KV2's 500 SERIES
User Guide

· TCL · QD8 · MPA

Contents

Important Safety Instructions

Before using your TCL, QD8, MPA be sure to carefully read the applicable items of these operating instructions and the safety suggestions.

1. Read all product instructions.
2. Keep printed instructions, do not throw away.
3. Respect and review all warnings.
4. Follow all instructions.
5. Clean only with dry cloth.
6. Install in accordance with KV2 Audio's recommended installation instructions.
7. Only use accessories specified by KV2 Audio.
8. An experienced user shall always supervise this professional audio equipment.

Installation Safety Instructions

To install the KV2's 500 Series modules into a 500 series chassis:

1. Turn off the power to the 500 series chassis.
2. Unpack the module and ensure the rear connector is free of debris.
3. Align and slide the module into the 500 series chassis, ensuring the connectors on the back properly seat with the connectors in the chassis.
4. Install the included screws to secure the module to the chassis. Both metric and standard screws are included. Use the correct screw type for your chassis.
5. Power on the chassis.
6. Enjoy!

WARNING! Do not hot swap 500 series modules! Doing so can potentially cause damage to the 500 series module or chassis. Always power down the chassis when installing or removing 500 series modules.

Connector Wiring

TCL (MONO)

#	Pin Description	#	Pin Description
1	**Chassis GND**	8	Input -
2	Output +	9	Not Used
3	Not Used	10	Input +
4	Output -	11	Not Used
5	**Audio GND**	12	PSU +16 V
6	Not Used (Normally Compressor stereo link)	13	**PSU GND**
		14	PSU -16 V
7	Not Used	15	PSU +48 V

QD8 (STEREO)

#	Pin Description	#	Pin Description
1	**Chassis GND**	8	Input - (ch. A)
2	Output + (ch. A)	9	Aux input - (ch. B)
3	Aux output + (ch. B)	10	Input + (ch. A)
4	Output - (ch. A)	11	Aux input + (ch. B)
5	**Audio GND**	12	PSU +16 V
6	Not Used (Normally Compressor stereo link)	13	**PSU GND**
		14	PSU -16 V
7	Aux output - (ch. B)	15	PSU +48 V

MPA (MONO + AUX)

#	Pin Description	#	Pin Description
1	**Chassis GND**	8	Input -
2	Output +	9	Aux input -
3	Aux output +	10	Input +
4	Output -	11	Aux input +
5	**Audio GND**	12	PSU +16 V
6	Not Used (Normally Compressor stereo link)	13	**PSU GND**
		14	PSU -16 V
7	Aux output -	15	PSU +48 V

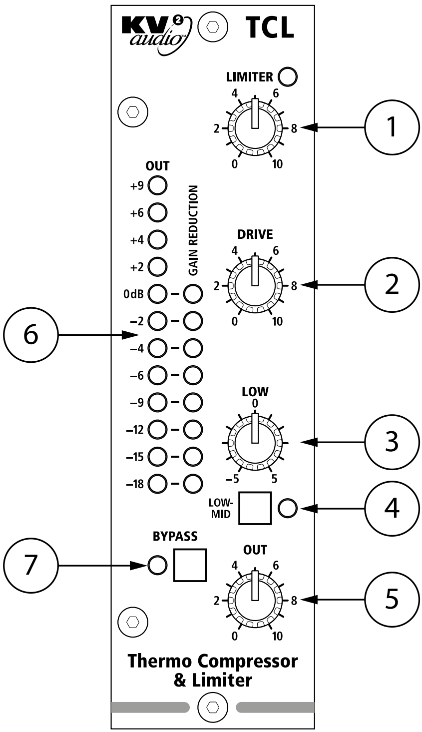
TCL - part number KVV 987 478

TCL - 500 SERIES

Superior RMS - Audio Thermo-Compressor with Limiter

Introduction

The TCL is a unique analog thermo audio compressor featuring a superior natural RMS detector and dynamic voice bass enhancement control. In another KV2 first its patented design offers an exceptionally musical and natural compression that is a component part of its real-time input, rather than a typical side fed gain reduction circuit attempting to either follow or anticipate compression demands. Its additional Bass balance control circuit allows the user to smooth out and minimise any proximity effects from a changing microphone position, giving the perfect voice color to every performance, from beginning to end. Featuring source intelligent attack and release times, whatever the dynamic content, with just a few simple steps the results from the 500 Series TCL are truly exceptional.



