QUICK USER GUIDE

EQ V.8

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EQ V.8
12 Band Equalizer
Compressor - Limiter - Echo - Delay - Roger
with Noise Reduction function.

(here in after referred to as "device").

Specification

- ADC/DAC resolution 24 Bits
- Audio bandwidth:
  - 5.2kHz (Normal Mode)
  - 3.2kHz (DX Mode)
  - 8kHz (Wide for CB)
  - 8kHz (Bypass)
- Mains hum suppression (50Hz & 60Hz) - >30dB
- Input: switchable with the ability to turn on bias voltage for the microphone
- Microphone amplifier: two independent, one with a balanced input, the second with an unbalanced input. Adjustable gain 0 + 40dB
- An additional option is the Integrated + 48V phantom power supply for the Condenser Microphone (maybe in the range of + 35 + 48V, does not affect the operation of the Condenser Microphone)
- Microphone available: Electret, Dynamic, Condenser
- Adjustable noise Gate can be disabled. Threshold adjustment range: -70 to -30dB
- Background noise suppression efficiency: up to 20dB
- Compressor: Ratio adjustable 1:1 - 10:1
- Narrow band compressor:
  - Attack - 1ms - 44ms (depending on frequency range);
  - Release - 5ms - 219ms (depending on frequency range);
- Wideband compressor:
  - Attack - 5ms;
  - Release - 50ms
- Soft-saturation limiter (will be described below)
- Up to 200mV output level
- Up to 2 x 30mW(32 Ω load) "MON" output power
- 12-band graphic equalizer with center frequencies:
  - *90/150/220/300/430/600/850/1200/1700/2400/3300/4700 Hz*
- Equalizer level adjustment range ± 15db
- Switchable processor Echo effects with multipath emulation:
  - Adjustable echo level 0 to -20db, 1dB step
  - Duration: 5 - 100%, 5% step
- Roger function (end of transmission tone marker):
  - Adjustable sine wave tone frequency
  - Duration : 150ms
- Real time output spectrum monitor:
  - Dynamic range: 50dB

Fast Fourier transform points: 256

- ◊ 4 operating modes:
  Bypass/Normal/DX/Wide
- ◊ 4 presets with independent parameter set
- ◊ PTT function
- ◊ Accepted for Up / Down functions (not for all transceivers)
- ◊ Indication of Input signal:
  level/compressor level (3dB step)
- ◊ Microphone level meter with 40 dB dynamic range
- ◊ Compression level meter with 20dB dynamic range
- ◊ Input signal asymmetry meter (waveform unbalance) with +/-10dB range
- ◊ Multi-stage phase shifter (to eliminate waveform umbalance):
  Stages: 0 - 32;
  Frequency: 50 - 1000 Hz.
  +10dB boost feature for low sensitive microphones
- ◊ Control - touch and with the help of rotation / pressing the encoder.
- ◊ Supply voltage - 4.5-5.5V
- ◊ Maximum consumption current - 500mA
- ◊ Maximum dimensions of the case - 170 x 90 x 80 mm (without cable and connectors)
- ◊ Weight - 0.63kg
- ◊ User interface based on uGUI library.

