Proceedings of the Institute of Acoustics

PC-BASED SOUND INTENSITY MEASUREMENT

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1. INTRODUCTION

The measurement of sound intensity is still a relatively new phenomenon in acoustics, with the first commercially available instruments becoming available in the early 1980's. These were developments of existing frequency analysers, and were characterised by their lack of portability, and dedicated nature. As the field developed, measurements taken with these systems were commonly transferred to computers, scientific computers initially and PCs later, for further processing and results presentation.

This history accounts for the common conception that the measurement of sound intensity is an expensive business, using esoteric hardware and analysers.

Virtual instruments have slowly eaten into the traditional instrument market, and now offer unrivalled performance and flexibility for traditional acoustics measurements, such as environmental noise, building acoustics and general frequency analysis. But what of the particular measurement requirements of sound intensity, with its strict phase matching and dynamic linearity?

This paper discusses the architecture of distributed virtual instruments, and their application to intensity measurements, highlighting the benefits to be gained from this approach, particularly in the context of sound power determination according to ISO9614.

Calibration issues are also discussed, and a commercially available sound intensity measurement system described.

2. THE VIRTUAL INSTRUMENT

The concept of the 'virtual instrument' is not new, and it owes its definition to work done during the standardisation of an automated test equipment (ATE) specification which later became the VXI specification (VMEbus eXtensions for Instrumentation) [1].

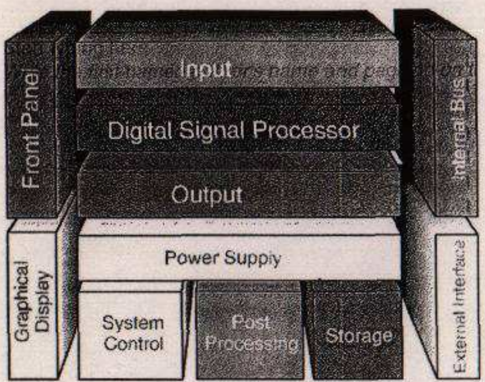


Figure 1 : The basic building blocks

Loosely described as a *'combination of software and hardware resources to achieve measurement functionality'*, the virtual instrument has taken many forms, some less efficient and successful than others.

To understand the concept, it is necessary to examine each function within the measurement instrument, and then establish the benefits or otherwise of implementing these functions in software, possibly running on a general computer platform, rather than dedicated instrument hardware.

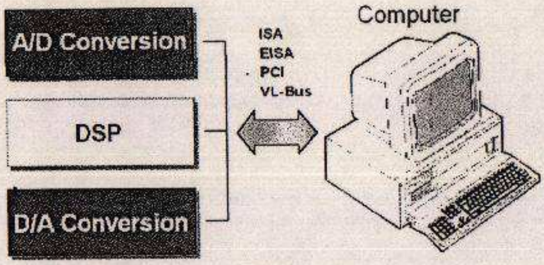
Figure 1 shows how a traditional measurement instrument may be built up from generic building blocks. The main instrumentation functions are found in the input, output and signal processing stages, and all other functions can be regarded as 'services' that manage these resources to yield the measurement result.

In its simplest form, a virtual instrument can simply replace the user interface on the measuring instrument by lifting off the front panel and graphical display, and using the host computer for these functions. This has benefits in terms of display resolution, colour and control interface, but in practice this is no more complicated than connecting a PC to a sound level meter via an interface (commonly RS232C serial) and remotely controlling it from a software package. All of the measuring functions are still handled by the instrument, and there is little benefit in terms of cost or measurement flexibility. The PC becomes no more than a storage device and instrument controller. Testing such instruments for accuracy also places little demand on the host computer, as the actual functions required by the standards are realised in hard- and firmware. These include A-weighting networks, RMS or peak detectors, gain etc.

However, to realise the true benefits of a computing platform, instrument functions can be distributed according to their timing priorities, which brings with it implications in measurement accuracy. Such an approach can be described as a 'distributed virtual instrument' [2], [3].

