JOEMEEK
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USA

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SIX Q

Joemeek User Guide

visit us on the web at joemeek.com

Joemeek is manufactured and marketed under the direction of:

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CE

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JOEMEEK reborn – the legend grows

The Next Generation of Joemeek studio processors represents a quantum leap in the history of the Joemeek legend. Long regarded for its “Big Sound”, the original Joemeek gear was both revered and reviled for its somewhat “quirky” nature. Now we have taken the best of what made the Joemeek products sound great, refined it, distilled it, added to it and repackaged it.

Properly and robustly engineered for predictable, controllable performance, the new range retains the famous Joemeek sound, with its wide, flat frequency response extending from subsonic to ultrasonic. It also uses genuinely low noise circuitry, with lots of headroom (immunity to overload). Accurate calibration and metering, together with clear panel labelling, give you complete confidence in what’s going on. While some equipment pays lip-service to quality and “professional rules” but fails to deliver, the Next Generation Joemeek products are founded on good solid electronic and audio engineering, and withstand direct comparison with the very best names in mixers and outboard gear.

The Joemeek range provides everything you need to get your performance onto tape/disc.

About the Designer

The Next Generation of Joemeek has been completely re-engineered by renowned audio electronics consultant Allan Bradford. With his background in physics and 30 years experience with the design of instruments, mixers, processors and amplifiers, Allan’s unique range of expertise ensures that Joemeek remains at the forefront of music technology.

sixQ Controls at a Glance

48V PHANTOM POWER switch - feeds 48V power to the microphone XLR connector. Most condenser microphones require phantom power to operate.

IRON switch - selects transformer coupling of the Mic (XLR) input. The LED lights when active.

PAD switch - selects 20dB attenuation of the Mic (XLR) input. The LED lights when active.

LINE switch - selects the Line and INSTRument (1/4” jack) inputs instead of the Mic (XLR) input. The LED lights when active.

Ø switch - reverses the phase of all inputs.

INSTRument input – high impedance input. Plugging in here overrides anything plugged into the Line input.

PREAMP GAIN - sets the amount of audio amplification. Too little gain and the sound will be too quiet; too much and the signal could become distorted.

MAKE UP GAIN - restores the level of the signal after compression.

COMPRESSION METER - 4-Led bargraph indicates the amount of gain reduction in dB, which is taking place at any given moment.

Compressor ON switch - turns the compressor on. The LED lights when active.

LF - controls the volume of Low Frequencies or “Bass” in the audio spectrum. 15dB of boost or cut is available at the selected frequency.

LF FREQ - sets the frequency at which the LF control operates, anywhere from 40Hz to 650Hz.

MID - controls the Middle frequencies in the audio spectrum. 15dB of boost or cut is available at the selected frequency.

MID FREQ - sets the frequency at which the MID control operates, anywhere from 300Hz to 5kHz.

PEAK LED - lights 6dB below clipping.

HPF - “high-pass filter”. Mainly for use with microphones, this helps remove stage rumble, handling noise and “pops”. The LED lights when active.

COMPRESS - sets the level of signal (or “Threshold”) above which the signal starts to be compressed.

SLOPE - sets the compression ratio applied to signals above threshold.

ATTACK - sets how quickly the compressor responds to peaks above threshold.

RELEASE - sets the time taken for the signal to return to its normal size after compression. In general, the longer the time, the less obvious the compression.

HF - controls the volume of the High Frequencies or “Treble” in the audio spectrum. 15dB of boost or cut is available at the selected frequency.

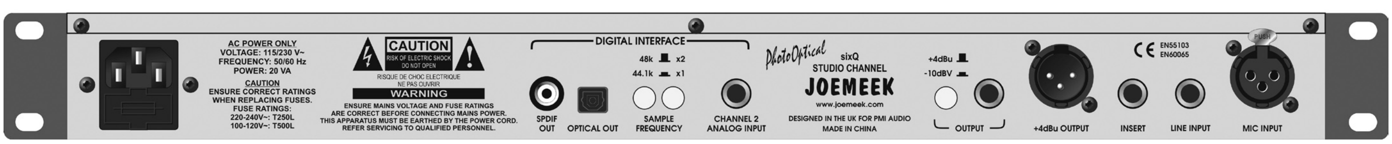
6kHz switch - sets the frequency at which the HF control operates, in for 6kHz, out for 12KHz. The LED lights when in.

EQ ON switch - turns the Meequalizer on. The LED lights when active.

OUTPUT GAIN - the volume control or “Fader” for the output of the sixQ.

VU METER - 8-Led bargraph shows the output signal level in dB at any given moment.

+4dBu/-10dBv switch - selects the operating level of the 1/4” jack output, either to the professional +4dBu level, or to the -10dBv semi-pro level.



