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FROM DIALS TO DIGITS

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INTRODUCTION

Many people say that the science of acoustics dates from 1877 the year in which the first major acoustics book Lord Raleighs 'The Theory of Sound' was published, this gave a framework for all that has followed and is still used today as a basic reference work. Others say acoustics really started during the war when MIT, Malvern and others did so much work on electronics and produced the miniature 'deaf aid' valves, making portable equipment possible. Yet others say the starting time was when transistors were first fitted into Sound Level Meters in the late 1950's and naturally people who qualified later than about 1980 are sure that acoustics only became 'scientific' after the computer chip became available.

All of these people have their point because each of these events marks a milestone in acoustic measurement. I, as an electronic engineer by training automatically assume that if you cannot measure, it is not a science but black art. I am sure that many theoretical mathematicians will disagree, but for a dirty fingered engineer measurement is everything.

In this paper, I want to look at how the measurement art has developed and attempt to show that there is far more similarity than difference in the road from dials to digits.

AN ACOUSTIC MEASURING SYSTEM

Any acoustic measuring system has 3 essential components. A Microphone a Process unit and a Read-out device, these three being present in all acoustic measuring systems from the first valve operated Sound Level Meter right up to the very latest devices which consist simply of a card fitted inside a personal computer connected to an external microphone.

Let us look at these 3 blocks in turn and see how there has been a logical progression from the early days right up to today and see if this allows us to predict where the future of acoustic measurement will lie. Also, let us see if we can predict blind alleys and areas where commercial producers have the chance of making the biggest errors.

THE MICROPHONE

In any discussion on the history of acoustics, the microphone must loom large. To many people, it is just a very expensive piece of metal stuck on the end of the system and they give almost no thought to this first element in the chain, but in reality it is the most important link. The cost of a precision grade microphone today is much more than 50% of the cost of a simple complete Sound Level Meter. In other words, the microphone costs more than all the rest of the instrument put together, why this should be so puzzles many people.

To perform acoustic measurements, the change of air pressure we perceive as 'noise' must be converted to an electrical signal by the microphone for subsequent processing. In general the performance of the microphone dictates the system performance, as the accuracy of measurement cannot be better than the accuracy of the microphone, except in a few very special cases.

All conventional microphones work by the sound waves moving a membrane. This membrane then either operates a mechanical device or changes the physical state of the system in some way. Clearly, the less the moving parts in the system, the lower the chance of mechanical resonances affecting the transfer function. The ideal transfer function for a microphone is such that it will transduce all the frequencies of interest at any pressure level of interest from the mechanical to the electrical domain.

Clearly such perfection is not practical and one measure of the microphone is how well its transfer function compares with the ideal. There are other measures such as fragility and performance in adverse conditions, but for simple measuring power, expressed as the closeness to the ideal transfer function the Air Condenser unit is supreme. Invented by Wente in Dresden in about 1920 it has not changed significantly since its invention. For many years, the major manufacturer of Air condenser measuring microphones has been Bruel and Kjaer. They have by their own efforts achieved a position in the world where in the late 1980's, they have of the order of three quarters of the world's acoustic measurement market. Many people believe that this was originally based on their microphone technology and there is a lot of truth in this. For many years, they were the de facto standard for high performance measuring capsules and with a few exceptions, other Precision Sound Level Meter manufacturers did not make their own microphones, although many tried.

Today, at least four Sound Level Meter manufacturers have their own microphones, Bruel and Kjaer in Denmark, Cirrus Research in the UK, Larson Davis in the USA and RION in Japan. As the microphone is the fundamental element in the measuring chain, having control over this is generally felt to give some insurance against errors of external suppliers. There are also at least two independent suppliers of measuring microphones and these supply those companies who do not have the capability or wish to manufacture their own capsules.

THE AIR CONDENSER MICROPHONE

The sound waves impinge on the membrane and it moves with the wave. This movement is very slight but it changes the capacitance of the device by a very small amount and we detect this change of capacity by measuring the voltage produced. Conventionally, the charge equation was satisfied by imposing a polarisation voltage on the backplate and while low voltage units are available, the most common voltage used is 200 volts. Unfortunately, the 200 volts is across a very small air gap and while dry air can support the resultant voltage stress, damp air can not. Thus in conditions of high humidity, the air starts to ionise which first leads to increased electrical noise in the system followed by total failure. To get round this, companies in a triumph of engineering over a poor concept, coated the membrane with quartz to make it impermeable, rear vented the capsule, added dehumidifiers and generally made a very simple principle very complex.

