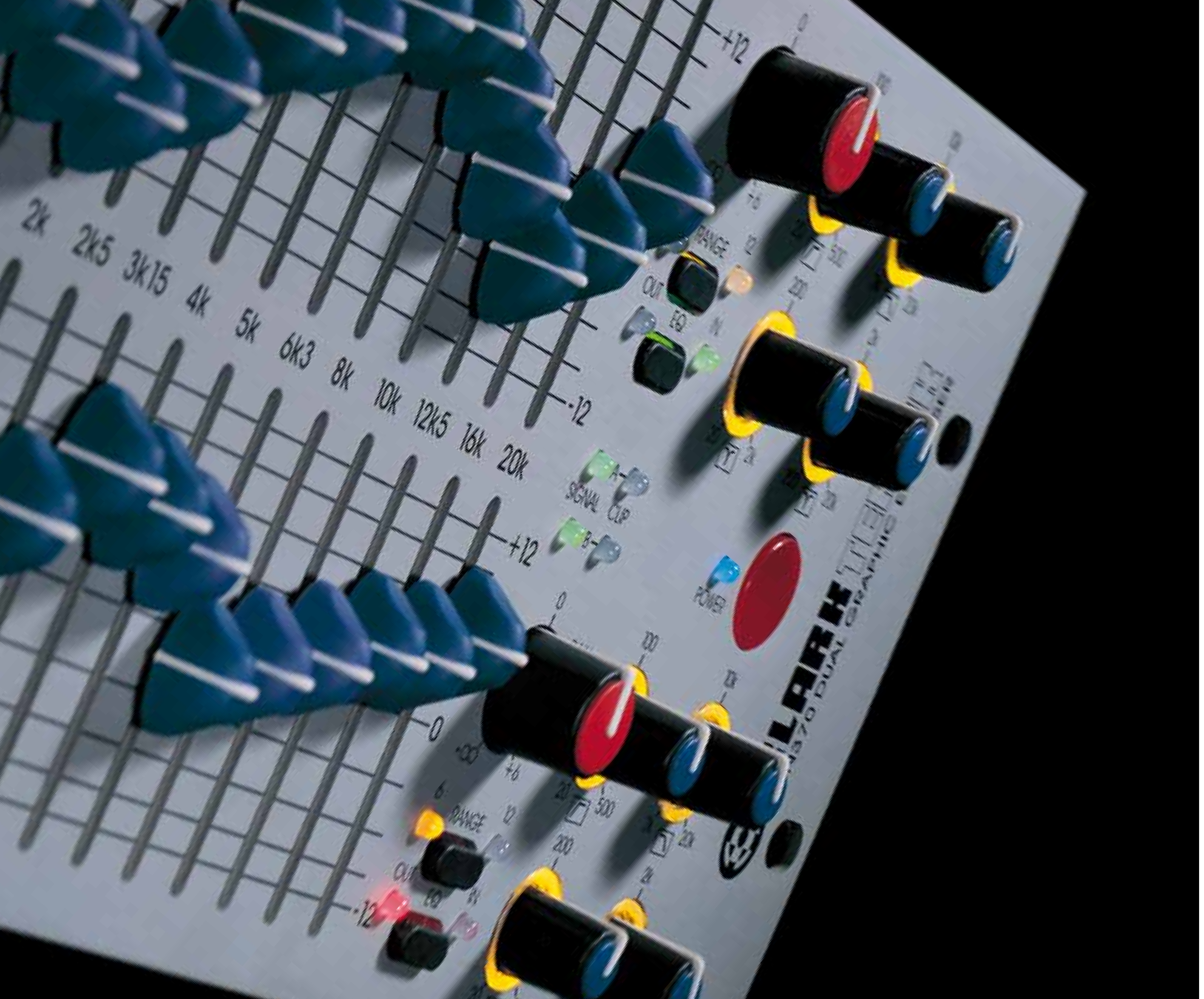
Helix DN9848E System Controller Remote Control Software (RCS) Add-In provides on-line remote control and off-line system configuration, either via wired or wireless Ethernet technology. The remote control software allows intuitive control of every function of an individual unit or units, it also (and 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 an ELGAR computer file.

Show Command Component

STS - Solo Tracking System

Helix offers the ability to link to all Midas consoles in the Heritage, Legend and Siena range via the Solo Tracking System (STS). This means that when you press any solo key on the console, the EQ for that input or output (outputs only for Siena) is instantly shown on the Helix DN9340E Dual EQ or a wired or wireless PC ready for immediate control. Once displayed on the your chosen user interface you naturally have complete access to all the Helix EQ functions allocated to that input or output. The graphic EQ portion of Helix will also be displayed on a DN9331 RAPIDE Graphic Controller if connected into the system.

A RS-232 connection is supplied on the rear panel of Helix DN9331 and DN9340E for this purpose, and up to 64 channels of Helix can be interconnected using standard CAT5 cables.

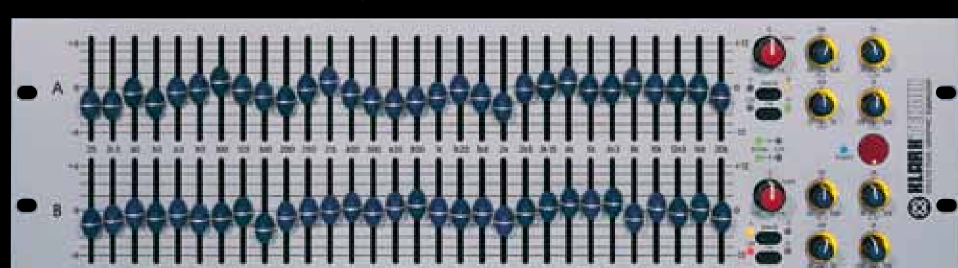


DN370 - Dual Channel 30 Band 1/3 Octave Graphic Equaliser

The Klark Teknik DN370 is the latest evolutionary step in a process of design refinement that goes back over 30 years. With DN370 we've started from the ground up and produced a unit that is totally without compromise, and one that we believe is the finest professional graphic equaliser in the world today. It also perfectly complements the existing Klark Teknik range of equalisers, both analogue and digital.

Our aim is simply to provide discerning professional users with the best possible solutions for system control. Our market research shows that the graphic equaliser is still the most commonly-used EQ device in fixed and mobile live sound applications, as well as many installations, mainly because the physical user interface provides instant access and controllability in even the most demanding environments. To this end we have completely re-evaluated the role of the graphic EQ, focusing exclusively on providing a new feature set that better reflects the needs of modern users.

Like all Klark Teknik units, DN370 is engineered for a lifetime of hard use and carries our 3-year international factory warranty.



Architect's & Engineer's Specification

The Equaliser shall provide ±12dB of boost and cut at 30 1/3 octave ISO centre frequencies from 25Hz-20kHz, selectable to ±6dB for increased fader resolution.

The equaliser shall meet or exceed the following performance specifications:

Frequency Response ± 0.5dB (20Hz-20kHz)
Distortion (THD+N) <0.003% @1kHz, +4dBu
Dynamic Range >114 dB (20Hz-20kHz unweighted, ±12 dB range)

The equaliser shall allow one adjustable second-order low pass filter and one adjustable second-order high pass filter per channel, and two adjustable overlapping notch filters per channel.

The unit shall have an 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. The slide potentiometers shall have protective covers to inhibit the ingress of dirt and dust.

All audio connections shall be electronically balanced and use XLR and Phoenix style connectors. Input and output transformers shall be available as an option.

The unit shall be capable of operating from a 100-240V ± 10% 50/60Hz a.c power source.

The equaliser shall be the Klark Teknik Dual Channel model DN370, and no alternative specification option is available.

Technical Specification

Inputs	**Two**
Type	Electronically balanced (pin 2 hot)
Impedance (Ω)	20k
Maximum input level	+22dBu
Outputs	**Two**
Type	Electronically balanced (pin 2 hot)
Minimum load impedance	600 Ω
Source impedance	<60 Ω
Maximum output level	+22dBu into >2kΩ
Performance	
Frequency response relative to signal at 1kHz	±0.5dBu 20Hz-20kHz
EQ out	±0.5dBu
EQ in (flat)	±0.5dBu
Distortion (THD+N)	< 0.003% @ 1kHz, +4 dBu
Dynamic range	>114dB (20Hz-20kHz unweighted, ±12dB range)
Overload indicator	+20 dBu
Gain	- ∞ to +6dBu
Equalisation	30 Bands
Centre Frequencies	25Hz-20kHz, 1/3 octave
To BS EN ISO 266:1997	Tolerance ±5%
Maximum Boost/Cut	±12dB, ±6dB
High Pass Filter Slope	12 dB/octave
Low Pass Filter Slope	12 dB/octave
Notch filter attenuation	>17dB, Q=32
Terminations	
Audio	3-pin XLR and 6-pin Phoenix
Power	3-pin IEC
Power Requirements	
Voltage	100-240V a.c ±10%
Consumption	<60W
Dimensions	
Height	133mm (5.25 inch - 3U)
Width	483mm (19 inch)
Depth	205mm (8 inch)
Weight	
Nett	5.8kg
Shipping	7.0kg

Options
Input and output balancing transformers

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

DN360 - Dual Channel 30 Band 1/3 Octave Graphic Equaliser

Now approaching some 23 years in continuous production, the DN360 dual graphic EQ has achieved ubiquity in professional audio circles. With nearly 36,000 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.

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 600Ω +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 a.c 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 (Ω)	
Balanced	20k
Unbalanced	10k
Outputs	**Two**
Type	Unbalanced (pin 3 hot)
Min. load impedance	600Ω
Source impedance	<60Ω
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	
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	483mm (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
SIGNAL PROCESSING BY DEFINITION

DN1248 *Plus* - Microphone Splitter

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 3-year international factory warranty, and you have a unit that exceeds the expectations of even the most demanding users.

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 110 to 240V ±10%, 50 to 60Hz a.c 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	110V to 240V a.c ±10%, 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.

SIGNAL PROCESSING BY DEFINITION

DN530 - Creative Quad Gate

With the unrelenting absorption of signal processing into digital mixing consoles, outboard gear now has to contribute something extra special merely to justify its space in the rack. The new DN530 Creative Quad Gate does just that. It brings some exciting new tools to the audio party. The DN530 features "Transient Accenting", a creative feature providing an easy way to enhance the attack envelope beyond merely opening the gate. This can provide up to 12dB of additional transient energy and its primary application is to provide additional impact on drums and percussion instruments, although it can be used to enhance the impact of many instruments, including acoustic stringed instruments such as guitar and piano.

Transient Accenting is unique in that it allows the operator precise control over the amount of enhancement applied to each channel of processing.

The DN530 brings a creative flare to drum channels with its unique accent feature which largely eliminates the need for excessive EQ or compression.