Overview

The JOEMEEK sixQ is like having one channel of a professional recording studio in one box. It takes microphones or instruments, amplifies them, compresses and equalizes them ready to be recorded. Simple to use yet extremely powerful, the sixQ will bring out the best in any microphone or instrument and give the gloss of a professional studio production to all your performances. As well as recording it will also be found useful for live work.

Think of each channel of the sixQ as four separate items of equipment:

- **The Preamplifier**
- **The JOEMEEK Optical Compressor**
- **The Meequalizer**
- **The Fader**

Preamplifier

This is the all-important front end to the sixQ. Its job is to accept any type of microphone, instrument or other source of audio signal, and make it loud enough. Microphones often need rather a lot of amplification, while guitars, keyboards and CD players need less. Mics need to be connected to low impedance inputs, while instruments prefer high impedance inputs. To ensure correct impedance matching, the inputs are split into an XLR connector for Mics, and 1/4" jack "Line" and "INSTRument" connectors for everything else. A switch on the front panel decides which input connector is active, the XLR or the 1/4" jacks. The LED next to the switch lights to show that the Line inputs (jacks) are selected. In other words:-

Switch out (LED off) = "Mic"
Switch in (LED on) = "Line" or "Instr"

Both Mic and Line inputs are electronically balanced. Note: although the Line input is not normally used for microphones, it can also be suitable for some high output unbalanced microphones, such as battery powered Electret types.

The rear panel Mic input (XLR) is balanced and wired as follows:
Pin 2: + (hot)
Pin 3: - (cold)
Pin 1: ground

The Line input (jack) is balanced and wired as follows:
Tip: + (hot)
Ring: - (cold)
Sleeve: ground

The front panel Instrument input (jack) is balanced and wired as follows:
Tip: + (hot)
Sleeve: ground (NB: use a mono jack plug).

Note that if something is plugged into the Instrument input, anything plugged into the rear panel Line input will be cut off.

Phantom power
Most high-quality studio mics are "Phantom powered", which is to say they have electronics inside them, which get their power from the preamp. Most mics require a supply of 48 Volts, so Phantom Power is often labelled "48V". The "48V" switch turns this power on or off and a red LED lights when active. When switching the Phantom Power on, quite a loud thump may be produced, so it is a good idea to turn down the Output Gain (or to momentarily select the Line input), when pressing the switch.

When using dynamic or ribbon mics, do not turn Phantom Power on. It probably won't do any harm but it certainly won't do any good, so leave it off! Consult the microphone handbook if you are unsure what kind of mic you have.

The main control, labelled "Input Gain", covers a range of amplification from 10dB to 60dB. In many other preamps the action of the Gain control is rather uneven, with the 40dB to 60dB range being crammed into the last 1/6th of a turn. All Joemeek preamps use a specially designed control that ensures smooth operation over the whole range of rotation. The (θ) symbol next to the 25dB mark, means unity gain, or 0dB, for a signal in the Line input. Hence for Line inputs the range of gain adjustment either side of this mark, is +35dB, -15dB.

The PEAK LED lights 6dB below clipping, so occasional brief flashes are OK but if it's on all the time you need to back the Input Gain off!

HPF means "high-pass filter". Mainly for use with microphones, this helps remove stage rumble, handling noise and "pops". The LED lights when active.

Technical stuff
Very low noise - does it matter? Yes and no, it all depends what you are doing - what really matters is "signal-to-noise ratio". All electronics produce a certain amount of background noise - it's in the nature of things. Providing there is only a relatively small amount of noise, the signal will cover it up, or "mask" it. So providing the signal is much bigger than the noise, you won't be aware of the noise. In other words the "signal-to-noise ratio" needs to be a big number, ideally such as 80dB or 90dB.

So how do you achieve that in practice? The trick is to keep the microphone as close to the sound source as possible without overloading it, so as to get as much signal out of it as possible. Then you set the Gain control to give only as much gain as is needed to get a decent level into the recorder.

Of course when there is no signal going on, you may hear the background noise of the electronics. In that case, given the amount of gain in a typical studio monitoring system, this noise "floor" should ideally be in the region of -80dBu or lower, in order for it not to be noticed.

The sixQ microphone preamplifier uses state-of-the-art electronics and has an equivalent input noise of around -128dBu (with 150ohm input load). Despite all the hyperbolae and obfuscation, the theoretical best possible performance for silicon-based electronics is about -132dBu. So the preamplifier design used in the sixQ and all other NextGen Joemeek products approaches this limit. To improve significantly on this would require highly specialised electronics and probably a vat of liquid Nitrogen to cool it!

The maximum gain available from the preamp is 60dB, in which case the noise floor will be -68dBu. This is actually quite noisy - if you record that noise onto a digital recorder and play it back you can definitely hear it. In practice of course, you do not record and play back "silence" and the rest of the mix will probably be more than 70dB louder than this noise and will mask it completely. Even so it is generally a good idea not to use gains greater than 40dB or 50dB and indeed, it should rarely be necessary to do so.

