AUDIO WAVEFORMS & METERS

Mark Yonge MIBS attempts to demystify the subject of audio metering.

You may have noticed that audio metering is no longer a local matter, for local engineers. It hasn't been for years. The business of sound recording is now increasingly international so that audio metered using one convention must be acceptable when using another. There are already signs that, in the UK, delivery requirements are moving from comfy British tradition to newer frameworks.

Audio level metering tends to be accepted as a fixed thing; it isn't usually questioned. Sound engineers rely on a fixed reference for what they do while concentrating their efforts on the more interesting variables of programme production. Unfortunately, deeply-ingrained traditional practices in different parts of the world are different, but their users frequently use similar words to describe quite different phenomena. In consequence, terms like 'peak' and 'zero-level' are often used without qualification, despite the fact that their vagueness could make them worse than useless in any interchange discussion.

There's another reason to be interested in metering. Increasingly, we are expected to use equipment that is built for a generalised international market and simply doesn't have the metering that a UK professional, for example, might expect. Some considered translation and adaption will be necessary!

In what follows, we are considering the amplitude of a single audio channel. Other forms of metering for stereo and surround signals, or special metering for 'loudness' will have to wait for another article in another edition of *Line Up*.

The Audio

There are two reasons for using a programme meter: firstly as a technical level gauge, and secondly as a guide to consistent balance. Most of us do a bit of both these things, but the applications are really quite separate. I believe that the usefulness of the second flows from the success of the first, not the other way round.

A real audio waveform is a naturally spiky, and often asymmetric, thing. You can see this on an oscilloscope or by looking at the waveform display in your favourite hard disk editor. Even comparatively low-frequency sounds, like speech, can have momentary waveform excursions towards the extreme.

Programme meters will typically show a more-or-less smoothed version of the waveform and will not show extremes at all. The difference between the indicated meter peaks and the actual waveform peaks needs to be consciously inferred, but is not well understood.

Programme meters traditionally related these high and low limits to some nominal mid-point reference so that the reference signal itself would not be corrupted by distortion or noise. Today, digital alignment tones can be at any level up to the maximum without distortion, but a mid-level alignment is still useful as a means to compare meter dynamics without being confused by faster or slower meter types.

Interchanged Audio

In a single box, or a closed system, any metering system can be used without challenge provided it performs the necessary service. The need for predictable characteristics arises when audio from one system is sent to another. It makes enormous sense for the source and the destination to be able to discuss audio levels with a common basis for understanding so that the chain can be made predictable. The general term for sending audio from one place to another is 'interchange' and it really doesn't matter whether a digital audio signal is passed from source to destination as a recording on a physical medium (tape, CD-R, flash RAM), or streamed over a satellite, or uploaded to the client's FTP site via the Internet; the audio modulating the PCM coding range has the same need to interchange predictably.

Typical spiky audio waveform

On the other hand, a sine-wave tone is almost entirely unlike an audio signal. It has no spiky excursions and is very stable in amplitude. It's useless for simulating an audio programme, but handy to calibrate and confirm the gain of amplifier(s) in the audio chain.

Audio Boundaries

All recording and transmission media for audio – analogue or digital – have boundaries limiting the maximum and minimum audio levels that can be carried successfully. The limits for each medium are pretty well understood and a programme meter is intended to provide a guide to the available working space for audio. It will indicate when signals are too high with distortion a likely outcome, and when the signals are so low they may be swamped by noise.

At one time, meters were calibrated in terms of analogue line levels, expressed in dBm, dBu, or millivolts. This made sense because all these meters were built using analogue technology and they could measure an electrical signal in a way that was impractical with the recorded quantities actually interchanged. Consequently, for a long time it was convenient to use line levels as a proxy value for the real quantities that were interchanged.

However, outside the studio where the calibration was performed, the proxy relationship becomes unreliable. These in-house habits are hard to break and lead to error in interchange discussions:

"What's the level of the tone on this tape?"
"Zero level."
"Oh. What's that?"
"Err, 0.775 volt?"

