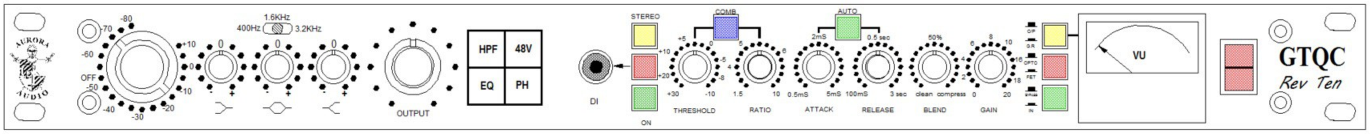
AURORA AUDIO

GTQC

Product Manual



AURORA AUDIO INTERNATIONAL
10358 Las Lunitas Avenue
Tujunga, CA 91042

EMAIL: Info@auroraaudio.net
Telephone: 323 428 9532
Web Site: http://www.auroraaudio.net

GTQC V10 Instruction Sheet

Introduction:

Well done for purchasing a hand-built, all discrete class A circuitry GTQC channel strip!
It is designed to give you years of superb sounds and service.

Unpacking Instructions:

Carefully remove the unit from the custom foam packing. For best noise performance always use a grounded 3 core / 3 pin a.c. cord. The power supply will automatically detect the voltage in your location.

If you connect the GTQC to an unbalanced source or destination, either use the preamps ¼" jacks or wire to the XLR pins 2 and 3 only. Leave pin 1 high. This will help prevent ground loops.

The rear jack socket behind the pre-amplifier is mono and should be connected with a mono jack. If a balanced TRS jack is plugged in, there will be a 6dB loss as there would be no signal on the ring contact.

The rear jack socket behind the compressor is also a mono jack. This is the side chain linking connector to enable the GTQC to link to other GTQC compressors and link their side chains together for multi-channel compression.
As this connector carries a dc voltage the cable connecting the two units together should be connected with the power off.

Wiring of this connector is tip = dc control volts, and sleeve = 0v (ground) common.
The voltage ranges from 0v (no compression) to 10v (maximum compression)
Either a mono or TRS cable can be used for the stereo link connection.

Operating Guide:

1. Preamplifier and Equalizer

The GTQC can accommodate any signal you care to throw at it, including high-level line inputs. The sensitivity switch provides gain adjustment from +80dB to –10dB and the level pot allows for fine adjustments. The level pot has an audio taper and is –20dB at half rotation. I would recommend using the level pot between ¾ and full rotation and never below ½ rotation unless part of a deliberate fade.
If you have to turn the pot below ½ way, the sensitivity switch must be adjusted at least 4 clicks to restore the level with reduced headroom!
Operating the unit with the level pot near maximum ensures that you keep the headroom in the designed 26dB region.

The D.I. input has 10 Megohm input impedance and around 10dB gain. It can be used (to great effect) with musical instrument pickups, but works equally well with high-level signals like a D.A.T. or CD player. The same gain structure rules apply.

The impedance switch on the rear panel selects either 300Ω or 1,200Ω input impedances. Most of the time you will find that the 1,200Ω input works best with dynamic and condenser microphones but very low impedance microphones (e.g. ribbon type) may work better with the 300Ω input. The 300Ω input provides 6dB additional gain if sourced from a low impedance, but if a higher impedance microphone is used (e.g. close to 300Ω), the series impedance will create a 6dB attenuator that negates the 6dB gain. The switch enables the user to experiment with which input impedance best matches the microphone.

Using analogue equipment in a digital world!

E.G. Analogue versus Digital levels

In my technical/design background in analogue circuitry, spanning over 30 years, the levels of audio were calibrated in dBm, a throwback from the telephone and communications era where 0dBm was 1mW dissipated into a 600 ohm load = 0.775 volts. 0dBm was later changed for the more convenient 0dBu which is a voltage into any specified impedance.

In a broadcast studio, Peak Program Meters were used that were calibrated from 2 to 7. Mark #4 equated to 0dBu and Mark #6 equated to +4dBu. The level +4dBu is 1.228 volts a.c. and also the 0VU reference point on a VU meter. This is, coincidentally, #6 on the PPM meter and a typical line up level for an analogue tape machine.

Most consoles and pre-amplifiers have a maximum output level before clipping of around 26dBu. This gives them 22dB headroom above 0VU = +4dBu. Driving the console and pre-amplifier "hotter" than +4dBu output reduces the headroom proportionately.

At the other end of the scale, the consoles/pre-amplifiers usually have +80dB gain and produce noise figures in the –45 to –48dBu region and an Equivalent Input Noise of -125 to -128dBu. The noise floor from a 200 ohm source at 20 degrees C is –129dBu so the amplifier is adding 1dB of noise to the absolute noise floor. As the gain is reduced, the difference between the signal and the noise floor widens as the noise is pushed further down.