Insert Point

This is simply an unbalanced "Send and Return" jack on the rear panel. It allows you to patch any other pieces of equipment into the signal path, such as an effects processor or noise gate. To use it you will need a "Y" lead wired as follows:

Tip: send
Ring: return
Sleeve: ground

When no jack is inserted, the socket is internally linked, or "normalled", so that the signal flows uninterrupted. Note that the Insert Point is after the Preamp but before the Compressor and EQ.

METER
The meter displays one of two things, depending on the setting of the "PRE" switch.

With this switch out, the Meter shows signal level at the outputs, after the Output Gain fader. Note that this is relative to the selected operating level of "+4dBu" or "-10dBv". In other words if you have selected "+4dBu" and the meter reads "0", then you have +4dBu coming out of the 1/4" jack output socket. If you have selected "-10dBv" and the meter reads "0", then you have -10dBv coming out of the output jack.

Pressing "PRE" allows the output of the preamp to be metered directly, rather like the "PFL" button on a mixing console. This is useful for adjusting the gain of the Preamp.

Compressor

The hardest device to understand, yet one of the most useful, the PhotoOptical Compressor is what gives Joemeek products their unique character. Its job is to make quiet sounds louder and loud sounds quieter, or in other words to reduce the dynamic range of the programme material. It's a bit like manually riding the volume control, except the compressor does it automatically, responding far quicker and more accurately than you ever could by hand. The compressor is applied in several ways:

1. Make Sounds Stand Out
Because compressors make loud sounds quieter, you can boost the volume of the quiet bits without the loud bits getting even louder. That means you can raise the average level of an instrument or vocal in the mix, which has the effect of lifting it and bringing it forwards. This can actually improve vocals for example, bringing them out in front of a mix, making them sound denser, more even, and more confident!

2. Crank Up The Volume
Raising the average volume of whole mixes means they can be heard in noisy environments, such as vehicles and factories. Boosting the average level is what makes radio stations sound LOUD and the same technique is used on TV commercials too, which is why they always seem annoyingly louder than the movie you were trying to watch!

3. Protection
Fast response times are generally used to control brief transients. In other words if an occasional peak sticks its head above a maximum permitted level, the compressor clobbers it; this is known as limiting and a compressor designed solely for this purpose is known as a Limiter. Limiters are primarily used to protect recorders and monitor systems from overload, radio transmitters from overmodulation, etc. The Joemeek compressor is not primarily intended for this purpose as the Attack is not really fast enough to satisfy radio station requirements, although it is generally good enough to protect recorders and monitors, where the effect of transients is less critical. Normally you should not hear a limiter operating but if it is driven hard constantly, it can render a mix somewhat flat and lifeless.

4. Accommodation
The dynamic range of the human ear is phenomenal, extending from the threshold of hearing (eg: a pin dropping onto soft carpet) to threshold of pain (eg: standing next to a jet aircraft) - some 120dBA in all. By contrast, vinyl, cassette tape and radio broadcasts all have a dynamic range of about half that. Since the advent of the CD, the dynamic range of the medium is far less of an issue and compressors are used more to give a certain "feel" to a production. AM and FM radio however, is still very much compressed to fit its restricted dynamic range.

5. Modification
A compressor can change the dynamics, or "envelope" of the track and it is here that the Joemeek Compressor excels!

Types of Compressor
Most compressors work in essentially the same way: a volume-controlling element or "gain cell" is inserted into the audio signal path. The level of the signal at any given moment is measured and that information is used to control the gain cell. So if the signal gets bigger, the volume is turned down. Various types of gain cell in common use include FETs, valves (tubes), light-dependent-resistors (photoelectric), digital potentiometers and voltage-controlled-amplifiers, better known as VCAs.

The sixQ Compressor is a unique recreation of the sort of photoelectric compressor used by record producer Joe Meek in the 1960's. Using modern components for consistency and reliability, it nonetheless reproduces faithfully the same punchy sound that was so characteristic of the pop records of that time.

Compression Ratio
What?? OK, it's simpler than it sounds. If the input gets 10dB louder but the output only increases by 5dB then the compression ratio is "2 to 1". If the input goes up 10dB but the output only goes up 1dB, then the compression ratio is "10 to 1". In a theoretically ideal compressor, this ratio is the same for any size of signal above the threshold but for that to be true, the gain cell and its control circuitry must be perfectly linear over a very wide range. In practice only compressors based on VCAs and digital potentiometers are likely to behave in this way.