DN530 Architect's & Engineer's Specification

The Noise Gate shall provide four (4) complete channels of creative transient-accentuating gating in a standard 1U 19" rack mount chassis. Each channel shall have an electronically-balanced input and output on 3-pin XLR connectors.

Each channel shall have a rotary Threshold control to set the signal level at which the Noise Gate opens. The channel metering shall have a green "THR" LED to indicate when the signal exceeds this threshold.

Each channel shall have rotary controls for setting the Attack, Hold and Release times of the Noise Gate. The channel metering shall have a yellow "HLD" LED to indicate when the Hold portion of the dynamic control envelope is active and an orange "REL" LED to indicate when the release portion of the dynamic control envelope is active. The Hold function shall also have Intelligent Threshold Shift (i-TS) to provide threshold hysteresis to avoid repeated opening and closing of the Noise Gate in response to low frequency signals close to the threshold level set by the user.

Each channel shall have an Accent rotary control to accentuate the initial transient that opens the Noise Gate by adding an overshoot characteristic with up to +12 dB of gain as the Noise Gate opens. The channel metering shall have a blue "ACC" LED to indicate when the Accent function is adding gain to the initial transient. The Accent shall also function independently of the noise gate when the range control is set to 0dB.

Each channel shall have a latching pushbutton switch with an associated orange LED labelled "DUCK" to reverses the operation of the gate so that it closes when signals exceed the threshold set by the user and opens when they go fall below the set threshold. When the Duck function is active, the Accent rotary control and "ACC" LED shall be disabled.

Each channel shall have a rotary Range control to adjust the amount of gain reduction applied to signals below the threshold level. Each channel shall also have a red "SHUT" LED to indicate when the Noise Gate is fully closed.

Each channel shall have a latching pushbutton switch with an associated red LED labelled "BYPASS" to remove the Noise Gate from the signal path.

Each channel shall have a Clip LED included in the channel metering to indicate when the input signal is exceeding the clipping point of the Noise Gate.

Each channel shall also have a sidechain control section with a bandpass filter controlled with a rotary control labelled Frequency to set the filter centre frequency. The sidechain control section shall have a latching pushbutton switch with an associated green LED labelled "FILTER" to put the bandpass filter in the sidechain signal path. The sidechain control section shall also have a latching pushbutton switch with an associated yellow LED labelled "EXT" to select the external sidechain ¼" TRS Jack input as the signal source for the sidechain instead of the signal that the Noise Gate is acting on.

The Noise Gate shall have a Solo Bus to allow monitoring of the sidechain control input signals for each channels. The Solo Bus shall have input and output XLR connectors on the rear panel to facilitate the serial connection of units and to connect to a line input on a mixing console for monitoring via its Solo Bus console headphone amp.

The sidechain control section of each channel shall have a latching pushbutton switch with an associated yellow LED labelled Solo to enable the post-filter sidechain signal onto the Noise Gate solo bus.

The Noise Gate shall meet or exceed the following specifications:

Distortion:	0.05% (1kHz 0dBu)
Dynamic range:	116dB (20Hz-20kHz unweighted)
Frequency:	±0.5dB (20Hz-20kHz, relative to 1kHz)
Noise floor:	-100dBu gate closed (20Hz-20kHz unweighted) -94dBu gate open (20Hz-20kHz unweighted)
Attack time:	30us-10ms
Hold/Release time:	2ms-2 secs
Max output level:	+22dBu into 600 ohms

The Noise Gate shall have an integral switch-mode power supply capable of operating from a 100 – 240 V a.c. ±10%, 50 to 60 Hz AC power source and have an IEC 60320 C14 mains inlet with integral fuse. A blue LED labelled "POWER" shall be included on the front panel to indicate when the unit is powered on.

The Noise Gate shall be the Klark Teknik Model DN530 and no alternative specification option is available.

Technical Specification

Inputs	**Four**
Type	Analogue, electronically balanced female XLRs (Pin 2 hot)
Impedance	10k ohms
Maximum input level	+22dBu
Common mode rejection	Typically, -80dB at 1kHz
Outputs	**Four**
Type	Analogue, electronically balanced male XLRs (Pin 2 hot)
Output impedance	<60 ohms
Maximum output level	+22dBu
Signal drive capability	<600 ohms
EXT SIDECHAIN inputs	**Four**
Type	Analogue, electronically balanced Jack sockets
Impedance	20k ohms
Maximum input level	+22dBu
Common mode rejection	Typically -60dB at 1kHz
SOLO BUS input	**One**
Type	Analogue, electronically balanced female XLR (pin 2 hot)
Input impedance	20k ohms
Maximum input level	+22dBu
Common mode rejection	Typically -60dB at 1kHz
SOLO BUS output	**One**
Type	Analogue, electronically balanced male XLR (pin 2 hot)
Output impedance	<60 ohms
Maximum output level	+22dBu
Signal drive capability	<600 ohms
Performance	
Max gate gain reduction	< -90dB
Max accent gain	+12dB
Max signal level any input or output	+22dBu
Noise at main output with unity gain	-94dBu
Distortion at 1KHz 0dBu with steady unity gain condition	< 0.05%
Signal delay	0 seconds
Frequency response	±0.5dBu (input to output), 20Hz to 20kHz
Dynamic range	>116dB (22Hz to 22kHz unweighted)
Gate	
Threshold	-50dB to +25dB
Range	-4 (minus infinity) to 0dB
Attack	30µs to 10ms
Release	2ms to 2s
Hold	2ms to 2s
Accent	0dB to +12dB
Filter	
Sidechain filter	40Hz to 16kHz
Terminations	
Audio	3-pin XLRs (male and female) and 1/4" TRS balanced jack sockets
Power	3-pin IEC
Power requirements	
Voltage	100VAC to 240VAC ±10%
Frequency	50Hz to 60Hz
Consumption	<25W

Dimensions	
Height	44.5 mm (1.75"), 1U high
Width	483 mm (19")
Depth	305 mm (12")
Weight	
Net	4.6 kg
Shipping	5.6 kg
Operation	
Temperature	+5°C to +40°C
Storage	
Temperature	-20°C to +60°C

Due to a policy of continual improvement, Klark Teknik reserves the right to alter the function or specification at any time without notice.

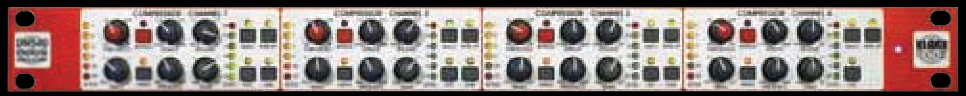
KLARK TEKNIK
SIGNAL PROCESSING BY DEFINITION

DN540 - Creative Quad Compressor

Compressors were originally devised to reduce the dynamic range of audio signals. To do this they use gain modification that makes adjustments to signal levels automatically, dependant on the level and dynamic of the signal itself, and upon the character and control settings of the particular compressor unit.

Throughout the history of dynamics processing many different types of compressor have been produced using many different types of gain technology. Each type has a distinctive sound. Users have found many applications which benefit from these often unintentional compression audio artifacts, taking the use of compressors far beyond simple dynamic range reduction, sometimes even generating the complete reverse; dynamic enhancement. It is with these creative applications in mind, that the DN540 has been designed to offer a latency-free, comprehensive and compact dynamics processing package.

With its unique presence feature the DN540 will be especially useful on vocals to eliminate the "pumping" effects of spill and to add "air" to the sound without introducing noise.



DN540 Architect's & Engineer's Specification

The Compressor shall provide four (4) complete channels of creative presence-accentuating compression in a standard 1U 19" rack mount chassis. Each channel shall have an electronically-balanced input and output on 3-pin XLR connectors.

Each channel shall have a rotary Threshold control to set the signal level at which the Compressor applies compression and a rotary Ratio control to set the amount of compression applied above the set threshold.

Each channel shall have rotary controls for setting the Attack and Release times of the Compressor. Each channel shall have a latching pushbutton switch with an associated green LED labelled "AUTO" to select rms-sensing Auto compression mode instead of the default peak-sensing Manual compression mode. In Auto compression mode the Attack and Release rotary controls shall be disabled.

Each channel shall have a rotary Presence control to dynamically accentuate a broad range of frequencies centred around 5 kHz. The purpose of this function shall be to allow high-mid frequency content often lost when compression is applied to be retained.

Each channel shall have a rotary Gain control to apply make-up gain to the Compressor.

Each channel shall have a latching pushbutton switch with an associated orange LED labelled "HARD" to change the compression characteristic from the default soft knee setting to a hard knee setting.

Each channel shall have a latching pushbutton switch with an associated red LED labelled "BYPASS" to remove the Compressor from the signal path.

Each channel shall have a six (6) segment attenuation meter to show the amount of gain reduction being applied by the Compressor.

Each channel shall have a six (6) segment signal level meter which by default shall show the compressor output signal level. A latching pushbutton switch shall be included on each channel with a green LED labelled "MTR I/P" to select input level metering instead.

Each channel shall have a latching pushbutton switch with an associated yellow LED labelled "EXT" to select the external sidechain ¼" TRS Jack input as the signal source for the Compressor sidechain instead of the signal that the Compressor is acting on.

The Compressor shall have a channel link facility whereby adjacent channels can have their sidechains summed with the left-handmost channel acting as the master control. Channels 1-3 shall have a latching pushbutton switch with an associated yellow LED labelled "LINK" to allow linking to the adjacent right-hand channel. The presence and gain controls shall remain independent when the channel linking facility is enabled.

The Compressor shall meet or exceed the following specifications:

Distortion:	0.05% (1kHz 0dBu)
Dynamic range:	116dB (20Hz-20kHz unweighted)
Frequency response:	±0.5dB (20Hz-20kHz relative to 1kHz)
Noise floor:	-94dBu (20Hz-20kHz unweighted)
Attack time:	0.1ms-20ms
Release time:	50ms-2 secs
Maximum output level:	+22dBu into 600 ohms

The Compressor shall have an integral switch-mode power supply capable of operating from a 100 – 240 V a.c. ±10%, 50 to 60 Hz AC power source and have an IEC 60320 C14 mains inlet with integral fuse. A blue LED labelled "POWER" shall be included on the front panel to indicate when the unit is powered on.

The Compressor shall be the Klark Teknik Model DN540 and no alternative specification option is available.

