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Digital Audio – It's Not Perfect!

Digital Audio signals are now common in most audio production and broadcast facilities. With new technology often being greater than the cat's meow, we all heard that once the old analog audio was a bunch of bits, ones and zeros, nothing bad could happen to it. We could move these bits through transmissions lines, down the chain of equipment, convert it, copy it and reproduce it back to analog without degrading the audio. It was invincible, just like a super hero.

Stop the train! Now that we've all had some experience with digital audio we are discovering this isn't true. There are issues that cause digital audio equipment not to work together and issues that degrade the resulting audio.

Analog vs. Digital Audio Analyzing

Having worked with analog audio and equipment for many years, we have gained an expertise and comfort level that is admirable. We can hear noise and distortion and link these audible symptoms to likely causes, making the needed changes. We have our trusty tone generator and oscilloscope for isolating those more difficult challenges. We know how to measure levels, frequency response and distortion to gauge the audio performance. We are in our comfort zone.

Unfortunately digital audio changes all this. The old cause and effect relationship we are familiar with is gone. Likewise our trusty tone generator and oscilloscope offers little help in diagnosing a digital signal or equipment defect. Can you teach an old analog dog a new trick?

Digital Audio.=Can you teach an old analog dog a new trick?

Sure, you can! Transitioning from analog to digital audio analyzing and troubleshooting is not as hard as you may think. Sure, it is going to take some study so you know what is in those digital bits. Secondly, it is going to require some new and exciting analyzing tests and instrumentation so you can analyze those digital bits.

Some Study
+ New Equipment
= Analyze Digital Bits

Digital Audio Bit-by-Bit

An analog to digital converter changes the audio signal to digital values by sampling the audio level at fixed intervals of time. Sampling is like taking "snapshots" of the analog audio signal level over time. Sampling happens at equally separated intervals measured in the number of samples taken every second, expressed in hertz (Hz) or in the thousands of hertz (kHz). Digital audio is commonly sampled at 44.1 kHz or 48 kHz or at doubled rates of 88.2 or 96 kHz for professional recording.

Assigning a digital value to the audio level at each sample interval is called "quantization." This requires that the amplitude range of the audio waveform be divided up into level steps. A quantized binary value encoding system, Pulse Code Modulation (PCM), has been adopted for overall improved system performance. PCM quantifies linearly all quantizing intervals by means of a fixed scale over the signal amplitude range. PCM makes use of a two's complement system to distinguish positive and negative binary coded values.

The number of bits used to form the PCM digital words (bytes) that are used to represent each of the sampled audio levels can vary from 8 to 24 bits. The bit word length determines the number of quantizing level steps (resolution) and the dynamic range. Each bit provides approximately 6 dB of range. An 8 bit digital audio word length provides 48 dB of dynamic range (quiet to loud audio range) while 16 bit provides 96 dB and 24 bit provides 144 dB.

