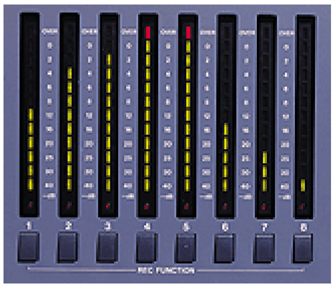
technique FAQ
METERING

meter rules

FREQUENTLY ASKED QUESTIONS: METERING



Audio metering is one of the most confusing and complex aspects of sound recording. Technical Editor **Hugh Robjohns** answers some of the most common questions on the topic.

There are literally dozens of different audio metering systems in common use around the world — and they often appear to read completely differently when supposedly displaying the same audio signal! However, there are perfectly good reasons why this should be the case and the differences are mainly due to the historical development of the various metering systems and their interpretation. Having said that, not all meters are equal and it's still a case of 'horses for courses' when choosing which system to use in particular applications.

Q What are the meters really for?

All audio material has a certain dynamic range — the difference between the highest and lowest acceptable levels. We typically arrange for the loudest peaks to be below the maximum level which the system can handle and for the quietest signals to be kept well above the noise floor. If signals roam beyond these boundaries then your ears will usually tell you something is wrong, irrespective of whether you are using analogue or digital systems. However, metering can help to make the process of setting optimum signal levels much quicker and easier, warning you of potential problems before they occur.

Beyond the technical considerations of avoiding overloads and maximising signal-to-noise ratio, the majority of level meters found on recorders and consoles are really only intended as an *aid* to balancing sound levels — the human ear should always be the final arbiter because, self-evidently, if it sounds right it *is* right! Simply matching peak meter levels between different sources (especially recordings made at different times and in different studios) certainly won't result in a consistent, balanced sound — and I know of several blind audio engineers who can balance and control programme levels better than many sighted engineers without the benefit of level meters at all!

Q What's the difference between VU and PPM meters?

RTA & Level-history Metering

It is often important to know how the programme level varies in different parts of the frequency spectrum, and this is the role of the real-time spectrum analyser or RTA. Just as with programme level meters, there are many different measuring standards for RTAs covering meter ballistics, numbers and widths of the separate measuring bands, and many other parameters. Another class of meter shows the history of programme levels — how the signal level has changed throughout a track or programme — and there are specialised metering systems which have evolved to provide this data, The VU (Volume Unit) meter is amongst the simplest of meter designs, and it has been used since the very beginning of the audio broadcasting and recording industry. It was designed to display an approximation to the RMS voltage level of electrical signals — RMS ('Root Mean Square') voltage is a complicated-sounding engineering measure of the average voltage level of electrical signals. For a sine-wave tone, a VU meter does give a true reading of RMS voltage level, but with complex signals, such as most audio, this approximation is less accurate, and the VU meter will usually read slightly lower than the true RMS value. However, it still provides a useful tool for most practical recording tasks.

too.

Because the VU meter measures 'average' levels, a sustained sound reads much higher than a brief percussive one, even when both sounds have the same maximum voltage level: the reading is dependent on both the amplitude and the duration of peaks in the signal. In addition, the standard VU response and fallback times (around 300 milliseconds each) exaggerate this effect, so transients and percussive sounds barely register at all and can cause unexpected overloads.

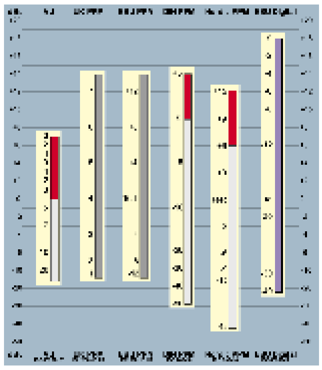
VU meters are inherently cheap, though, whether in the form of a moving-coil meter or as a bar-graph of LEDs. This is principally because there is no complex peak-sensing driver circuitry involved — as a consequence, VU meters tend to be used in order to cut costs where there is a requirement for a large number of meters, or where the meter needs only to provide an indication that sound is reaching a particular channel (such as on a multitrack recorder or large console).

Occasionally, you might notice the VU meters on different equipment reacting differently to an identical audio signal, particularly when professional and budget units are used side by side. This is because, though VU meters are supposed to be sensitive to both the positive and negative half-cycles of audio signals, many budget units are sensitive only to one half of the waveform. This can lead to considerable differences between VU readings, as many audio signals are asymmetrical.

'Peak Programme Meters' or PPMs are considerably more expensive than VU meters, partly because of the much more elaborate circuitry and partly because of the precisely defined characteristics of the physical meter itself. Yet even PPM displays aren't designed to catch the very fastest of transient peaks, and are often termed 'quasi-peak' meters for this reason. They only show transients which are sustained for a defined time — the specifications state that 'Type I' meters have an integration time of 5mS, whereas 'Type II' meters have double this figure. The result of this is that the levels of transients will usually exceed the PPM reading by between 4dB to 6dB. This design encourages overall programme levels to be driven slightly higher (giving better signal-to-noise performance) and assumes that overloading the briefest transients will be inaudible — a fairly reasonable assumption in most good analogue audio equipment. The differing integration times of the Type I and II meters simply reflect alternative opinions on the audibility of transient distortion.