Technical Specification

Inputs	**Four**
Type	Analogue, electronically balanced female XLRs (Pin 2 hot)
Impedance	10k Ohms
Maximum input level	+22dBu
Common mode rejection	Typically, -80dB at 1kHz
Outputs	**Four**
Type	Analogue, electronically balanced male XLRs (Pin 2 hot)
Signal drive capability	<600 ohms
Output impedance	<60 ohms
Maximum output level	+22dBu
EXT inputs	**Four**
Type	Analogue, electronically balanced Jack sockets
Impedance	20k ohms
Maximum input level	+22dBu
Common mode rejection	Typically -60dB at 1kHz
Performance	
Maximum signal level any input or output	+22dBu
Frequency response	±0.5dBu (input to output), 20Hz to 20kHz
Dynamic range	>116dB (22Hz to 22kHz unweighted)
Noise at main output with unity gain	-94dBu
Distortion at 1kHz 0dBu with steady unity gain condition	<0.05%
Signal delay	0 seconds
Compressor	
Threshold scale	-50dB to +25dB
Ratio scale	Minus infinity (-4):1 to 1:1
Attack scale	0.1s to 20ms
Release scale	50ms to 2s
Presence scale	Minimum (flat) to maximum (-3dB at 5kHz)
Gain scale	0dB to 18dB
Terminations	
Audio	3-pin XLRs (male and female) and 1/4" TRS balanced Jack sockets
Power	3-pin IEC
Power Requirements	
Voltage	100VAC to 240VAC ±10%
Frequency	50Hz to 60Hz
Consumption	<25W
Dimensions	
Height	44.5 mm (1.75"), 1U high
Width	483 mm (19")
Depth	305 mm (12")
Weight	
Net	4.6 kg
Shipping	5.6 kg
Operation	
Temperature	+5°C to +40°C
Storage	
Temperature	-20°C to +60°C

Due to a policy of continual improvement, Klark Teknik reserves the right to alter the function or specification at any time without notice.

KLARK TEKNIK
SIGNAL PROCESSING BY DEFINITION

DN100 -
Active Di

The 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.

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 Ω load.

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

The output shall be transformer balanced and isolated, with a source impedance of 150 Ωs, 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 Ω 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

Inputs	**Four**
Type	active electronic, balanced or unbalanced
Impedance	1M Ωs nominal, balanced or unbalanced (jack connectors) 20K Ωs (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 Ωs
Connector	3 pin XLR
Max. Level	10dBu with load >2k Ωs
Min. load	600 Ωs
Performance	
Noise	-100dBu, 20Hz to 20kHz unweighted, with input terminated by 10k resistor
Frequency response	+0.5/-1dB 20Hz to 20kHz
Distortion (THD+N) output	<0.01% @ 1kHz, +4dBu
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

The Helix DN9848E displays its filter steepness as “bandwidth” in octaves – what are the corresponding values expressed as “Q”?

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

What is AES/EBU?

AES/EBU is the term used for a professional digital audio transmission system, jointly specified by the Audio Engineering Society (AES) and the European Broadcast Union (EBU), and published by the former as their AES3 standard (at the time of writing, the current version is AES-2003). It allows the transmission of two channels down a shielded twisted-pair cable using time division multiplexing (TDM) with one sample from each channel being transmitted within the sample period of the system.

Typically, XLR connections are used for AES interfaces, and because of the TDM format, one XLR cable carrying AES/EBU data can replace two regular analogue connections. The clock for the data transmission is embedded in the data, using a process known as bi-phase mark encoding or Manchester encoding. This means that the incoming data can be used a the clock source for the master clock within a unit equipped with an AES/EBU interface, and this is the preferred mode of operation, as it guarantees that the unit is synchronised to the incoming data stream.

For large digital transmission systems using AES/EBU interfaces, such as those encountered in broadcast and studio installations, a distributed clock signal operating at the sampling frequency is distributed to all units, separate to the AES/EBU signals. This is generally known as ‘word clock’ and allows all connected units to be synchronised on a sample-accurate basis. Word clock is most commonly connected using 75 ohm, BNC connectors.

As the DN9340E is primarily designed for live applications, it will always take the clock reference from the incoming data stream when an external clock reference is selected. However, the provision of a word clock input on the AES/EBU interface allows the unit to be used as analogue to digital converter, synchronised to a system word clock if one is available.

Why do I need to set the destination wordlength for my digital outputs?

The correct setting of the output wordlength is necessary to avoid distortion caused by truncation of the audio data. If a 24-bit audio signal is transmitted to a 16-bit device (such as a DAT recorder), the lower 8 bits will simply be ignored or “truncated”. This results in an error with an amplitude (on average) of half the size of the least significant bit (LSB) of the 16-bit signal. Because the size of the error for each individual sample will depend on the actual data in the input (24-bit) waveform, the error will be related to the input signal, and will therefore appear as harmonic distortion. This distortion, once created, can never be corrected by subsequent processing.

In order to avoid this situation, we need to add a random noise signal with an amplitude of “half an LSB” (called “dither”) to the 24-bit waveform BEFORE we truncate it to 16 bits. This has the effect of randomising the error so that it no longer relates to the input signal (although the error still has the same total energy). Our 16-bit signal now has random noise at the _ LSB level, instead of harmonic distortion at the _ LSB level - which is very much better to listen to…

For the mathematically inclined, this is rather like rounding numbers. As an example, 7.9 and 7.1 will both truncate to 7 exactly, but we know that this is not the “minimum error” answer. If, however, we add 0.5 (equivalent to _ LSB) before we truncate, we get 7.9+0.5=8.4 ~ 8 and 7.1+0.5=7.6 ~ 7 which is the answer we expect. Note that if you do this “wrong” at the start, and get 7 in both cases, it doesn’t help to add the 0.5 afterwards! This is also true for the audio - once you have caused distortion by truncating, you cannot remove it by adding noise.

So, in practice, it is always safest to set the wordlength to 16-bit. This will ensure that any 16-bit (or better) device will connect up OK and will receive a correctly-dithered, low distortion signal. Only if you are absolutely sure that the destination device actually makes use of the additional bits should you select 20-bit or 24-bit operation to achieve the maximum dynamic range available from the unit.

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

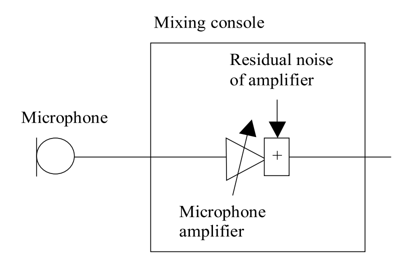


Figure 1 shows a microphone connected directly to a console. In this simple case we bring the microphone signal into the console, and immediately amplify it in the first active stage - the microphone amplifier. To take a practical example, a common dynamic vocal microphone subjected to an SPL of 110dB 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

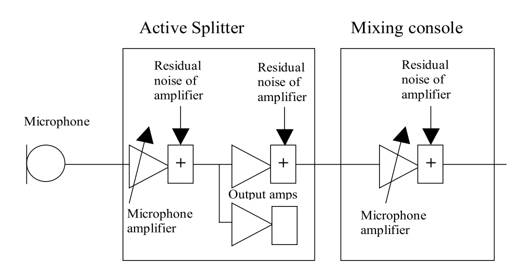


Figure 2 shows the same signal connected using an active splitter system. The splitter contains a variable-gain microphone amplifier, which then feeds a number of independent output amplifiers. One of these is then connected to the console input, which itself has a variable-gain microphone amplifier. 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.

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…

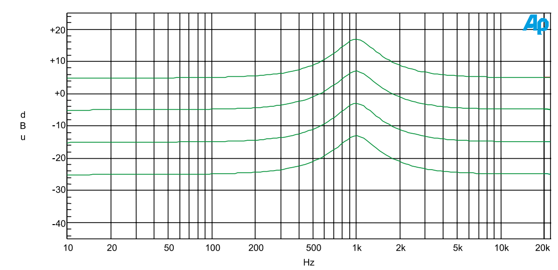
What is Dynamic Equalisation (T-DEQ)?

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

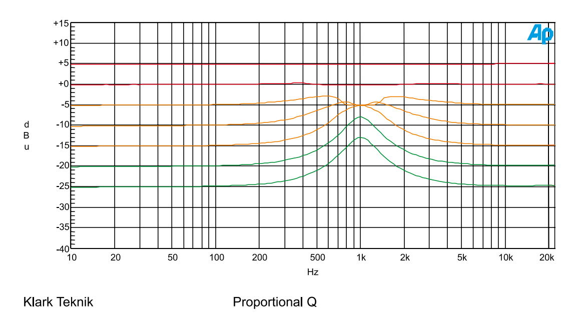


This shows a series of responses for the parametric EQ with different input levels. As expected, there is no change in the shape of the curve with different input levels. If the input is 10dB louder, the output is 10dB louder at every frequency.

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

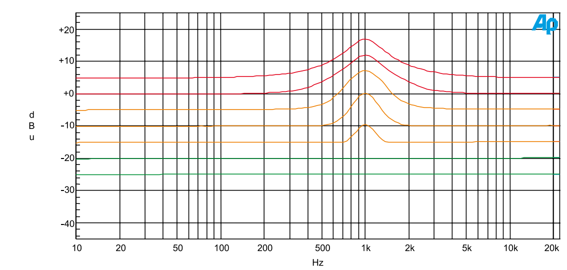


In this example, [low threshold] is -20dBu, [low level] is +12dB, [high threshold] is set to -5dBu, and [high level] is 0dB. Thus the lowest trace shows an input at -25dBu with a standard parametric boost of +12dB at 1kHz. The -20dBu trace shows an identical response, as expected. However, once above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

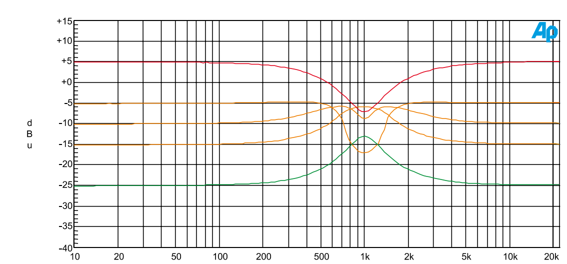
Figure 3 – Helix with boost at high signal level



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.

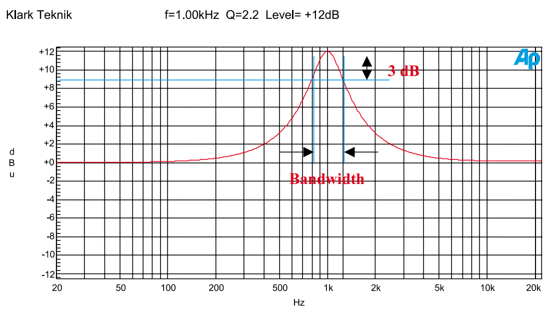
Helix with boost at low level and cut at high level



FAQ (frequently asked questions)

What is the difference between the various Q types on the Helix DN9340E Equaliser?

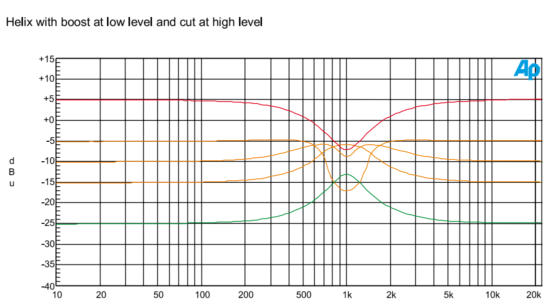
The "Q" of an audio equaliser describes the steepness of the filter - the degree to which it will affect signals either side of its nominal or "centre" frequency. In general, the Q of a peaking filter is defined mathematically as , centre frequency / bandwidth where the bandwidth (in Hz) is the range 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.

