

Zen 10 Stereo AGC

The Zen Level Zen 10 audio level controller by David Reaves Audio and Aqua Broadcast offers advanced, customizable Automatic Gain Control (AGC) for consistent and sonically pleasing audio levels. Developed by David Reaves, whose fascination with AGC began in his youth, the Zen 10 refines AGC principles to create a seamless, unobtrusive audio experience.

The Zen 10 is the ideal solution for precision audio level control in various applications, including:

- Radio broadcast pre-processing
- Audio production
- TV loudness control
- Streaming
- Podcasting

If you're good at creating content that uses audio, you know that the people listening to you are going to want your sound levels to be consistent. The thing that does this for you is an "Automatic Gain Control," or AGC.

What sets the Zen 10 apart is its unique "Zen" approach, knowing when not to adjust audio. If the audio level is already correct, no change is made, preserving the integrity of your sound.

This philosophy helps prepare audio for the next stages of a broadcast chain, making any processor sound better while maintaining consistent, high-quality output.

In a world of multi-platform delivery—streaming, podcasts, network contribution—consistent, great-sounding audio is essential. The Zen 10 excels across all these platforms, ensuring your content always sounds its best.

With support for analog, AES3 digital, and Dante AES-67 digital audio I/O, and monitoring via a front-panel headphone jack, the Zen 10 can be controlled through its front panel or a web interface with dual colour LCD displays.

Offering presets for quick setup and customizable settings, the Zen 10 features ten customizable control systems and up to five adjustable frequency bands, providing precise, dynamic control for any audio environment.

Experience the power of Zen Level—elevate your audio while maintaining its natural integrity!

AQUABROADCAST

Aqua Broadcast is well known as a FM Transmitter and Audio product manufacturer, but also offers a comprehensive range of R&D services in both hardware and software design.

Our experience with Broadcast audio has led us to develop unique algorithms for FPGA-based Digital Audio Processing and RF measurement. Our team of highly qualified engineers, ensure the highest levels of innovation, precision, and technical excellence in everything we create. Recent projects we have successfully completed and delivered include SFN (Single Frequency Network), Tunnel re-broadcast and break-in as well as Defence, and Solar FM radio solutions.

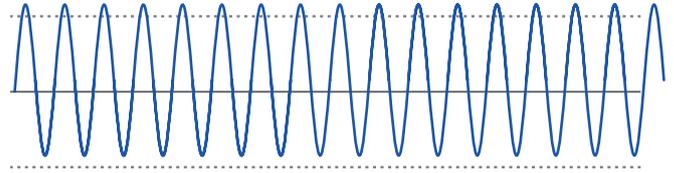


How the Zen Level AGC System Works

by David Reaves

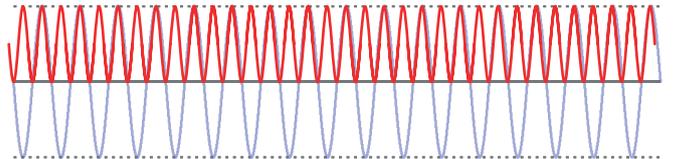


Here's how level control works in every Zen Level AGC! We'll start with a sine wave input, shown in blue:



To create a high-quality, average-based control signal, the audio is first multiplied times itself, **squaring** it, shown below in red.

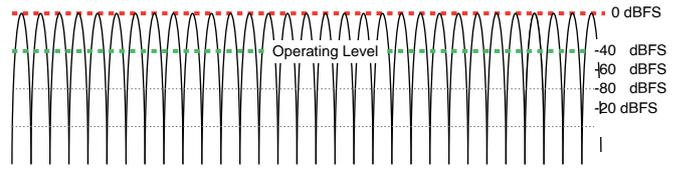
The result is a signal that, unlike its input, is only positive-going. In the case of a sine wave, it doubles its frequency and cuts its magnitude in half. This can be clearly seen in the graph. (For typical complex audio waves, the result wouldn't be so obvious.)



