Calibrating Cinema Sound Systems

by Peter Soper, R&D Engineer

Cinema Sound systems have always had a target "calibrated" level, such that when a "reference" noise signal is fed into the system, the overall gain should be adjusted so that the measured SPL is calibrated to a reference SPL level.

The traditional method involves a "reference" pink noise signal and the target calibrated SPL level was 85dB SPL, Slow, C-weighted, measured in the room.

Back when these levels were chosen, 85dB SPL was selected because it was sufficiently loud enough to easily overcome any HVAC or other environmental noises while leaving what was considered "more than adequate" headroom before system components would clip. This practice paid very little attention to how a signal of a higher level would behave when fed into the system, and because of tape saturation, optical septum effects and other self-evident non-linear aspects that come into play as analog signals get pushed towards their limits, it was generally expected that in the interest of sonic quality, soundtracks would not get mixed a whole lot louder than what could comfortably be reproduced by a system that was happy playing at 85dB with some assumed amount of headroom above that.

Meters in those days also had inherent limitations. Peak Program Meters (PPM), and Volume Unit (VU) meters were prevalent, but only the best and most expensive meters offered true "peak" or even true "RMS" reading modes, so much of the industry was left working with these less expensive (PPM and average responding) tools that did not fully capture these values of importance for the signals they were being used to measure.

Even today there are efforts to mimic the meter behavior of the analog average reading meters from the past with modern digital meters. This creates a problem because this introduces an error into the measurements. For example; the RMS value of a sin wave is $(\sqrt{2})/2$ times the peak value, or .707x where the average value of a sin is $\varpi/2$ which is .636x, while PPM meters simply do not respond fast enough to capture the value of absolute signal peaks.

A square wave is the only case where the peak, RMS, and Average values are the same.

Measuring noise and music with an 'average responding meter' will create serious errors, so essentially, average measurements of noise signals are irrelevant.

With the advent of digital audio now becoming the predominant means for cinema audio tracks to be delivered, there is a need to revisit and reconsider all of the implications and expectations that come along with it.

The first and perhaps most important aspect of digital audio to consider is the fact that digital storage and playback systems tend to be very linear over their entire dynamic range. Digital systems are not prone to soft compression and gentle distortion onset as they get pushed towards the limits of headroom, like analog tape used to be; instead they act just fine and reproduce signal exactly as they are fed right up to the very top of their range, "Full Scale Digital" (hereafter referred to as FSD), beyond which they tend to have a very hard and unforgiving clip behavior, as it is simply impossible for a digital audio system to reproduce a signal that exceeds FSD.

And it doesn't matter whether we are talking about 16 bit or 24 bit data—FSD is FSD, the bit depth only effects the sample level resolution below FSD, so while a strong case can be made that 24 bit is more sonically accurate than 16 bit, and that the noise floor (specifically quantization artifacts) are much lower on a 24 bit stream than a 16 bit stream, the upper bound of the dynamic range is the same in either case.

While at first these seem to be radically disparate levels, they do make more sense if the phonographic soundtrack program is measured with a PPM, since 6dB down from 100% modulation on a noise signal with a 14dB crest factor should correspond to an RMS value of roughly "10% of full modulation."

Since we know that peak swing hitting FSD is now 100% of full modulation, we should be able to dial in our reference noise to 10% of (or −20dB from) that level, but even this is slightly more complicated than it first appears.

Consider a single tone sine wave which is operating at a level which just barely touches FSD but does not clip (*Figure 1*). This would seem by definition to meet the "100% of full modulation" criteria, however, when this same sine wave is measured using an RMS meter, the value will be 3 dB below FSD. This is because a sine wave's level measurement is most commonly based not on its peak value, but rather on the RMS value, which for pure tones just happens to be 3dB below the peak, another way of saying this would be to state that a sine wave has a 3dB peak to RMS value or "crest factor."

And accordingly, when this 100% modulation signal has its level reduced to 10%, or −20dB, the resulting signal would now have a peak value of −20dB but an RMS value of −23dB when referred against FSD.

Figure 1

The entire discussion can become problematic when we shift from talking about single frequency sine tones and consider instead pink noise.

In the case of pink filtered random noise, the crest factor is no longer a fixed and predictable 3dB, but instead it varies over the course of time, and depending on several different factors, a pink noise signal might have a crest factor ranging anywhere from 10–16 dB or even more.

In order to work with numbers that accurately relate to the power content of the signal, we will need to use its RMS value, which in the case of a digital data stream conveniently equals the standard deviation of the sample value distribution which makes up the data stream.