Analogue Interchange…

Medium	Interchange Method	Interchange Units
Analogue tape and film	Physical	nWb/m magnetic flux
FM radio (and TV sound)	Wireless transmission	% FM deviation
Analogue optical film sound	Physical	% Variable-area optical track width
Analogue disc	Physical	cm/s groove velocity

See what I mean? The level on a tape could have been correctly expressed in nanoWebers per metre, or decibels relative to full scale for a digital tape, but never in volts. The misunderstanding shows a once-useful craft tool now sadly divorced from engineering reality!

In recent years, analogue media have faded and digital audio interchange is now dominant. Unlike analogue interchange units, digital interchange units are easy to measure directly. It's also convenient that, although the physical recording media can vary a lot, almost all digital audio interchange is referenced to a single PCM coding scale. Digital inputs and outputs, such as AES3 connections, use the PCM coding range directly, while analogue-interfaced equipment uses PCM at the convertors. Recorders using data-compressed formats, such as MP3, AAC or MPEG, are referenced to PCM external to the codec. This means that we don't have to wrestle with the notion of special signal levels in differently encoded forms. It's sufficient to measure the PCM signal.

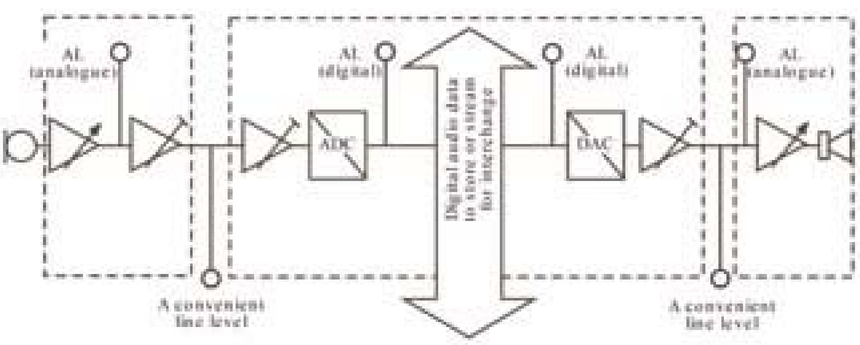
All PCM levels are referenced to the maximum coding level of 0 dB FS (full scale). This is the same whether using 16, 20, or 24 bits of digital resolution – the higher resolutions simply add more Least Significant Bits to the digital sample to improve the noise floor. Because audio signals are now interchanged almost entirely in the digital domain, I suggest that the primary reference for audio levels should clearly be a point in the PCM coding range, expressed in dB FS. Proxy values in dBu should only be considered as a local expedient.

From time to time it's good to ask the question, "What exactly are we measuring?" It's easy to get suckered into believing that what's displayed on our meter is directly relevant to the job we're trying to do. We need to be careful, because the quantities we meter may not be the quantities we interchange in the recording.

All types of audio meter to date have measured the amplitude of the audio waveform using various meters of defined characteristics. All of them provide the technical level gauge, but rely on expert interpretation to complete the judgment of audio balance. Where all users share a common set of rules to guide their judgement, the quality of interchange is good. When programmes move across cultural boundaries, the quality of interchange tends to deteriorate.

The meters we use today derive from mostly-sane decisions made many decades ago. The fact is, though, that most programme metering in current use was originally designed using electro-mechanical movements and an analogue electrical input. Because we still use those meter characteristics (even if often within a digital emulation) an understanding of those meters can be illuminating.

As we have already seen, a real audio programme is AC and can vary hugely in shape and symmetry. A moving-iron meter can measure an AC signal, but is too slow to be responsive enough for audio. A moving-coil movement is lightweight and fast-responding, but can only measure a DC signal. However, rectifying the audio waveform produces DC to drive a moving-coil meter; either to make a needle waggle, or to make a mirror waggle a light beam on a ground-glass scale. In the early-to-mid



A schematic programme chain. Programme metering relates to Alignment Level throughout

…and Digital Interchange

Medium	Interchange method	Interchange units
Digital audio tape: (DAT, DTRS, DASH, ProDigi, etc.)	Physical	dB FS
Digital videotape: (Many variants)	Physical	dB FS
Computer files: (BWF, Wave, AIFF) stored on hard drives, optical discs (CD-R, DVD), data tape, flash memory cards, USB memory sticks, Internet servers, etc.	Physical, network, Internet	dB FS
Streams: (AES3, S/PDIF, IEC 60958, AES47)	Connection, network	dB FS

1930s, rectifiers were a leading-edge technology and design choices were not trivial. Two ways to produce a rectified audio signal emerged.

