Report to the National Measurement System Directorate, Department of Trade and Industry

New Directions - Digital Signal Processing: recommendations for the Software Support for Metrology programme 2004-2007

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April 2003

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ABSTRACT

This report sets out the conclusions of a study into the metrology requirements of Digital Signal Processing (DSP) that will be needed to meet future challenges within the National Measurement System (NMS). It forms part of the "New Directions" project, within Theme 5 of the current Software Support for Metrology programme.

The intention is that the reports should provide useful input to the process of formulating the next Software Support for Metrology (SS*f*M) programme, which will begin in April 2004.

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1. INTRODUCTION

1.1 THE BACKGROUND

This report sets out the conclusions of a study into the metrology requirements of Digital Signal Processing (DSP) that will be needed to meet future challenges within the National Measurement System (NMS). It forms part of the "New Directions" project, within Theme 5 of the current Software Support for Metrology programme. The "New Directions" project encompasses a wide range of topics: legal metrology, new developments in mathematics and software engineering, bioinformatics and measurements for biotechnology, soft metrology, and the subject of the current report, DSP. Separate reports are being produced for each of the five topic areas and it is the intention that each report should stand alone and that overlap between the topic areas should be avoided. The intention is that the reports should provide useful input to the process of formulating the next Software Support for Metrology (SS*f*M) programme, which will begin in April 2004. Further information about the SS*f*M programme can be found at the website address: www.npl.co.uk/ssfm .

1.2 METHODOLOGY

During the formulation of the current SS*f*M programme, a number of potential topics were identified for the subject of this study, including for example, error modelling, uncertainty evaluation, validation and traceability requirements. There was also a clear need to consult with a relevant universities in order to get a proper understanding of the state of the art of DSP in the academic world to the extent that it is relevant to applications in metrology. Also of interest are the metrology requirements for DSP and the extent to which there are issues in DSP to be addressed by the SS*f*M programme. The study will attempt to identify topics or actions that should be taken forward by the SS*f*M programme.

The study began with a planning phase in which NPL decided who needed to be consulted in the study and appropriate consultation meetings were set up. In particular, it was expected that representatives of each metrology area currently using DSP would be consulted. Members of the SS*f*M Club were consulted at their meeting held on 22 October 2002. A questionnaire was also designed and sent to members of the SS*f*M Club, other relevant NMS Clubs and selected DSP users both within NPL and externally. Dr. Ted Chilton, a Course Organiser and lecturer on the "Advanced Signal Processing" short course at the University of Surrey, was invited to give a seminar on DSP at NPL on 21 November 2002. In addition to being a Senior Lecturer at Surrey, he has acted as signal processing consultant to many research/industrial organisations.

Additional research had to be carried out by the authors, through existing literature, DSP course notes and through a series of brief meetings with DSP users/experts in and outside NPL.

1.3 STRUCTURE OF THE REPORT

The aim of this report is to identify those signal processing issues which are of most concern to metrologists and to make recommendations for the Software Support for Metrology programme which should lead to work that will assist metrologists in applying best practice in their DSP-related measurement activities. In formulating the recommendations it has been recognised that one must take into account not only the state-of-the-art in modern signal processing, but more traditional topics, such as the application and implementation of Fourier Transform algorithms, and correlation and convolution algorithms, must also be included within the remit of the report. Metrology often has to adopt a conservative approach to the choice of technologies for particular applications, to ensure that the measurement process can be shown to be on a firm footing. It can take some time for the leading edge of research in the academic community to be adopted as routine in measurement laboratories. This process is an inevitable result of the metrologist's need to understand fully the sources of uncertainty in the measurement process.

The report summarises the consultation with the DSP user community, the metrological requirements to be met by DSP and the state of the art in DSP, identifying any best practices within the DSP field that might need dissemination within metrology. DSP-related issues to be addressed by SS*f*M are summarised.

The report attempts to answer the following questions:

- What do metrologists know?
- What should they know?
- Do they understand their own limitations?
- How can SS*f*M help?