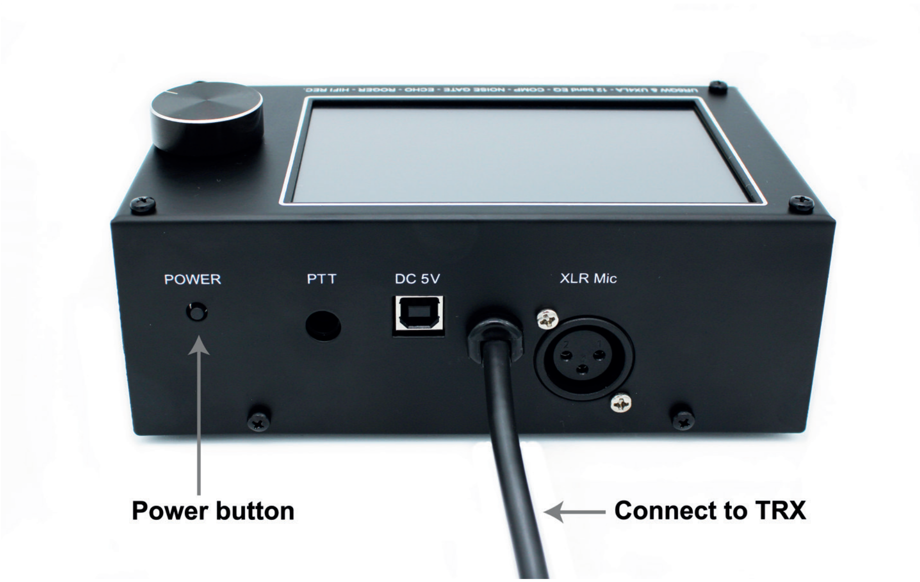


Figure 1

A brief description interconnections

The transceiver usually has a classic equalizer that is not able to equalizer adjacent frequencies in a narrow range. As a result, if a dynamic microphone is connected directly to the transceiver, the signal may have a dull tone at low frequencies and may whistle at high frequencies. Any expensive studio microphone can't sound any better without audio processing than a cheap one with audio processing. The V.8 has with a host of audio processing blocks to make your signal stand out! Of course, having complex signal processing blocks, you need to study a lot of information in order to clearly understand how it works and how to set it up correctly. In the V.8 device, there is no need to configure a bunch of blocks, we did it for you, leaving only a part available to the user, where you can change the settings.

The device has two independent low-noise microphone amplifiers with two independent inputs for various types of microphones. This is a universal device, any microphone can be connected to it:

- Elecret
- Dynamic
- Condencer (48V PS)
- Hand mic

The first amplifier has an unbalanced input, this is a 3.5mm jack on the front of the unit, signed “MIC”. You can connect an Electret or Dynamic Microphone to it. The Bias + 3V power supply can be disabled via the menu (the procedure will be described below).

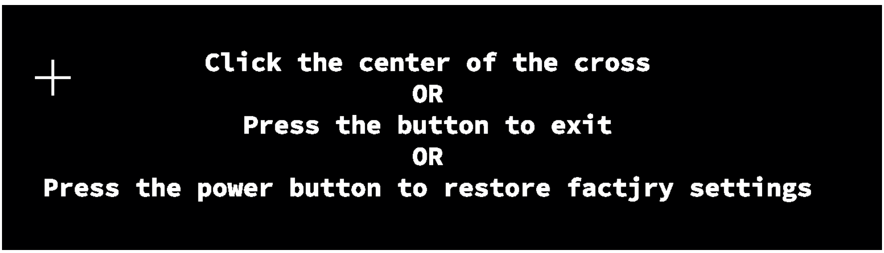
The second amplifier has a balanced input, this is an XLR socket on the back of the unit, signed “XLR Mic”. It is intended for connecting high-quality studio microphones, Dynamic or Condenser.

To the left of the “MIC” jack is the “MON” jack, with the same 3.5mm jack. This is an audio output to which you can connect headphones and check your signal generated by EQ V.8.

Attention!!! This line does not transmit audio from the transceiver! Please note that on the air your signal may sound different from the “MON” output, depending on the filters and transceiver settings!