This way of discussing signal levels also helps us more accurately evaluate power requirements for the space, and various system elements needs as well.

Since we know that the 10% modulation of a sine signal is −23dB RMS, we can measure the pink noise with a true RMS meter and set it to this same value as the sine signal.

It seems that reference noise signals have been created and put to use without fully considering some of these issues, as the RMS and crest factors vary quite a bit between different so-called "reference" noise signals.

In an effort to provide some clarity to all of this discussion, Meyer Sound has created a pair of "reference" pink noise signals for two distinct purposes.

The first which we have been calling our "−20dB" pink noise is set with a fixed crest factor of 12.3dB and an RMS value very close to what we feel is the correct figure when including sine wave crest factor at 100% modulation and reducing that level by 20dB. Which is to say the RMS value of our "−20dB" signal is, in fact −23dB referred against a peak value at FSD.

The following graph (*Figure 2*) shows the crest factor of eight different "reference" noise signals with the RMS value relative to FSD being on the left hand side and the peak values on the right of each color bar, and clearly very few of them are compliant with the −23dB FSD value that we should expect for a signal which is 10% of full modulation (the TMH "Hollywood Edge" reference noise being the closest next to the Meyer −20dB pink noise, the only reference noise signal we have measured which meets this definition.)

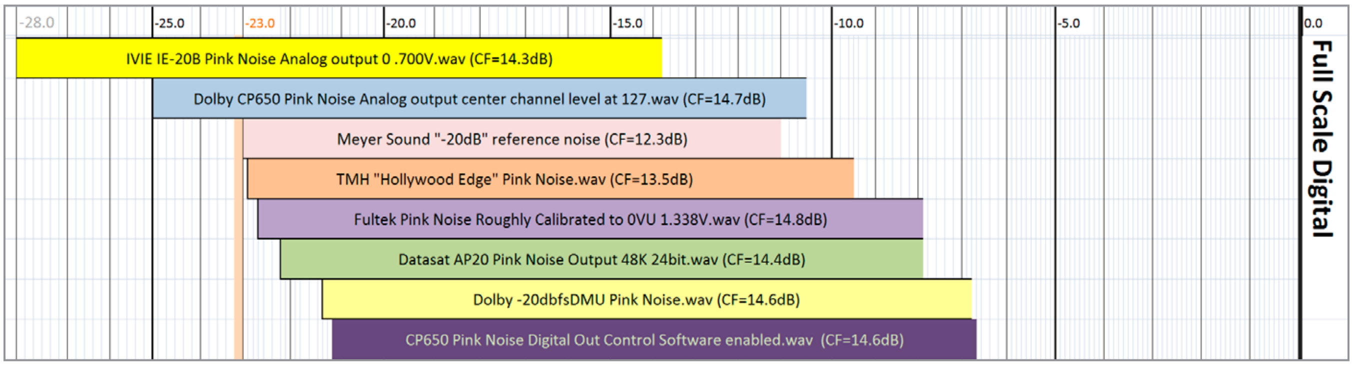

Figure 2: RMS to Peak Ranges of Various "Reference" Noise Signals

Regardless of which reference signal is chosen, there is some amount of signal swing which these calibration reference signals do not even exercise. Conversely, the selection of calibration signal will also result in different peak SPL capabilities for the system when driven to peaks that approach FSD.

This means that simply applying the calibration reference signal does not give us any information about how the system is going to behave in response to signals which might make use of this unused headroom, and that can be a serious concern if a system is constrained by power compression or clipping at levels below FSD but above the peaks inherent in these reference signals.

In order to fully exercise the full headroom and peak capability of the system, we have also created what we call the "−12dB pink noise" (*Figure 3*) which has the same 12.3 dB crest factor as our "−20dB" noise signal did, but now with an RMS value which is actually −12.3dB from FSD.