A design based on a valve precision rectifier followed by a valve logarithmic amplifier was clever but achievable. Thermionic valves were very expensive in the 1930s when these meter designs were standardised, but it was not expected that you'd need many of them – one or two per Control Room, probably – and this approach led to the BBC and DIN PPMs. An alternative technology was invented in the USA in 1927 – the copper-oxide rectifier. This remarkable solid-state device had limitations in terms of voltage handling, but they didn't matter in audio applications and the simplicity of manufacture offered a cheap but rugged solution that could be built in comparatively large numbers. This is the SVI better known as the VU meter. The historical development and the characteristics of these disparate meter types is discussed at depth in a separate document available in the 2008 archive section of the *Line Up* website.

So, What Does 'Peak' Mean?

Not much on its own. You need to be very clear about what sort of peak you mean – other people's assumptions will sometimes be different. True waveform peaks are clearly different from PPM indications, which are different to the peak indications on VU meters. For a typical, real-world audio signal, the true waveform peaks will be 6 to 8 dB higher than the PPM peaks, and much higher than the VU meter indicates; all depending on the character of the audio in question.

It's not widely known that the analogue waveform can actually exceed the digital coding range. If the peaks of the waveform fall between two samples, then when reconstructed the waveform will be higher than the samples on either side. If these samples are already at 0 dB FS, the reconstructed waveform will exceed maximum coding level. Clipping distortion will result if that waveform is subsequently re-sampled, perhaps in a sampling-frequency converter or oversampling or delta-sigma D-A converter. In practice, this only happens at high audio frequencies or with impulsive sounds. (See AES-R7-2006).

What About Headroom?

Headroom simply refers to the unused and available space above signal peaks, but the term is almost meaningless without a careful definition of 'peak.' Headroom is a metering issue. Headroom above a meter indication does NOT imply that the 'unused' space is empty – it will be more or less full of the fast-moving waveform peaks that the meter is unable to indicate.

Trained ears provide us with the ability to produce a consistent listening level with reference to our programme meter readings, regardless of whether we use all the dynamic range available or not. In CD mastering, by contrast, the headroom above waveform peaks is ruthlessly minimised by 'normalisation' and/or the use of compression, regardless of whether a consistent listening level is produced.

International Exchange of Recordings

The need to resolve different programme-level measurements is not new. The EBU and the ITU have been working for years to resolve such conflicts. The audio Rosetta Stone is to be found in ITU–R Recommendation BS.645, 'Test signals and metering to be used on international sound programme circuits.' This sets out some formal definitions that pin down a working relationship between all conventional programme meters in broadcast use, based on a sine-wave line-up tone.

Term	Abbr.	Definition
Permitted Maximum Level	PML	The level of a sine-wave equivalent to the permitted maximum programme-signal indication of a PPM. The programme sound should be controlled by the broadcaster so that the amplitude of the peaks of the programme signal rarely exceed the peak amplitude of a sine-wave signal at the permitted maximum level. (This meter mark is traditionally used as a line-up reference level in DIN territories.)
Alignment Level	AL (*)	The level of a sine-wave signal 9 dB below a sine-wave at permitted maximum level. This level can be used to align sound-programme circuits and equipment.
Measurement Level	ML	The level of a sine-wave signal 12 dB below a sine-wave at alignment level (and 21 dB below a sine-wave at permitted maximum level). This level can be used for measurements at all frequencies.
Maximum Coding Level		The level of a sine-wave whose peaks can just be accommodated by the full coding range of the digital system in use.
Full Scale	FS	The full range of numbers available in a digital coding system. A sine-wave at maximum coding level is 0 dB FS

(*) For each type of programme meter, there is a scale marking corresponding to Alignment Level. This is the only metering point that is consistent for all programme meter types. It is directly related to a digital reference level expressed in dB FS.