The squared signal is then **logged**, to bring it into a **decibel** (dB) domain. Using decibels for control allows complicated mathematics to be performed on the control signal which, along with the initial squaring of the signal and post-logging filtering, can yield processing that is surprisingly pleasing to human hearing.

In the chart below, our squared sine wave has been converted to a log range of 80 dB on the vertical scale, and is shown in solid black.

The top, red-dashed line, represents 0dBFS. Incoming 16 or 24-bit audio will not exceed this magnitude, but internal to our DSP control system, an additional 8 bits (an extra 48 dB) is added to the top of the range, giving us plenty of headroom to make calculations.



The dashed green line represents 20 dB below the clipping level of our 0dBFS sine wave. This is our arbitrary **operating level** and internally, all control is referenced to this as a neutral, **±0 dB gain** point. [Externally, input and output operating levels can be set to reference anywhere from -26 dBFS to -14 dBFS]

Being able to accurately measure wide level variations is great, but real-life situations rarely demand even 30 dB of control. There's a way to artfully control access to all this gain, and it's called **"release gating."**

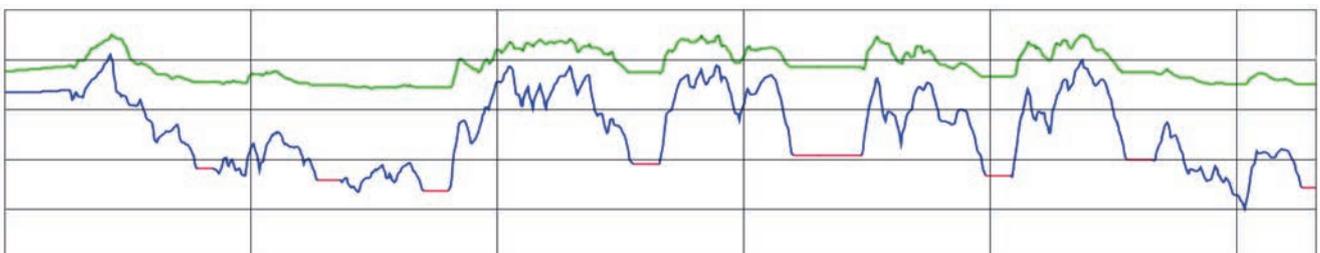
The idea is that if there is verifiable audio at a level somewhat lower than normal level, but still above our **gating threshold** point, our system can allow the gain to increase ('release'), to bring that level up.

On the other hand, if the incoming audio is below this gating threshold, as would be the case with background noise or quiet sections of a musical piece, then the logic would not allow release. So the gain will not increase, but rather hold steady, exactly where it was before the gating occurred.

For the Zen Level processors, this gating threshold is user-adjustable, from 10 dB below the -20 dBFS operating level (-30 dBFS), to 30 dB below operating level (-50dBFS).

The chart below illustrates how release gating affects audio control. It's a graph of a simple voice track, about 26 seconds long. The horizontal lines are spaced at 6 dB apart. The vertical lines are spaced five seconds apart. The green and the blue lines represent a comparison of two otherwise identical control signals with different gating thresholds. The lower each line goes, the more gain will be added to the controlled audio signal.

The green signal uses a gating threshold of -10 below operating level, while the blue signal's threshold is 20 dB lower, at -30 below operating level. Even though all other control parameters including input level, attack, release, etc., were the same, the signal using the lower gating threshold is clearly more active. Gating occurs every time the logged waveform goes below the threshold and occurrences can be as short as in the millisecond range, hundreds or even thousands of times per second; longer instances are clearly shown in red on the -30 signal.