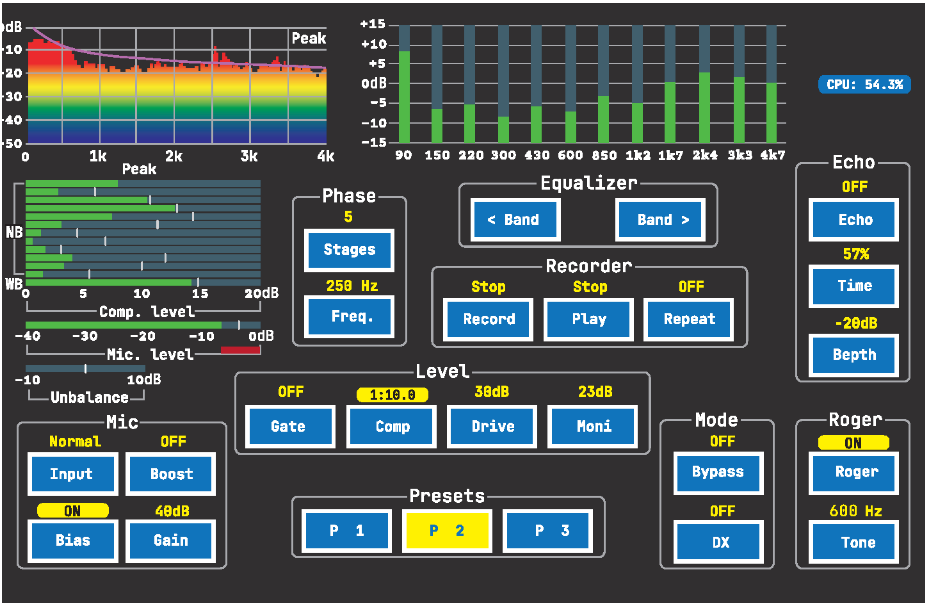
Calibralion and factory setting

Click and hold encoder knob 4 second. After this, the following window will appear:



a) to calibrate - press the center of the cross
b) to reset all settings to factory default - press the power button
c) to exit the current menu without changes - press the encoder button

Display menu



The first time you turn on the device. Microphone connection

Connect the main cable of the device to the transceiver in figure 1 marked “Connect to TRX”. Connect a Microphone or hand mic to the device. If necessary, connect the pedal (6.3mm jack on the rear panel).

Disable in transceiver standard Compressor and Equalizer!!!

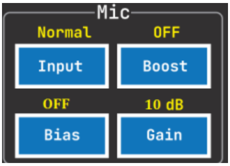
Connect the USB cable to the device and to a 5V voltage source. The device is connected with a USB cable to any 5V source (computer, USB power supply, Power Bank, adapter 12> 5V, etc.).

Press and hold 3s **Power button** to turn on the device.

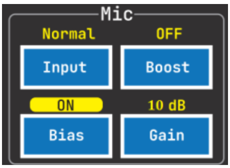
Press Presets > P1: 

You can choose any of the three presets P1, P2, P3. Presets have independent settings. Any parameter can be changed, so you can use several microphones connected to different inputs at the same time, for example, the Dynamic Heil PR-781 is connected to the XLR input for Preset P2 and the standard Hand mic is connected to the 8pin input for Preset P3. In just a second, you can switch between microphones. Note that the 3.5mm input jack is physically connected in parallel with the 8pin jack. Therefore, do not connect two microphones to the 3.5mm and 8pin input at the same time (although this is not prohibited).

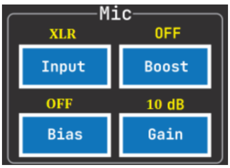
Select correct MIC setup (only then connect the microphone):



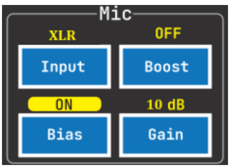
Unbalanced input for connecting a Dynamic microphone jack - 3.5mm, hand mic - 8 (6/5/4) pin connector.



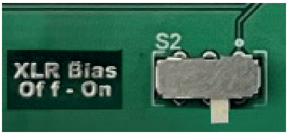
Unbalanced input for connecting an Electret microphone - 3.5mm jack, hand mic - 8 (6/5/4 ..) pin. Attention - in this mode, there is + 3V voltage on the 3.5mm jack.



Balanced input for connecting a Dynamic Microphone, jack XLR.



Balanced input for connecting a Condenser microphone, jack - XLR. Attention in this mode, the + 48V voltage on the XLR input jack.



If you have not Bias knob, when you select XLR input, you must set S2 to ON position (it is located on the main board inside the device).

If you use Dynamic mic connected to XLR, please set S2 to OFF position!

Attention!!! Never connect a Dynamic Microphone to the XLR input, if you have Mic XLR Bias turned on, it can kill your microphone! Never plug or unplug the Condenser Microphone while the device is on, first turn off the power of the device !!!