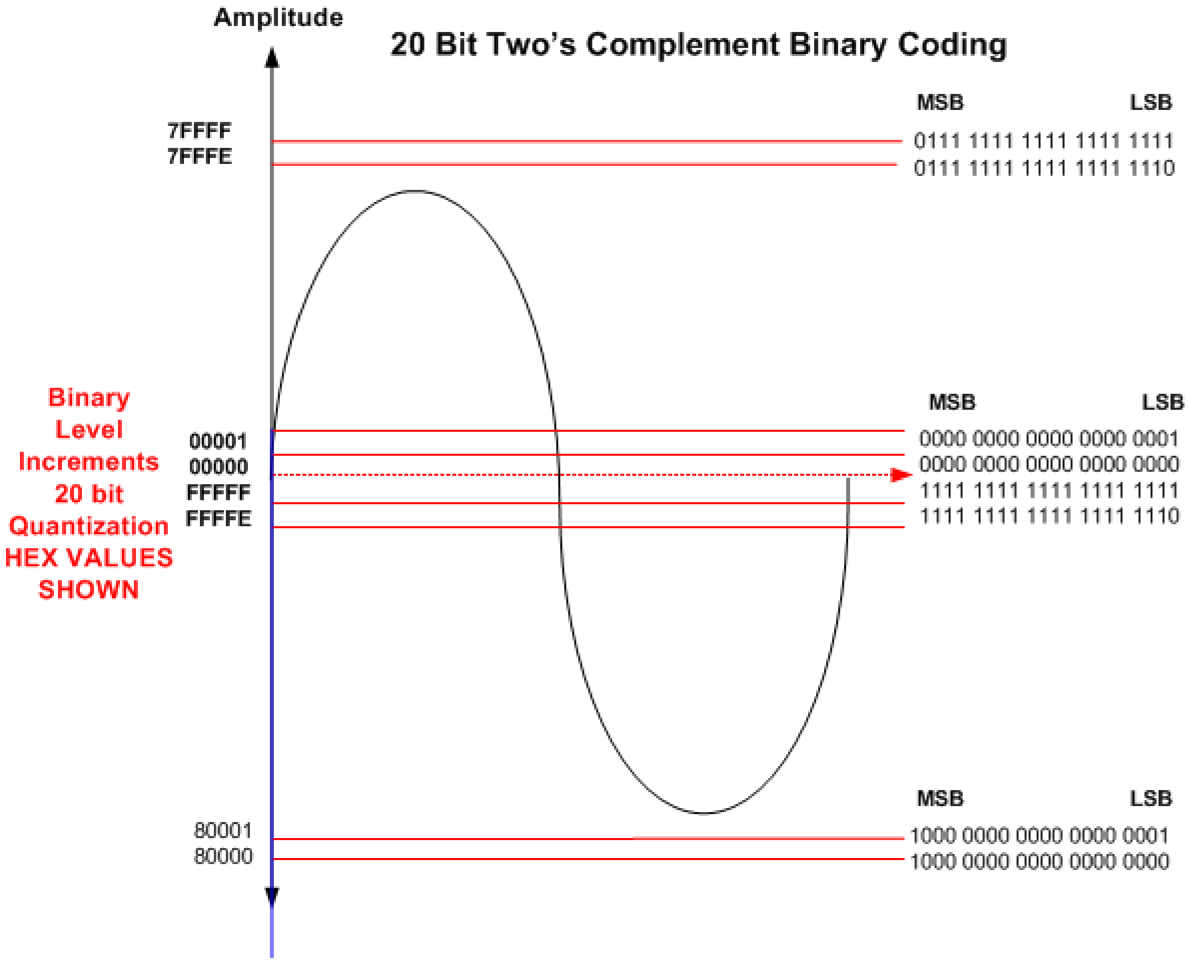

Figure 1. Pulse code modulation (PCM) uses a two's complement system to distinguish positive and negative binary coded values with word lengths from 8 to 24 bits.

In a digital audio system, the maximum audio level corresponds to 0 dBFS (dB full scale) which is assigned the largest digital code word. Manufacturers have adopted the familiar 0 VU level equal to + 4 dBu as a standard operating level (SOL). This level corresponds to -20 dBFS, in which the digital values are well below the largest digital code word value. This provides 20dB of range for audio peaks to go above 0 VU before digital clipping occurs.

PCM digital data is encoded using a second scheme called bi-phase mark coding (BPM). Bi-phase coding insures a DC balanced data line, as each bit begins with a transition and ends with a transition. If the data bit is a "1," a transition also occurs in the middle of the time slot. A data "0" has only the transitions at the beginning and end of the time slot and does not have a transition in the middle. Bi-phase coding doubles the data rate or frequency, as each data bit has two time intervals (clock cycles). A balanced line enables the receiver to properly detect logic high and low levels and the transition between them.