Application

- The TCL is the ultimate compression tool for various studio and live applications with instruments and voice.
- The TCL perfectly suits Musical theatre - giving consistency, richness and depth to head worn microphones.
- The TCL is ideal for corporate

presentations - giving a full, warm, full voice timbre and allowing tone matching for lecturn and lavalier microphones.
- The TCL can be used live on many musical instruments, both classical and contemporary.

Controls

1) LIMITER

Serves to set the maximum input level. Any subsequent levels above the set level are hard limited.
This ensures that any connected devices further down the signal chain are not affected by an overloaded input The limiter's active state is indicated by the limiter LED.

2) DRIVE

This control sets the drive level for the input of the compressor's gain reduction circuit.

3) LOW

Serves as a low frequency enhancement circuit by rerouting some signal from the second in line thermo-compressor through a low pass filter, providing a full and warm bass enriched sound, even under compression.
Select the required bass enhancement by sweeping the pot in a clockwise direction, where the center position presents a flat frequency response, with additional bass enhancement available beyond that point.
The combined output of its signal path after gain reduction and the bass enhancement circuit blend, monitor and balance themselves perfectly in real-time to maintain the all important voice color within a 20dB range.

4) LOW-MID Switch

This switch allows two alternate settings of the low pass filter frequency in the thermo-compressor LF enhancement circuit. This can be set to the operators preference depending on personal taste, microphone type and individual voice characteristics. (Male / female etc)

5) OUT

Adjusts the output level.

6) METERS

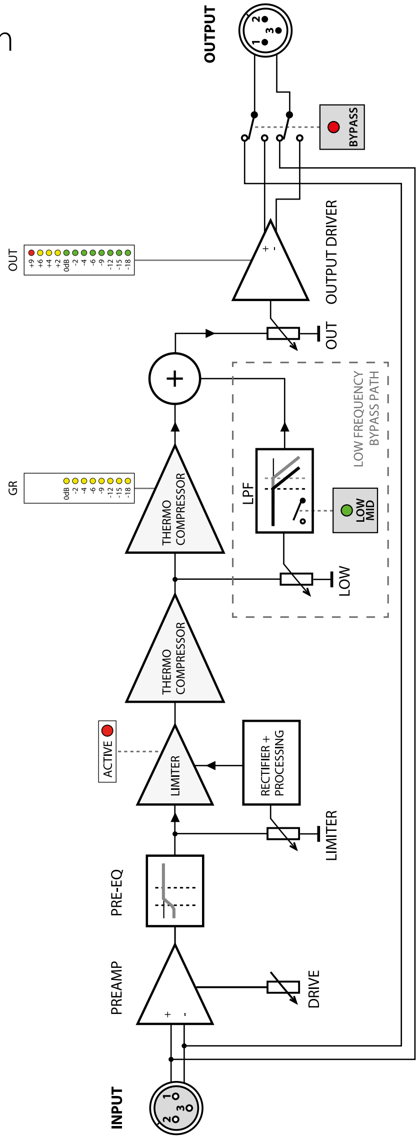
Output level meter – a 12 segment, 3 color peak reading meter serves to monitor the output signal level. If the red PEAK light flashes, the signal level is too high and should be reduced to prevent possible overload distortion.

Gain reduction level meter – an 8 segment level meter displaying the amount of gain reduction performed by the compressor.