3. THE DISTRIBUTED VIRTUAL INSTRUMENT

The distributed approach offers several benefits both in user interface and also measurement capability, as the resources required by the instrument function (i.e. Input, Output and Digital Signal Processing) can be quickly reconfigured by software. This then allows many different types of measurements to be performed, without any change in hardware, or use of additional instruments. This is illustrated in Figure 2.



Examples:- Generator = DSP + D/A
Analyzer = A/D + DSP + D/A
Sound Level Meter = A/D (+DSP)

Figure 2 : The distributed virtual instrument

In practice, this involves a further transfer of responsibility to the host computer, where calculations of e.g. $L_{Aeq,T}$ are made in software, rather than instrument firmware. In fact, an instrument can be achieved with the hardware performing little more than analogue-to-digital conversion, before the software takes over.

The implication of this is that the hardware itself is no longer a measuring instrument, but simply a resource component of a measuring system. The PC becomes an integral part of the process, and therefore any measurements of instrument accuracy or calibration must include the software element.

To illustrate which elements in the measuring chain are best handled by dedicated hardware, and which parts are best suited to interrupt-driven computing, the data handling process is shown in Figure 3.

At the physical interface, e.g. acoustic pressure, all processes need to occur in real-time, in other words, a process must be dedicated to a particular function at all times. Such processes include A/D conversion, filtering, etc.

Data acquired from this interface can then be treated in a block-oriented fashion, assuming it is buffered to avoid loss of data. Calculations such as Fast Fourier Transformation are examples of block-oriented processing.

The results of these calculations can then be passed on to a display processor to yield the results to the user, and further post-processing calculations may be made, for example, sound power levels may be calculated from sound pressure measurements, after spatial averaging.

In a dedicated measurement instrument, certainly no more than 10 years ago, all of this functionality would be achieved in hardware and firmware, with computers performing no more than archiving of results output from these instruments.

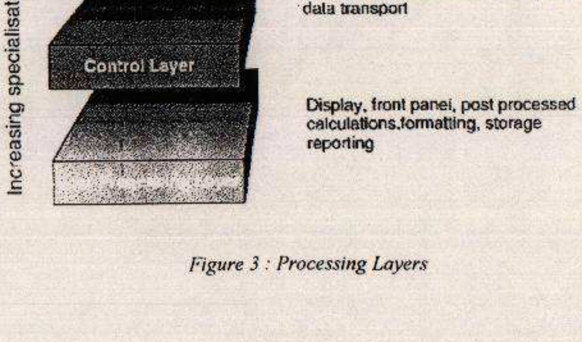


Figure 3 : Processing Layers

With today's commodity computing power, and by selecting a sensible distributed architecture, the PC will take care of certainly the second two stages in the chain, and in many cases, the block oriented layer. It is now possible to calculate an FFT using floating point arithmetic on a Pentium processor faster than some dedicated DSP chips. However, the designer must consider how much extra work the processor is being asked to do, such as display, peripheral management, etc.

In practice, then, the real-time layer still requires dedicated hardware in many cases, and the nature and size of the sound & vibration market will still dictate specialised functions not addressed, or even acknowledged, by the PC development departments.

4. THE VIRTUAL INTENSITY ANALYSER

The measurement of sound intensity imposes specific requirements on a virtual instrument. As the measurement of sound intensity using the two-microphone technique is essentially a measurement of phase, the acquisition hardware must be carefully phase-matched. However, once the data is digitised, the rest of the process is purely calculation, generally performed in the DSP part of the virtual instrument. The only other consideration must be to detection of overloads, preferably in the real-time layer, and this is not only in the analogue chain, but also in the digital portion. When working in very reactive environments, it is possible to experience digital overload from the calculation of intensity, depending on the word-length used in the DSP.

Beyond the actual calculation of intensity, the rest of the analysis and results presentation is possible on the PC platform itself, integrating the measurement with the post-processing, often in real-time.