Getting Your Bits in a Row

Some form of organization is needed so the receiver can reassemble and identify the assorted bits of information contained in a digital audio data stream. Organization involves assembling the data into blocks. Each block consists of 192 frames of audio. Each of the 192 frames can be divided into two sub-frames for two channel audio. Each frame is produced at the digital audio sampling rate so each frame contains one digital value. In a 48 kHz audio sampling rate, each frame is 20.833 uS (microsecond) with each frame lasting 4 mS (millisecond).

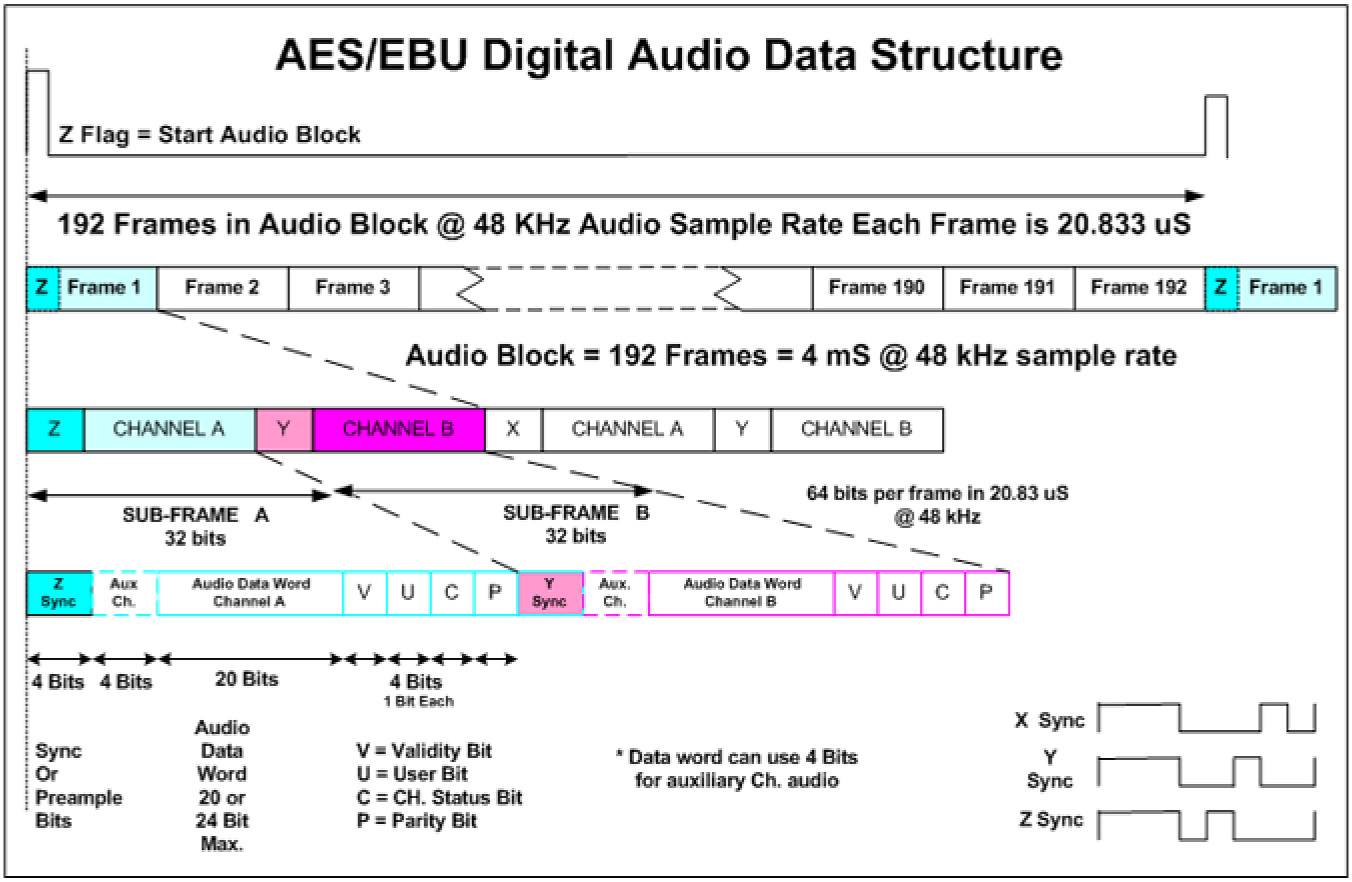

Figure 2. Structure of AES digital audio.

