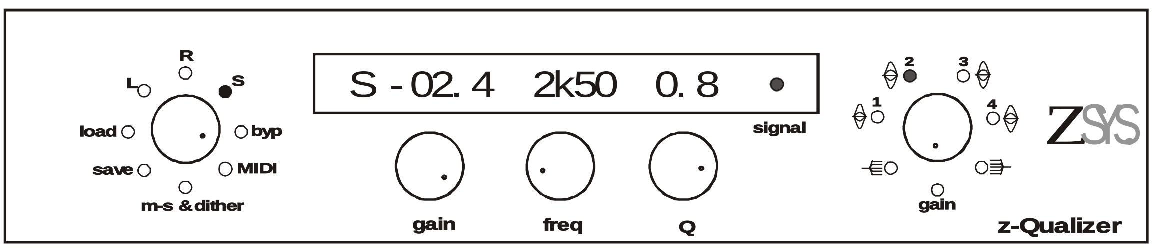
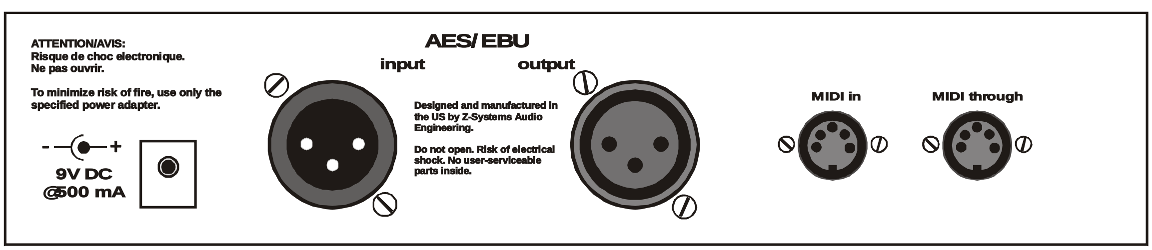
z-Qualizer Operations Manual

Welcome!

Congratulations on purchasing the z-Qualizer, the first low-cost outboard digital EQ capable of world-class mastering performance. The z-Qualizer comes from a distinguished lineage of mastering processors that Z-Systems has been making for nearly a decade. You've already heard our EQ algorithms on countless recordings and now you have the chance to hear what they can do for your very own projects.

We urge you to read the following manual before using the z-Qualizer. We're confident that anyone can learn to use it on their own, but it might be helpful to read the manual and learn some of the finer points before diving in.





Powering Up

Plug the DC power adapter that was included with the unit into an AC power outlet and connect the other end of the DC power adapter to the DC power inlet on the back of the unit. The z-Qualizer will power up quickly. Next, connect the z-Qualizer to an AES/EBU standard digital audio source. The **signal** LED will light to indicate the presence of a valid incoming signal.

Gain and Channel Offsets

We should mention a general rule at this point. The z-Qualizer is a stereo unit and it can be operated in either dual-mono or stereo-linked modes. Dual-mono means that the EQ parameters can be adjusted independently for the left and right channels. In stereo-linked mode, the left- and right-channel EQ parameters track one another. The **L** and **R** positions on the left knob are used to select between the left- and right-channel parameter banks, whereas the **S** position puts the unit into stereo-linked mode. The left-most edge of the alphanumeric display will always show whether the unit is in **L**, **R**, or **S** mode.

When the unit powers up, the left knob will be set to **S**, for "stereo" and the right knob will be set to **gain** as indicated by the respective LEDs surrounding the left and right knobs. In this mode, the only parameters showing in the alphanumeric display will be an **S** (indicating the z-Qualizer is stereo-linked) and a number above the **gain** knob indicating, in dBFS, the amount of gain (or attenuation) applied simultaneously to the left and right channels:

S   +00.0

Turning the **gain** knob adjusts the stereo gain. To apply different gains to the left and right channels, rotate the left knob to **L**, for "left channel." Observe that the alpha-numeric display changes to show an **L** indicating that the z-Qualizer is now in dual-mono mode, with the left-channel parameters currently active. Rotating the **gain** knob changes the left channel gain. The procedure is similar for the right channel. If you turn the left knob back to **S**, you'll see that the stereo gain is still the same as before. The z-Qualizer is designed so that the left, right, and stereo gains can be different. This allows you to dial in an offset between the left and right gains and then maintain that offset by manipulating only the stereo gain.

Equalization

The equalization in the z-Qualizer is performed with floating-point arithmetic and uses very special proprietary low-distortion algorithms developed. There are six bands, including two first-order (6 dB per octave) shelves and four parametrics (bell curves).