2. QUESTIONNAIRE

2.1 QUESTIONNAIRE KEY QUESTIONS

As part of the study, a questionnaire was disseminated via e-mail to various contacts, including SS*f*M Club members, EMMA (Electronic Materials Measurements and Applications) Club members, ComNet (Communications and Measurements Network) and ANAMET. It was also circulated amongst DSP users selected by the authors and/or colleagues of the authors.

The questionnaire was formulated in order to gain an understanding of the following points:

- DSP applications;
- Definition of DSP;
- Type of DSP hardware and software used;
- Existence of staff specifically trained in DSP;
- Pitfalls facing inexperienced DSP users;
- Identification of specific training needs;
- Consideration of effects of the measurement system on the measurement;
- Traceability requirements;
- Existence of DSP quality system.

Some of the key questions asked were:

- What applications do you have/expect to have in the future for digital signal processing in your current business/project/research?
- How do you/your organisation/research team define digital signal processing?
- What hardware do you use for signal acquisition? Is it purpose-built or do you just combine existing proprietary equipment to do what you want? Who typically designs and implements DSP systems in your organisation?
- Do you design your own transforms or DSP algorithms? Do you use modelling software and if so, which? Do you test algorithms by modelling? Do you test transform algorithm/code by repeated inversion to look for errors?
- Do you have staff who have specific skills/training in DSP? Do you think there is any benefit in having specifically trained staff or do you expect that DSP should be part of the range of skills needed by all physicists and engineers?
- What do you think are the main pitfalls facing inexperienced DSP users? Can you recount any specific problems/mistakes etc. from your organisation's or team's experience?
- Can you identify any specific training needs for yourself or others within your team, which you think would be of benefit to you and your colleagues?
- Do you consider the effect that the measurement system has on your measurements, e.g. do you deconvolve instrumentation/transducer responses and so on? If so, how?
- Do you make relative measurements or absolute measurements? Are traceability issues important to you?

- Does your organisation have a quality system that governs the development of DSP systems (both hardware and software)?

More details of this questionnaire may be found in Appendix A.

2.2 QUESTIONNAIRE RESPONSES

Overall, eleven replies were received. Although this represents a somewhat disappointing outcome, many comprehensive replies were received from both metrologists and non-metrologists. A synthesis of the replies is presented below.

The recipients of the questionnaires spanned the following areas of research:

- Active control of sound and vibration
- Optical radiation measurement
- Aerospace
- Medical ultrasound
- Spectroscopy
- Metrology and its application to the control of manufacturing processes
- Defence
- Telecommunications
- Marine acoustics

Generally, the recipients described DSP as the “processing of sampled data” or the “extraction of information from discrete-time signals”. Most definitions acknowledged the discrete nature of the data being processed and the fact that DSP techniques were being used to extract useful information from the data.

The type of equipment being used included:

- Bench-top measurement systems, e.g. Vector Network Analysers, oscilloscopes, etc.
- PC-based systems
- Purpose-built equipment, including in-house design of circuitry using commercially available chips; sub-contractors are sometimes employed.

The recipients cited the following sources for their DSP algorithms and transforms

- MatLab®
- Mathematica®
- Mathcad®
- Labview®
- Text books
- In-house

Two recipients did not test the transforms or algorithms they used.

Three out of the eleven recipients did not have a DSP expert as part of their team, saying that they would hire a private contractor when necessary. The remaining eight employed people with training in DSP and believed it beneficial to do so.

Three recipients stated that they thought that basic DSP should be part of the skills possessed by scientists and engineers. Four recipients did not comment on this, whilst four believed that DSP-related problems should only be tackled by qualified members of staff.

When asked about potential pitfalls of the use of DSP, the recipients cited many of the issues covered in Sections 3.2 to 3.6. One recipient mentioned the importance of keeping records of the ‘raw’ data.

Four recipients identified the need for training, including continual development. Five said they had no need for training, mostly because they already had employees sufficiently skilled for their DSP requirements. Only two recipients stated that they attempted to keep up to date with the latest advances in DSP.