7) BYPASS

Serves as a TCL module bypass.
Please Note - In bypass mode all meters and LED indicators will still show the lastest operation mode, but the audio path is unaffected by any of the settings.

Block diagram



Specifications

System Acoustic Perfomance

-3 dB response	15 Hz to 100 kHz
Total Harmonic Distortion	<0.002%

Signal Input

Input Channels	1
Input Impedance	20 kΩ
Max. Input voltage	+26 dBu
Features	Limiter, Compressor, Bass enhancer

Signal Output

Signal Output Channels	1
Max. Output Voltage	18 dBu (600 Ω) RMS
Output Impedance	600 Ω

Features

Limiter	Adjustable
Drive	Adjustable
Bass enhancement	Adjustable
Low-Mid	Selects the bass enhancement LPF frequency
Output	Adjustable
Indicators	12 segment VU meter, 8 segment gain reduction
Bypass	Yes

Power Requirements

Power Consumption	120 mA
Operating Voltage	±16 V

Physical Dimensions

API Series 500 format	One slot

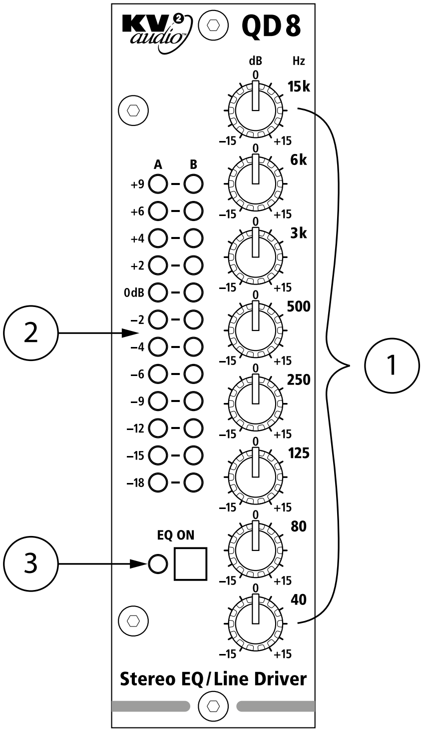
QD8 - part number KVV 987 477

QD8 - 500 SERIES

Stereo 8-band Master Equaliser with Line Driver

Introduction

A revolutionary George Krampera 8-band Equaliser specifically designed to highlight a desired band without adversely affecting the overall sonic landscape. Whilst not intended for tuning out room modes, the frequency bands and filter types have been precisely chosen after extensive research, to perfectly align with the science of human hearing and our perception of achieving the ultimate spectral balance. In a Live scenario, faced with a number of different sources, program types, or performance genres at a festival or concert, the equaliser is very musical, intuitive and an exceptionally quick way to shape any mixed content to the sound you want to hear. In the studio, it can be either the simple enhancement of an instrument / vocal track, or the final mastering polish to take an ordinary mix and make it something very special.



NOTE

As the QD8 is a stereo (two channel unit) it will require a series 500 Rack that facilitates dual channel input and outputs. It is also recommended to insert the QD8 as the last unit in the series 500 chain, directly before the amplifiers/speakers.

Application

- For use in Recording Studios as a master equaliser, single instrument, or group equaliser.
- Front of House PA system master equaliser, the unit is also equipped with line drivers designed to maintain audio signal integrity over long cable lengths (up to 200 m).
- Single stereo zone or dual mono system equaliser, to sonically match different fill speakers and zones with the main FOH PA when used in multiples.
- To reduce the harshness in some digital recordings, particularly around the 6 kHz region.

Controls

1) 8-Band Equaliser

The Equalizer boosts or cuts frequencies at 40, 80, 120, 250, 500, 3000, 6000 and 15000 Hz by +/-15 dB.

2) Output level meter

A 12 segment, 3 color peak reading meter serves to monitor the output signal level. The signal should average around '0dB'. If the red PEAK light flashes, the signal level is too high and should be reduced to prevent possible overload distortion.