Each frame can carry two audio channels. In a 2-channel mode, the samples from both channels are transmitted in consecutive sub-frames. Channel 1 is in sub-frame A and channel 2 is in sub-frame B.

In addition to the digital audio word data bits, each sub-frame contains additional data. Each sub-frame consists of 32 bits, which includes 20 or 24 bits of audio word data bits and 8 bits of additional data. Each sub-frame includes bits for preamble or sync data, auxiliary data, audio data word bits, validity (V), user (U), Channel status (C) and Parity (P) data bits. Considering that each sub-frame consists of 32 X 2 bits, occurring in 20.833 us (FS = 48 kHz), the bit rate increases to 1,536,024 X 2 = 3,072,048 bits per second.

The first four bits of each sub-frame consists of four preamble bits or sync bits. These bits identify the start of a new audio block and each sub-frame. A "Z" sync bit arrangement marks the start of the first frame in the 192 frame block. The sync word "Y" indicates the start of every B sub-frame. The sync word "X" indicates the start of all remaining frames. The sync bit arrangement is used by a digital audio receiver to identify that start of the audio blocks and sub-frames.

Analyzing Frequency Accuracy

At the heart of any digital system is a clock. This is a crystal oscillator or voltage controlled crystal oscillator circuit. The oscillator output determines the resulting audio sample rate and audio data rate. A perfect circuit would be exactly the desired frequency and each cycle of the clock waveform would be identical in duration or time.

The clock isn't a perfect circuit, as the crystal is not perfectly accurate. Crystals are rated in accuracy described by a PPM (parts-per-million) rating. This indicates the maximum number of cycles the frequency may deviate for every 1 million cycles or hertz. A typical crystal rating is +/- 20 PPM. If the crystal frequency was 1,000,000 Hz (1 MHz), the generated frequency would be within +/- 20 Hz (1,000,020 - 999,980). The 20 PPM rating is additive. A crystal of 2,000,000 could deviate +/- 40 Hz, while a 3,000,000 Hz crystal could deviate +/- 60 Hz and so on.

In digital audio terms, a crystal frequency of 12,288,000 Hz is commonly selected. This is 256X the ideal sample rate of 48,000 Hz. A 20 PPM error at this frequency calculates to an error in frequency of +/- 246 Hz. Since this is a maximum error, one would expect typical operational errors in PPM or Hz to be much less.