Level setting

Gate - adjustment of the squelch threshold. Where 30db is the maximum limit of even very loud sounds, 70db is min. restriction, **OFF** - Gate is disabled and does not work.

Gate needs to be configured at the very end of your experiments. Do not speak into the microphone, starting to set the Gate below 70db on a scale, the noise will begin to decrease, achieve a sufficient reduction in environmental noise. At the same time, some letters should not be cut off in the conversation. If this happens, raise the Gate higher.

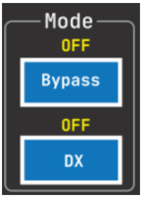
Comp - adjusts the compression level.

Where **1:1.0** - no compression. **1:1.10** maximum compression.

Drive - adjusts the level of the output signal that is supplied to the transceiver. Drive settings must be made by speaking into the microphone and controlling the

ALC level in the transceiver.

Moni - volume level control for headphone output, front panel 3.5mm “Mon” jack. This output is only needed for rough pre-setting of the device. Use it if your transceiver does not have monitor function.



Mode section

Bypass ON - the signal from the microphone, without pre-processing, goes to the output of the device.

Bypass OFF - the signal from the microphone passes through all enabled processing blocks and goes to the output of the device.

DX OFF – standard mode for normal signal, ESSB.

DX ON - the band is narrowed, pre-emphasis is introduced, the compressor operation is changed to a harder one - ideal for working in noisy conditions with DX-stations or to stand out in a pile-up.

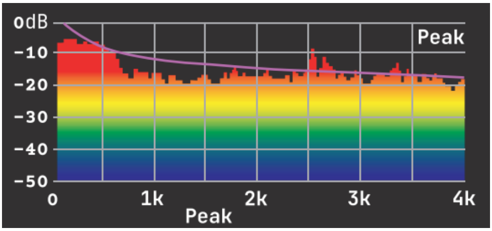
Roger sections

The signal of the end of transmission with a given frequency. It will work when the pedal or hand mic is connected to the device.

Attention!!! If you use TX switching on the transceiver side, the Roger function will not work.

Click on the **Roger** button to turn it ON or OFF

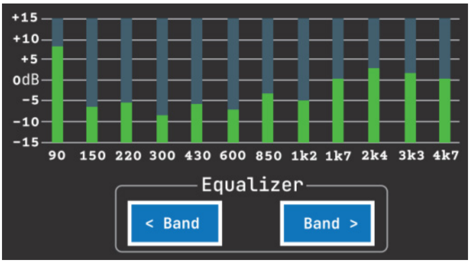
Click on the **Tone** button and set with Encoder required frequency



Amplitude-frequency response

On the graph you can see the frequency response of your signal in the frequency range 0-4 kHz, in real time. Perhaps in new firmware a line will be added that has the frequency response of pink noise. It's very simple, if you don't know or can't set up the equalizer, then just set it up so that your signal has a pink noise response and it will be perfect.

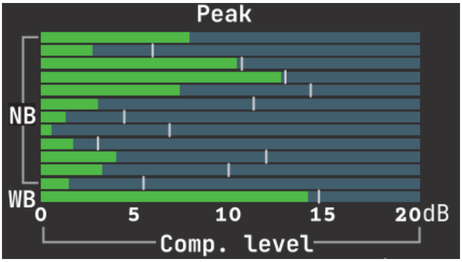
Also on the graph you can see the main peaks in the signal, which dominate and can make the signal difficult to read. And this will help you set the equalizer correctly and have sound like a studio radio.



12-band graphic equalizer

EQ The equalizer has the ability to adjust the signal amplitude within ±15dB. This makes it possible to get a good result even with a complex signal, which has a very uneven frequency response.

How to setup. Click to knobs **Band** or click to graphic, the active indicator will be red. Rotate the encoder to change the value. To select the required frequency, press the **Band** button. If you need reset equalizer to flat responce, click on the graph and hold for 5 seconds.