There is another principal method of obtaining a charge on a surface, the 'electret' principal and a microphone can be made using this technique. The electret condenser capsule compares to the polarised air condenser unit as a permanent magnet compares to an electromagnet. The electret or permacharged unit has a permanent charge embedded in either the membrane or the backplate and this allows the charge equation to be fulfilled, without an external polarising voltage.

The first obvious reference to an electret condenser microphone was in the late 1960's, but much of the modern work was described by Sessler and West at Bell Telephone Research laboratories in New Jersey. Naturally, the main aim of Bell was to produce a capsule for telephone use and thus price was more important than measuring performance. The work of Sessler and West led to a series of plastic membrane units which had spectacular acoustic performance compared with contemporary capsules of other technologies and this led to a revolution of microphone techniques.

The first commercial measuring electret capsules were produced by General Radio Inc. (now called Genrad) and a paper on their construction was presented at ICA Madrid 1977. The Genrad capsules still had charged membranes and while offering performance advantages in certain areas, were not quite as stable and smooth as the best polarised Air condenser units. However, Genrad were able to claim quite correctly that their unit was indeed a true measuring microphone and the advantages, particularly in terms of field performance, made them almost the de facto standard for overnight measurements, as they would stand up to high humidity which the conventional polarised units could not. At the 1977 ICA several engineers, believed to be from Denmark, objected to the claims made by Genrad for their capsule and made statements to the effect that the electret condenser unit would never replace the polarised air condenser unit. These objections no longer look so sensible.

In 1979, Cirrus Research introduced a charged membrane 1/2 inch capsule to meet the requirements of the then new IEC 651 grade 2. As production techniques improved, many units met the more stringent requirements of class 1 performance and thus a type 1 capsule was introduced which was simply a selection of the type 2 units which when new, met type 1 performance. The drift and aging characteristics were not really adequate for long term type 1 performance but the MK 202 was used by many Sound Level Meter manufacturers who did not have their own production capability, as it was the only reasonably priced unit available to them.

In about 1984 Bruel and Kjaer introduced a new electret capsule which they called a 'permacharged' air condenser unit. This new unit, while brilliantly engineered, was just a conventional electret unit with the charge embedded in a layer of polymer on the backplate and not on the membrane. Because a metal membrane could be used the technique is capable of producing a unit with a long term stability of the same order as a polarised unit. In 1986, Cirrus Research introduced a similar unit, the MK 224 with similar performance. Today, these are the only two known companies with backplate electrets but others will clearly follow as they master the new technique.

The advantages of the backplate electret techniques are many, but in particular, they do not suffer as badly as externally polarised units from the effects of high humidity. Conversely, they do lose their charge if exposed to high temperatures for long periods and thus long term operating temperatures should be kept as low as possible. Further, as there is no need to generate a high voltage to polarise the capsule the price of preamplifiers should be less. The advantages are particularly important for long term outdoor measurement where the electret will probably totally replace the polarised unit. Even today, the MK224 electret unit outsells its polarised cousin the MK221 by about 50 times.

It is clear that the electret will remain the de facto standard for measuring microphones at instrumentation level and it is only a matter of time before one of the current producers makes a 'standard' microphone in this technology.

To look ahead at other technologies it may be possible to make semiconductor capsules where the electret layer on the backplate is replaced by a layer of semiconductor material. With a high level of silicon integration, there could be a very large array of transistors on the backplate itself and in principle this can produce microphones of a much higher sensitivity that todays maximum of 100 millivolts per Pascal. The operating principle would probably be by frequency modulation of an oscillator somewhat similar to todays carrier system. However, it is difficult to see any principle that does not use a stretched thin membrane as the front element of the device. While this thin membrane has to be used, microphones will be expensive.

THE READ-OUT UNIT.