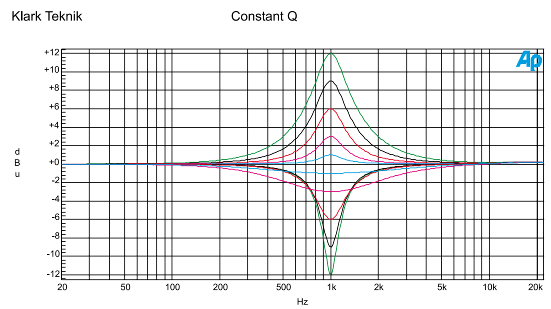
So far so simple - but why the different types? This is due to the way in which the Q of the filter varies (or not) when the gain control is adjusted. There are three modes available in the Helix system, which we term Proportional, Constant, and Symmetrical Q.

Proportional Q



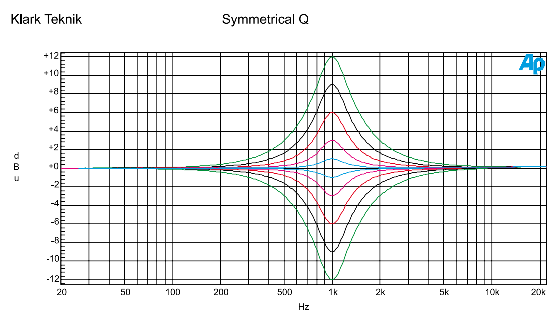
Proportional Q is the mode of operation familiar to users of the Klark Teknik analogue graphic equalisers such as the DN360. As the amount of cut or boost is increased, the Q also increases. This has the effect of making the equaliser "focus" more tightly as the amount of EQ is increased. This allows 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.

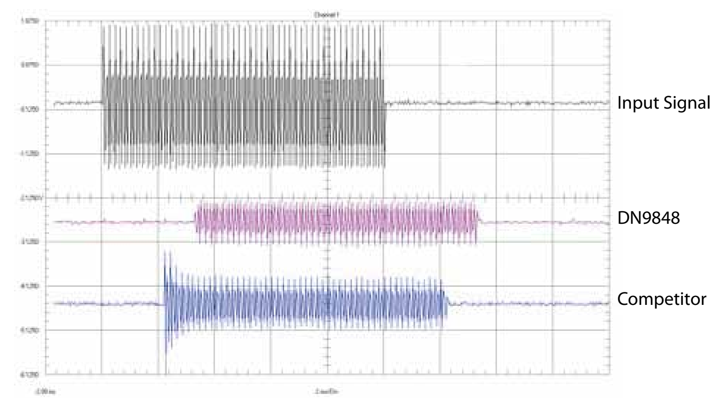
White Paper - paper one

The Use Of Look-Ahead Limiters In Loudspeaker Driver Protection

The limiter in a loudspeaker processor is the last line of defence in protecting the speaker drivers from damage, and as such it has a very specific and critical job to do. One of the chief modes of loudspeaker failure is driver over-excursion, and unless the limiter is designed to act instantly in response to sudden increases in level, it will allow through brief transients that can cause damage through over-excursion. All dynamics processors take a finite amount of time to respond to a change in input level, and unless additional steps are taken the result is that the input signal is initially let through at anything up to its full level, until the gain element in the limiter can act in response to the increase in signal level.

The DN9848 exploits the fact that digital signal processing works on a sample-by-sample basis (the signal data samples are clocked through the unit at the sample rate of 48 kHz) and that there is a small delay through each processing block, and literally 'looks-ahead' further back in the signal chain to sample the data for the limiter side chain, so that the limiter can apply the required gain reduction on an instantaneous sample-by-sample basis, so that the limiter never lets through any dangerous transients.

In the example below a 10 kHz tone burst of 10 ms duration has been used as the input signal and the output of both a DN9848 and a leading competitor are shown. Note the very large transient of the competitor unit which does not have a look-ahead limiter.



Signal source: Audio Precision System One

Settings: Waveform: Burst – Normal. Frequency: 10.0 kHz. Burst: 10 ms. Interval: 100 ms. High Level: +10.0 dBu Low level: -40.0 dBu

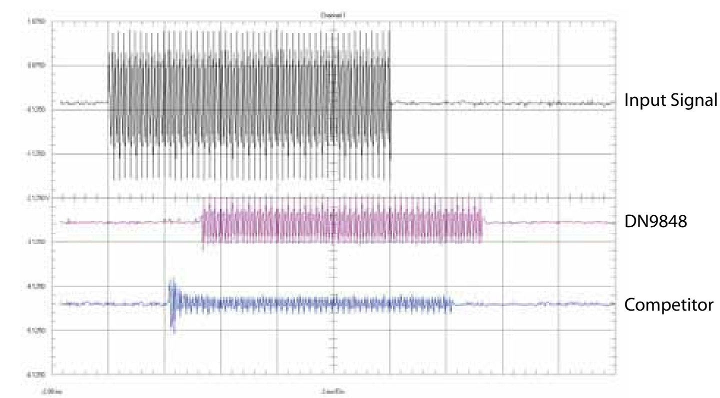
DN9848 settings: HPF: 1.25kHz Lnk-Ril 24dB/Oct. LPF: 20kHz Lnk-Ril 24dB/Oct. Limiter Threshold: 0.0dBu Release: 50ms Response: Hard Knee

Competitor settings: HPF: 1.26kHz Lnk-Ril 24dB/Oct. LPF: 22kHz Lnk-Ril 24dB/Oct. Limiter Threshold: 0.0dBu Attack & Release: Automatic

All other settings are default on both units.

As with all units that use sigma-delta ADC and DAC converters, there is a propagation delay from input to output, 3.2 ms for the DN9848 and 2.1 ms for the competitor unit, the additional delay in DN9848 is caused by the use of sigma-delta converters for both analogue-to-digital and digital-to analogue conversion (the competitor unit uses a different method of digital-to-analogue conversion), which allows the DN9848 to achieve its superior dynamic range.

In order to prevent driver failure, the competitor product's limiter threshold needs to be reduced such that the peak of the transient is at the same level as the threshold of the DN9848's look-ahead limiter, with a major effect on efficiency of speaker systems, as the effect of reducing the limiter threshold is to limit the amount of continuous output power available, which means more amplifiers and more speaker cabinets to achieve the same SPL. In the example below the competing unit's limiter threshold has been reduced so that level of the transient peak matches the threshold of the DN9848's limiter:-



The limiter threshold of the competitor's unit has been lowered to -10.0 dBu to avoid the risk of driver damage from the initial transient, at the cost of greatly reducing the efficiency of the PA system. The look-ahead capability of the DN9848's limiters allows the thresholds to be set at the levels required to protect the loudspeaker drivers, without the need to be concerned about transients being passed by the limiters. This allows the PA system performance to be maximised by safely exploiting the full operational range of the loudspeaker drivers.

KLARK TEKNIK
SIGNAL PROCESSING BY DEFINITION

White Paper - paper two

Phase adjustment on the Klark Teknik DN9848 Loudspeaker Processor

To meet the demands of a wide range of situations, the Klark Teknik DN9848 provides two all-pass filters with complementary control parameters for fine-tuning the phase response on each output. Although some crossover filter designs, e.g. Linkwitz-Riley types, are inherently phase-aligned at crossover, others such as Butterworth or Bessel responses may require manual phase alignment. Even Linkwitz-Riley filters may not produce accurate phase coherence when HPF and LPF are combined to produce a band-pass output. In addition, the phase response of the drive units and cabinets (especially horn-loaded types) may require compensation to achieve correct acoustic phase, even if the electrical phase is correct. The DN9848 filters provide straightforward tuning control in all cases.

The first filter of the DN9848 is presented as a "phase shifter" for which you can specify a particular phase shift at a reference frequency, namely a HPF or LPF (typically the cross-over point) or one of the 6 PEQs. The plots in Fig. 1 show the effect of these controls on the filter response when set to a 90° phase shift at references points equivalent to 20 Hz, 300 Hz, 1 kHz and 20 kHz. Referring to the figure, the overall response always remains the same shape i.e. tending from +180 at low frequencies to 0° at high frequencies, but is shifted along the frequency axis to achieve the required phase shift at the specified reference point.

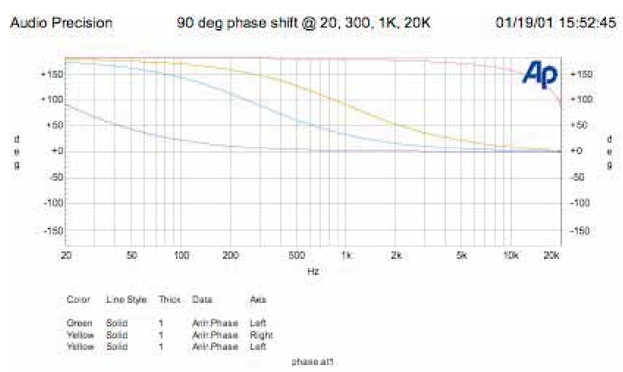


Figure 1.
Response of the Phase Shifter filter for a phase shift of 90° at reference points equivalent to 20 Hz, 300 Hz, 1 kHz and 20 kHz

The second filter, presented to the user as an "all-pass filter", enables the user to set the Order and Q of the phase shift, at a particular Frequency. The Order can be switched to Off (no filter), 1st order (90° shift) or 2nd Order (180 ° shift).

Fig. 2 shows the response of the filter for a 1st order shift. In this mode, the Q control is disabled. As can be seen, the filter behaves in an identical manner to the phase shifter i.e. the response tends from +180 at low frequencies to 0° at high frequencies, shifted along the frequency axis according to the chosen frequency. In effect, this is a phase shifter for which the frequency is entered directly, rather than being referred to a HPF/LPF etc.

Figure 2:
Response of the All Pass Filter for 1st Order phase shift at frequencies of 300Hz, 1kHz and 10kHz

Fig. 3 shows the response of the filter for a 2nd order shift, with the Q control set to 1, at frequencies of 300Hz, 1kHz, and 10kHz, and also Q set to 6 (max) and 0.4 (min) at 1kHz. As can be seen, the filter response now tends from 360° at low frequencies to 0° at high frequencies, and Q controls the rate at which the phase changes (i.e. the slope) around the transition point. With low Q, the phase changes gradually across the whole frequency range. With high Q, the phase changes rapidly in the transition area, and is unchanging at 360°/0° over the remainder of the frequency spectrum. Hence, the 2nd order all-pass provides the user the additional control of shaping the phase shift 'window'.