<u>Stereo-linked mode:</u>

Make sure the left knob is in the **S** position and turn the right knob to the **low shelf** position (one click counter-clockwise from the **gain** position). The **low shelf** LED will light and the alphanumeric display should look like:

S   +00.0   1K00

Rotate the **gain** knob to change the gain of the low frequency shelving equalizer. Rotate the **freq** knob to change the frequency *below* which the shelving action takes place. This frequency is at the nominal 3 dB point of the curve. The first-order equalizer is extremely gentle and quite natural-sounding.

<u>Dual-mono mode:</u>

Rotate the left knob to put the z-Qualizer in left-channel mode. The alphanumeric display now reads:

L +00.0 1K00

Here you can display and adjust the left channel shelf independently. Similarly, turn the left knob to the **R** position and adjust the right channel. If you then return to stereo-linked mode, the **L** and **R** filters retain their settings until you change a parameter in **S** mode. For example, if the **L** and **R** gains are different for the low shelving filter, if you return to stereo-linked mode and adjust the gain, it gets copied to both the **L** and **R** filters.

Important: You will notice that the “L” indicator in the display is blinking. This is to remind you that the z-Qualizer is in dual-mono mode. The same is true for the “R” when you are adjusting the right-channel parameters.

<u>The other bands:</u>

The **high-shelf filter** is accessed by turning the right knob one click clockwise from the **gain** position, and performs exactly as above for stereo-linked or dual-mono equalization. The **freq** knob controls the frequency *above* which action occurs.

The four **bell-shaped parametrics** are numbered as **1**, **2**, **3**, and **4** on the right knob. Turn the right knob to any of these positions and you will see:

S -00.0 1K00 0.4

This indicates, in order, stereo-linked or dual-mono mode, boost/cut, center frequency, and quality factor, or Q. As always, the label is above its corresponding knob, and as above, turning the left knob to **L** and **R** will allow you to control the left or right channel separately. Q is the inverse of bandwidth. It is the product of center frequency divided by the 3 dB down bandwidth. Thus, a Q of 0.4 produces an extremely wide curve and will be rarely used. A Q of 0.6 or 0.7 corresponds with the bandwidth of a typical midrange EQ in an analog equalizer.

M/S & Dither

<u>Dither</u>

Turn the left knob to **m-s & dither**, The alphanumeric display will appear as follows:

24 un ENC:N DEC:N

As in all z-Qualizer menus, the labels appear above the knob which affects the parameter. For example, in this menu, you can change the wordlength and dither by rotating the left knob to any of the following:

24 un
24 di
20 un
20 di
16 un
16 di
16 p2
16 p3

The last two **POW-r dither** options are only available at 44.1 or 48 kHz. The sample rate cannot be altered as the system is always slaved to the incoming rate.

<u>Using the dither and wordlength settings</u>

When the z-Qualizer equalizer is feeding additional digital processors, *nearly always set the output wordlength to 24 dith*. This is the maximum wordlength available in AES/EBU and will send the highest resolution signal to the following device. The dither in the z-Qualizer is a carefully-randomized floating point dither which removes quantization distortion. The dither noise level at the 24 dith setting is at approximately -141 dB below full scale, so we doubt you will consider this to be an audible problem! (To put this in perspective, most people consider analog tape hiss below about -80 dB to be inaudible, and the noise floor of the best analog audio console in the world is around -90 to -100 dB below full scale. Microphone preamplifiers can do a bit better, but in the real world, noises add, and the practical full band noise of the quietest typical musical recording is rarely better than about -70 dBFS, excluding fadeouts).

However, the ear can easily hear signal as much as 20 dB or better below the wide-band noise level, depending on the frequency and masking. That's why we have to use dither noise: to eliminate low level distortion caused by DSP processing. Without dither noise, the sound of quantization distortion can reduce stereo imaging, and make the sound cold and harsh, something to be avoided in most cases. The **undithered** menu choices actually truncate the output of the z-Qualizer without concern for quantization distortion.

The other dithered wordlength settings are only to be used if the z-Qualizer is connected **directly** to a following device which truncates the wordlength. For example, use **20 dith** if the z-Qualizer is directly connected to a 20 bit ADAT. Use **16 dith** or **16 pwr** if the z-Qualizer is connected directly to a 16-bit storage medium such as DAT or CDR. A router may be used, but no additional DSP processor should be between the Z Sys and the 16 or 20 bit device.

The only exception to the dither rule: you might choose to select the **24 undithered** option if you wish to bypass the z-Qualizer without turning the left knob to the **byp** position (in an automated session, for example, where one tune passes through flat without any alteration). In this case, select **24 un** and make sure that all the gains and equalizer levels are set to 00.0 dB. In the **24 un** setting, with everything neutralized, the z-Qualizer is bit-transparent, (*i.e.*, it will pass all incoming AES/EBU signals without data alteration). In the **20 un** (**16 un**) setting, the z-Qualizer truncates its output to 20 bits (16 bits). These last two settings will rarely be used. Perhaps you wish to cause some serious digital grittiness (assuming you are fond of the authentic sound of early digital audio), or you need to truncate the signal because a preceding device has not done so. Proceed with care when choosing any of the truncation settings; their use is extremely rare.