PPMs are also characterised by a slow fallback from displayed peaks, which is intended to make it easier to register the peak level visually — Type I meters should take between 1.4 and 2.0 seconds to fall 20dB whereas Type II meters should fall 24dB within 2.5 to 3.1 seconds. Furthermore, Type II meters also incorporate a delay of between 75mS and 150mS before the fallback occurs — effectively a peak-hold condition — which helps reduce eye fatigue.

In an attempt to combine the best aspects of both VU and quasi-peak meters, some bar-graph level displays are available with a VU response shown as a solid bar, accompanied by a floating dot above it which registers the PPM level. This floating dot often has a temporary or permanent hold function to ensure that the maximum peak level is observed.



Q Why do some meters have very different scales to others, and how do they compare?

A variety of different audio metering scales have been developed by different industries in different countries, each of which optimises the display for a specific set of applications. While the VU meter has now become fairly standardised — zero point at +4dBu with a decibel scale ranging non-linearly from 20dB below this point to 3dB above — the PPM meter has a number of recognised scaling systems. Type I PPMs are available in German DIN and Nordic N10 variants, while Type II PPMs can commonly have either a BBC or an EBU version (see the diagram, right, for a comparison of these standards). As a rule of thumb, the scales for Type I meters generally display a considerably greater dynamic range than those of Type II, and their calibration encourages more of the system headroom to be exercised in normal use.

Where multiple meters are used together (such as in stereo or multi-channel systems), each meter's dynamic response must match to within a tenth of a second and their amplitude responses should be within 0.3dB of each other in the critical areas of the meter range. In order to calibrate VU and PPM meters, it's best to use a mid-frequency sine tone (typically 1kHz), as these signals are the most accurately read by meters which are not truly peak-reading.

Q How does all this relate to digital metering?

Unfortunately, the nature of digital systems is such that even the briefest of transient overloads is clearly audible, so neither VU nor PPM metering is suitable. The majority of digital recorders, mixers and converters therefore use true peak-reading meters whose displays are derived from the digital data stream. As these don't rely on analogue level-sensing electronics they can be extremely accurate.

Analogue meters all have a nominal alignment point — the zero reference — with a notional headroom above. The idea is that signal peaks are routinely allowed into the first 8dB or so of this headroom, though peaks of +12dBu will usually start to cause distortion which becomes more and more noticeable with increasing level until clipping occurs, usually at between 18dBu and 22dBu.

Digital systems, however, have no headroom above the maximum quantisation level, and therefore a notional headroom must be created by choosing a 'zero' point well below this. Digital meters are scaled such that the maximum quantisation level is denoted as 0dBFS (full scale), so the alignment level is always a negative value below this point.

In Europe 0dBu has been standardised by the EBU to be -18dBFS, in order that a signal peaking in analogue equipment at the top of the EBU-standard PPM scale — and therefore with true peaks at around +16dBu — remains a little below the digital full scale

Metering And Loudness

Although the VU meter was designed to provide some indication of volume, level meters in general display information about signal voltages rather than their perceived loudness. This is why it is important to realise that meters are only an *aid* to judging the acoustic balance of audio material. However, there are specialist metering systems designed to measure and display the absolute loudness of a programme, taking into account the value. Just to be awkward, though, the American SMPTE organisation set their standard for 0dBu at -20dBFS instead...

These calibrations assume uncontrolled source dynamics where unexpected transient peaks might use the full dynamic range available. However, in post-production and mastering situations, where programme levels have been carefully controlled, these standard alignments typically cause most mixes to peak only below -10dBFS. Consequently, there are strong arguments for adopting a different alignment strategy in these circumstances — I generally align 0dBu to -12dBFS for mastering work, for example. However, there is as yet no standard calibration specifically for post-production applications.

characteristics of human perception. This kind of metering is becoming increasingly important as broadcasting organisations are now transmitting hundreds of channels via satellite, cable and the Internet and it is impossible to monitor all of them acoustically. Also, with the growing use of sophisticated multi-band compressors, it is possible to create audio material which appears completely normal on quasi-peak meters yet sounds extremely loud. This is starting to cause problems at programme junctions and, in the cinema, has lead to an increasing number of complaints about excessive playback volumes. Responsible post-production houses are starting now to monitor and regulate the true perceived loudness of films using special loudness meters. The most familiar programme loudness meter is, probably, the Dorroughs unit, but Thames Television and the ITC have also come up with a specification for displaying the relative loudness perception of typical audio programme material.