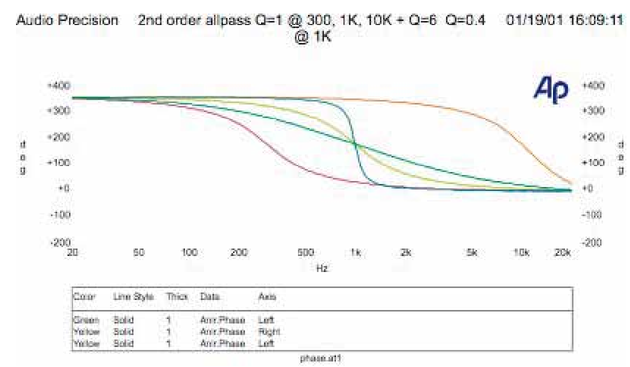


Figure 3:
Response of the All Pass Filter for 2nd Order phase shift with Q=1 at frequencies of 300Hz, 1kHz and 10kHz and also with Q=6 at (max) and Q=0.4 (min) at 1kHz

Note: Both filters are all-pass with a flat amplitude response between 20Hz and 20kHz; only the phase response changes with frequency.

Transient Accenting White Paper

Gate Uses and Abuses

Gates were originally devised to reduce noise in the silent passages of music programme especially during the process of multi track recording on analogue tape.

They still get used to reduce noise but they have found many other uses and creative applications over the years

Applications include:-

- Noise reduction
- Removing compressor breathing noise
- Reducing spill from adjacent sound sources
- Gating ambience reverb for effects
- Gating one instrument to sync it with another
- Ducking one source to make room for another in a mix
- Reduction of ringing on drums
- Increasing the definition and punch of drums

Very often the process of gating produces unwanted side effects especially if the gate is not set up well or does not operate sympathetically with a particular type of source material.

These can include:-

- Chatter (when the gate is indecisive and keeps opening and closing on sustained notes)
- Clicking (when the attack is set too fast or when it's envelope shape is unsuitable and generates high order harmonics each time it opens)
- False triggering (when the gate opens on microphone spill as well as its intended source).
- Noise enhancement (when noise is removed totally, the transition from gate shut to gate open and back modulates the background noise so much that it draws the listeners attention to the noise when it is present)

Providing controls for threshold, attack, release, hold and ratio, plus good metering, will help operators set the gate up for differing signal sources and minimise most of the negative effects above. However, this may involve a degree of compromise between the desired gating effects and the unwanted side effects. The DN530 will fair better than most because design choices made during its development were based on listening tests carefully tailored to match real world applications.

Chatter

Chatter can normally be eliminated by increasing the hold time but this may allow the gate to stay open longer than is desirable. Adding hysteresis to the threshold control helps enormously and allows hold times to be reduced without signal chatter. The DN530 has 4dB of hysteresis built in which is enough to eliminate chatter on all normal instrument types.

Clicking

The key to silent gating is the shape of the gain transition curve that is used to fade up the signal level when the gate opens (attack) and fade it back down when the gate closes (release). Many gates use linear transitions which, applied to low frequency signals, generate high order harmonics that sound like extra clicks (in time with the music source). The ideal shape is log (like an audio fader) so that the initial transition from shut to mostly open is fast and the final adjustment to fully open is progressively slower and slower. The exact reverse applies to the gate closing; this needs to start slowly and then speed up to close the gate fully. With these shapes no harmonics are produced during an attack, only a fundamental frequency (quarter cycle) that can be controlled by the attack time.

The tonality of the gate opening transition can be adjusted using the attack control to be slightly higher in pitch than the LF content of the sound it is processing to accentuate the start of each note; or set to be the same, in which case the transition will not be heard at all. If the attack is made slower still the start of each note will be softened which may be useful as an effect. Release times are typically much slower so audio frequency clicks are rarely heard but the log shape is still the best because it makes the fade out much less noticeable. The attack and release characteristics in the DN530 are log and fully adjustable to ensure effective gating that can be tonally transparent or used to add (or reduce) punch and definition.

False Triggering

Often microphones pick up as much spill from other instruments as they do sound from the intended source. This will cause the gate to open at times when it should be shut. Traditionally hi pass and low pass filters have been employed on gates side chains in an attempt to limit the frequencies spectrum that will trigger the gate to open. This type of filter seldom works well in this application because they are not easily manufactured with steep enough transitions from pass band to stop band. Also set up is difficult because you typically need to adjust them together to form a band pass filter.

A much better solution is to use a band pass filter in the first place set up with a high enough Q to make it very selective. Most instruments (especially drums) have a resonant frequency and false gate triggering can be massively improved by tuning a single band pass filter as described to find this resonance. False triggering is eliminated because the frequency spectrum and resonance from the spill does not produce enough energy at the tuned frequency to open the gate; only the intended source will.

The DN530 side chain filters are a high Q band pass types as described above. Set up is made by a simple single control (per channel) and this is made even easier because you can listen to the filter output (without interrupting the source material) on a separate solo bus.

Noise Enhancement

We all have the ability to block out constant back ground noises and often using a gate to eliminating background noises altogether defeats this mechanism; drawing our attention to the noise when the gate opens. Typically adjusting the gate range down to 10 or 15dB produces much more natural sounding results.

Transient Accenting

Many instruments have a percussive start to notes that are played. These transients can be enhanced or reduced by careful adjustment of the attack time as described above. Additionally the DN530 has the ability to accentuate this transition even more through application of its Transient Accenting capabilities described below:-

Every time the gate opens an accent is applied. This is a controlled boost in the signal level that lasts approximately 30ms. The amount of boost applied is determined (in dB) by the accent control and speed at which it is applied is determined (in ms) by the attack control. The boost is gradually removed during the 30ms accent period returning the signal level to normal.

If the gate is being used creatively the effect of the gate opening transition can be accentuated, which is particularly useful on drums improving definition and punch.

The accent effect is totally independent of the range control so it is possible to reduce the range very low or even off (0dB) and still achieve noticeable enhancement of transients.

Some benefits from Transient Accenting are listed below:-

- Increased tonal shaping effectiveness – gates can be used to modify the leading edge of percussive sounds to add definition and punch. This effect can be massively increased by accenting if required.

- Reduction of gated breathing – the noise enhancement issues associated gating background noise can be reduced or totally eliminated while retaining the tonal shaping and punch that a gate can add to percussive sounds.

- Reduction of delay – when gates are used to re shape the transient start of a sound they remove a small amount of sound at the start. This effectively causes a slight delay. Digital gates can capture this sound by using look ahead techniques but they are not time machines; the way they work is to delay the signal until the gate has opened. This eliminates any losses from the transient but delays the signal even further. Using Transient Accenting with the gate range at minimum (0dB) ensures absolutely no sound is lost and no delay is introduced; however controlled accentuation of the transient is still possible.

- Reduction of resonant howl round – with high amplification levels it is possible for a drum to resonate in sympathy with the amplified version of its self similar to microphonic howl round. It is possible for this sound to be so loud that once open a gate will not shut again. If the sound levels are reduced and the drum transients are accentuated the possibility of howl round is substantially reduced. The Transient Accenting only lasts for 30ms so any risk of resonant build up is restricted to this time period only.

The Transient Accenting feature contained in the DN530 adds another tool for the sound engineer that can be used to enhance and compliment the application of gating to many types of sound source.

KLARK TEKNIK
SIGNAL PROCESSING BY DEFINITION

Compressor Uses and Abuses

Compressors were originally devised to reduce the dynamic range of audio signals. To do this they use a gain cell that makes adjustments to signal levels automatically dependant on the level and dynamic of the signal itself, and depending on the character and control settings of the particular compressor unit.

Over the years many different brands of compressors have been produced using many different types of gain cell technology; each with a distinctive sound. Users have found applications that suite many of these taking the use of compressors far beyond simple dynamic range reduction, sometimes even generating the complete reverse; dynamic enhancement.

Applications include:-

- Protecting systems from amplitude overload
- Improving the power of signals when an artist performance is inconsistent
- Maintaining a stable signal level to help an instrument sit better within a mix
- Changing the percussive timbre of an instrument to produce more punch
- Compressing a mix so that it maintains a stable signal level (typically trying to make it sound louder)
- Changing the percussive timbre of a mix to make it sound more whole and add punch
- Frequency conscious compression to remove unwanted sibilance or popping

Very often the process of compression produces unwanted side effects especially if the compressor chosen does not match the application it is being put to very well.

These can include:-

- Breathing (when high frequency environmental or processing noise levels jump up and down sounding similar to someone breathing in the background)
- Pumping (when level modulation can be heard on an instrument making it sound unnatural)
- Amplitude flutter (a faster oscillating modulation that makes sounds "gritty")
- Source inter-modulation (when one sound source within a mix causes amplitude changes to another; for example, the natural decay of cymbals can be modulated by the crack of a snare drum in a drum mix)
- Transient enhancement (an unwanted accentuation clicks at the start of a sound)
- Dull sound (when all the natural attack and brilliance is stripped away from the sound source)

Providing controls for threshold, attack, release, ratio, and providing responsive metering can go along way toward helping with suitable parameter set up for any signal source such that most of the negative effects above can be reduced or eliminated. However, this is a complex and critical task which often involves a degree of compromise between the desired compression effects and the unwanted side effects, which can be time consuming.

Simple Compression

For many every day compression jobs an RMS compressor with automatic time constant set up (attack and release) is the best solution providing quick and easy set up. The RMS averaging process slows the time constants on relatively steady state signals reducing distortion and pumping; and when large signal changes occur they automatically speed up capturing and constraining the bulk of any large sound level variations.

Very often a Ratio control and Threshold control combined with the automatic time constant adjustment just described is all you need to set up a good sounding compression. The DN540 auto compressor operates in exactly this way providing simple fast set up on straight forward compression jobs.

Advanced Compression

RMS compressors are not fast enough to capture everything on transient material because the RMS averaging process always adds some delay; thus they are not suitable for ultimate protection against system overloads etc. Also their creative use to tailor percussive instruments is very threshold dependant and often results in attacks that are either too fast or too slow for the desired effect (unless the sound source is extremely regular, which is not typical with most musicians).

For more difficult compression duties a compressor with fully adjustable attack and release is a better choice. With this style of compressor there is no averaging process delay (RMS detector) so the action of attack or release can start the instant there is a change in signal amplitude that requires it. The user must define the rate of response and can adjust this precisely to match the sonic effect required on the source material. Normally this also results in high distortion on constant signal levels because the compressor attacks and releases on every cycle but advanced compressors utilise windowing methods whereby the time constants set are greatly increased on steady state material. Another technique used is to automatically add a little hold before any release commences.

The above techniques are integral to the semi-linear attack and second order release characteristics of the DN540 "normal mode" compressor (i.e. when auto is off).