Some compressors have a control to set the ratio anywhere between 1:1 (ie: no compression), and 20:1 (which would be regarded as a "brick wall limiter"). In the sixQ the "Slope" or "Compression Ratio" is variable from 1:1 (ie: no compression) to 10:1. Slopes around 3:1 are gentle for vocals while higher slopes are hard for drums and guitars. However that's not all there is to it.

Variable Ratio
In the Joemeek optical compressor the compression 'threshold' is not clearly defined and the compression ratio varies with the amount of compression applied. Suppose the 'Slope' control is set to 5:1. For signals only just exceeding threshold, the ratio is little more than 1:1. As the compressor is driven harder, the ratio rises to 5:1, at least up to a point. It is a feature of the Joemeek compressor that the compression ratio actually reduces again during large transients and, adjusted correctly, this helps to retain brightness that is often lost with other types of compressor. This is why vintage compressors often sound more lively than their modern counterparts.

Controls

'COMPRESS' sets the level of signal (the "Threshold") above which the signal starts to be compressed. Turning the compression control clockwise lowers the compression threshold, and drives the compressor harder.

'SLOPE' sets the average compression ratio applied to signals above threshold. Lower settings (anti-clockwise) have less effect. Turning the control clockwise increases the ratio and makes the effects of compression more dramatic. At maximum (10:1) the Joemeek compressor effectively becomes a limiter.

'ATTACK' sets how quickly the compressor reacts to peaks above threshold. Turn this control anticlockwise for a quick response. Slower (clockwise) allows the fast leading edge of percussive sounds to pass uncompressed for a moment, before the compressor reacts to control the gain. This example of "changing the envelope" of a sound exaggerates the percussive nature of drums and other instruments. Settings around mid-position are used where the compression needs to be less obvious. Vocals for example, require Attack times around 10msec for natural sounding results. Faster attack times (anti-clockwise) in conjunction with large amounts of compression, result in extreme "pumping" effects.

'RELEASE' sets how long the compressor goes on squashing the sound for, once the signal has dropped below threshold. If it stopped instantly there would be very noticeable modulation or "pumping" of the sound. So we may want it to stop compressing less abruptly and that is what the Release control is for. Generally, the longer the Release time, the less obvious is the compression. Of course some "pumping" might actually be desirable as a special effect and that is another way in which the envelope of a sound can be modified. The sixQ Release is variable from 100mS up to 3 seconds giving a wide variety of effects.

How the compressor behaves actually changes with programme content and volume. So experiment with the controls with different kinds of material to discover the range and depth of effects that can be achieved. The 'COMP' in/out switch allows comparison between compressed and uncompressed sound (blue LED lights when active). Remember that the 'MAKE UP GAIN' is there to restore the level of the signal after compression. Correctly adjusted, there will be no change in volume as the Compressor ON switch is operated.

The Compressor is after the Preamplifier and the Insert Point, and before the Meequalizer.

Meequalizer

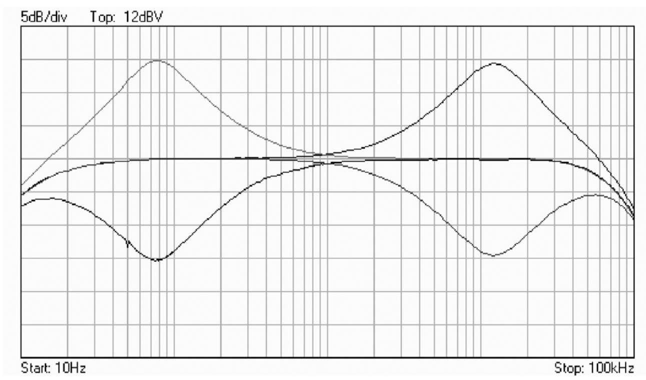
The sixQ "Meequalizer" is a highly effective, versatile and musically rewarding three-band equalizer, or tone control system. Each stage allows boost or cut of up to 15dB around the frequency in question. The "EQ" switch turns the equalizer on, and the green LED lights when active.

The LF band can be tuned or "swept" anywhere between 40Hz and 650Hz. This effectively covers the whole range of low frequencies. It may help to think of it as like a graphic equalizer, only instead of lots of frequency bands, you have just one, but it can be moved to cover any given frequency band. Cutting can be used to reduce unwanted LF noise, such as hum or rumble. Boosting can bring out the warmth and body of bass lines and (especially around 80Hz) kick drums.

