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DIGITAL SOUND LEVEL METERS, A CRITICAL EVALUATION

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1. BEFORE DIGITS

In most fields of physical measurement, converting from an analogue display to a digital one is simple, but in sound level meters, or slm, the conversion was a long process. To understand the evolution of the digital slm, an understanding of the concepts and the history of the analogue unit is helpful and this is briefly discussed. The slm attempts to measure twenty times the log of the root mean square of the weighted ac signal from the microphone. Classic slm did not in fact make this measurement, instead they gave a reading which was an approximation to the rms curve and generated the log by having a log shaped scale. The rms approximation was usually performed by either a series of semiconductors or by a 'lossy' full wave rectifier which gave an rms curve within the very large tolerances of the slm specifications IEC 123 and IEC 179. Most 'rms' circuits had the disadvantage that no output resulted from very small inputs and this, together with the log shaped scale limited the readable dynamic span to a typical figure of about 16dB. Most instruments would only meet the full linearity requirements over a very limited temperature range and this was usually marked 0 to +10 on scale with the less accurate portion being below scale zero. Clearly, to simply replace the analogue meter with a digital voltmeter was not practical and several steps were needed before this could be done. The most vital was to include a log converter in the instrument such that the scale was linear in decibels and at the 1974 ICA, such devices were shown but this was far from the only problem. The Slow (S) and Fast (F) time constants in slm which should be the rms averaging time, were usually limited by the physical speed of the meter needle. Typically, the rms circuit had a faster time constant than was correct and the mechanics of the needle brought the response time down to the requirements of IEC179. This often resulted in instruments which would meet the 'A' weighting specification, but on 'Lin', the low frequencies had an additional drop-off due to the too small integrating capacitor in the rms circuit. By 1975, the basic problems were well understood and circuits which could perform true rms and true log were available.

2. EARLY UNITS

The first digital slm were large and heavy with LED displays. They were simply existing designs with rms-log circuits added. The first International standard which accepted that such devices existed was IEC 651 dated 1979 which paid lip service to tests on a digital device. It was obvious that a digital slm on 'F' response was not an easy device to use and to allow it to be pulse tested, a 'hold' function was included, which locked the maximum reading during the measuring period. While the digital slm has been improved and liquid crystal displays have replaced the LED, many current units are still made in the same way, even though they are difficult to use for untrained users - the very users who demand them because they are 'modern'. Manufacturers have succumbed to the customer demand for digital units, but many analogue units are still available. This early technology will probably continue for another decade before fading away.

When the concept of Leq became fashionable, the digital slm came more into its own. For most signals, Leq becomes more and more steady and SEL slows its rate of rise with increasing measuring times and for these instruments a digital display is better than an analogue one. The resolution of a digital display is usually 0,1dB and unless an analogue scale is very large, it cannot match this. With the realisation that the exponential average of sound level could be easily calculated from the short linear integral, or short Leq, the circuitry of conventional slm and integrating-averaging slm, or Leq meters as most people call them, became more common and most Leq meters included a sound level function and digital displays became the norm.

3. DIGITAL CIRCUITRY

By the late 1980's while digital read-outs were overtaking their analogue cousins, the circuitry was still analogue based. This is not to say that there was not a digital processor in the system. All short Leq meters used an embedded digital microprocessor to process the final Leq dc signal after quantisation. Mainly the signal being processed was the log-rms signal that would on a simple instrument go to the display, but was now passed to an A to D converter and thence to the processor.

Figure 1 Leq meter block diagram

The chain of events after frequency weighting is shown in figure 1. The absolute value of the signal is first squared by a 2log-antilog converter, then integrated and finally passed to a log divider, generating Leq. In instruments from classic acoustic companies, this process was mainly analogue, not only because an analogue solution was more in keeping with their experience, but also because International standards could be met.