Almost all recipients (ten out of eleven) carried out absolute measurements and nine expressed an interest in traceability, yet four were not concerned with the effect that the measurement system had on their measurements.

Four recipients stated that their organisation had a quality system that governed the development of DSP systems (both software and hardware). Others mentioned that they had an ‘informal’ quality system.

2.3 DISCUSSION

Overall, it appeared that the organisations that have staff trained to a high level in DSP clearly saw advantages in this. For obvious reasons, these organisations were more likely to design their own DSP software and hardware, therefore minimising problems that may occur with inexperienced staff adopting a ‘black-box’ approach to commercially available measurement software and hardware.

Since there were not many replies from NPL employees, the questionnaire was later disseminated to NPL staff via e-mail, in hope that more information could be gathered from DSP experts on site. The lack of replies as well as consultation with measurement scientists at NPL however suggests that DSP-related metrology issues are not given the priority they deserve.

3. DSP REQUIREMENTS IN METROLOGY

3.1 DATA ANALYSIS CONSIDERATIONS

The objective of data analysis is, broadly, to extract information contained in a signal that direct observation may not reveal. Sometimes, only 'gross' characteristics of a signal may be required (e.g. mean value, mean square level, total energy) in which case simple processing may only be required and accounting for other information in the signal might complicate matters. In the context of metrology, where traceable measurements are an issue, it may be necessary to probe more deeply in an attempt to extract features of interest.

Generally, data analysis will generally involve three phases:

- Data acquisition
- Data processing
- Data interpretation

Generally, the first two phases will introduce artefacts in the raw unprocessed data that may be observed in the third phase. The aim of this section is to focus on the first two phases, so as to highlight how signal conditioning and processing can affect the original data. Conditioning and processing are used in the context of "improving" the data but can in fact introduce errors making traceability an issue.

This section will be concerned with methods of analysis of time histories which have been obtained when conducting a measurement and/or attempting to quantify the behaviour of a physical system. This discussion will be limited to basic procedures and the treatment of the subject will be superficial.

Broadly speaking, the types of signals under investigation can fall into two categories:

- Deterministic
- Random (i.e. not predictable exactly)

Sub-categories of deterministic signals include:

- Periodic signals (e.g. sine wave)
- Non-periodic signals (transient and 'almost' periodic)

Sub-categories of random signals include:

- Stationary signals (e.g. white noise)
- Non-stationary signals

Very often, processes are mixed and the above demarcations are not easily applied and consequently, the analysis procedure to be used may not be apparent. If we restrict ourselves to the above categories, the classes of non-stationary random signals and 'almost' periodic deterministic signals are often difficult to analyse, and will only be commented on briefly.

The signals that are processed in modern metrology are almost all electrical in nature, often being electrical outputs from transducers. Useful information extraction procedures may be performed in the time domain. However, transform methods are often used since they can offer simplification both in theory and physical interpretation. Frequency domain methods in particular are commonly used.

3.2 ANALOGUE TO DIGITAL CONVERSION

3.2.1 Fundamentals

The signals recorded from physical phenomena are generally continuous in nature and are referred to as analogue data. Developments in digital computation methods have ensured that most signal processing is achieved on special purpose analysers (e.g. digital oscilloscope or dedicated PC with card). This implies that the original data is sampled, and the analysis of discrete (in time) data requires the metrologist to have a good grasp of the fundamentals of digital signal processing methods.

Analogue to digital conversion comprises two distinct processes, sampling and quantisation, which must be considered independently. After sampling, an analogue signal is transformed into a discrete signal. After quantisation, the discrete signal results in binary data. In Figure (3.1), $x(n\Delta)$ is the exact value attained by the time history $x(t)$ at time $t = n\Delta$, where Δ is the sampling interval for uniform rate sampling. $\tilde{x}(n\Delta)$ is the representation of $x(n\Delta)$ in a computer and differs from $x(n\Delta)$ since a finite number of bits are used to represent each number.