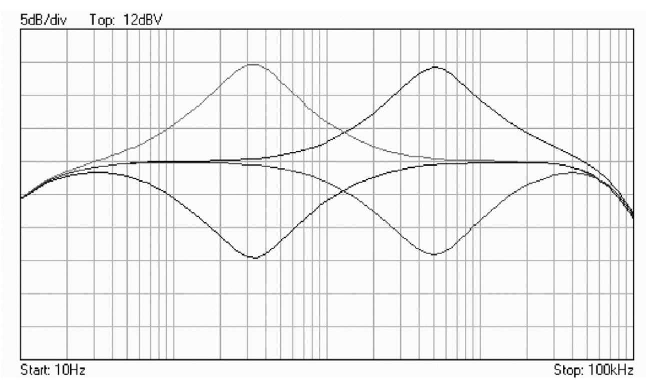
The Mid band can be tuned or "swept" anywhere between 300Hz and 5kHz. This effectively covers the whole range of mid frequencies. Cutting the Mid can reduce sibilance, boominess or other annoying resonances. Boosting can bring out the body of a vocal, or the harmonics of instruments. Increasing or reducing the "presence" of an instrument or vocal in this way, can appear to move the sound forwards or backwards in a mix.

The HF or treble section is centred at either 6kHz or 12kHz. Boosting the 12kHz band gives a sense of "air" or "sparkle" to vocals, instruments and mixes, without boosting harsh upper-mid frequencies. Alternatively with bass instruments, cutting this band will reduce HF noise such as hiss and crackle. The 6kHz setting is very effective at controlling sibilance and reducing harshness, or indeed creating it, for example by boosting the harmonics of electric guitars.

The Meequalizer is after the Preamplifier, the Insert Point and the Compressor.



LF and HF frequency response



Mid frequency response (at 300Hz and 5kHz)

Technical stuff

Each section of the Meequalizer has a peaking or "bell" shaped frequency response, which will be found to be musically more satisfying than conventional "shelving" equalizers. The use of bell curves at LF and HF also avoids boosting subsonics and ultrasonics which can have adverse effects on other studio equipment, such as recorders, monitor amplifiers and speakers. The "Q" value of the peaking filters is 0.9 (or 1.6 octaves). Zero phase distortion ensures the best possible audio coherence.

Output Stage

'OUTPUT GAIN'. This output volume control provides up to 10dB of gain and also goes right down to nothing, so acting as a fader to fade a sound out completely.

CONNECTORS

Two outputs are provided, jack and XLR, so you can simultaneously feed (say) a recorder and a monitor amplifier. The switch adjacent to the 1/4" jack selects the output operating level to either +4dBu (suits most professional studio equipment) or -10dBv (suits semi-pro or hi-fi equipment). The XLR output is always +4dBu. Check with the handbook for whatever you are feeding, to find out which level is required.

The XLR output is balanced and wired as follows:
Pin 2: + (hot)
Pin 3: - (cold)
Pin 1: ground

The jack output is balanced and wired as follows:
Tip: + (hot)
Ring: - (cold)
Sleeve: ground

Balanced or Unbalanced
To run the XLR output unbalanced, it will be necessary to ground pin 3 of the cable connector.
To run the1/4" jack output unbalanced, just plug in a mono jack plug. Either way, this increases the gain of the "+" signal by 6dB, so there is no drop in level compared with balanced gear.

All outputs on all NextGen Joemeek products are properly balanced, which is to say there is a signal on both pins! In this way the maximum possible common-mode rejection of interference, can be achieved at the receiving end.

VU METER
The LED VU Meter shows signal level at the outputs, after the Output Gain fader. It covers the range -24dB to +12dB in eight steps. Note that this is relative to the selected operating level of "+4dBu" or "-10dBv". In other words if you have selected "4dBu" and the meter reads "0", then you have +4dBu coming out of the output sockets. If you have selected "-10dBv" and the meter reads "0", then you have -10dBv coming out of the output sockets.

Digital Interface

The Joemeek Digital Audio Interface provides high quality digital audio outputs compatible with most digital recorders, as well as Digital Audio Workstations and mixers. S/PDIF format is available from the optical and RCA phono connectors, while the transformer coupled XLR connector provides an AES3 compatible output.

The Joemeek Digital Interface has highly stable onboard master clocks for low-jitter, hi-fi results. Internal sample rates of 44.1kHz, 48kHz, 88.2kHz or 96kHz are selected by means of rear panel switches. 44.1kHz is the standard used for audio CD's, while 48kHz and 96kHz are widely used in recording studios. In general, the higher the sample rate, the better the audio fidelity, but the more disk space is required for the recording. For example, recording at 96kHz requires twice as much storage as 48kHz. Consult the manual of your recorder or DAW as to what sample rates it will accommodate.

To avoid distortion, care should be taken not to overdrive the input to the Digital Interface. The red LED next to the Output Gain control is labeled "PEAK FSD" which stands for "Full Scale Digital". Occasional flashes are OK but if it is on all the time, turn something down!