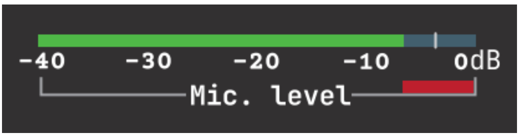
Compressor Level indicator

The device has a two compressors connected in series.
First is multi-band compressor (NB).
Second is single compressor (WB).

The signal is split into 12 bands and each band is processed by a separate compressor, which operates only in its own frequency range. This flattens out the frequency response much better than with the single-band compressor which applies in most transceivers. For example, if your voice has an excess of low frequencies in the 100-300 Hz range, then this will not affect the compression in the 300-4700 Hz frequency range and, as a result, the signal will be clearer and more transparent.

NB – Narrow Band Compressor indication. Shows the compression level for each of the 12 bands. The highest line is 90 Hz, slightly below 150Hz etc.

WB – Wide band Compressor indication. This is the second compressor that works together with the external 12-band compressor and is turned on after the equalizer. So that even if you adjust the equalizer incorrectly, your signal will not have distortions and overloads. When you change the Comp Level, the parameters of two compressors change synchronously.

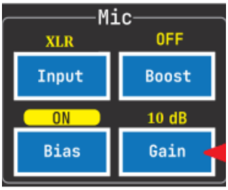


Microphone level

-40 -30 is weak level	-30 -10 is normal level	-10 – 0 is excess level

To change the Mic level, press the **Gain** button and rotate the Encoder.

In this case, you need to speak into the microphone, in the same way as you would do when working on air.

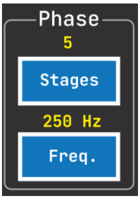


If your microphone has low sensitivity and even at a Gain level set of 40dB and Mic Level is not enough, turn on an additional 10dB amplifier by pressing the **Boost** button.

But having a total gain of 50dB at the microphone input is a bad solution, the noise level will be excessive, the input will be sensitive to RFI and the requirements for your APS and shielding will increase. Therefore, the best solution would be to use a microphone with a sensitivity of about 55 db or speak louder.

Signal symmetry indicator

You need to strive for the green pointer to be as close to the center as possible. Everything affects him, even intonation and you speak into the microphone

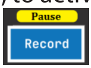


Phase rotation

Allows you to make the signal more symmetrical. It is recommended to enable this block only when all others are configured. Balance changes can be viewed using the Unbalance indicator described above.

Recording a voice TX memory.

You can record voice from microphone with 16s lasting, with very good quality. During recording, the signal goes directly to the recorder, bypassing all processing blocks (only mic amp works). And during playback, the signal goes through all processing blocks.

Click the **Record** once, to activate it. the recorder will be in standby mode and pause will be highlighted — 

Press the button again to start recording.

Press the button again to stop recording.

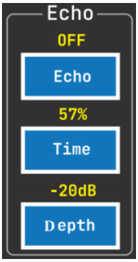
Press 

to start the playback. Playback is automatically terminated when all of the recorded content in the memory is played back.

Press 

and rotate encoder. If you select a value of **1s** or more, the recording will be played back in a loop. The cycle can be stopped by pressing a PTT or button Play/Record.

Where **1s** is standby time (in RX mode) 

