z-q2 operations manual

Introduction

Z-Systems model z-q2 6-band digital equalizer is based on the highly-successful model z-q1. For those who are familiar with the z-q1, here is a list of the extensive improvements we've put into the z-q2:

•**Complete ergonomic upgrade**. The z-q2 is much easier and more fun to use. The front panel has been simplified, keystroke and hand movement greatly reduced, especially the process of saving and recalling memories. All screens are now non-modal, which in plain English means that you can navigate instantly from any menu to any other. The system remembers the last mode of any button that you used. Left and Right channel equalization can now be separately and easily adjusted and inspected. The one-line display has some added features keeping it simple, yet powerful.

•**POW-R™ dither**. The reknowned dither from the POW-R consortium is now included. POW-R dither is so transparent that it yields nearly 20-bit psychoacoustic performance from a 16-bit master with virtually no penalty in tonal balance, depth, space or width.

•**MIDI control**. Memory recall can now be recalled by an external MIDI controller, and the entire state of the memories can be stored or recalled with Sysex.

•**M/S**. A bit-transparent M/S encoder and/or decoder can be inserted in the signal path.

•**Range of gains** has been optimized for very fine 0.1 dB equalization near 0 dB, and more coarse as the gains become extreme.

•**Range of frequencies** has been extended to 20 kHz for that ultimate "air control".

•**Range of Qs** now includes a fine range around 1.0, to include the most common and musically useful Qs.

•**Power-Down state**. The z-q2 writes its current state into memory whenever power is lost.

•And all the original low-distortion Zelniker equalization algorithms are still here!

z-q2 Operation

Connect AC power of the proper voltage range with an IEC power cord to the back of the z-q2. If in doubt, verify the internal power selector switch has been set to the proper voltage for your country **before plugging in power!**

System Menu

Dither, Wordlength and Sample Rate Display

Connect the z-q2 to an AES/EBU standard digital audio source whose sample rate is 32 kHz, 44.1 kHz, 48, 88.2, or 96 Khz. The **AES LOCK** LED will light to indicate it's locked to the incoming signal. Now, press the **SYSTEM** button once. You should see this display:

in:1 44k 24 dith

If the system is locked, the sample rate appears in the left side, and the wordlength and dither mode appears in the middle. As in all z-q2 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 middle knob to any of the following:

24 dith 24 bits 20 dith 20 bits 16 dith 16 bits 16 pwr2 16 pwr3

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.

Using the dither and wordlength settings

When the z-q2 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-q2 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 wideband 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 **bits** menu choices actually truncate the output of the z-q2 without concern for quantization distortion.

The other dithered wordlength settings are only to be used if the z-q2 is connected **directly** to a following device which truncates the wordlength. For example, use **20 dith** if the z-q2 is **directly connected** to a 20 bit ADAT. Use **16 dith or 16 pwr** if the z-q2 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 bits** option if you wish to bypass the z-q2 without pressing the BYPASS button (in an automated session, for example, where one tune passes through flat without any alteration). In this case, select **24 bits** and make sure that all the gains and equalizer levels are set to 0 dB. In the **24 bits** setting, with everything neutralized, the z-q2 is bit-transparent, will pass all incoming AES/EBU signals without data alteration. In the **20 bits** (**16 bits**) setting, the z-q2 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-q2 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, probably inaudible, much less loss than with competing noise-shaping processes.

MIDI

Connect the MIDI output to the input of a MIDI sequencer, and the MIDI input to the output of the sequencer. Press the **SYSTEM** button again to arrive at the MIDI screen, which should look like this:

midi #06 read no

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

SYSEX read and dump. The second and third menu items are used for sysex dumps and reads.

For a sysex restore of all 99 presets from a MIDI sequencer. When the second menu says **read**, then you may rotate the third knob until the display says **yes**, preparing the z-q2 to receive a previously stored sysex from a computer sequencer. After setting the screen to display

read yes

start your sequencer playing the sysex dump. When the dump is proceeding, the left-hand portion of the display counts down from 99 to 1 to indicate the memories which are being restored. This takes only a few seconds.

For a sysex dump of all 99 presets to a MIDI sequencer, rotate the middle knob (clockwise or counterclockwise, depending on where it was last set) until the menu says **dump**. Then start the sequencer recording, and rotate the right-hand knob until the menu shows a blinking **yes** indicator. During the dump, the sequencer should indicate that it is receiving midi data. After all 99 memories have been sent to the sequencer, the display returns to **no**.

Midi Program Change. Connect the output of a MIDI sequencer to the MIDI in jack. Any time the Z-Sys sees a program change whose value is from 00 through 99, it will load the corresponding memory. In the **preset load** screen both a and b memories change to the same when the program change command is received. Any other screen will display the results of the program change. For obvious reasons, response to program change is disabled in either the SAVE or LOAD screens.