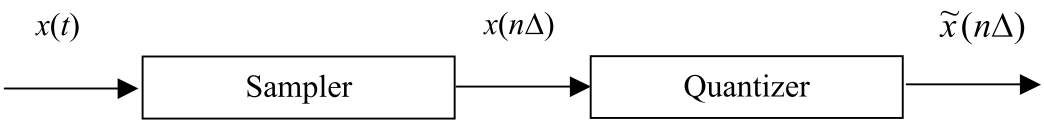

Figure 3.1: Analogue to digital converter

Note that in practice, actual analogue to digital converters do not physically consist of two separate components.

3.2.2 Aliasing

It should be noted that the spectrum of a series of regularly spaced pulses in time, with spacing Δ seconds, is a set of regularly spaced pulses in frequency, with separation $1/\Delta$ Hz.

The spectrum of a discrete signal is the true spectrum superposed by replicates spaced at intervals of F_s where $F_s = 1/\Delta$. For a broadband signal, the spectra overlap. This is called aliasing.

In practice, all physical systems will have a limited bandwidth. The sampling frequency must therefore be chosen to be sufficiently large in order to avoid aliasing. Where this is not possible, the signal must be low-pass filtered to restrict its bandwidth. The bandwidth must be less than half the sampling frequency. This is known as the Nyquist limit.

Shannon's theorem states that: A signal may be exactly reconstructed from a sampled counterpart, provided the sampling frequency is at least twice the highest frequency of interest in the signal.

3.2.3 Anti-aliasing filter considerations

For critically sampled signals, large attenuation is needed. It should be noted that high order analogue filters are difficult to design and are prone to severe phase distortion around the cut-off frequency (phase departs from linearity). It is usually best to raise the sampling frequency as far as possible. This will increase the data rate. The data can be low-pass filtered digitally and down-sampled later.

It should be noted that even if there are no signal components above the Nyquist frequency, there may be high-frequency noise present which, if not filtered out before sampling, may appear as lower frequency components in the sampled spectrum due to aliasing.

Problems can occur if no effort is made to use an antialiasing filter whilst acquiring analogue data. Since sampling frequencies can easily be set to several times the maximum frequency of the signal under test, a metrologist may assume that no high-frequency component will appear as a lower frequency component in the sampled spectrum. However, even if the signal of interest is at low frequency, in general high frequency noise is present in the measurement system, which can lead to high frequency noise above the Nyquist frequency appearing in the lower end of the spectrum. Clearly, if this noise is significant, it may add a further source of type A and type B uncertainties. In the Ultrasonics Group of the Centre for Acoustics and Ionising Radiation at NPL, there is no awareness of whether or not an antialiasing filter is used when acquiring pulsed ultrasonic waveforms amongst measurement staff. The effects of aliasing arising from high frequency noise are not investigated.

Recommendations:

- Educate users on aliasing, the use of antialiasing filters and methods to examining of aliasing – this may be difficult to do without other DSP equipment or good wideband analogue instrumentation.
- If no antialiasing filter is used, ensure that the effect of high-frequency noise beyond the Nyquist frequency on the spectrum of the signal is examined.
- If an antialiasing filter is used, ensure that its attenuation and phase shift characteristics are known so that it is possible to examine whether it significantly affects the analogue signal being acquired in the frequency range of interest.
- Educate users on aliasing, the use of antialiasing filters and methods of examining for aliasing.

This may be accomplished through use of analogue instrumentation when trouble-shooting in early stages of the experimental design.