Linear attack provides a constant rate of attack (in dB against time) such that large changes in program signal level take a little longer to compress than smaller ones. However, on material with more constant signal levels the attack rate of the DN540 "normal mode" compressor automatically reduces. This appears as a curvature in the linear attack rate characteristic as it nears completion, hence the term semi-linear.

This makes the compression very transparent providing some dynamic control but without unduly effecting the intentional dynamic content of the source material.

It can be used on difficult instruments like acoustic guitar with slower attack time settings and relatively fast release to keep equal perceived loudness within a mix without producing excessive amplitude flutter or distortion.

It can also be used with faster attack times to capture dynamic instruments like electric bass guitar without adding excessive distortion on constantly compressed passages.

Adding "soft knee" noticeably delays the onset of attacks, which can be particularly useful on drums where compression can be applied to emphasise transients giving more punch while retaining a good deal of artistic dynamic from the drummer.

Thus when suitably adjusted the normal compression mode is suitable for any task from capturing fast transients in order to provide system protection; producing subtle compression of dynamic range without changing timbre or removing intentional accents made by the artist; to deliberate thickening of transient sounds.

Soft Knee Compression

Many vintage compressors exhibit a soft knee transition between linear gain transfer, on signals below threshold, to compression on signals that are above threshold. This normally occurs because of non linear errors in the gain cell. Modern compressors normally have much better gain cell linearity so any soft knee has to be generated in a separate process that operates outside the gain cell.

The DN540 soft knee generation is set up so that it does effects the gain cell like vintage compressors. This means it bends the compression ratio at the onset of compression as you would expect but it also bends the attack and release character. This provides very natural sounding compression as found on many vintage compressors and generates more full bodied punch and definition when attach times are deliberately set very slow.

Dynamic Enhancement

Many instruments have a percussive start to notes that are played. These contain the bulk of the signal harmonics that are recognisable and that we use to distinguish one instrument from another. Without this initial attack most instruments sound quite similar - and very dull! Unfortunately this is what tends to happens on compressed sounds sources.

Compressors capture much of the percussive start and reduce it in level more than they reduce the remainder of the sound. It's not as extreme as totally removing the start of notes but it still strips much of the harmonic content and removes presence from the sound.

This can be corrected by using equalisation to boost the upper frequencies but this is dangerous in sound reinforcement because when the instrument is silent and the compressor relaxes there is no gain reduction; but the upper frequency boost remains increasing noise and making microphonic feedback much more likely.

The Dynamic Enhancement in the DN540 was developed to correct the tendency for dull sound during compression without changing the sound of uncompressed signals. It works by reducing the ratio of the compressor in a relatively broad range of frequencies centred on 5kHz and the effect is continuously variable so as much presence accentuation can be added as needed or to suite taste.

When Dynamic Enhancement is used, any applied compression acts on high frequencies to a lesser extent than low frequencies and by a predetermined amount set by the user. The transient start to sounds are be greatly reduced in the low frequency content, which is where all the power is (that needs to be controlled), but the harmonic content is preserved in a more natural and dynamic state.

The presence band is an area where we are most perceptive to sounds and reducing compression ratios in this area allows much greater compression to be applied at the other frequencies for what ever reason without sounding unnatural.

Dynamic Enhancement compression can produce results that are very similar to multiband compression but with only one addition control required (as opposed to many) it is much more straightforward to set up and uses far less rack space.

Some additional corrective benefits from Dynamic Enhancement are listed below:-

Reduction of source inter-modulation – it is very common for the pop/rock singers in to stand in front of drum kits and unless suitable screens are placed between them there will be a lot of spill from the kit. Often you can hear the compressor on the vocalist modulating the spill from cymbals which sounds very unnatural. This can be eliminated by using the Dynamic Enhancement to stabilise the higher frequencies; and if that makes the vocals sound too bright the high frequencies can be reduced a little using EQ reducing the spill and reducing the chances of high frequency howl round.

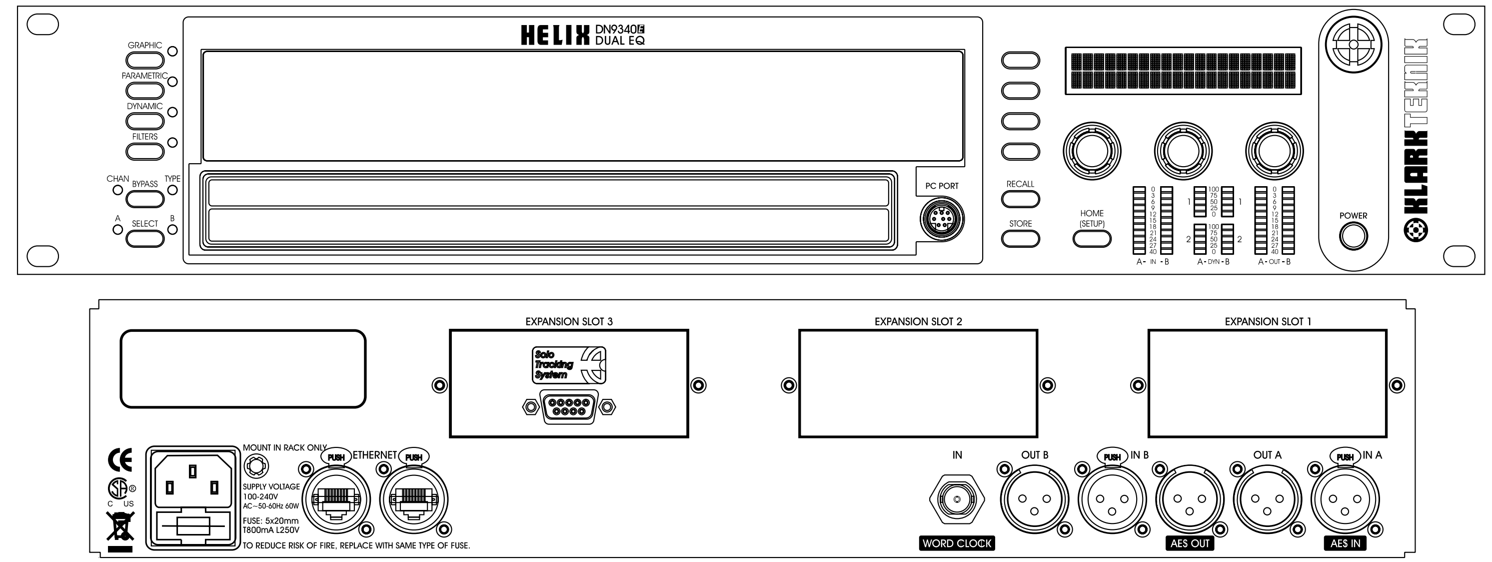
- Reduction of breathing – most processing noise occurs in the presence band and it is much more noticeable when it is modulated by the compression of a sound source. Just as above presence masking can eliminate this effect.

- Reduction of pumping – because human hearing is so sensitive to presence band frequencies, stabilising their dynamic response with presence accenting can mask the pumping effects at lower frequencies.

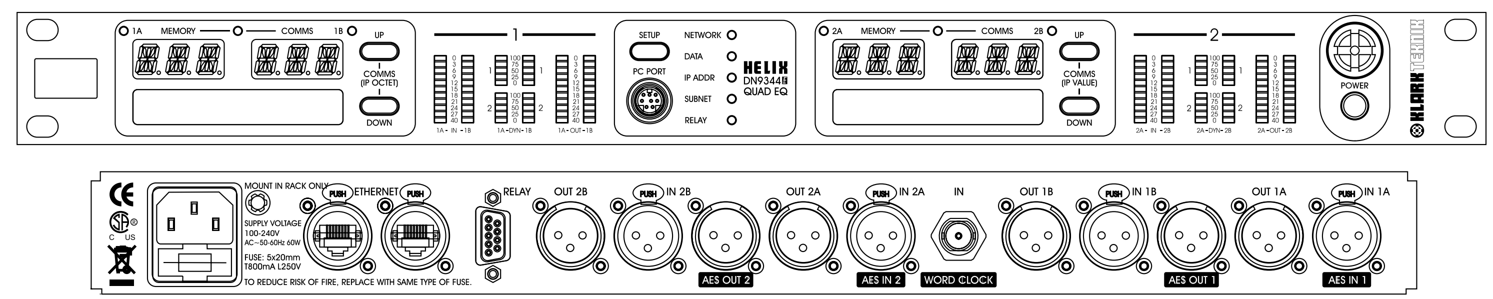
- Reduction of dull sound and increased brilliance – it is often tempting to boost high frequencies to get sounds to cut through a mix but in sound reinforcement this increases the likelihood of howl round. Dynamic Enhancement can help increase of the brilliance of compressed sounds by correcting their tendency to sound dull, and with increased use it can be used creatively to add even more high frequency energy without increasing the risk of howl round. This can be used to great effect on vocal both sweetening and improving their ability to cut through a crowded mix.

The Dynamic Enhancement feature contained in the DN540 adds another tool for the sound engineer that can be used to enhance and compliment the application of many types of compression as outlined earlier in this paper.