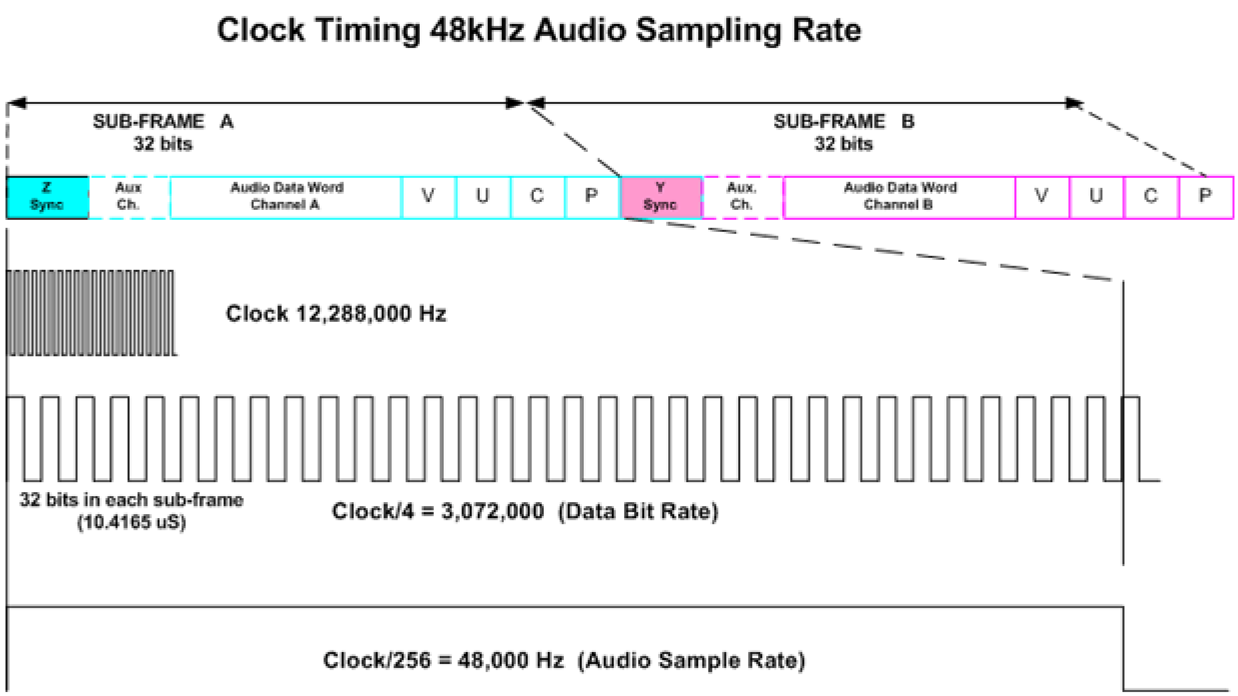

Figure 3. Clock/Data/Sample rate relationships in a 48 kHz digital audio signal. A typical clock is 256 times the audio sample rate and the data bit rate is ¼ the clock frequency.

Figure 4. A frequency measurement insures the digital audio signal is based on an accurate clock frequency and indicates any unwanted variations from the norm.

In digital audio systems, some frequency error is tolerable since the clock frequency is imbedded into the audio data stream and used to recreate a matching clock frequency by subsequent digital audio equipment. However, good maintenance and troubleshooting practices should include a frequency measurement of the digital audio signal(s) including the sample rate frequency (Fs) and clock frequency (256X Fs). Figure 4 shows an AES/EBU digital audio analyzer indicating a frequency measurement. Periodic measurement insures that when trouble strikes, you know good from bad!

When multiple AES digital audio signals are created by separate clocks, differences in clock frequencies and sync timing exist. These differences present challenges to digital audio equipment designed to switch between or process multiple inputs. To produce multiple AES digital audio signals at the same frequency and timing, master clocks or digital audio reference (DARS) signals can be used to synchronize oscillators and sync timing.

What is Jitter?

Discussion about clock frequency and timing errors would not be complete without talking about jitter. With a perfect clock square-wave each subsequent clock cycle would be identical in time, with positive and negative parts of the cycles the same duration. The clock would be a symmetrical square-wave with each of its transitions occurring in exact time increments from the previous transition.

Again, the clock is not perfect. Clock cycles may fluctuate in time with cycles being slightly shorter or longer than previous cycles. Clock positive and negative times may be slightly longer or shorter causing transitions to occur at slightly different intervals in time. These variations are called "jitter".

VU/Peak Program Material Meter

Figure 2. Digital audio levels in comparison to analog levels.

The DA795's VU/PPM meter provides comprehensive level measurements on both the left and right digital audio channels when measuring/monitoring program audio material. In all, three measurements are indicated including:

1. VU (Main VU Bars - Also numeric VU readout at right of display above left VU bar and below right VU bar)
2. Instantaneous Program Peak Meter (Small bar above or to the right of main bar)

3. Peak Capture or Hold Meter - Captures and displays maximum peak program levels for left and right channels. Reset by selecting the "Max" field and clicking.