The perception of loudness depends not only on the level of a signal, but also on its frequency and bandwidth — the wider the bandwidth, the louder a signal seems to be, even if its peak level remains constant. The ear is known to be most sensitive around the 2-4kHz region, so signals in this frequency band will sound much louder than low- or high-frequency signals of similar peak level. For example, band-limited 1/3-octave noise signals at 100Hz and 10kHz can be almost 15dB higher in level than noise centred on 4kHz, yet all will be perceived as sounding equally loud!

Q What is the significance of the 'Over' light on a digital machine?

Over indicators are found on A-D converters, mixing consoles and some digital meters, and are supposed to illuminate when the signal exceeds the maximum quantisation level. In the case of the A-D this is when the analogue input is greater than the available quantisation range and, on a digital console, it is when some signal-processing operation results in a sample value larger than the maximum quantisation level. Both situations are clearly defined, because the excessive source signal can be directly compared with the quantisation level, and are therefore entirely valid situations in which to light the Over indicator.

However, the Over light can often be less meaningful when you are merely metering already-digitised audio without the benefit of the original source as a reference. The only way in which most digital meters can detect overloads in the audio data stream is by watching for consecutive samples with the maximum quantisation value. Commonly, four consecutive maximum-value samples are interpreted as an overload, but some meters include an option for the user to specify this number.

This manner of interpretation is ambiguous, however, because four maximum quantisation values in a row does not necessarily imply an overload. A peak-level low-frequency signal could easily create four maximum-value samples quite legitimately, whereas a high-frequency signal could be severely distorted with just a couple of peak-value samples. Most decent stand-alone meters use oversampling techniques to provide greater accuracy in their calculation of what represents an overload within the digital data stream, but even this is not foolproof. If in doubt, set the Over light to activate with a single full-scale value — even a single peak-value sample means that you're awfully close to overloading.

Q What is a phase meter?

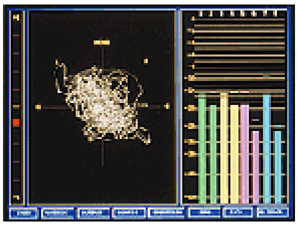
When monitoring a stereo signal, the coherence between the two channels (ie. how similar they are) greatly affects its mono compatibility. The obvious worst case is when the two channels carry identical signals with opposite polarities — summing the two channels to derive a mono signal would therefore result in silence!

The phase meter was developed to provide an indication of the relative phase of the two channels and thereby provide some measure of mono compatibility. There are no specific standards defining the characteristics of phase meters, but a common scale has nevertheless evolved. This shows same-polarity signals at the right-hand end of the scale (marked '+1' or sometimes '0 degrees') and opposite-polarity signals at the other end (marked '-1' or '180 degrees'). Phase meters should be reasonably independent of input signal level, and should maintain an accurate phase reading even with both input signals as low as -20dBu.

In general, phase-meter readings above zero and in the positive half of the scale indicate acceptable mono compatibility, whereas negative readings warn of a potential compatibility problem. The ballistics of this type of meter are undefined, however — some flit about quickly whereas others take a more leisurely approach.

Q What does an audio vectorscope show?

The audio vectorscope is, in principle, an oscilloscope where the left and right sides of a stereo signal modulate the position of a dot along the display's X and Y axes respectively. The resulting two-dimensional pattern, called a Lissajous figure, is characteristic of the amplitude, frequency and phase relationships between the two signals. Most audio vectorscope displays work like this, though usually the X and Y axes are rotated by 45 degrees in order to provide a more easily understandable correlation between the displayed image and the stereo positioning of the audio signal — identical signals on both channels will produce a vertical line (representing a central, mono signal).



Most vectorscope displays are relatively poor at providing accurate absolute levels, so they are frequently used in conjunction with conventional bar-graphs. However, the vectorscope display does provide a wealth of information about stereo source positioning and the relative phase and level of the two channels. It takes a little while to learn how to interpret a vectorsope image, but once mastered it is far superior to any other stereo metering system.

The image takes on recognisable characteristics and shapes when the audio signals are sourced in particular ways. For example, stereo recordings made with spaced mics are very easy to distinguish from ones made with coincident pairs, or multi-miked recordings. Alignment errors between channels and numerous other subtle fault conditions become extremely obvious on an audio vectorscope well before they become audible to most people.

Q What on earth is a jellyfish display?

The challenge of monitoring stereo signals is trivial compared to that of monitoring multi-channel surround sound. Adjacent bar-graph meters for 5.1 surround monitoring provide information about relative signal levels, but are hard to interpret in terms of the spatial positioning of a surround signal.

The 'jellyfish' display was designed to address this problem, and has a number of notional loudspeakers which are depicted in appropriate positions on the screen. The surround signal is drawn as a moving outline, the size and shape of which relates to the amplitude of the signals destined for each speaker output — the greater the signal on any channel, the closer the trace moves towards that speaker icon. This results in 'fingers' stretching towards the various speakers, resembling some bizarre animated jellyfish — hence the name of this display mode. SOS

Glossary

http://www.sospubs.co.uk/sos/regular_htm/glossary.htm

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