Newcomers to the field tended to digitise the signal further forward in the measuring chain and points 'A', through 'D' in figure 1 show the various logical points which can be used. Each of these has advantages and problems. For example, if the signal is digitised at point 'D', the voltage ratio is twice that of the original signal as the signal has been squared. However, digitising at this point can ease a significant temperature drift problem in the integrator and final analogue log divider, a limiting factor in some designs. Once the signal is digitised at point A, there is really very little analogue circuitry left, except the weighting and amplifying circuitry and the stage was set for an 'all digital' approach, the first of which was the 'card in a computer'.

4. COMPUTER CARD SYSTEMS

It was inevitable that someone would use an A to D converter in a personal computer to make an acoustic measuring system and from about 1982 it was possible to make such a system. The signal was normally provided by a conventional slm and read to 0,1dB resolution then stored and displayed. The limitation of these systems was that the A to D converter was not capable of sampling the original ac signal for reasons of both speed and dynamic span. Using an already logged dc signal removed both these problems and compliance with standards was mainly a function of the slm used as an input device.

It was not long before the slm was dispensed with and a card fitted inside a pc with simple amplifiers and a fast 'sample and hold' A/D converter. The card could sample the data, typically at 44,1kHz and could then feed these samples to the main cpu where the data could be used to generate the 'F' and 'S' response as well as true Leq or SEL. However, while possible, this was not a sensible use of technology, so usually an FFT algorithm was used to give the frequency information in the form of a classic FFT or third octave bands. Clearly such a system has many advantages over the classic 'mahogany box' solution of a dedicated instrument, the main one being flexibility at lower cost. There were and remain many disadvantages. Initially, the speed of the IBM PC was inadequate for many applications and even today, many computers have peripherals which are not really fast enough for some card systems. The main technical problem however was that the resolution of the available cards; a maximum of 16 bit is not adequate for the acoustic world. Users had been used to 120dB dynamic spans and the sub 50dB of a 16 bit card was therefore inadequate. Even today, very computer literate engineers insist that the resolution of a 16 bit processor is 20 log $2^{16.}$ In acoustics, this of course is invalid, as until a true 0,1 dB resolution is reached, the least significant bits must be discarded. Many engineers insist that techniques such as 'dithering' can allow a higher resolution to be realised, but they forget that the instrument must pass a stringent pattern approval test which uses steady level sine waves to test the linearity. None of these techniques will improve the resolution with such a signal source, all needing some 'randomness' to operate. The result is that all these single channel systems fail the tests for compliance. The resolution problem was solved by using both 16 bit channels of a stereo system in parallel with different gains so that the digital outputs could be combined to give a longer digital word .

Another disadvantage is the slow speed and limited size of any mass storage . With a 9mS, 2 gigabyte disk there are still often difficulties if the raw 44,1 kHz data is to be recorded, in other words, if the device is to be used as a digital tape recorder, when about 300 megabytes an hour are stored. If the measurement needed is known, there is no need to record the actual noise, but this negates one of the main claims of such a system. If you can store the raw data, it can be re-processed to give any acoustic index, chosen after acquisition. This is very important when the descriptor to be used will depend on the noise itself. For example, a recording of environmental noise may use Leq, L10 and L90. However, if during the night single noise events occurred, the peak level, L0,1, Lmax or a third octave event spectrum may be helpful. If the raw data is not recorded, this may not be possible. A tape recorder can be made to start if the noise goes over a threshold and this technique is common, but may not be aceptable.

A further difficulty arises with the 'card and computer'. IEC 651 requires the whole instrument to be tested from -10 degrees to +50 degrees and over a large humidity range. Clearly, if the computer is considered part of the instrument, it should be so tested. This leads to conceptual problems which can be overcome if test organisations accept that the computer need not be tested over the whole temperature range if it is housed in a controlled climate room and a long cable used to the microphone. However, the most important practical difficulty is that even today a source of ac power is needed for all except trivial measurements and this sensibly limits these systems to laboratory use, except for very dedicated computer users. Conversly, inside the laboratory, these systems will probably very soon become the norm.