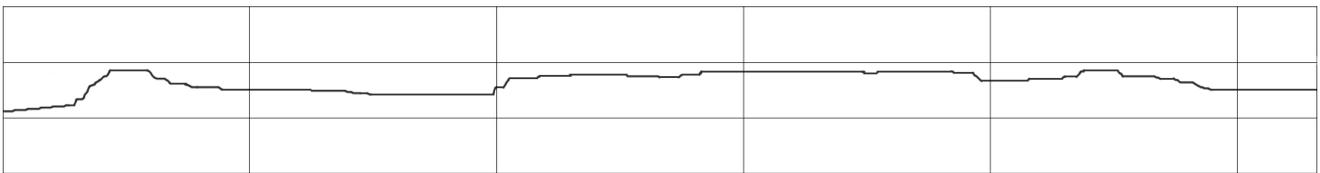


In the late 1950s into the mid-1960s, the Columbia Broadcasting System's research and manufacturing division, CBS Laboratories, investigated sophisticated new methods of level control. Some of these ideas were patented and incorporated into products which would be used by CBS's own radio and TV stations, and then offered for sale to other broadcasters.

The company's "Audimax" line of automatic level controllers featured several of those ideas. One of them was the previously described "release gating." Another, even more ingenious, was the "gain platform" concept.

Gain Platforming was intended to allow consistent, useful level control over a range of about 20 dB, while minimally affecting the dynamic aspects, the impact, of the original audio. Conceptually, it was likened to a very skilled human, who would only change the gain when the audio left a predetermined dynamic range "sweet spot." CBS researchers determined that this sweet spot should be about seven decibels (the Zen Level platform can be set to anywhere from 2 to 10 dB). So as long as the audio stayed within this 'window,' nothing would happen. But when the incoming level shifted either higher or lower outside this range, the system would correct for it, momentarily changing the gain, moving the sweet spot to a new location appropriate to the new conditions. Then the audio was again left alone.

When viewed on a strip chart, the gain pauses and variations looked like a series of "platforms," flat lines where no gain change occurred, interrupted only occasionally by a brief gain change in one direction or another. And that's where the name came from. [shown below: our previously-seen voice track using the -10 release gate, with a 4 dB gain platform]

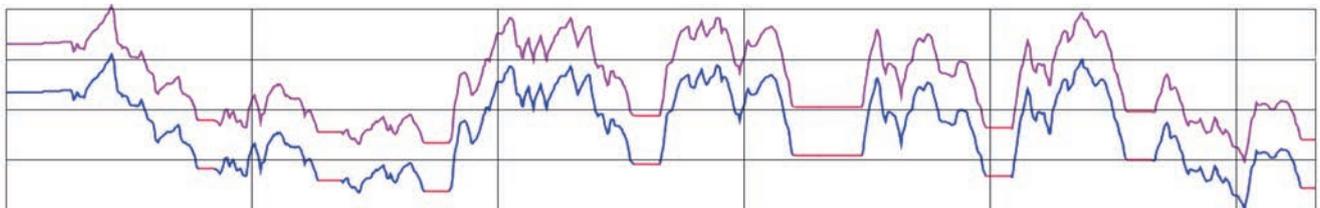


This gain platform concept differed dramatically from any audio level controllers which had come before; previously, level control had always been either increase (release) OR decrease (attack) in its active range. The Audimax introduced a third possibility: Do nothing. (Which is where the **Zen** name comes from: doing something by doing nothing.)

So now you know what the gain platform does, and why it does it. Here's how it works in the Zen Level:

The gain platform starts with a squared, logged control signal that has already had filtering applied to enable the attack and release timing. Naturally, this release timing was subject to interruption by release gating.

A copy of this signal is made, and it is shifted upwards, biased by the number of decibels of platform desired. [shown below: the voice track using the -30 release gate, with the new copy placed six dB above it.]



Now the two signals, the original (blue, above) and the biased version (violet), are analyzed and compared, in real time, to create a third signal: The control signal.

As long as the original signal is rising, the control signal follows it.

If the original signal starts to fall, the control signal holds, not changing. Control has entered a "platform."