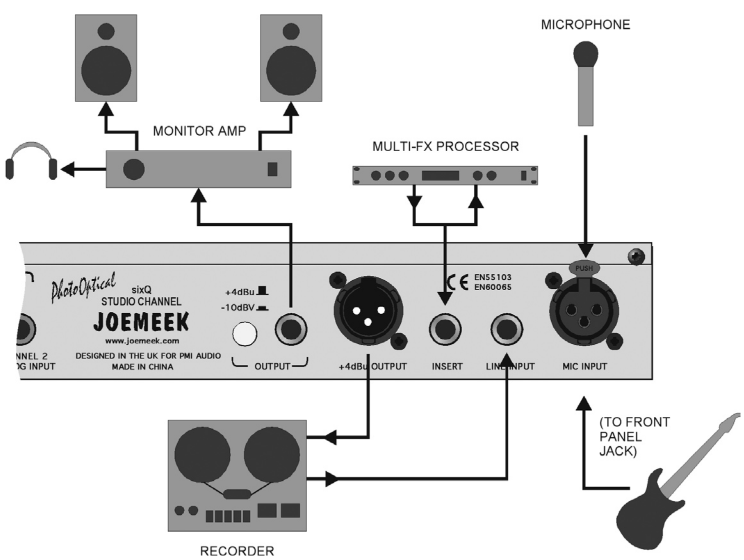
Note that the Digital Interface is a two-channel device, with channel one fed by the sixQ's internal circuitry. In order to utilize the second digital channel, an external analog input is provided in the form of a balanced 1/4" jack on the rear panel wired as follows:
Tip: + (hot)
Ring: - (cold)
Sleeve: ground

In this way the output of the sixQ and another analog source can be fed into one digital input of a recorder or digital workstation.

Using the sixQ

GETTING CONNECTED

The figure shows the sixQ being used instead of a mixing desk in a recording setup:



- A microphone is connected to the Mic Input
- A guitar is connected to the Instrument Input
- The insert point is being used to divert the preamplified signal through an external effects processor
- The recorder output is connected to the Line Input for playback. Previously recorded tracks may also be replayed via the Line Input, to permit compression and equalization

POWER SUPPLY
Connect the sixQ power cord to the AC connector on the rear panel and switch on the mains supply. NB: ensure that the sixQ is set to the correct mains voltage for your region – either 115V or 230V. Orientate the fuse holder / mains voltage selector draw so that the required voltage appears at the top. If in doubt consult a competent engineer.

Using the Preamp

Turn the 'INPUT GAIN' control to minimum and connect the input source. If you are using a condenser microphone, remember to switch on the 48V Phantom Power. Set the 'OUTPUT GAIN' to "0dB". Turn up the 'INPUT GAIN' until the microphone sound registers on the VU Meter, adjusting it so that the meter reads between "0" and "+3" on sound peaks. When the red LED (labeled "Peak") lights, the sixQ is within 6dB of clipping. Occasional flashes are OK but if it is on all the time, turn the Input Gain down!

Remember you can check the preamp gain at any time by pressing the "Meter Pre" button.

Using the Compressor

Start with the Compressor and Meequalizer off and adjust the input and output gain so that the VU Meter reads around 0dB.

Set ‘COMPRESSION’ and ‘ATTACK’ fully anti-clockwise, with ‘SLOPE’ and ‘RELEASE’ at mid-position. Press the ‘COMP’ push-button and turn up the ‘COMPRESSION’ control until the compressor GR meter starts to read 2dB or 4dB on audio peaks. You should now be able to hear the compressor working as the volume diminishes. Use the ‘MAKE UP GAIN’ control to restore the signal to its previous (uncompressed) level. Alter the SLOPE and listen to how the severity of the gain reduction changes. Try changing the Attack and listen for percussive sounds getting louder. Reducing the Attack and Release times should emphasise this even more and the compressor should start to “pump” audibly.

The overall result of compression depends on the combined settings of the Compress, Slope, Attack and Release controls. Experiment with different combinations to discover what best suits the material you wish to compress. Watch the GR meter and don’t overdo things - it’s possible to apply 20dB of gain reduction before you realise it!

Use the Compressor ‘ON’ switch to make comparisons between compressed and uncompressed signals.

Using the Meequalizer

Always start with the Meequalizer boost/cut controls (LF, MID and HF) set to “0” (the control knobs set vertically, in their centre notches). This setting is also known as “flat”.

You need to be careful about too much boost or “lift”, since boosting takes the sixQ closer to overload. The sixQ has generous overload margins but when a lot of boost is used, it may be necessary to compensate by reducing the Output Gain or the Input Gain (the latter will affect the Compressor setting though).

Keep an eye on the VU Meter when adjusting the EQ. When the red LED (labeled “PEAK FSD”) lights, the sixQ is within 6dB of clipping. Occasional flashes are OK but if it is on all the time, turn something down!

The way to use the LF FREQ and MID FREQ controls, is to apply quite a lot of boost, then sweep the frequency until you “tune in” to the sound you are interested in. Once you find it, adjust the amount of boost or cut to give the desired effect.