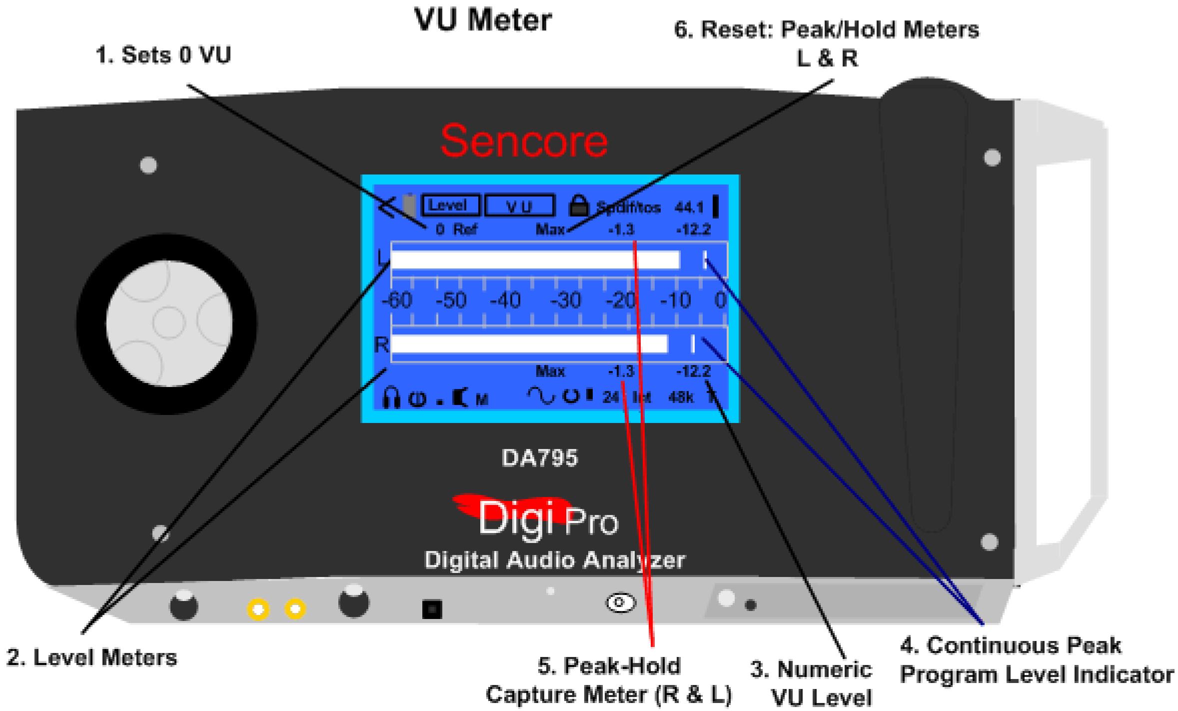

Figure 3. The DA795's VU Meter measures audio program levels and peak levels on both digital audio channels.

Field Descriptions:

1. **VU Reference Level:** Scales the digital signal level dB FS to which the meter's OVU point is referenced. The level can be adjusted from 0 to 24. This scales the 0 VU from 0 dBFS to -24 dBFS. (Standard setting would be "20 Ref" resulting in a -20dB FS = 0 VU)
2. **VU Graphic Bar Level Meter:** Indicates VU level of left and right digital audio.
3. **VU Numeric Level Meter:** Indicates VU level of left and right digital audio.
4. **Peak Program Meter:** A single bar the bounces above the main bar indicating peak program levels.
5. **Maximum Peak Level Meter:** Captures and updates the maximum or peak level of the audio program material.
6. **Max Reset:** Resets Maximum Peak Level Meter. Highlight field and click to reset.