Reminder:- *Increasing the gain amplifies the signal AND raises the noise floor.*
Running the device at hotter levels than usual also reduces the headroom.

In the digital world measurement criteria differ. Instead of using a reference level that relates to a particular power or voltage (like 0dBm) the 0dBf reference is the maximum signal that the analogue to digital converter can accept before the onset of clipping.

The 0dBf level is usually somewhere in the region of +18dBu to +24dBu in the analogue world…. It is **NOT** the same as 0VU (+4dBu) on an analogue VU meter.

It's very important to use an A to D input level that maximizes the headroom and minimizes the noise in the analogue world.

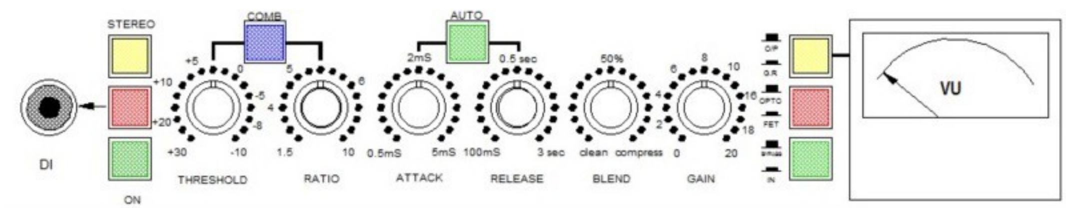
***E.G**. If an attempt was made to drive the console or preamplifier high enough to hit the 0dBf (+24dBu) reference level on the A to D, the amplifier would be running at over 20dB greater than it's normal operating level.*
This raises the noise floor by 20dB (ten times louder) and reduces the analogue headroom to around 2dB. A microphone normally needing 40dB gain would need 60dB gain and any peaks would drive both the pre-amplifier and the A to D into clipping. Not good!

Depending on the reference level recommendations of the A to D manufacturer, the analogue levels on its input should be typically around –18dBf. This will optimize both the signal to noise ratio and the headroom of the analogue signal. A degree of variance, say –16dBf, is acceptable but higher levels will begin to degrade the analogue performance with no improvement to the quality of the sound.

When using condenser microphones requiring +48v, be sure to turn the volume control down before pressing the 48v switch as applying this voltage may involve a switch on thump, especially if set to high gains.

2. Compressor Details

The compressor can either be used on its own or in conjunction with the preamplifier + equalizer



The drawing above shows the GTQC compressor section.

The six control potentiometers function as follows :-

***NB** There is one further small detent at either end of the pot rotation.*

Threshold (Grey knob) :- A 17 detented potentiometer with a total range of 40dB, i.e from -10dBu to +30dBu.

Ratio (Grey knob) :- A 17 detented potentiometer covering a range of ratios from 1.5:1 to 10:1.

Attack (Blue knob) :- A 17 detented potentiometer covering attack times from 500 microseconds to 5 milliseconds.

Release (Blue knob) :- A 17 detented potentiometer covering release times from 100 milliseconds to 3 seconds.

Blend (Grey Knob) :– A non-detented potentiometer that sweeps between no compression (ACW) to full compression (CW).

If the meter is switched to O/P then it will read the input level if the Blend control is set to no compression (ACW).
Gain (Red knob) :- A non-detented potentiometer covering gains from 0dB to +20dB. The silk screened dB incremented scale is accurate and allows for fine level adjustments.

The eight illuminated switches are as follows :-

NB. *The first item is with the switch off, the second with the switch on.*

Bypass/In (Green switch) :- With the switch deselected, there is a hard (relay switched) connection between the input and output XLRs. This also applies when the unit is switched off/powered down and means that it could be left in a circuit path without disturbing continuity.

When bypassed and the power is on, the VU meter and side chains are disabled to prevent confusing meter information as the audio circuits would be all powered up.

When the button is pressed in, the the hard bypass is released and the audio passes through the unit. The VU meter should indicate the output level where 0VU = +4dBu = 1.228v ac.

Side Chain Off/On (Green Switch) :- With the switch deselected the side chain is disconnected from the audio path so that no compression takes place and all the audio paths work at unity gain. As there are three alternative paths (described in detail later) this offers ample opportunity to use their different sounding configurations to “sweeten” or “modify” the sound passing through the compressor.

When the button is pressed in the side chain is activated and the degree of compression will depend on the relationship between the input signal level, the value of the threshold and ratio settings, and the mode of compression selected.

Opto/FET (Red Switch) :- The default control element is a photo-resistive opto coupler whose degree of attenuation is controlled by a d.c. voltage generated by the side chain. When the button is pressed, the opto path is switched out of circuit and replaced by one using a FET (Field Effect Transistor) as the control element. Each element has a particular sound footprint especially when compressing hard.