How to use the POW-r Dither

POW-r dither is a psychoacoustically-optimized dither. The two choices in the z-Qualizer are **POW-r 2** and **POW-r 3**, at 16 bits only, idealized for CD and DAT recording. **POW-r 3** has the strongest noise-shaping, and yields the highest resolution. We can't hear the effect of this shaping, but if you feel the shaping of **POW-r 3** is too strong for you, then by all means choose **POW-r 2**. **POW-r 2** may yield slightly less depth, space and resolution, but the loss is barely perceptible, and much smaller than with competing noise-shaping processes.

MS encode and decode

This versatile feature can be used in several ways. Let's return again to the alphanumeric display:

24 un   ENC:N   DEC:N

The middle knob changes encode from **no** to **yes**, the right hand knob changes MS decode from **no** to **yes**. The MS encoder in the z-Qualizer is located in front of the gain, left and right channel adjustments and in front of the stereo equalizers. The MS decoder is located after all processing and before the dither. In the z-Qualizer, the MS encoder is at unity gain, and the decoder drops the gain by exactly 6 dB, which turns out to be perfectly symmetrical---but you don't need to know how the math works to take advantage of **MS** mode. If the mode is set to **ENC:Y DEC:Y** and all equalization and gains are set to 0 dB, the system remains perfectly bit transparent, since its MS encoder and decoder are exactly symmetrical, down to the last mathematical bit. You can leave the system in **ENC:Y DEC:Y** mode if you wish and operate it as a stereo equalizer without any concern for losses.

MS stands for *Mid-Side*, or *Mono-Stereo*. When in **MS Y, Y** mode, internally the z-Qualizer becomes an MS-style equalizer instead of a stereo equalizer. However, the input remains stereo and the output remains stereo. It is useful to know that an MS encoder contains the same circuit as a decoder. This means that you can use the encoder to decode and the decoder to encode if you wish. For example, you could feed an MS signal into the z-Qualizer, decode it to stereo, manipulate the left and right balance and eq, and then reencode it to MS for further processing in MS mode. There are about 16 permutations you could think of, so the limit of flexibility is completely up to your imagination!

Taking advantage of M/S

The most common use of MS equalization is to deal with center channel information which needs separate EQ than the sides. If you feed a stereo recording into the z-Qualizer and set the mode to **MS Y,** Y, the **left channel gain** control becomes the **M gain** (or center gain), and the **right channel gain** control becomes the **S gain** (or side gain). These controls can manipulate the width of the stereo image, while also reducing the ratio of the mix of center-located instruments to side-located instruments. Try it. Feed in a good stereo recording with a center-located vocalist. Turn the **right knob** to **gain** and turn the left knob to **L**. Turn the gain knob counterclockwise until the **L** gain is at -95 dB. The vocalist (and all center instruments) should virtually disappear, and you will be left with a widely separated, out of phase mono representation of the instruments and stereo vocal reverb. See what happens if you turn down the R gain instead.

For equalization, you can cheat the frequencies for the M channel up or down separately from the S channel to manipulate the recording in creative ways. All the L labels are interpreted as M and the R labels as S. In **MS Y,** Y mode there is no way to adjust full channel balance because the L and R gain controls have become M and S controls, respectively.

The **presets** (see below) remember the M/S state, so, for example, you can work on one tune in MS mode and another in stereo mode.

Presets

Loading Presets

Rotate the **left knob** to the **load** position. The alphanumeric display will look something like this:

a=01 load

Rotate the **gain knob** (which is directly below the a=01 display) to choose the memory number from 00 to 99 from which to load, memory 00 is a factory default, which is set to: *MS N, N; 24 bits; all gains and eqs at 0 dB*. Memories 1 through 99 are user-adjustable. To return the z-Qualizer to flat settings, simply use the following procedure with preset #00.

Let's assume we want to load the contents of memory #34. Rotate the **gain knob** until the alphanumeric display looks like this:

a=34 load

Then rotate the **Q** knob (below the word **load** in the alphanumeric display). This loads memory #34 and places an arrow next to the preset number to let you know the memory has been loaded:

a=34< load

It's that simple. If you change the preset number from the one that's currently loaded, the arrow disappears, indicating that you have to **load** again if you want this new memory. Of course, the previously loaded memory is still in the equalizer until the next **load**. Try it. It's easy and intuitive.

The **preset load** function remembers the last memory you were working on, and it will come back to that mode when you return to this screen.