3.2.4 Errors in sample and hold devices

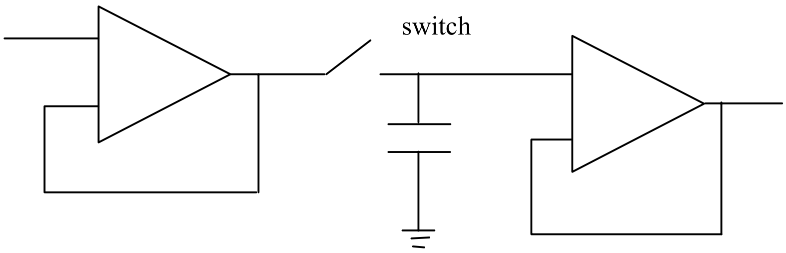

Figure 3.2: Sample and Hold amplifier

Sampling occurs when the switch in the circuit in Figure 3.2 is closed. There is a time constant associated with the circuit and the capacitor does therefore not charge instantly. When the switch opens, the voltage is held at the output stage. Sampling is therefore not instantaneous. The period when the switch is closed is called the tracking mode. The resulting spectrum is then modified and spectral energy is biased towards the low end of the spectrum. The input signal may change during tracking mode, and the acquired voltage is therefore an average of the signal voltage. Choosing a short acquisition time relative to the sampling frequency alleviates both.

Switch closure and opening timings are subject to variation due to clock jitter and delay (aperture time) variation (this depends on the prevalent line voltage). This will tend to corrupt the high frequency end of the spectrum (aliasing is increased). A bandwidth of less than $F_s/2$ should be used. The switch may also inject extra charge into the capacitor. This effect will increase with frequency.

The voltage at the output stage changes when the switch changes. This is referred to as pedestal or offset voltage. During the hold mode, sampled voltage decays (droop). Increased capacitance is therefore needed, which increases the acquisition time. This will require a lower sampling frequency.

The above two statements are contradictory. A compromise must therefore be sought.

Often, sample and hold amplifiers are incorporated into analogue-to-digital converters and the manufacturer will have attempted to include the effects in the specifications. Whether these specifications can be trusted is another issue.

As part of 2000-2003 Electrical Metrology programme, a project (D4.2) was proposed to establish an NMS ADC/DAC (analogue-to-digital converter/digital to analogue converter) Characterisation Facility. The rationale for this project was to provide testing facilities for establishment/verification of data converter specifications. The facility would test converters according to existing international standards and would also develop new testing methods for these devices. Because of the wide frequency range of data converters and the related accuracy levels of these devices, some prioritisation for the measurement ranges of these devices was first required.

The DTI commissioned Scientific Generics Ltd. to produce a scoping study including the need and possible industrial usage of such a facility. The report entitled "Scoping Study of UK Industry Requirements for ADC and DAC Characterisation Services" reported the following: "The overall conclusion from this study is that no requirement was found for a national characterisation facility for ADCs and DACs that could be operated as a commercially viable service to UK industry" [1]. Respondents to Scientific Generics were drawn from a variety of industries using data converters from a wide range of industries with an equally wide range of expectations for the performance of these devices. Notwithstanding the conclusions of the report, it is possible that facilities solely in support of metrology-based applications could be justified in terms of the standardisation of data converters for high accuracy use.

3.2.5 Quantisation

Once a sample-and-hold device has acquired a voltage, it must be mapped onto a binary code. There are different ways in which this can be done, but it usually depends on whether the signal consists of only positive values or whether it is bi-polar. The number of bits used depends on the required signal-to-noise ratio.

Quantisation is a non-linear, non-invertible process, which maps a given amplitude at a fixed time to an amplitude taken from a finite set of values. The signal range is divided into L intervals by L+1 decision levels. Typically, most mappings use linear or uniform quantisation steps, especially when the signal is to be further processed by digital systems. However, for some data storage and transmission purposes non-linear and time-variant quantisers are used [2] (Proakis and Monolakis).

In practice, analogue to digital converters can handle only a finite range. If the dynamic range of the signal is larger than the range of the quantiser, the samples that exceed the range are clipped. With the analogue to digital converter's normal range the instantaneous quantisation error is in the range:

$$-\frac{\Delta}{2} \le e_q(n) \le \frac{\Delta}{2} \tag{3.1}$$

where Δ is the quantisation step size. In the case of clipping, the error becomes larger than the step size.

During coding, the analogue-to-digital converter assigns a unique binary number to each quantisation level. If there are L levels, then L different binary numbers are needed. With a binary word length of b+1 bits, 2^{b+1} different binary numbers can be represented. The step size or resolution of the converter is:

$$\Delta = \frac{R}{2^{b+1}} \tag{3.2}$$

R is the quantiser range. A number of binary coding schemes exist, but the most common is the two's complement representation. In practice, the coding process has no effect on the behaviour of the quantisation process.