5. ALL DIGITAL INSTRUMENTS

One of the main reasons for making a 'card in a computer' system as opposed to a similar dedicated instrument, is the ease and cost of development. The cost of pc software using a high level language is usually less than embedded code having a similar function. The development costs of a computer and card are spread over far more users than would be the case if the final device was dedicated to acoustics,so the computer and card cost far less than a high grade slm. However, the ease of use of a general purpose slm, is greater than a pc based system which does the same task. For this reason, designers started to look at ways of having a direct sampling slm which mimicked the 'card in a pc'.

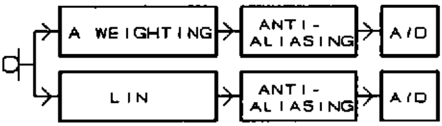

Figure 2 Twin channel unit

Current 'digital' instruments vary in the amount of digitisation. Today, it is more practical to use an analogue 'A' weighting circuit with a second channel for Lin or 'C' functions as in fig 2. This leads to a lower cost unit and if only 'A' weighting is required, one channel can be switched off to gain battery life. Some instruments simply quantise the ac signal after a simple anti-aliasing filter and use true digital filters to give the frequency weighting desired, in other words use just the lower channel of fig 2. This approach is the most flexible as within the limits of the processor used, several weightings can be generated in parallel and the device is not restricted to the two conventional weightings of 'A' and 'C'. However, the obverse of this coin is that digital filters tend to have a poorer 'in use' signal to noise ratio and use more power than passive 'A' weighting. In an 'all digital' realisation, after filtering, almost any function can be produced and typically the short Leq is generated and from this the 'S', 'F' and 'I' values are found from a simple 'look-up' table. These exponential values can be more accurate than an analogue realisation as no component tolerances are involved. Any reasonable low cost CPU can be used and low power versions exist of many designs, but if frequency analysis is needed, a DSP is usually used with an FFT algorithm and while these are still high current devices, they will soon be available in a low enough current consumption to allow use in a hand held instrument running on low cost batteries. Anything less than a 12 hour battery life cannot really be considered sensible, especially when measurements are being made in remote areas at night.

6. THE FUTURE

Probably most new digital techniques will be used first in fixed installations such as airport noise monitors where ac power is available, filtering into hand held units as lower power chips become cost effective. Typically a new microprocessor device is first developed with little regard to current consumption and only after the design has become mature, do very low power versions appear. Even then, it takes a significant time for the low power versions to fall to a low price level. In most fields, high cost, heavy, highly complex instruments have given way to mass produced, light, low cost yet accurate units and there is no reason the think acoustics will be different - it will just take longer. Today, about 50% of the cost of a Type 1 digital slm is the microphone and while work on all digital microphone designs is continuing, a true replacement for the traditional air condenser unit is not yet in commercial sight.

For the next generation of slm, particularly for Type 2, the twin channel analogue 'A' weighted instrument will probably become the most common as it will have lower cost and use less power than the 'all digital' design. Most users really do not care what technology is used, their motivation is acceptable measurement performance at the lowest cost. To the user 'cost' is total cost including batteries, added measurement time, instrument weight and ease of use. 'Acceptable measurement performance' is usually the minimum allowed by the regulations they are trying to follow and usually a type 2 unit will be acceptable. It is a pity that many designers forget this and have an 'arms race' of the latest technology. Their task should be a careful balance between what can be done in the way of 'features' and what the user wants in terms of 'ease of use', which suggests techniques like 'user configurable' instruments.

7. SUMMARY

Digital techniques have revolutionised the slm and eventually the low cost of an 'all digital' design will make the current units obsolete. For this to be reality, the current consumption and cost of microprocessors and their support chips must be reduced.

However, for many years yet, the mixed technology units will remain the most cost effective for general use, but almost all will have an embedded microprocessor.

Mid range instruments will have more and more features available and become more accurate. The very expensive, over-specified and over-weight instruments that have led the field for decades will become more and more an anachronism. Commercially successful units will be those which are simplest to operate.

The long term main drive to reduce the user cost will be the development of a true all digital microphone, as yet probably a decade away from mass production.