Output Level/Gain reduction (Yellow Switch) :- The VU meter default mode (switch deselected) is the output level from the compressor. If the side chain is switched off and the gain control set to 0dB this reading is also the input level to the compressor.

When the switch is pressed, the VU meter needle will rise to the 0VU point and will then move backwards, towards the -20VU section of the dial, to indicate the dB's of gain reduction being applied. Any electrical meter is always more accurate at the full scale deflection end of the scale so small mechanical and electrical discrepancies (around 2dB) may apply in the -20dB region. (i.e ±10% accuracy)

Auto Release Off/On (Green Switch) :- If the switch is deselected the attack and release times are set by the two blue potentiometers. The user should always experiment with the settings of these controls to see how they affect the sound.

When the Auto button is pressed the release time is dependent on the nature of the program material passing through the compressor. Generally the release times are longer but, if the material is short bursts of sound, the auto timing capacitor will have less stored energy and will produce shorter release times. If the material is long, flowing, rifts then the capacitor will charge up more and produce longer release times. The actual timing value changing constantly with the nature of the material passing through the compressor.

Combined Off/On (Blue Switch) :- When deselected the control element is either the opto or FET path depending on whether the red FET switch is selected.

If the Combined switch is pressed, relays reconfigure the internal circuit paths so that both control elements are in circuit. The signal passes first through the opto path and is then applied to the FET path for additional control. As both paths are unity gain before compression, there will be little difference in compression until both sections start working hard and then it will be seen to compress as much as another 6dB. (i.e double the single control element value).

The Combined mode's main function is not just to squash the living daylights out of a signal but rather to provide the user with a third option of sound signature. When the Combined switch is used, try having the Threshold and Ratio controls at minimum and then gently rotating the Threshold until the gain reduction meter indicates that the side chain is beginning to work. Listen to the color of the sound at this point and the effect on that sound by adjusting the other controls. It's a very useful feature!

Stereo Link Off/On (Yellow Switch) :- (Assumes that the GTQC is linked to another GTQC) .... If not selected the compressor channel works completely independently and can be assigned to different circuit paths with different degrees of compression.

If the Stereo button is pressed the two GTQC compressor sections will work together as a stereo unit (i.e. as a stereo bus compressor) with whichever channel has the most side chain voltage (i.e. the most compression at that instant in time) taking control over both channels and providing identical amounts of compression.

The control will pass seamlessly from one channel to the next depending on which of the two is working hardest and, as a consequence, the stereo image should not drift from center.

NB. For the stereo link to function properly, both compressors should have the same control element selected (i.e opto/FET/combined) and the bypass and sidechains should both be switched to the “on” position.

← **Switch Source to Preamp Output (Red Switch)** :- The compressor normally works independently from the microphone amplifier/equalizer but, if the red switch is pressed, the compressor's input is sourced from the pre-amplifier output creating a complete channel strip function. The pre-amplifier output XLR is unaffected and can be used as a pre-compressed output.

5. Compressor Specifications

Input Impedance (Compressor) = 10Kohm balanced and floating, transformer coupled.

Input Impedance (Mic Pre) = XLR Input = 1200 ohms.
Transformer balanced and floating

D.I Input = 10Mohm into a “super-transistor” class A amplifier.
Unbalanced mono jack socket input disables the rear XLR

Output Impedance = <75ohm balanced and floating, transformer coupled

Output drive capability = +26dBu into any load of 600 ohms and above

Frequency response = 20Hz to 20KHz ± 1dB

Total Harmonic Distortion = < 0.1% @ +20dBu O/P @ 1KHz (Typically 0.025%)

Noise (Compressor) = < -75dB filtered 20Hz to 20KHz

Equivalent input noise: (Preamp)	At +80dB gain < -125dB (typical -127dB) (Input terminated 200 ohms, measured 22Hz to 22KHz
Side chain voltage =	Typically between 0v minimum and +10v maximum (Greater d.c. voltages provide greater compression.)

6. Tips for users :-

Ah, what to expect here?!!! I'm only the designer and feel sure sure that, with minimal practice, you will be able to email me with tips on good settings for particular instruments or vocals! I look on the unit as a tool with which I have provided more than adequate variations for you to sculpt the sound you desire!
As a very general guide/starting point, I suggest the following :-

Threshold = minimum (fully counter clockwise)

Ratio = minimum (fully counter clockwise)

Attack = maximum (fully clockwise)

Release = minimum (fully counter clockwise)

Balance = maximum (fully clockwise)

Gain = 0dB (fully counter clockwise)

Bypass = pressed in

VU = selected to VU (not pressed)

Sidechain = selected to OFF (not pressed)

Source Select = Selected (pressed)

Preamplifier Optimal Gain :-

With the preamp output potentiometer set to maximum, use whatever source material you have and slowly rotate the red gain knob until the VU meter indicates 0VU. Leave the gain settings there as this is the optimal level for best noise and headroom.