Experiment with combinations of settings of EQ and try to picture how the audio signal is being affected. Use the EQ ‘ON’ switch to make comparisons between EQ’d and non-EQ’d signals.

Using the Output Stage

Final adjustments to the output level can be made with the OUTPUT Gain control, again keeping an eye on the VU meter and PEAK FSD LED. Note how this control can also be used to fade out the signal completely.

Troubleshooting

1) No Power (no lights work)
- Is the power cord plugged in (both ends)?
- Is the mains power on?
- Is the mains voltage set correctly for your region?
- Has the mains fuse blown?

2) The microphone doesn’t work
- Is it connected to the correct (XLR) input on the rear panel?
- If it is a condenser microphone, is the phantom power switched on?
- Is the ‘Line’ switch out (LED off)?
- Is the ‘Input Gain’ control turned up?
- Is the ‘Output Gain’ control turned up?

3) The line input doesn’t work
- Is the source connected to the correct (jack) input on the back of the unit?
- Is the ‘Line’ switch in (LED on)?
- Is the ‘Input Gain’ control turned up?
- Is the ‘Output Gain’ control turned up?

4) The compressor doesn’t work
- Is the Compressor ‘ON’ switch in (LED on)?
- Is the ‘Compress’ control turned up enough?
- Is the ‘Slope’ control turned up enough?
- Is there enough signal, as set by the ‘Input Gain’ control, to drive the compressor?

5) Too little or too much compression
- Turn the ‘Input Gain’ control up or down respectively, to adjust the signal level to the compressor

6) The Meequalizer doesn’t work.
- Is the EQ ‘ON’ switch in (LED on)?
- Is ‘Input Gain’ control turned up?
- Is ‘Output Gain’ control turned up?

7) Too much noise
- Is the ‘Input Gain’ control too high? Try moving the mic closer to the source
- Is the ‘Output Gain’ control too high (eg: when lots of compression is being used)?
- Is there too much EQ boost?
- Is the noise already present in the input signal? (Try removing the input)

8) Sounds distorted
- Is the ‘Input Gain’ control too high?
- Is the ‘Output Gain’ control too high?
- Is there too much EQ boost?
- When using the compressor, is the Release control set too low?

Technical Specification

Input impedances	Mic: 1.2kohm; Line: 20kohm
Pre-amp overall gain	0dB to 60dB
Common mode rejection	70dB
Equivalent input noise	-128.5dBu (unweighted)
Distortion	0.001% (below Compressor threshold)
Frequency response	15Hz to 70kHz (-3dB)
Maximum input before clipping	Mic: +21dBu; Line: +45dBu
Headroom before clipping	+21dBu
High Pass Filter	12dB per octave cut below 80Hz
Compression threshold	-6dBu to +22dBu (variable)
Compression ratio	1:1 to 10:1 (variable)
Compressor attack time	1 msec to 100 msec (adaptive)
Compressor release time	0.1 sec to 3 sec (adaptive)
EQ Boost & Cut	+/-15dB (each band)
EQ “Q”	0.9 (1.6 octaves)
LF Frequency	40Hz to 650Hz variable
MID Frequency	300Hz to 5kHz variable
HF Frequency	6kHz/12kHz switchable
Nominal output levels	+4dBu/-10dBv
Output impedance	75ohm
Output Level switch	12dB attenuation
Noise Floor	-85dBu (typical, with ~40dB mic gain)
VU Meter	Analogue movement
Power supply	115V / 230V ac mains, 50/60Hz
Power consumption	30W
Mechanical	482W x 44H x 220D (overall)
Weight	2 kilos

Joemeek Limited Warranty

THIS PRODUCT IS FOR PROFESSIONAL USE ONLY

PMI Audio Group warrants that all products will be free from defects in material or workmanship:

A: For a period of (3) three years from the date of purchase (hereinafter the labor warranty period), PMI Audio Group will repair or replace this Product if determined to be defective. After the expiration of the labor warranty period, the Purchaser must pay labor charges.

B: In addition, PMI Audio Group will supply, at no charge, replacements for defective parts for a period of (three years) from the date of purchase. During the labor warranty period, to repair the Product, Purchaser must return the defective Product, freight prepaid, or deliver it to PMI Audio Group Service Center. The product to be repaired is to be returned in either its original carton or a similar package affording an equal degree of protection. PMI Audio Group will return the repaired Product freight prepaid to the Purchaser. PMI Audio Group is not obligated to provide Purchaser with a substitute unit during the warranty period or at any time.