As the paper title suggests, the change from mechanical meters to digital read-out has been the most obvious change in acoustic measurement. Every individual will have a view on the relative merits of the two displays, but

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when Bruel & Kjaer decided to make only digital instruments, most of the world followed them blindly. It is clear that for a steady signal or a steady reading such as would be given by an Integrating Averaging Sound Level Meter (Leq meter), the digital read-out has no equal for resolution over the big dynamic spans used today. Conversely, there are often situations where digital resolution is not only not needed but is a positive nuisance such as reading the variation of noise of an intermittent source where 'eyeball' averaging can give much information. Also, at the type 2 level of general purpose measurements, the exactitude given by digital resolutions of 0,1dB have no meaning, an analogue display with a 30dB span is usually much simpler to use.

The first practical Sound Level Meters had scales which were between 16 and 20 dB long, with the scale being linear in volts over most of its length. This gave excellent resolution over the top few decibels but it became very poor at the bottom of the scale. In 1971, Pulsar Instruments then in California, produced a meter with a scale linear in dB and having a 30dB scale length and constant resolution over the scale. This principal was taken up by Genrad and some time later by B & K. These 'long scale' instruments incorporated a 'log converter' to generate an output equal to the logarithm of the input and this caused the complexity of the circuit to increase. For example, the prototype Pulsar meter with a scale linear in volts had 3 transistors and 2 operational amplifiers. The same unit with the scale linear in decibels had 7 transistors and 5 operational amplifiers, the difference being simply to provide a linear decibel scale. Today, as electronic components are very cheap, most analogue sound level meters use this technique unless really low price is the most important parameter.

In the late 70's an RMS LOG chip became available which was used by many manufacturers to simplify their circuitry. Today, with IEC 651, it is very difficult to use these chips and still meet the standard at type 1 level because of problems with the decay time constant, which would be too fast at the top of scale and too slow at the bottom. Thus the major companies developed their own circuitry and now only third tier companies use this type of chip in new designs. However, such chips are still to be found in units in current production from the major companies.

With rms/log converters, the read-out could be made digital simply by connecting a digital voltmeter module to the output of the circuitry. However, this phase lasted only a few years as the larger companies started adding other data to their read-out. The first major addition was a quasi analogue scale which purports to give some idea of the fluctuations of the signal. However, most of these units have a resolution of about 2 dB and this becomes little more than an attractive gadget. The 2 dB limitation comes mainly not from the circuitry but from the limitation of the number of connections it was practical to make to the edge of the Liquid Crystal Display. Today this limitation has to some extent been resolved and at least two instruments, have fully addressable matrix displays which can be addressed just as though they were a computer screen. Another read-out which was useful was the LED matrix which was used on several instruments for frequency or level display. Initially this was an attractive solution as such displays could be made without incurring a huge tooling cost. However, LED matrices suffer from the problem of poor resolution even more than the quasi analogue display of current LCD units and as well uses many times the power, thus it is clear that this route is a blind alley. Today, there are only a few units left with this type of display, mainly being used for data storing units where the display can be turned off without affecting the functioning of the system.

In future, as portable computers are fitted with colour LCD screens, it is probable that the best acoustic analysers will be fitted with similar colour displays, allowing say third octaves of Sound Level, Max level, Leq etc. all on the same display. Or, the time history of several parameters could be plotted on the same display, in fact anything that can be done today on a portable computer will probably be possible on tomorrows analysers. It is now possible to buy an LCD colour television set for about $250, while the screen used in the device when purchased alone costs about $500. As the devices become more available displays will be priced at more sensible levels. Certainly at Cirrus laboratories, we have often bought complete units to take out individual blocks to make prototypes, confidently predicting that the device would eventually become available; sometimes we were correct. One of the worst mistakes was in 1976 when it was predicted that a certain digital watch would become a market leader and thus the chip it used would become readily available. In the event, every unit made required the purchase of a digital watch from which was removed the complete works. It is perhaps just as well that our meter was not a runaway seller.

THE PROCESS UNIT.

While this is in the middle of the chain, I have left it until last. In the description of the microphone and the read-out, several references have been made to circuit problems and it becomes clear that the discussion of a measuring system can only take place on the basis of a whole system. However, it is in the process part of the chain that the biggest advances have taken place. Many acousicians however, ignore the process unit because it is not easy to see how it affects the overall instrument, being simply a 'black box' into which the microphone feeds, which in turn feeds the read-out unit.