Level Meter — dB FS

The DA795's Level Meter provides a measurement of an AES/EBU or S/PDIF signal into Input 1. Both the left and right digital audio channel levels in dB are measured simultaneously. The meter provides digital numeric value readouts. (Default: O dB rms is referenced to the full digital scale 0dB FS) Several other measurement units are available including, dB peak, % rms, and % peak. The % measurements are referenced to full scale. Typically, the Level Meter is used to calibrate digital audio equipment levels while the VU/PPM meter is used to monitor audio program material levels.

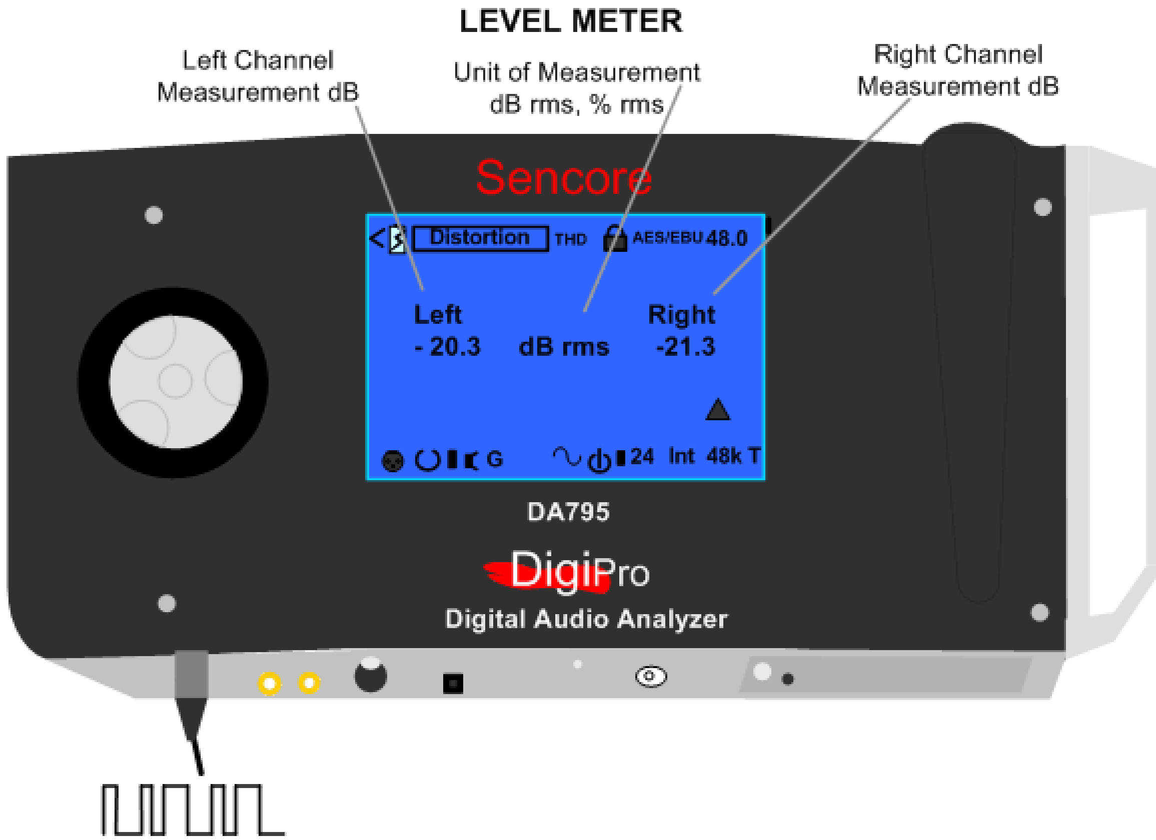

Figure 4. Digital Audio left and right Level dBFS meter of the Sencore DA795.

Your Solution to Digital Audio Analyzing

The DA795 provides comprehensive digital audio level measurements. The DA795 fully equips you for all digital audio troubleshooting and performance testing needs. For more information on analyzing digital audio signals with the DA795 DigiPro, call 1-800 Sencore (736-2673) or visit www.sencore.com.