From the comments made above, it will be seen that for an ideal analogue-to-digital converter the only performance limitation is the quantisation error. This can be reduced by increasing the number of bits employed in the converter. Real converters differ from the ideal in a number of ways. Firstly, they may have an offset error, i.e., the first transition may not occur at exactly +1/2 LSB (least significant bit). There may be gain error, i.e., the difference between the values at which the first and last transition occur may not be equal to the full scale – 2 times the LSB. Linearity error refers to the fact that the difference between transition values may not all be equal. For large differential linearity errors, some codes may actually be missed.

Analysing the quantisation noise performance of analogue to digital converters can only be carried out in a deterministic manner in very simple cases. Generally, a statistical analysis is required which aims to evaluated the signal-to-quantisation noise power level. To do this, one makes a series of assumptions, many of which may not be justified in practice; e.g., that the error is uniformly distributed across the quantisation level, that the error sequence is a stationary white noise sequence, that the error and the signal of interest are uncorrelated, and that the signal sequence has a zero mean and is stationary. It is not surprising that many converters do not meet the noise performance predictions based on these assumptions, so that a 16-bit converter may have in reality only 14-bit performance.

Referring to Figure (3.1), the output $\tilde{x}(n\Delta)$ can be written as $\tilde{x}(n\Delta) = x(n\Delta) + e(n\Delta)$. The ADC is described by its dynamic range or quantisation signal to noise ratio given by:

$$SNR = 10\log_{10}\left(\frac{Signal\,Power}{Error\,Power}\right) \tag{3.3}$$

The signal power is given by $\langle x^2(n)\rangle$ where $\langle\ \rangle$ denotes the temporal average.

The ‘error power’ or quantisation noise is $\langle e^2(n)\rangle$

Assuming that the error is random and has a uniform probability density function (i.e. that the noise is white), each additional bit improves the signal-to-noise ratio by approximately 6 dB (e.g. a 16-bit quantisation offers a 96 dB signal-to-noise ratio).

Quantisation converts a continuously varying signal to a set of discrete values. Where the quantisation level is small compared to the signal amplitude, the effect is often approximated as added white noise.

All signals will have some analogue noise due to limitations of the recording system. Quantisation noise should be kept at a level lower than prevalent analogue noise. Binary information stored in bytes – increase from say 16-bits to 24-bits, increases costs and storage overhead. It will also decrease the possible sampling frequency.

The minimum number of bits per sample consistent with the desired signal-to-noise ratio should therefore be used. The signal should be amplified prior to conversion to occupy as much headroom as possible (use low-noise amplification techniques). Voltage limit should not be exceeded, otherwise clipping will occur, resulting in severe spectral distortion.

Once sampled, analogue values must be mapped onto a set of binary values. Since data is to be stored and processed by a computer, mapping must produce binary codes that are readable by standard computer software. Two formats are usually used: unsigned and signed binary integers.

Recommendations:

- Metrologists need to have an appreciation of the errors associated with the digitisation process, and also be able to evaluate the performance of analogue to digital conversion systems. Guidance on selecting suitable DSP hardware and the design of DSP systems should include specific information on error contributions arising from the digitisation process itself.
- When accurate measurements are required, the DSP hardware should be characterised or calibrated using traceable methods. Depending on the application, such characterisation may need the system signal-to-noise ratio to be determined, as well as its integral and differential linearity, phase performance, etc.

3.3 DETERMINISTIC DATA – FOURIER METHODS

3.3.1 Introduction

In section 3.1, three types of deterministic signals were mentioned: periodic, transient and ‘almost’ periodic. Periodic data and transient data may be analysed using the Fourier methods, namely the Fourier series, the Fourier transform and the Discrete Fourier Transform (DFT), a special case of which is the Fast Fourier Transform (FFT).