The first simple Sound Level Meters contained valves which required not only a supply to operate the circuitry, but a second supply to operate the valve filaments. I can still remember the agonising over whether in a particular design an extra valve could be fitted, with all the ramifications of reduced battery life. Even in the mid 1960's one UK manufacturer was living so far in the past that it advertised 'Transistorised' Sound Level Meters, as though by that time there were any other sort.

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With the transistor came the situation that there was no longer a direct cost link between the number of active components, I.E. Valves or Transistors, and the selling price. Adding more transistors did not always mean adding a corresponding number of support components. The link between active parts and price became even more tenuous when during the 1960's integrated circuits started to be put into instruments. By the 1980's integrated circuits were being used to replace mechanical components such as multibank switches. Now, integrated circuits are being dropped in favour of computer chips which themselves may have about 100,000 transistors inside, performing all the housekeeping functions of the older devices.

These 'complications' came in at about the same time as the acceptance of Leq as the preferred technique for industrial noise measurement. The first practical commercial meter which was produced by Computer Engineering Ltd, now CEL, used totally analogue techniques. Generating the integral at least doubled the circuitry of the simple meter and indeed in some designs the complexity became so high that reliability started to suffer. At ICA 1977 the 2218 was announced and this used digital techniques as well as analogue. While the first well known digital unit, it was actually predated by some years, as a California company Digital Acoustics had shown a unit at a previous Internoise.

Most initial devices used a computer chip simply to control the analogue circuitry, the Log and the Dose/Time calculations still being done by analogue circuits. This route was chosen as the early chips were slow, had poor instruction sets and required long learning times. As time went on, more and more of the calculations were passed to the computer. By 1979 a prototype unit made for the Open University was using look up tables to calculate the log and generate the SEL, still using analogue circuitry up to and including the rms converter. By 1983, the computer was being used to make calculations of L10 etc, to perform complex housekeeping tasks and to store the position of the front panel switches. One of the first of these units being the CEL 393 which was shown as a space model at the 1983 ICA.

It was at ICA 1983 that two routes for process circuitry started to appear. A French company Integra, now defunct, showed a new Leq meter with no read-out at all. This unit took a series of very short Leq's and stored them in its 44k internal memory. After the readings or rather acquisition had finished, the meter was then taken back to base and the data transferred to a host computer which could then recreate the acoustic climate and allow the actual measurements to take place. This division of acquisition and measurement had been predicted by Komorn in a 1979 Internoise paper.

As is now well known, Cirrus Research took the concept and added a conventional digital read-out and some conventional functions and produced a 'Short Leq' meter which could acquire and store 1 second Leq elements for up to 30 hours without needing data to be transferred to a host. While the Cirrus unit is still the only unit that can do this, other companies such as B & K and CEL used external add-ons to their conventional units to perform some of the same functions, usually by using a simple external computer. The new technique is usually know as 'outbox' processing as the instrument does little except store data.

There are two good reasons why the 'outbox' processing concept is of such value. Firstly, the computer chip which can be used in an instrument will always lag well behind the power of the chip in a conventional desktop computer for reasons of scale, power consumption and simple investment. It therefore seems foolish to effectively stop acousticians using the power of a desktop by attempting to perform all the calculations inside the meter. Also, attempting to calculate the noise statistics in real time pre-supposes that the user is sure he knows which index to use. If the data is simply stored, any index can be computed later. Thus an 'outbox' system such as Short Leq uses the power of the internal chip only to acquire data as accurately as possible.

As time passes, the point in the circuit where the computer takes over will go further and further towards the microphone in both 'inbox' and 'outbox' units, eventually replacing most if not all the analogue circuitry. The first units to do this are already available albeit in a form not quite as useful as a conventional instrument. Thoroughbred Digital Systems with 'CANAL' in the UK and 01dB SA with 'ARIA' in France have produced systems where everything is done by the desktop computer. In such a system the only parts external to the computer are the microphone and its associated preamplifier, everything else is either part of a digital subsystem on the add-on card or implemented in software. To meet International standards such as IEC 804 and IEC 651, it can be argued that the whole computer must therefore be considered as an integral part of the Sound Level Meter and indeed 01dB have submitted an 'ARIA' unit for EEC approval.