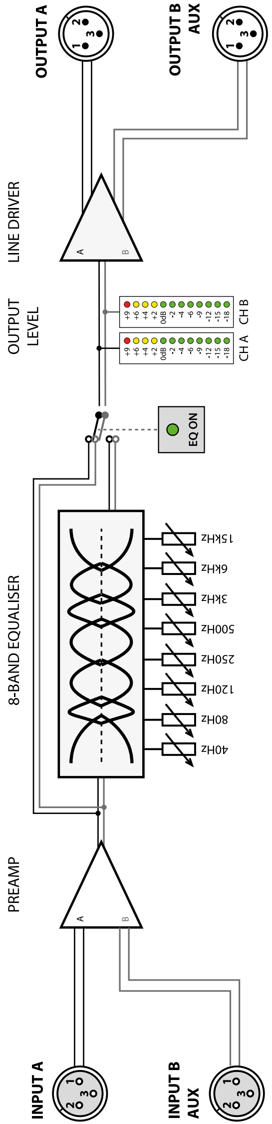
3) EQ ON switch

8-band Equaliser EQ enable switch. When depressed, the EQ is bypassed.

NOTE

It is recommended to insert the QD8 as the last unit in the series 500 chain, directly before the amplifiers/speakers.

Block diagram



Specifications

System Acoustic Perfomance

-3 dB Response	10 Hz to 100 kHz
Total Harmonic Distortion	<0.001%

Signal Input

Input Channels	2×
Input Impedance	20 kΩ
Max. Input voltage	+26 dBu

Signal Output

Signal Output Channels	2×
Max. Output Voltage	18 dBu (600 Ω) RMS
Max. Output Current	300 mA
Output Impedance	50 Ω

Features

Equalization	8-band EQ
Output	Line driver

Power Requirements

Power Consumption	160 mA
Operating Voltage	±16 V

Physical Dimensions

API Series 500 format	One slot (dual channel series 500 rack required)

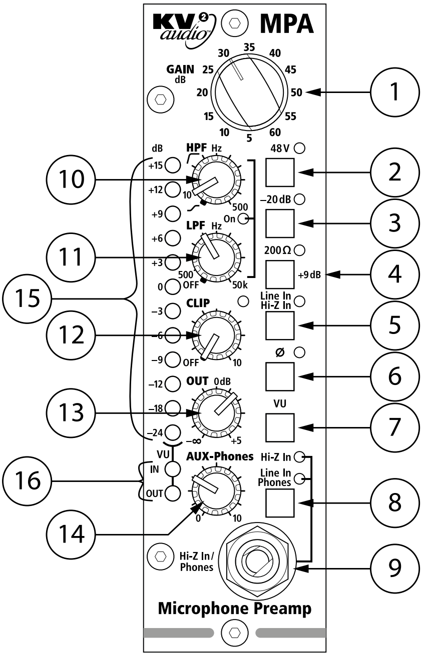
MPA - part number KVV 987 482

MPA - 500 SERIES

Microphone preamplifier

Introduction

The MPA is a Very High Definition microphone and musical instrument preamplifier with Class-A circuitry and a high quality transformer balanced output. A stepped Input gain selector is provided for easy and precise repeatability of settings. The preamplifier features adjustable high and low pass filters, with an adjustable limiter to control clip level and a transformer isolated high impedance line input. It also has a further auxiliary transformer balanced isolated output meaning a number of these units can be used together to make a multi-channel stage preamp, for splitting FOH and monitoring outputs, or for broadcast and recording duties -delivering them all through line drivers in pristine audio quality. Phase reverse and an in-built headphone amplifier, which doubles up as a third output, completes the feature rich unit. With the exceptional quality of all KV2's analog outboard electronics, the MPA can replace a preamplifier, Di-box/Splitter, line driver and headphone amplifier all in one box.