6. A PRACTICAL VIRTUAL INTENSITY ANALYSER

An example of such a configuration is the Symphonie system from 01dB (Figure 4). This system uses a small data acquisition unit containing A/D conversion, D/A conversion and powerful DSP modules, connected to the PC bus via a Plug-and-Play PC Card (PCMCIA) connection. The two analogue input channels are phase matched to 0.05°, and the two 18-bit A/D converters sample the incoming signal at 51.2kHz. The DSPs can be configured for FFT analysis or real-time digital filtering, and data is fed to the host PC (often a notebook computer) in real-time.

Figure 4 : The Symphonie data acquisition unit

Using the dBFA software, the system can be configured for general measurements of active or reactive intensity, along with sound pressure levels and mean pressure, as a function of frequency.

For measurements of sound power according to the ISO9614 procedures, the power module of dBFA takes care of all the required measurement acquisition, as well as calculation of all the necessary quality indicators.

As with most virtual instruments, the control panel for setting up the system is presented on the PC screen, where the user can input all the parameters associated with sound intensity measurement, such as microphones spacing, speed of sound, density, etc. In addition, any other parameters relating to the measurement can be set, such as the requirement for convolution correction to extend the high frequency limit, or phase compensation to extend the low frequency limit in certain circumstances.

A typical set-up menu is shown in Figure 5.

One of the strengths of the PC-based approach is the way the actual measurement can be incorporated into the procedures for sound power determination. Rather than, as in the case of dedicated instruments, acquiring the measurements and transferring them to a PC for later processing, the measurements become an integral part of the acquisition process, so the user is led step-by-step through the measurement procedure, and can monitor the progress in real-time. This is especially useful if the quality of the measurement is borderline between e.g. Engineering grade and Survey grade, and corrections for 'hotspots' can be implemented before the grade of measurement deteriorates too much. Similarly, methods such as probe reversal and stationarity checks can be performed before the measurement proper.

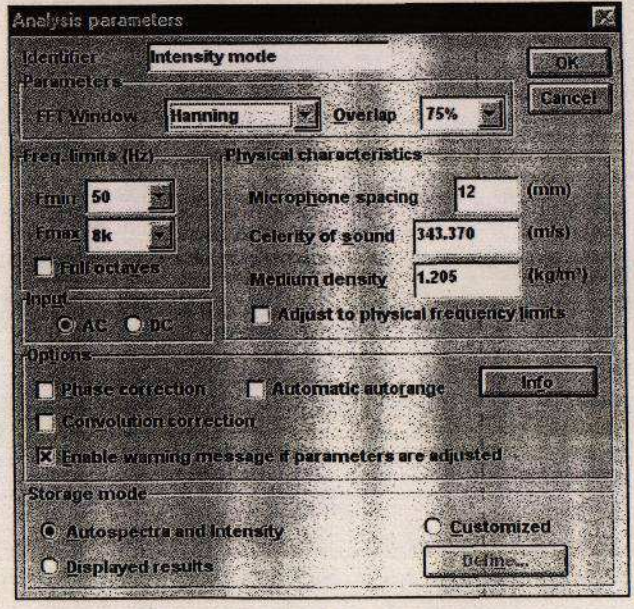


Figure 5 : Measurement parameters

With the sound power options of the dBFA software, the ISO9614 standard becomes integrated with the measurement, and there is no line drawn between the two stages. The safeguards for measurement quality are built-in, ensuring that all measurements are taken, and all indicators calculated before moving on to the reporting stage.

Data can be acquired using a standard parallelepiped surface around the source, the dimensions of which are suggested according to the standard, and point-by-point as well as the scanning method are supported.

Alternatively, a non-standard surface may be defined in difficult circumstances.

Acquisition is controlled remotely by the intensity probe handle, connected to the PC via either a standard serial port, or via a dedicated port on the Symphonie unit itself.

Although the main purpose of acquiring the data is for sound power determination, a by-product of the acquisition is that results can be presented in the form of a map, which can provide qualitative information about the sound radiation from the source. By using interpolation algorithms, contours and colour-scale plots can present the data in a form easily assimilated by eye, for noise control applications.