Saving Presets

Rotate the **left knob** to the **save** position. The alphanumeric display will look something like this:

a=01 save

You cannot save anything into **preset 00.** To save the current equalizer state into any memory, rotate the gain knob to choose the memory, then rotate the Q knob (in either direction), and the display now looks like:

a=01< save

To save the current state of the equalizer into a series of memories, just increment the memories with the left knob, rotate the right knob, and repeat. This is very useful when you have been doing a job and want to neutralize a bunch of memories.

MIDI

Setting the MIDI channel

Connect the MIDI output to the input of a MIDI sequencer, and the MIDI input to the output of the sequencer. Turn the **left knob** to the MIDI position and the alphanumeric screen will look like this:

ID=02 read dump

To change the midi channel, rotate the **gain knob**. The channel is preserved in non-volatile memory at power down.

SysEx Read

For a SysEx restore of all 99 presets from a MIDI sequencer, turn **the freq knob** (below the word **read** in the alphanumeric display) in either direction to prepare the z-Qualizer to receive SysEx data. Start your sequencer playing the sysex dump. While the dump is proceeding, the word read will change to a numerical countdown from 99 to 1 to indicate the memories which are being restored. This takes only a few seconds. When the SysEx restore is complete, the word **read** re-appears in the alphanumeric display.

SysEx Dump

For a sysex dump of all 99 presets to a MIDI sequencer, first start your MIDI sequencer recording. Then rotate the **Q knob** to start the dump. During the dump, the word **dump** will blink and the sequencer should indicate that it is receiving MIDI data. After all 99 memories have been sent to the sequencer, the display returns to **dump**.

MIDI Program Change

Connect the output of a MIDI sequencer to the MIDI in jack. Any time the z-Qualizer sees a program change whose value is from 00 through 99, it will load the corresponding memory. In the **preset load** mode the memory number will change to the one sent by the MIDI program change command. Any other screen will display the results of the program change.

Specifications

- Input/Output: AES/EBU (transformer-isolated, 110-ohm terminated)
- Input/Output precision: up to 24 bits
- Gain control: from -95 dB to +12 dB
- Gain resolution: 0.1 dB increments between 0 and 3 dB (gain or loss), 0.2 dB increments from 3 dB to 12 dB (gain or loss), 1 dB increments from –12 dB to -20 dB, 2 dB increments from -20 to -50 dB, 3 dB increments from -50 to -62 dB, 4 dB increments from -62 to -70, 5 dB increments from -70 to-95 dB.
- Filter types: 4 parametric, 2 shelving
- Center frequency resolution: 1/6th octave ISO from 28 Hz to 20 kHz
- Filter gain/cut: from -95 dB to + 12.0 dB with the same increments as the gain control.
- Filter bandwidths: Q=0.4, 0.5, 0.6, 0.7, 0.8, 0.9, 1.0, 1.1, 1.2, 1.3, 1.4, 1.5, 1.6, 1.7, 1.8, 1.9, 2.0, 2.5, 3.0, 4.0, 5.0, 6.0, 7.0, 8.0
- Shelf filter slopes: 6 dB/octave (first order)
- Channel separation: Effectively infinite
- Dither types: 24 bit, 20 bit, and 16 bit proprietary floating-point techniques, and 16 bit POW-r Dither (two types)

- Digital filter architecture: proprietary minimum roundoff-noise structure
- Dynamic range: better than 144 dB
- Ergonomic Stereo-linked and dual-mono operation
- Number of user-alterable presets: 99. Factory Preset: One (preset #00)
- THD+N: better than -135 dB
- Processor type: TMS320VC33 32-bit floating point DSP
- Digital audio demodulator/modulator: Crystal Semiconductor
- Sample rates supported: 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz
- Auxiliary-bit status handling: unit is transparent to channel status, validity, and auxiliary bits
- Power supply: 9 VDC @ 500 mA.

Z-Systems One-Year Warranty

All Z-Systems products come with an automatic one-year full warranty. We warrant to the original purchaser that the product purchased will be free of defects for a period of one year from the date of purchase. Z-Systems will, without charge, replace or repair, at its option, defective products or component parts upon delivery to the manufacturer. This warranty does not apply in the event of misuse or abuse as a result of unauthorized alterations or repairs. For warranty service work, simply contact the manufacturer to arrange for return and repair. Z-Systems will not be liable for any consequential damages, including, without limitation, damages resulting from loss of use.

An Invitation to You

After you have used your Z-Systems product for a while, call, fax or email us and tell us what you think. We enjoy hearing from you. Many of our new products, updates and modifications were designed as solutions to technical problems encountered by our end-users. Our enthusiastic customers help spread the good word about Z-Systems. We would like you to be one of them.

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