Application

- A Very High Definition studio recording preamplifier with low noise and increased dynamic range.
- A stage preamplifier and splitter with Line driver capability and Class-A circuitry.

- An esoteric Musical Instrument preamplifier with filters, Hi-Z input and Class-A circuitry.

Controls

1) GAIN

Adjusts the MPA's input gain from +5 dB to +60 dB with 5 dB steps. In conjunction with the 200 Ω setting a total gain of 74 dB is available.

2) 48 V

Supplies +48 V to pins 2 and 3 of the main input XLR for microphones, or devices which require phantom power.

3) -20 dB

Applies the -20dB PAD for the main XLR input.

4) 200 Ω

Switches the input transformer to accept a 200 Ω load impedance. The input sensitivity increases by +9 dB. This serves as an input for equipment requiring a 200 Ω loading and also as a galvanic isolator, for dynamic microphones, ribbon microphones or line level outputs. This setting helps to increase the signal to noise ratio of low level output sources (e.g. ribbon microphones).

5) Line In / Hi-Z In

This switch selects between the Main Mic input (XLR) and AUX input (XLR or front panel Jack). The AUX input sensitivity is reduced by -15 dB. The front panel Jack input sensitivity is reduced by -10 dB.

6) Ø

Applies output signal phase reverse. Both main XLR output and AUX output are reversed at the same time.

7, 16) VU

Selects the signal meter display between input and output levels. The 'IN' and 'OUT' LEDs as shown at 16) indicate the current state of metering.

8) Hi-Z In / Phones

Selects the front panel Jack function between the high impedance input and phones output.

9) Front panel Jack 6.3 mm

Serves as a high impedance input or Phones output.

10) HPF, FIX/TUNE (in conjunction with control 11)

The controls 10) and 11) are designed to be operated together as an overall HPF/LPF circuit. To realise any settings from control no 10) you must first turn on the overall HPF/LPF section using control no 11). Turning this control from the off click position fully clockwise to the 50kHz setting will give you an open, extended high frequency setting without an audible filter. This gives an option to use just the HPF feature on its own by adjusting control no 10) leaving no 11) fully clockwise.
The first position on the control 10) is fully anti clockwise. This is not the off position but in fact a special 'FIXED VOCAL PRESET' designed by George Krampera.
This preset has been optimised to maintain a rich and full vocal quality, retaining and shaping the most important bass frequencies, whilst reducing any problematic low frequency content or typical rumble.
The next part of the control is a tuneable high pass filter (HPF) selected by turning the control clockwise past the click. Select the required cut off frequency by sweeping from 10Hz to the maximum 500 Hz (when fully clockwise) The filters are adaptive, which means they change their characteristics depending on what frequency is selected. The lower the frequency that is applied, the steeper the filter is tuned.

11) LPF

A tuneable low pass filter (LPF). This can be applied to clean up sounds from any unwanted high-frequency content. Once switched on, select the required cut off frequency by sweeping anticlockwise from the maximum 50 kHz down to 500 Hz. The LPF drops by 12 dB per octave above the cut off frequency.
NOTE-To use this LPF in isolation without any additional HPF adjustments set the control above no 10) to the minimum tuneable setting of 10 Hz (Not the fully anti clockwise 'Fixed vocal preset' position).

12) CLIP

The Clip limiter control sets the maximum output level, with all levels above this point limited. After switching on the clip circuit with a clockwise turn, the red LED to the right will indicate when the circuit is actively limiting.
This ensures any further devices in the signal chain are not affected by undesired clipping or distortion.

13) OUT

Adjusts the output level from -∞ to +5 dB.