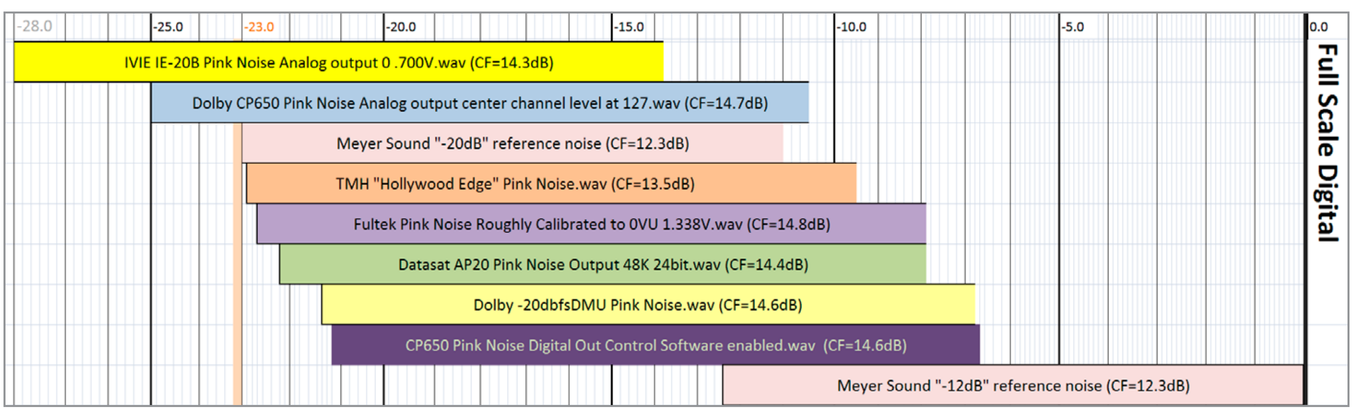

Figure 3: RMS to Peak Ranges of Various Noise Signals

So this signal does in fact use the entire dynamic range of the system on transients, and provides an opportunity to measure the transfer function to verify the system response is stable and not being degraded by power compression or clipping, while at the same time measuring the system output to look at both average and peak values allows us to confirm that the 12 dB crest factor is also preserved at this highest level of output.

This difference in RMS values for these different "reference" signals creates a situation where the selection of the calibration signal will influence your apparent calibration when one of the other signals is applied. This next chart (*Figure 4*) shows if we calibrated a theater to the true −20dB ref noise (−23dB rms), and if the mix rooms are calibrated to the other reference signals, we would measure the sound to be louder or softer depending on the reference chosen.

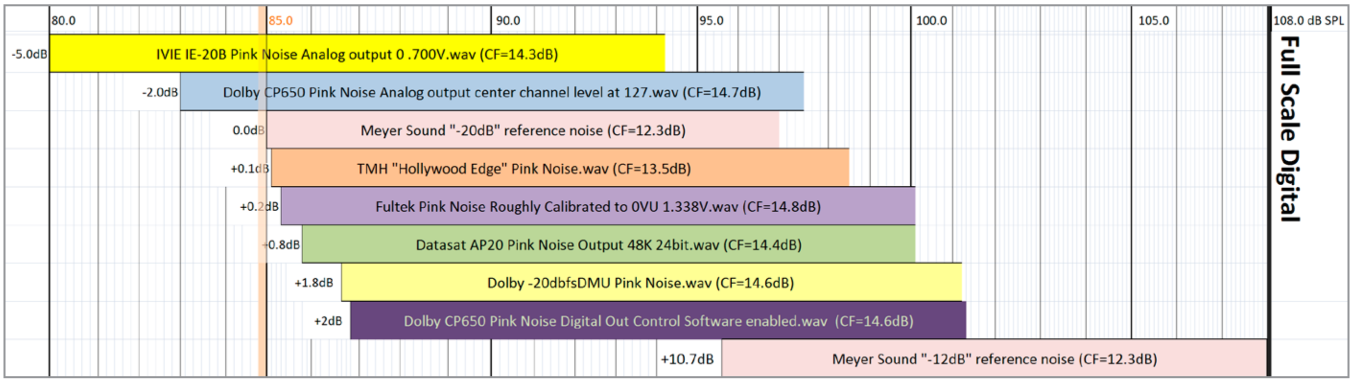

Figure 4: Comparative Expected RMS and Peak Playback Levels between Various "Reference" Noise Signals

With these two signals and the measurement techniques described above, we are able to calibrate systems to conform with any of these standard levels, as well as verifying the stability and performance of the system through the full dynamic range—all the way to FSD.

We feel that this is an essential aspect to consider in cinema system calibration, so that the full dynamics of the cinematic presentation's audio content can be translated to the listener in any high quality, linear playback system just as the content creators intended when they mixed it on the studio soundstage, without level alteration by the B-chain playback system.

Resources for Further Reading

IEC 61672: 2003 (Sound level meters)

IEC 60268-10 (Peak Program meters)

SMPTE 202-M (B-chain electro acoustic response)

SMPTE RP-200: 2012 (B-chain SPL calibration)

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