Finally, results are ready for documentation, which is simply a process of direct printing, or transferring tables and graphs to other Windows applications such as spreadsheets or word processors.

7.CALIBRATION ISSUES

Although the ISO9614 standard includes some checks on the performance of the measurement system, and its suitability for the measurement field, it is always advisable to perform some calibration check of the system at regular intervals.

This will take the form of two measurements:-

- Amplitude Calibration
- Phase Calibration

Amplitude calibration is performed using a standard pistonphone of calibrator, in the same way as for sound level meters. This form of calibration will normally be performed before and after each series of measurements.

Phase calibration in the field, tends to be more involved, and uses a special coupler (Figure 6), into which the two microphones are placed. By exposing the microphones to the same acoustic field, a measurement of phase mismatch, and hence residual intensity, can be performed. This then sets the limits of dynamic capability of the system.

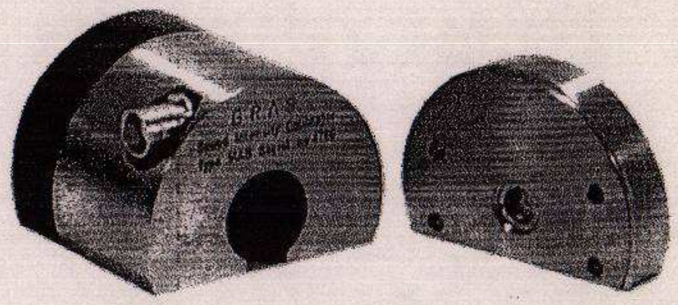


Figure 6 : The GRAS 51AB intensity calibrator

Again, the virtual instrument can simplify this process. For example, the Symphonie, with its D/A converters, can supply the necessary signals for driving the calibration coupler, and an additional software module, dBSONDE, steps the user through the calibration process.

The end result is a measurement of the effective dynamic capability of the system. This can then be used to check if the microphones have sustained damage, by comparison to the original calibration data, or alternatively, the residual phase data can be automatically compensated during the measurement process, within the limits of uncertainty defined in ISO9614.

Figure 7 shows a typical result with this software.

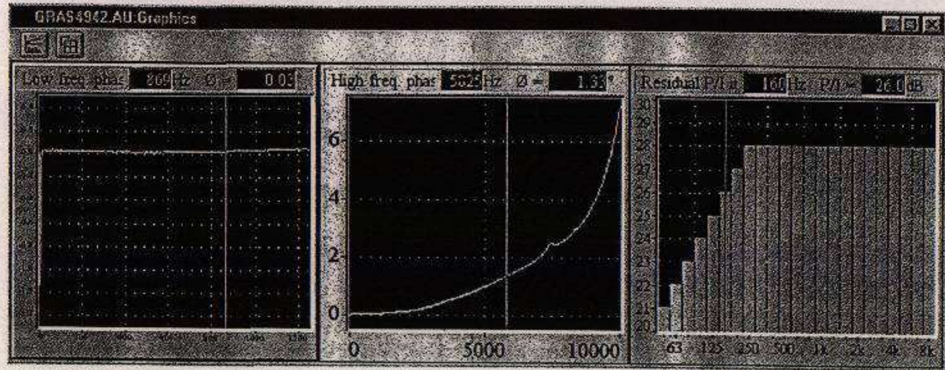


Figure 7 : Measurement of residual phase in dBSONDE

By careful design of both hardware and software, it is now possible to provide Class 1 accuracy for both the processor and probe to IEC1043, and combine the two into a Class 1 system.

8. CONCLUSIONS

With new virtual instrument architectures, it is now possible to measure sound intensity using PC-based systems. The attendant cost benefits of this approach in terms of office standardisation of hardware and software allow the development of very cost-effective configurations. Portable systems are now possible using notebook platforms, giving a complete measurement system for sound power determination to international standards, at a fraction of the cost of dedicated instruments.

9. REFERENCES

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