By far the most common digital signal processing algorithm in use in applied science, engineering and mathematics is the FFT. It provides an efficient means of identifying the frequency content of time and spatial domain signals. The development of computing technology, and especially the personal computer has meant that the FFT has now become a widely-available data analysis tool, which is no longer limited to the signal processing specialist [3] (Brigham). Brigham also lists a wide range of application areas in which the FFT is routinely employed. These include applied mechanics, sonics and acoustics, biomedical engineering, numerical methods, signal processing, instrumentation, radar, electromagnetics, communications, metallurgy, imaging and image restoration, nonlinear systems analysis, geophysics, quality control.

The FFT is not only important in itself but it also simplifies many other signal processing operations such as convolution and correlation and the estimation of power spectral densities. Frequency domain processing of time series and of two and three-dimensional data sets is now routine. In fact, much FFT processing is carried out within instrumentation itself, either by DSP chips or in firmware. Many modern oscilloscopes, for example, have on-board Fourier Transform routines, for example, to allow the user to convert time domain displays to frequency spectra.

3.3.2 The Fourier series and Fourier integral

Generally, if a signal $x(t)$ has a period T_p so that, for all values of t, $x(t) = x(t+T_p)$, $x(t)$ may then be represented as a sum of sines and cosines and written in complex form as:

$$x(t) = \sum_{n=-\infty}^{\infty} C_n e^{j\frac{2\pi nt}{T_p}} \tag{3.4}$$

where

$$C_n = \frac{1}{T_p} \int_{-\frac{T_p}{2}}^{\frac{T_p}{2}} x(t) e^{-j\frac{2\pi nt}{T_p}} \tag{3.5}$$

This is known as a Fourier series representation of $x(t)$.

If $x(t)$ is a transient, the discrete set of sines and cosines becomes a continuum:

$$x(t) = \int_{-\infty}^{\infty} X(f) e^{j2\pi ft} df \tag{3.6}$$

where

$$X(f) = \int_{-\infty}^{\infty} x(t) e^{-j2\pi ft} dt \tag{3.7}$$

is the Fourier transform of $x(t)$.

3.3.3 Windowing

An important consideration when calculating the transform $X(f)$ is that of data truncation or windowing.

The infinite integral in equation (3.6) may be replaced by a finite one if, for instance, $x(t)$ is only known or measured for a limited duration, say for T seconds. The following function of time may be defined:

$x_T(t) = x(t)$ $\quad |t| \le T/2$
$x_T(t) = 0$ $\quad$ elsewhere

$x_T(t)$ is a truncated version of $x(t)$. In the time domain, this is equivalent to multiplying $x(t)$ by a rectangular window $w(t)$ defined by:

$w(t) = 1$ $\quad |t| \le T/2$
$w(t) = 0$ $\quad$ elsewhere

so that $x_T(t) = x(t)\, w(t)$

The Fourier transform of $x_T(t)$ gives $X_T(f)$, which is related to the Fourier transforms of x and w by a convolution:

$$X_T(f) = \int_{-\infty}^{\infty} X(g) W(f-g) dg \tag{3.8}$$

where g is a frequency-related variable. $W(f)$ is sometimes called a 'spectral window'.

The convolution integral in equation (3.8) indicates that $X_T(f)$ is a distorted version of $X(f)$ (unless $W(f)$ is a delta function). This effect is simply demonstrated with a truncated cosine wave. If $x(t) = \cos(2\pi pt)$, then:

$$X(f) = \frac{1}{2}[\delta(f+p) + \delta(f-p)] \tag{3.9}$$

and

$$X_T(f) = \frac{T}{2}(\mathrm{sinc}[\pi T(f+p)] + \mathrm{sinc}[\pi T(f-p)]) \tag{3.10}$$

The appearance of spectral energy at frequencies other than p and $-p$ is referred to as leakage. The rectangular window is a poor window since the side lobes are large and do not decay rapidly. Numerous alternatives have been suggested to reduce leakage in Fourier transform calculations. By tapering windows to zero, side lobes are reduced but at the expense of widening the main lobe, hence increasing spectral smearing. See [4] (Harris) for a more comprehensive treatment of windows.