The control signal will not change, will not leave its gain platform, unless one of two things happens:

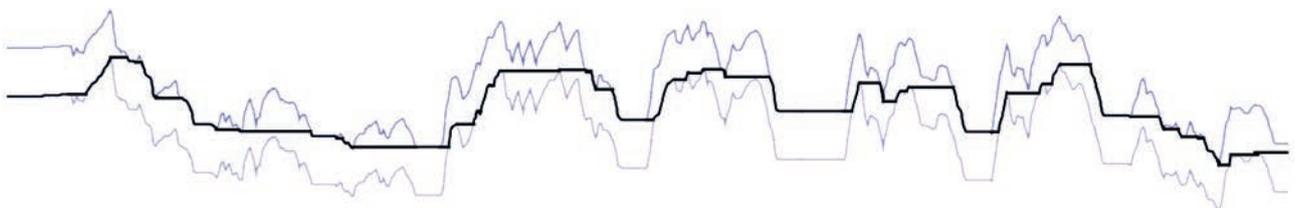
1) if the **original signal** again **rises above** the control signal, the control signal will follow it. Control will not follow the original signal when it falls, no matter how much. On the other hand...

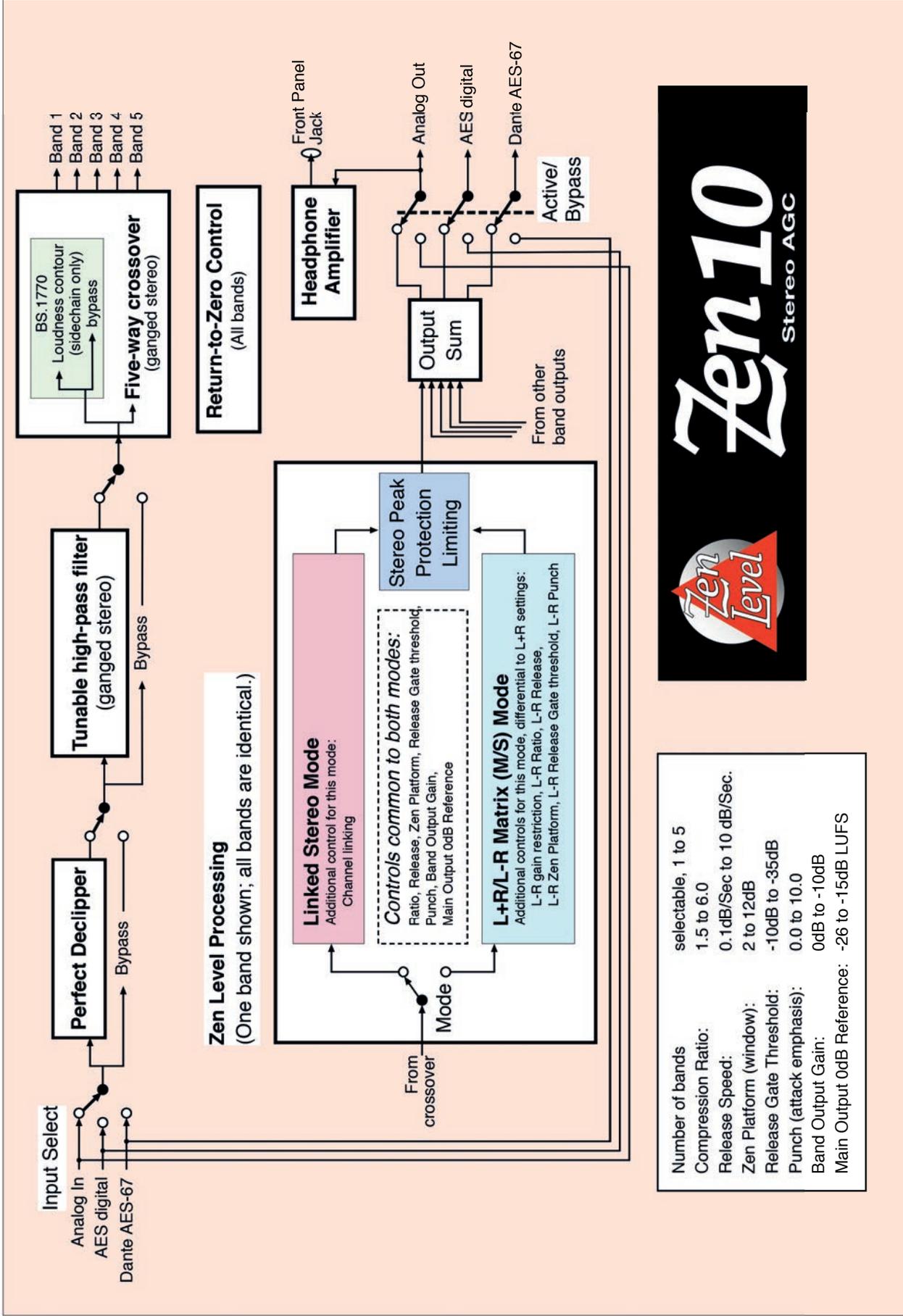
2) if the **signal with the bias falls below** the control signal, the control signal will follow it. If this signal should start rising, the control signal will hold, entering another gain platform.