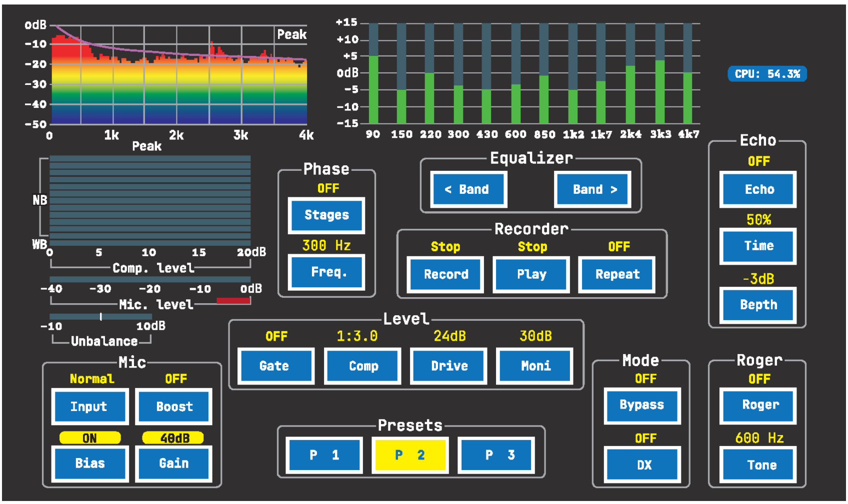


Echo processor effects with multipath emulation

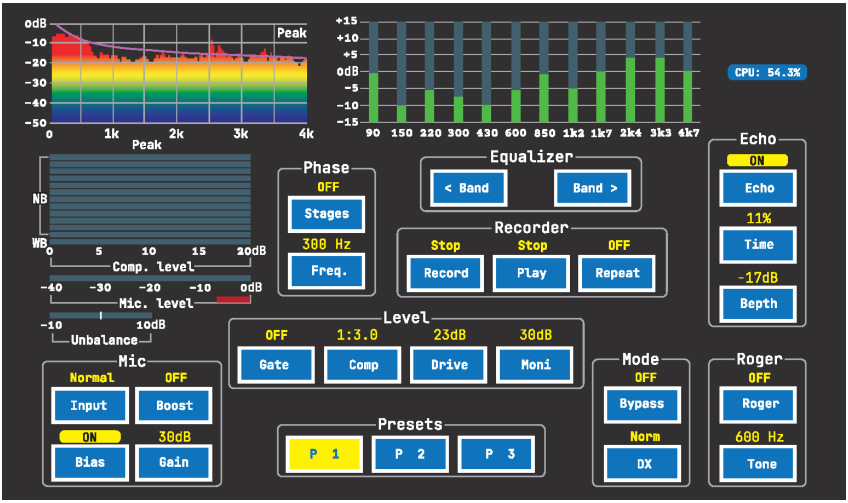
Intuitively easy to set up and does not require detailed description. To turn it on/off you need to press the **Echo** button.

After finishing the settings, click on the Power button and save the parameters.

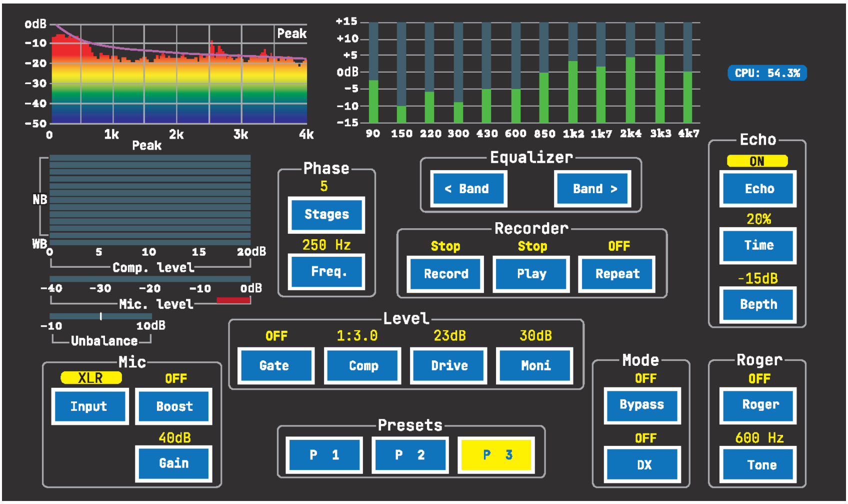
Some practical setup



Icom hand mic **HM-219** EQ/Level/Mic setup



Classic electret Mic without case. EQ/Level/Mic setup



Studio dynamic mic **Heil PR-40** EQ/Level/Mic setup

Attention! These settings are highly dependent on the type of microphone and your voice. They are given to make it easier for the user to first Equalizer setup. It's also not necessary to turn on Echo.

UR6QW stores

Ukraine

DB6QW stores

Germany

User support: **ur6qw.ua@gmail.com**

EBAY stores

hfvhfparts

XADO Store

Ebay distributor: **ux4la79@gmail.com**