MS Encode and Decode

This versatile feature can be used in several ways. Press the **SYSTEM** button again to arrive at this screen:

MS ENC:N MS DEC:N

The middle knob changes encode from **no** to **yes**, the right hand knob changes MS decode from **no** to **yes**. The optional MS encoder in the z-q2 is located in front of the gain, left and right channel adjustments and in front of the stereo equalizers. The optional MS decoder is located after all processing and before the dither. In the z-q2, 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 **MS Y, 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 MS Y, 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-q2 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-q2, 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-q2 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. Press the **GAIN** button and turn the middle 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, 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.

This completes the SYSTEM Menu.

presets

Loading Presets. Press the **presets** button once to arrive at the **load** screen,

which should look like this:

a=35 b=49 load

There are two nonvolatile memory “spaces”, labelled **a** and **b,** which can be compared. Rotate the left knob to choose the memory number from 00 to 99 for the **a** space, or the middle knob for the **b** space. 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.

Let’s assume we want to compare memory #34 against memory #23. To recall (load) memory a into the equalizer, rotate the right knob until the display looks like:

a=34< b=23 load A

This loads memory A, which in this example is preset #34. Rotate the right knob again (in either direction) to yield:

a=34 b=23< load B

This loads memory B, which in this example is preset #23. It’s that simple.

If you change a knob corresponding to a memory which is currently loaded, the caret and the letter after **load** disappear, 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. Press the **presets** button again to arrive at the **save** screen, which looks like this:

a=01 save

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

a=01 saved

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.

gain

Press the **gain** button (the LED next to the **gain** button illuminates) and see this screen:

G +00.0 L +00.0 R +00.0

Adjust the overall gain to avoid output clipping. Adjust left and right gains to taste. As mentioned previously, if a stereo signal enters the equalizer, and the MS encoder is on, the **L gain** controls the M level, and the **R gain** controls the S level.

Equalization

The equalization in the z-q2 is performed with floating point arithmetic and uses proprietary low-distortion algorithms developed by Dr. Glenn Zelniker. There are six bands, including two first order (6 dB per octave) shelves and four parametrics (bell curves).

Stereo (ganged EQ channels):

Press the **lo shelf** button (on the top left). The display should look like:

S +00.0 56 Hz

Rotate the left knob to change the gain of the low frequency shelving equalizer. Rotate the middle 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. The **S** stands for stereo, and when the screen has an **S** in it, frequency and level control for both stereo channels are ganged together.

Separate Left and Right Channel EQs:

Press the lo shelf button again. The display now reads:

L +00.0 56 Hz. Here you can display and adjust the left channel shelf independently.

Similarly, press the **lo shelf** button again to display and adjust the right channel.

Now, press **lo shelf** again to return to the **S display.** If you altered either the frequency or the level of one channel, you will see one or two corresponding

asterisks in the **S(tereo)** display, for example:

S +00.1* 56 Hz*

An asterisk indicates that **there is a different setting between the left and right channels**. This way you can instantly inspect if the left and right channels are identical. Just look for the asterisk to see if either frequency or level are different between channels. In the **S display,** you can set both channels to the same frequency (overriding any individual settings), without altering their level differences, as long as the asterisk remains next to the level in the S display. The equalizer buttons return to the display mode last used (Stereo, left or right), and the current channel display is also saved in the presets. For example, if you've been concentrating on the left channel of band #2, that is the way it will first display when you press that band, if it has been saved that way in a particular preset.

The other bands:

High Shelf: The top right button is the high shelf, and performs exactly as above for stereo (ganged), or individual left or right equalization. The frequency knob controls the frequency **above** which action occurs.

Parametrics: Press any of the remaining four buttons, to see:

S -00.2 70 Hz Q = 0.5

As always, the label is above its corresponding knob, and as above, pressing the button sequentially will allow you to control the Left or right channel separately. There may be up to three asterisks in the **S**(tereo) display to indicate if the level, frequency, or Q differs between left and right channels. Gang both channels instantly by moving any control knob until the asterisk is eliminated.

Q is the inverse of bandwidth. It is the product of center frequency divided by the 3 dB down bandwidth. Thus, a Q of 0.5 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.

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: TMS320C31 32-bit floating point DSP
• Processor performance: 60 MFlops
• 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: 110/220 VAC, 50/60 Hz. User selectable internal switch between 110 or 220 volts

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.

Z-Systems Audio Engineering
4641-F N.W. 6th Street
Gainesville, FL 32609 USA
(352) 371-0990 (voice) or (352) 371-0093 (fax)
email: z-sys@z-sys.com