Compressor Optimal Theshold :-

VU = selected to Gain Reduction (pressed)

Sidechain = selected to ON (pressed)

Then, using the gain setting previously established, gently rotate the Threshold control clockwise until you see the VU meter just start to indicate gain reduction. Leave the Threshold control there and adjust the Ratio, Attack and Release controls for the sound and compression you want. There's lots of scope for experimentation!

As the degree of compression increases and the signal gets smaller, use the gain control to restore the level back to the 0VU region (with the meter set to output level).

The Version 10 now includes a pan control to sweep between no compression to full compression.
This provides parallel compression with a single control knob.

There will also be a difference in output level if the difference between non-compressed and fully compressed signal levels is wide due to heavy compression. So turning towards non-compressed you would expect an increase in level that could be compensated for with the gain control or adjusting the compress ratio.

GTQC Problem solving page

The output sounds distorted and noisy!

It may be that you have too much input gain and/or the output level control is turned back excessively.

To check that you have set up for the correct gain structure, apply the input signal to the input of the pre-amplifier. Press the red “←” button on the compressor and also the green “bypass/in” button. Make sure all the other compressor buttons are off.

Take your audio output from the compressor output XLR. Now, with the pre-amplifier level control at maximum, rotate the pre-amplifier red gain knob clockwise until your input signal sends the VU into the red region past 0VU. It should just be “kissing” this region of the scale.

You now have the gain set correctly for the source you are using.

I don’t get any output from the unit!

There are several things to check if you have no output from the unit…

If you are using the compressor and the pre-amplifier, make sure the red “←” button is pressed. Also, make sure that the output level control for the pre-amplifier is at maximum.

Finally, the impedance switch on the rear of the pre-amplifier is designed to switch between 300Ω and 1200Ω input impedance. If this switch has been accidentally been moved to the center of its travel then the microphone input XLR will be disconnected from the pre-amplifier.

I get bad hum when using the DI input!

This is because you are either getting a ground loop with the device you are connected to or, if a guitar, the connecting cable is either not inserted properly at both ends or the pick-ups are getting interference from a strong electro-magnetic field… like lamp dimmer pulses in the building wiring.

If the issue is a ground loop then make up a cable to connect the device to the microphone input XLR. Connect the inner (signal) core to pin 2, and connect the shield to pin 3. Do not connect anything to pin 1.

Now, if the impedance switch is selected to 1200**Ω**, then the transformer-coupled input will isolate the GTQ2 ground from the ground (or rather, lack of ground!) of the device you are amplifying.

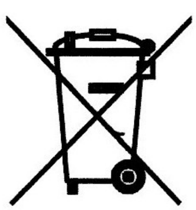
There is also a possibility of a ground loop when using the pre-amplifier unbalanced output. Should this occur... like if you connect it to an unbalanced guitar amplifier... then use a DI box with a transformer input to isolate the ground of the GTQ2 from the grounding issues of the guitar amplifier.

Warranty: ONE YEAR PARTS AND LABOR LIMITED WARRANTY

Aurora Audio International warrants this GTQC unit against defects in workmanship for a period of one year and parts for a period of one year from receipt by the original end user. This warranty shall not apply to damage resulting from misuse including water damage, in-transit damage, fire damage, improper maintenance, dropping the unit and operation or storage outside the environmental specification for the product.

Do not try to repair this GTQC. Only qualified Aurora Audio International technicians are authorized to repair this unit. WARRANTY VOID IF CASE IS OPENED

ROHS Directives



The RoHS Directive stands for "the restriction of the use of certain hazardous substances in electrical and electronic equipment". This Directive bans the placing on the EU market of new electrical and electronic equipment containing more than agreed levels of lead, cadmium, mercury, hexavalent chromium, polybrominated biphenyl (PBB) and polybrominated diphenyl ether (PBDE) flame retardants.

The restrictions took effect in the E.U from 1st July 2006.

It is very important that the owner of any piece of equipment that contains even microscopic amounts of the listed hazardous substances (in relation to the weight of the unit) realize that the responsibility of its disposal rests with them. The unit should not just be thrown away at the end of its lifetime, whether that's 10, 20 or 30 years hence.

Please contact us at the address below and we will provide you with the necessary information for proper disposal.

Aurora Audio, 10358 Las Lunitas Avenue, Tujunga CA 91042, USA
Phone +1 323 428 9532
Email info@auroraaudio.net **Web URL** www.auroraaudio.net