Conditions

1. Notification of claims: Warranty Service: If Purchaser discovers that the Product has proven defective in material or workmanship, then written notice with an explanation of the claim shall be given promptly by Purchaser to PMI but all claims for warranty service must be made within the warranty period. If after investigation PMI determines that the reported problem was not covered by the warranty, Purchaser shall pay PMI for the cost of investigating the problem at its then prevailing time-and-materials rate. No repair or replacement by Purchaser of any Product or part thereof shall extend the warranty period as to the entire Product. The specific warranty on the repaired part only shall be in effect for a period of ninety (90) days following the repair or replacement of that part or the remaining period of the Product warranty, whichever is greater.

2. Exclusive Remedy: Acceptance: Purchaser’s exclusive remedy and PMI’s sole obligation is to supply (or pay for) all labor necessary to repair any product found to be defective within the warranty period and to supply, at no extra charge, new or rebuilt replacements for defective parts. If repair or replacement fails to remedy the defect, then and only in such an event, shall PMI exchange to Purchaser a new or reconditioned unit. Purchaser’s failure to make a claim as provided in paragraph 1 above or continued use of the product shall constitute an unqualified acceptance of such Product and a waiver by Purchaser of all claims thereto.

3. Exceptions to Limited warranty: PMI shall have no liability or obligation to Purchaser with respect to any Product subjected to abuse, improper use, negligence, accident, modification, failure of the end-user to follow the operating and maintenance procedures outlined in the users manual, attempted repair by non-qualified personnel, operation of the unit outside of the published environmental and electrical parameters, or if such products original identification (trademark, serial number) markings have been defaced, altered, or removed. PMI excludes from warranty coverage, Products sold AS IS and/or WITH ALL FAULTS and excludes used products which have not been sold by PMI to the Purchaser. PMI also excludes from warranty coverage consumables such as fuses and batteries, tubes, etc.

4. Proof of purchase: The dealer’s dated bill of sale must be retained as evidence or the date of purchase and to establish warranty eligibility

Disclaimer of Warranty

EXCEPT FOR THE FORGOING WARRANTIES, PMI HEREBY DISCLAIMS AND EXCLUDES ALL OTHER WARRANTIES, EXPRESS OR LIMITED, INCLUDING, BUT NOT LIMITED TO ANY/OR ALL IMPLIED WARRANTIES OF MERCHANT ABILITY, FITNESS FOR A PARTICULAR PURPOSE AND/OR ANY WARRANTY WITH REGARD TO ANY CLAIM OF INFRINGEMENT THAT MAY BE PROVED IN SECTION 2-312(3) OF THE UNIFORM COMMERCIAL CODE AND/OR IN ANY COMPARABLE STATE STATUE. PMI HEREBY DISCLAIMS ANY REPRESENTATIONS OR WARRANTY THAT THE PRODUCT IS COMPATIBLE WITH ANY COMBINATION OF NON-PMI AUDIO PRODUCTS PURCHASER MAY CHOOSE TO CONNECT TO THE PRODUCT.

Limitation of Liability

THE LIABILITY OF PMI, IF ANY, AND PURCHASER'S SOLE AND EXCLUSIVE REMEDY FOR DAMAGES FOR ANY CLAIM OF ANY KIND WHATSOEVER, REGARDLESS OF THE LEGAL THEORY AND WHETHER ARISING IN TORT OR CONTRACT, SHALL NOT BE GREATER THAN THE ACTUAL PURCHASE PRICE OF THE PRODUCT WITH RESPECT TO WHICH SUCH CLAIM IS MADE. IN NO EVENT SHALL PMI BE LIABLE TO PURCHASER FOR ANY SPECIAL, INDIRECT, INCIDENTAL, OR CONSEQUENTIAL DAMAGES OF ANY KIND INCLUDING, BUT NOT LIMITED TO, COMPENSATION, REIMBURSEMENT OR DAMAGES ON ACCOUNT OF THE LOSS OF PRESENT OR PROSPECTIVE PROFITS OR ANY OTHER REASON WHATSOEVER.

Owners Registration Card

TO BE COMPLETED AT TIME OF PURCHASE

Name ____________________________________

Date of Purchase ____________________________

Serial Number ______________________________

Dealer's Name ______________________________

RETAIN FOR YOUR RECORDS

PLEASE DISPATCH AND RETURN
YOUR REGISTRATION
TO JOEMEEK WITHIN 14 DAYS
OF PURCHASE

Specifications and model numbers are subject to change without notice

Product Registration Information
Please Fill in the Below Sections and Return

Name: ____________________________________

Address: __________________________________

City: ________________ State: ____________ Zip Code: __________

Telephone Number: ________________ email Address: ________________

Model Purchased: ________________ Date Purchased: ________________

Serial Number: ________________ Dealer: ________________

Comments: ________________________________

____

What magazines do you read to influence your buying decision: (please check all that apply)

❑ MIX ❑ Electronic Musician ❑ EQ ❑ Sound on Sound ❑ Pro Audio Review ❑ Recording ❑ Pro Sound News ❑ Audio MIDI