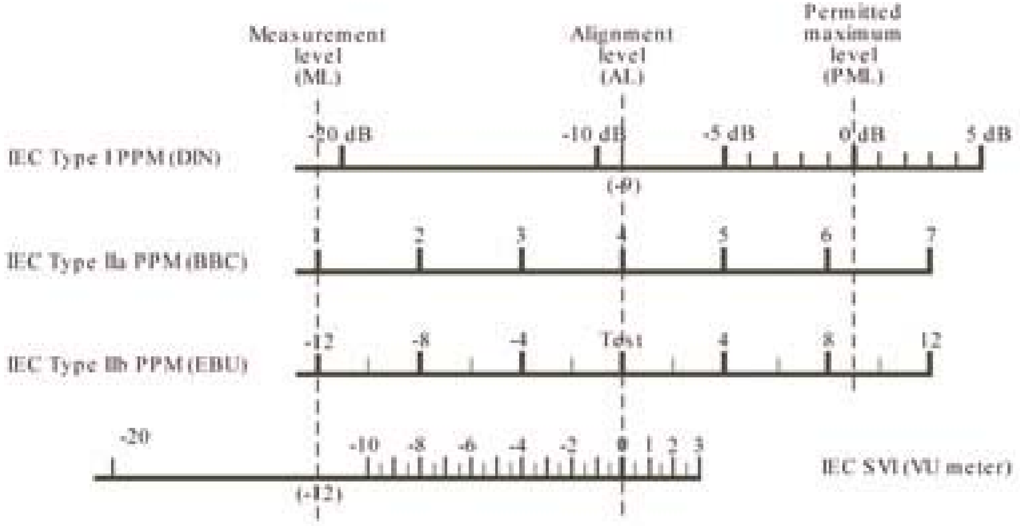
Note that not all transmission chains have a flat maximum headroom frequency response. Many analogue broadcast systems, and some digital ones, use pre-emphasis to boost high-frequency (HF) signals for the transmission so that the receiver can attenuate them again and thereby reduce HF noise introduced in the transmission (TX) path. This has the effect of reducing the TX headroom at high frequencies. A meter intended for use in such systems should probably show this HF headroom reduction in its overload indication. National Geographic's delivery specs, for example, currently require a 75-microsecond pre-emphasis in FM transmitters to be respected. This will be something to think about when specifying meters for a postproduction facility.

Digital Reference Level

Unfortunately, there are two of these. EBU R68 defines a digital audio reference level to be -18 dB FS. SMPTE RP155 defines a slightly different digital audio reference level, -20 dB FS. It is worth pointing out that, when these two figures were chosen, they were felt to represent the meter line-up level such that instantaneous waveform peaks with real programme would be safely contained within the PCM coding range without clipping.

It's my personal, if empirical, view that a faster-acting meter (PPM) with 18 dB headroom and a slower-acting meter (VU) with 20 dB headroom will tend to fill the dynamic range in a very similar way. This view will be contentious for some who will

Meter scale comparison for line up



see the 2 dB difference between EBU and SMPTE reference levels as a systematic error that always needs fixing.

Remember the golden rule:

Alignment Level = PPM 4
= DIN -9
= 0 VU
= -18dB FS

Be Ready

Don't assume that the metering techniques you learned at your mother's knee will be all you need to know in your career. Metering in the wild may not use standard meters at all. Don't assume it's a VU just because the scale says 'vu.' Don't assume it's a PPM when it says 'peak.' Check it using test signals. Where possible, use a meter you know and trust and use the VTR meters for line-up only.

When using new equipment, check that it's set up as you would want and expect. Carry a suite of test tones on CD (or any other medium that's appropriate). If you're using an analogue meter with a digital recorder, be sure that the meter Alignment level is set to correspond to the correct digital alignment level.

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