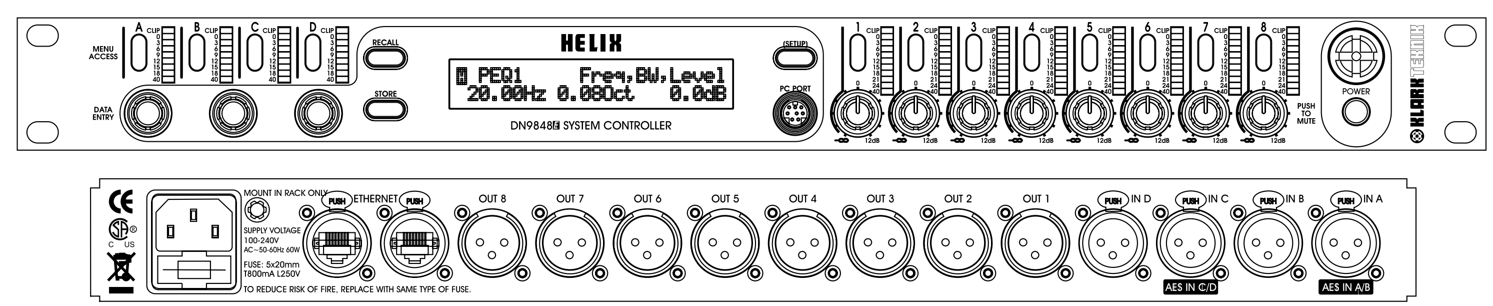
Helix DN9340E Dual EQ



Helix DN9344E Quad EQ



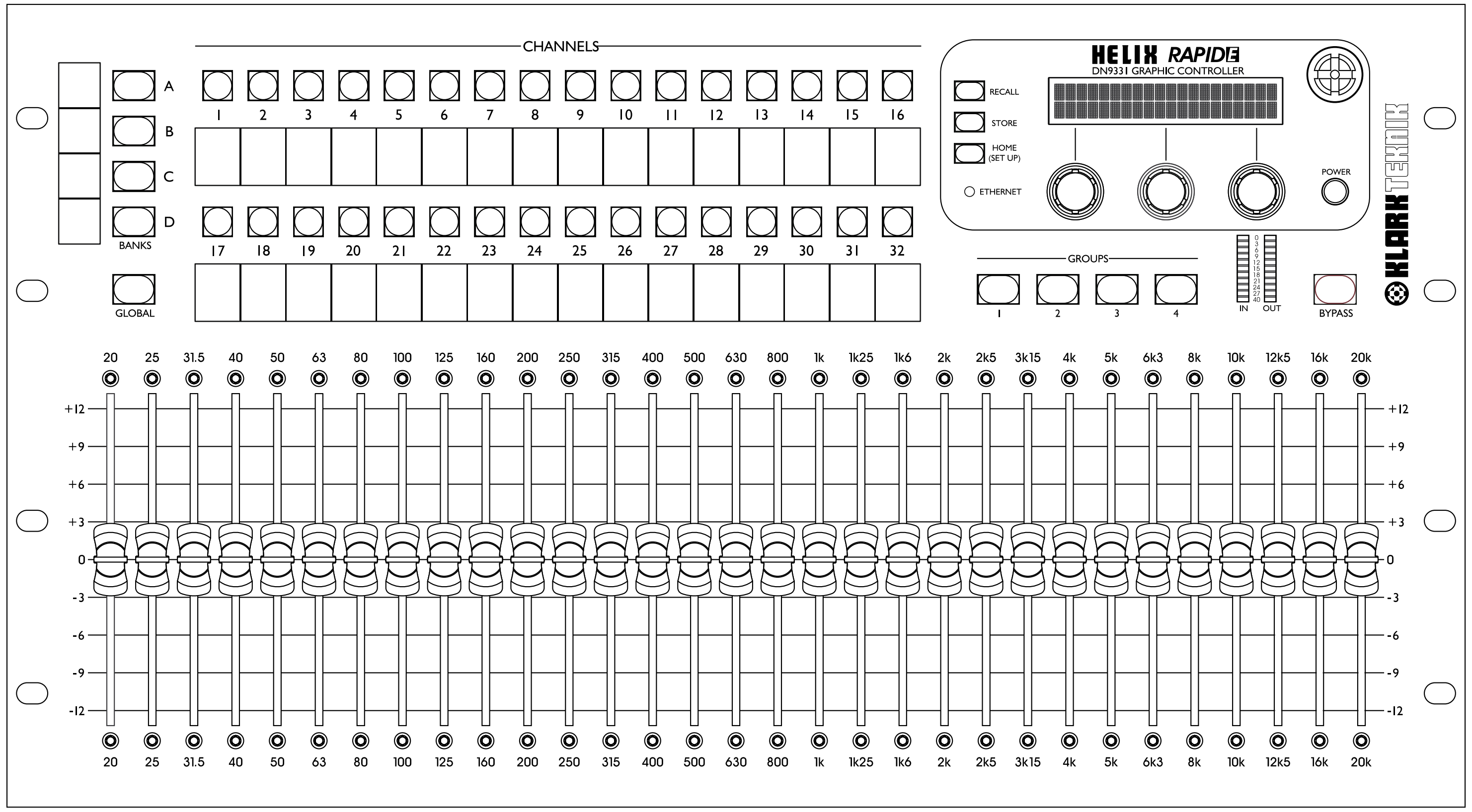
Helix DN9484E System Controller

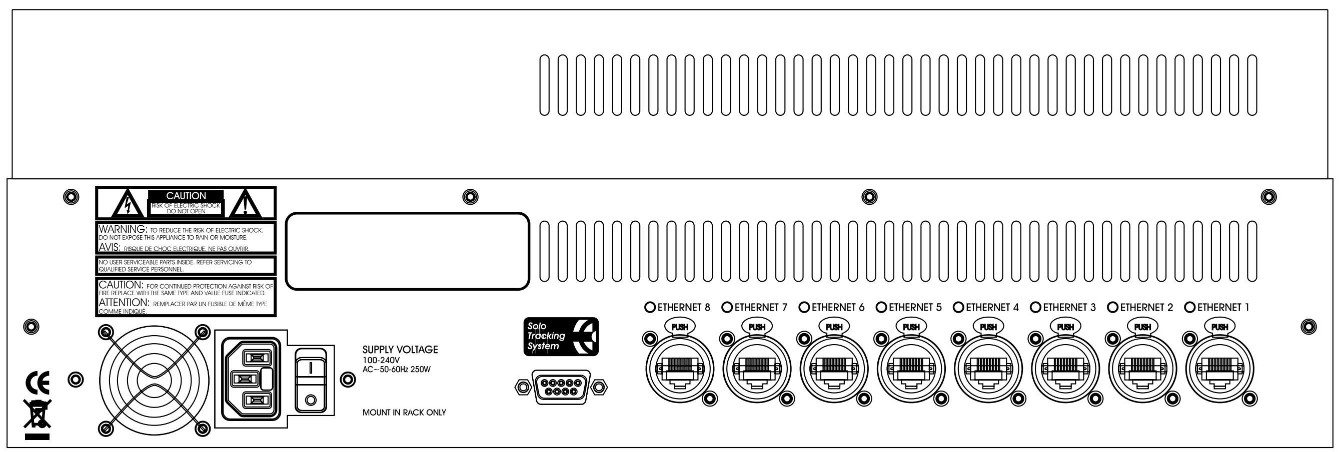


KLARK TEKNIK
SIGNAL PROCESSING BY DEFINITION

Line Drawings

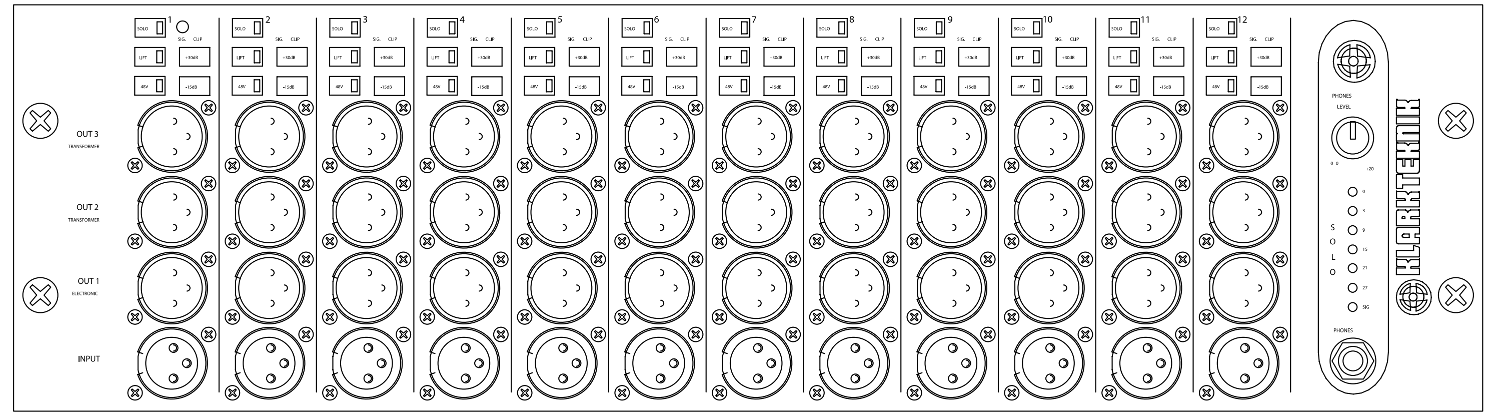
Helix DN9331 RAPIDE Graphic Controller

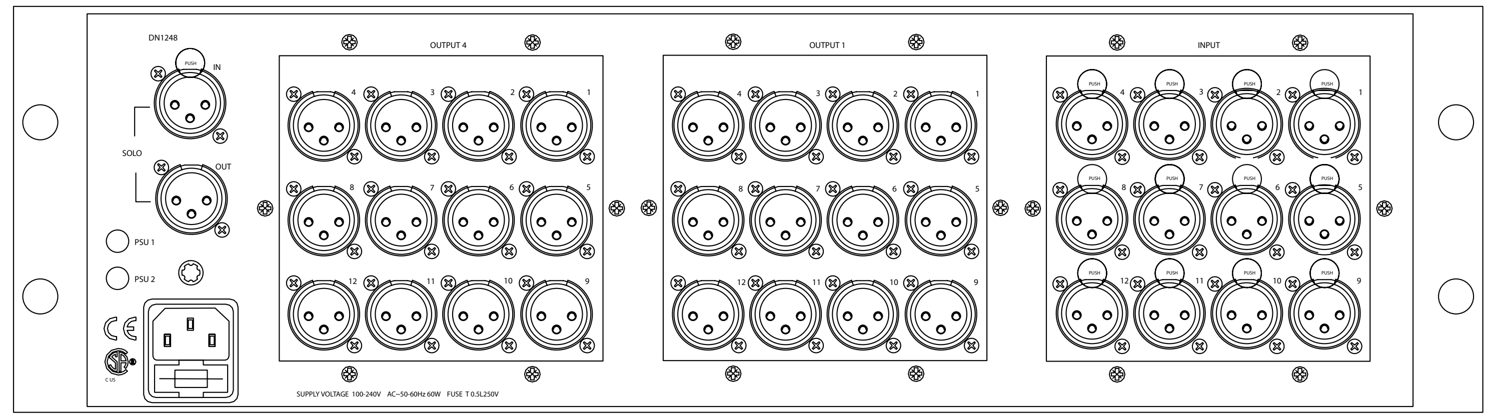




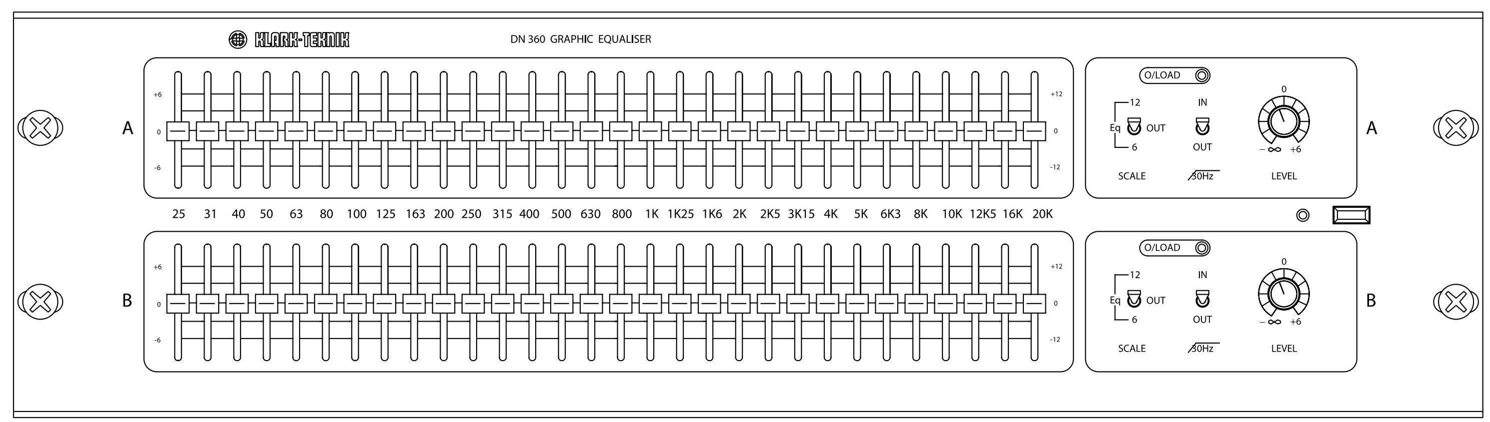
Line Drawings

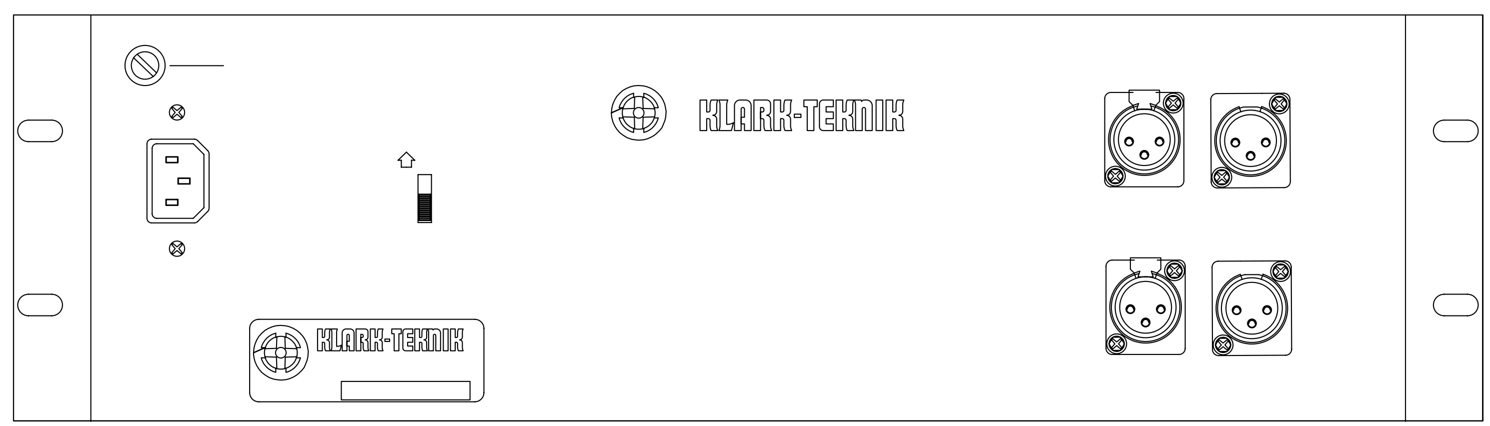
DN1248 Plus Mic Splitter





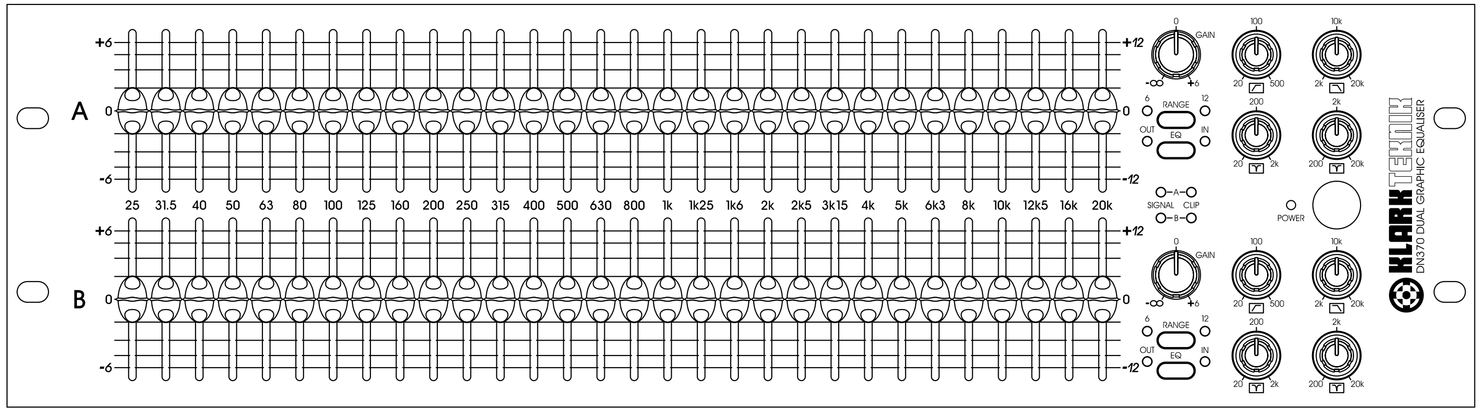