So the rule is that the control signal will always follow the rising of the original signal (attack) as long as it attempts to rise above the control signal, and follows the falling of the biased signal (release) but only when the biased signal attempts to fall below the control signal. In all other cases, the control signal remains unchanged.

[Now all we have to do is follow the rules and connect the dots. Our control signal is black.

Multiply it by -1 and the ratio factor (1-(1/ratio), and that is your Zen Level gain control]





Number of bands	selectable, 1 to 5
Compression Ratio:	1.5 to 6.0
Release Speed:	0.1dB/Sec to 10 dB/Sec.
Zen Platform (window):	2 to 12dB
Release Gate Threshold:	-10dB to -35dB
Punch (attack emphasis):	0.0 to 10.0
Band Output Gain:	0dB to -10dB
Main Output 0dB Reference:	-26 to -15dB LUFS



GENERAL

Input ACV @ 50/60Hz	100-240VAC, 50/60HZ
Power Connector	IEC13 3A filtered
Optional DC power input	48V
Dimensions WxDxH (inch)	19x10.5 x 1RU (1.75")
Dimensions WxDxH (cm)	48.3 x 27 x 44
Weight	2.5kg / 5.51lb

ENVIRONMENTAL

Maximum operating Altitude	15,000 ft / 4,420m AMSL
Temperature range	0 to +45°C working. -10 to +50°C storage
Humidity	95% @ 35°C, non-condensing

CONNECTIVITY

2x 1Gbit Ethernet (1 per CM5)
2x 100Mbit Ethernet with POE+ (1 per CM5, 1 shared with AES67 module)
4x USB 3.0 (2 per CM5)
1x USB 2.0 on the front panel
1x USB UART for primary CM5
2x HDMI 2.0 (1 per CM5)
Optional 2.4 GHz, 5.0 GHz IEEE 802.11 b/g/n/ac wireless on CM5
Optional Bluetooth 5.0, BLE on CM5

USER INTERFACE

Primary display	TFT LCD 320x240 RGB
Secondary display	TFT LCD 1424x280 RGB
Control Inputs	Analog joystick with 2 x silicone backlit buttons

ANALOG AUDIO INPUT

Stereo analog audio input	2 x XLR female, 10k Ω impedance level adjustable 0-24dBu
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ANALOG AUDIO OUTPUT

Stereo analog audio output	2 x XLR Male, 100 Ω impedance level adjustable 0-24dBu Optional bypass relay module for power-off signal pass through
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DIGITAL AUDIO INPUTS

1 x AES/EBU Audio, RJ45 StudioHUB+, 48kHz, 24bits, 110 Ω input impedance
1 x AES/EBU MPX, RJ45 StudioHUB+, 192kHz, 24bits, 110 Ω input impedance

DIGITAL AUDIO OUTPUTS

1 x AES/EBU Audio, RJ45 StudioHUB+, 48kHz, 24bits, 110 Ω output impedance
1 x AES/EBU MPX, RJ45 StudioHUB+, 192kHz, 24bits, 110 Ω output impedance

HEADPHONE OUTPUT

1/4" (6.3mm) female stereo socket
Software adjustable volume
Minimum load impedance 16 Ω