NEW TECHNOLOGY

There is no doubt that systems such as these will soon appear inside Sound Level Meters, in other words direct sampling of the AC signal in the meter. The miniaturisation of parts means that today it is technically feasible to put the converter and processor inside a conventional Sound Level Meter. The biggest problem is the memory requirements and the associated problem of battery power. A hand held meter is of little use if it has to be connected to a wall socket.

Today, in late 1989, 7 companies dominate the world of acoustic instrumentation internationally, so to speak one whale and 6 minnows. These companies have all grasped the nettle of computers and either store data inside the Sound Level Meter for processing by an external computer or attempt to use the simple computer inside the meter to calculate several parameters at once.

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COMPUTER DATA ACQUISITION.

Cirrus Research have chosen the 'outbox' route where data is acquired blindly into the meter memory doing as little processing as possible to the stored signal. Clearly, the more processing is done, the greater the restriction on the ability of the user to generate different indices. In a Short Leq Meter therefore, the only processing done is that the signal is integrated into very short bundles of energy. If these bundles are short enough, they can be used instead of the raw data for all statistical purposes. The main disadvantage of the method is the time limit of measurement due to memory size. On the CRL 2.36A 256,000 data elements can be stored which gives over 60 hours of acquisition of 1 second Leq elements. This equates to 'S' or Slow response on a conventional Sound Level Meter. While this is adequate for most people, 1 second is not a suitable acquisition period if there are very impulsive signals. The next standardised period is 1/8th second which is the same time constant as 'F' or Fast response. This then allows just over 8 hours of data to be acquired before the memory is full. However, if 10 milliseconds is required, the time comes down to a little over half an hour. The memory size used in this hand held battery unit is 8 times the size of that used on the first IBM PC and 16 times the memory of the Apple or BBC computer.

The stored data can be used for the production of any conventional acoustic index where PEAK is not involved, thus the user simply acquires data blindly into the meter and only after the acquisition does he have to determine what is to be measured. For example, traffic noise might be under investigation. Conventionally, it may be that L10 and L90 are the measures you require. However, after seeing the data, it is decided that the New Zealand L95 may be a more sensible background. With Short Leq, this is no problem; simply reprocess the raw data on file.

When a Short Leq meter such as the CRL 2.36A or a direct acquisition system like 'ARIA' acquires the data, it is particularly difficult to be sure that the values being stored are correct. A series of predefined data can check that the calculations are correct, but they are no help if the original acquired data is incorrect. How then can this be checked? The short answer is 'just as we always have'. The instrument or card is fed with a precise set of signals, such as those specified in IEC 651 and 804. In general, computer systems are no better and no worse for frequency performance e.g. 'A' weighting, than are conventional units and they can be checked in the same way. However, there may be special problems associated with pulse testing which have to be resolved. For example, a particular unit has a 250 microsecond sample rate of the (F)ast response. This sampling may beat with the 4kHz test signal used in IEC 804 and the unit can give a nonsense answer. While this could also occur with a conventional unit, most Sound Level Meter designers would be aware of the problem and give this sample rate and its sub and super harmonics a wide berth. However the intent of IEC 804 is that all reasonable signals should be correctly integrated and thus there is always the danger that a particular test signal will give aliasing errors. At Cirrus, we use a sweep technique to find these frequencies as the unit must be able to function to the specification with any signal in its operating range. It is not enough just to meet the tests in the standard. Standards by definition lag behind technology.

CONCLUSIONS

While the use of computer systems is going to become more widespread because of their unique ability to compute statistical data such as noise indices, they introduce new problems which can be hard to evaluate. The potential user at this time has no help from International standards in the evaluation of their performance and must rely on his own judgement.

Since about 1980, small computers have been available for use in the home and laboratory. These devices, however small and crude, appealed to acousticians because of the way they could help with the calculation of complex acoustic parameters. 1982 was a watershed year in the field in that the quasi 16 bit IBM PC became readily available and although the initial devices were little faster than the 8 bit machines of the day, the PC became the forerunner of a whole new generation of devices. Today, with the ready availability of fast 32 bit 80386 machines at prices of the same order as a Sound Level Meter, acousticians are thinking more and more towards the use of computers for routine measurements. Nothing we can do will change this.

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