DN360 Dual Channel 30 Band 1/3 Octave Graphic Controller

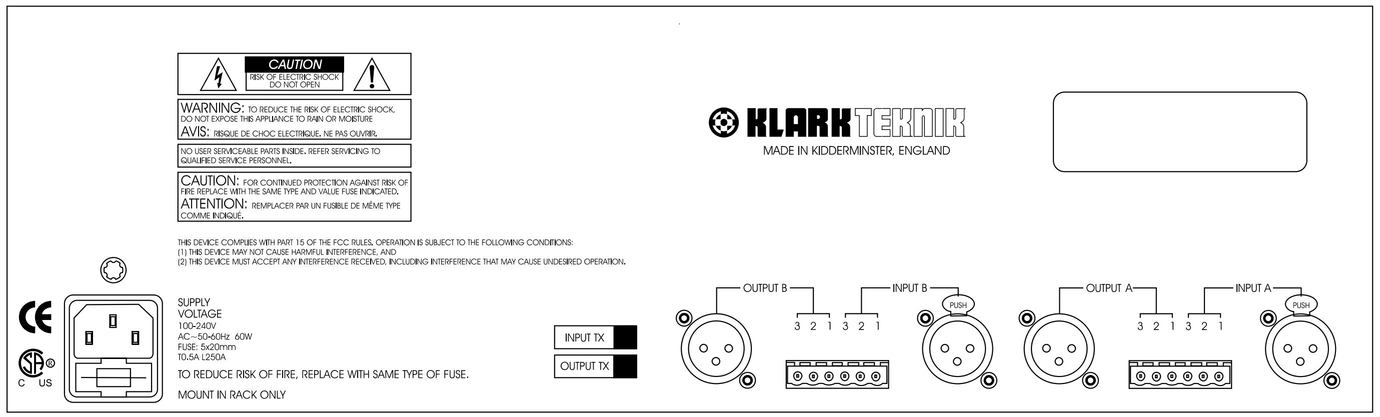




Line Drawings

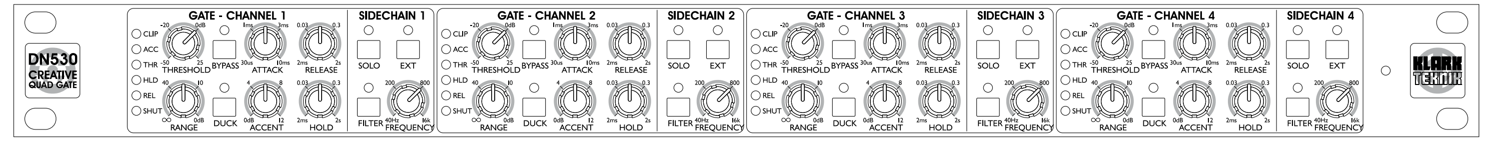
DN370 Dual Channel 30 Band 1/3 Octave Graphic Equaliser

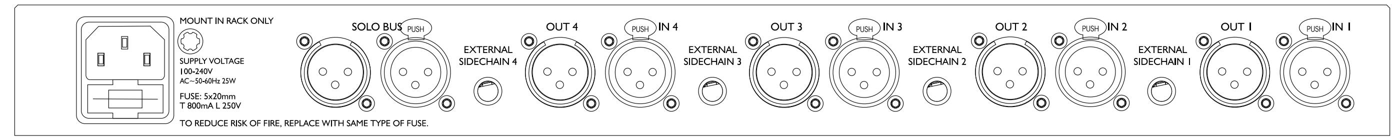




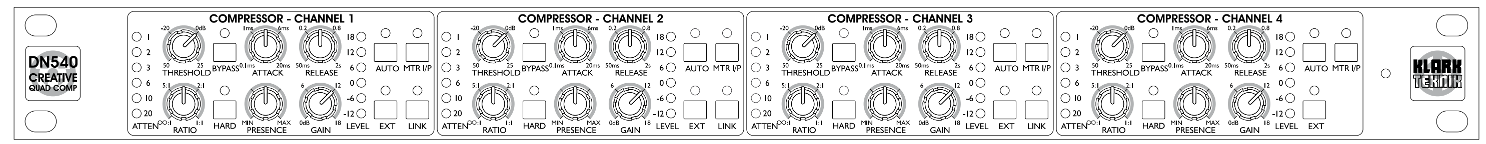
Line Drawings

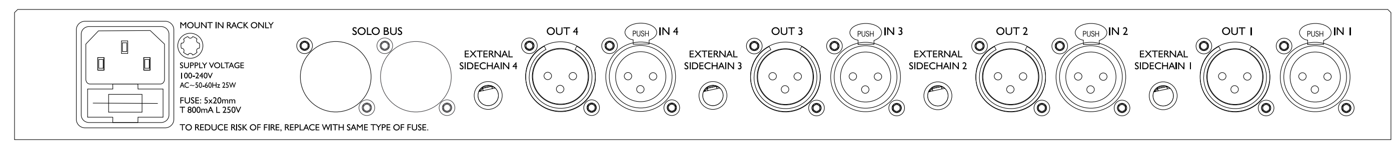
DN530 Creative Quad Gate



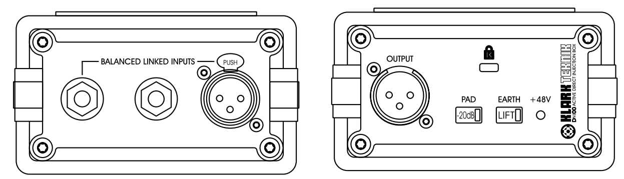


DN540 Creative Quad Compressor





DN100 Active Di



SIGNAL PROCESSING BY DEFINITION

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