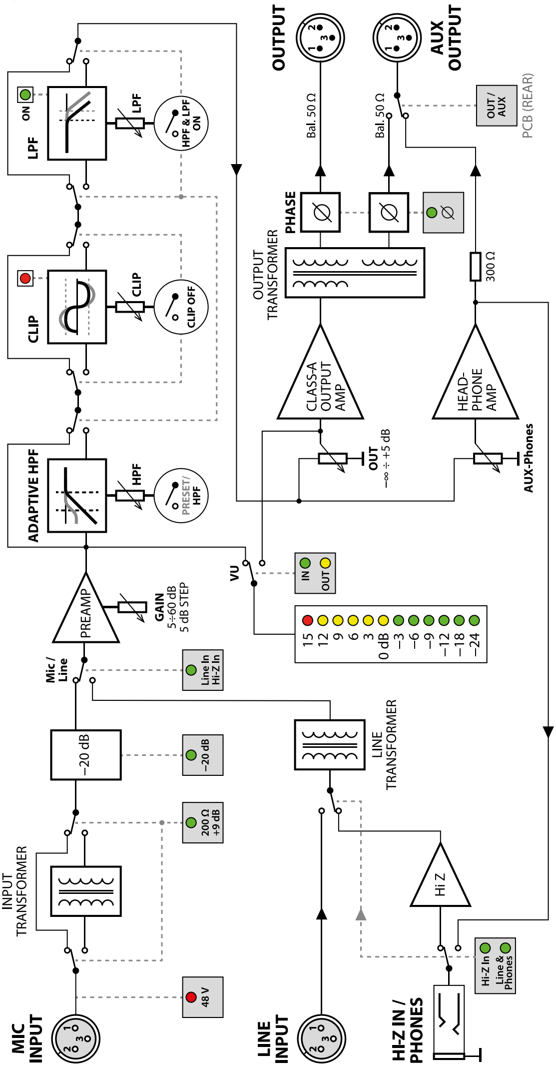
14) AUX-Phones

Adjusts the AUX and Phones output levels.

15) Input/Output level meter

A 12 segment, 3 color peak reading meter serves to monitor input or output signal levels depending on the position of the VU switch no 7). The signal should average around '0 dB'. If the red PEAK light flashes, the signal level is too high and should be reduced using the GAIN control to prevent possible overload distortion.

Block diagram



Specifications

System Acoustic Perfomance

-3 dB Response	10 Hz to 100kHz
Total Harmonic Distortion	<0.002%

Signal Input

Input Channels	3×
Input Impedance	2 kΩ / 200 Ω Mic (+9 dB), 1M Ω Hi-Z In
Max. Input voltage	+26 dBu
Mic input	Phantom power, -20 dB, Gain, Phase, LPF, HPF
Line input	Hi-Z, Input transformer

Signal Output

Signal Output Channels	3 (Main, Aux, Phones)
Max. Output Voltage	19 dBu (50Ω) RMS
Recommended loading	≥50 Ω balanced, ≥600 Ω AUX unbal., ≥10 Ω phones

Features

Input Level Control	+5 to +60 dB, 5 dB step, total gain 74 dB
High Pass Filter	Vocal preset / 10 Hz to 500 Hz
Low Pass Filter	500 Hz to 50 kHz
Clip	Adjustable
Indicators	LED, 12 segment VU meter
Phantom power	48 V

Power Requirements

Power Consumption	250 mA
Operating Voltage	±16 V

Physical Dimensions

API Series 500 format	One slot

Warranty

Your 500 SERIES is covered against defects in material and workmanship.
Refer to your supplier for more details.

Service

In the unlikely event that your 500 SERIES develops a problem, it must be returned to an authorised distributor, service centre or shipped directly to our factory. Because of the complexity of the design and the risk of electrical shock, all repairs must be attempted only by qualified technical personnel.

If the unit needs to be shipped back to the factory, it must be sent in its original carton. If improperly packed, the unit may be damaged.

To obtain service, contact your nearest KV2 Audio Service Centre, Distributor or Dealer.

KV2's 500 SERIES · Notes

The Future of Sound.
Made Perfectly Clear.

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KVV120155-00-03-0