When making a measurement of a transient signal, one is often faced with the problem of reflections occurring before the initial transient is given a chance to decay to a level below a specified threshold. This is the case, for example, in [5] (Esward et al), where a glass block is excited by a force transducer using a transient signal, and a displacement sensor is used on the block. Due to the block's finite size, reflections from the boundary contaminate the initial transient signal.

Recommendations:

- Metrologists need to have a good appreciation of the effects of windowing and its potential effect on the overall uncertainty of the measurement. Also, they should be aware of the various types of window available and which one, if any, should be used for their particular application. It should be noted that windowing, although sometimes making frequency-domain results look 'smoother' by getting rid of the Gibb's effect (i.e., a 'ringing' of the frequency response, which causes the harmonics to 'smear' across the spectrum leading to errors), does not necessarily reduce uncertainties in the measurement. One may want to consider using un-windowed data and quantify the effects of the unwanted signal on the overall measurement. Sometimes, redesigning the experiment so that the initial transient is given enough time to decay 'sufficiently' before arrival of the first reflection and/or so that the level of the noise is lower may be possible, and this is the first path that should be explored. If there is no considerable overlap in the frequency domain, consider filtering the noise using a Finite Impulse Response (FIR) filter, designed to have linear phase (see Section 3.5).
- In the case of periodic signals, the window should have the same width as the fundamental period in order to avoid spectral leakage.
- It is common practice to 'pad' a time domain signal with zeros in order to gain frequency resolution. In reality, this technique does not improve the resolution associated with the input signal and the only way to truly achieve better resolution in the frequency domain is to use a longer time window. (For a more comprehensive discussion on zero padding, see [6]: http://zone.ni.com/devzone/insights.nsf/webmain/81227DF4C2952C1A86256CA80053F322?OpenDocument).

3.3.4 The Fourier transform of a sequence

Calculations of Fourier coefficients are generally evaluated using a digital computer. One way of introducing the Fourier transform of a sequence involves the use of the mathematical notion of 'ideal sampling' of a continuous signal. If an analogue signal $x(t)$ is sampled every Δ seconds, it is convenient to model the sampled signal as a product of the continuous signal with a 'train' of delta functions $i(t)$ where

$$i(t) = \sum_{-\infty}^{\infty} \delta(t - n\Delta) \tag{3.11}$$

The sampled signal then becomes

$$x_s(t) = x(t)i(t) \tag{3.12}$$

The Fourier transform of $x_s(t)$ is $X_s(f)$ and, using the properties of the delta function, is written

$$X_s(f) = \sum_{n=-\infty}^{\infty} x(n\Delta)e^{-j2\pi fn\Delta} \tag{3.13}$$

Multiplying both sides of equation (3.13) by $e^{-j2\pi fn\Delta}$ and integrating with respect to f from $-1/2\Delta$ to $1/2\Delta$ gives the inverse transform

$$x(n\Delta) = \Delta \int_{-\frac{1}{2\Delta}}^{\frac{1}{2\Delta}} X_s(f)e^{j2\pi fn\Delta}df \tag{3.14}$$

Equations (3.13) and (3.14) relate the sequence of numbers $x(n\Delta)$ to the quantity $X_s(f)$, which is the Fourier transform of the sequence.

3.3.5 The Discrete Fourier Transform (DFT)

In practice, equation (3.13) is not generally used since a finite sequence is acquired. If the sequence consists of N points, the summation is carried out between 0 and N-1. In order to be suitable for machine computation, it is desirable to express the Fourier transform in discrete form by evaluating f at specific points, every $k/N\Delta$ Hz, where k varies between 0 and N-1. The discrete Fourier transform is usually written in the following form:

$$X(k) = \sum_{n=0}^{N-1} x(n)e^{-j2\pi n/N} \tag{3.15}$$

Multiplying both sides by $e^{-j2\pi nk/N}$ and summing over k gives the inverse transform:

$$x(n) = \frac{1}{N}\sum_{k=0}^{N-1} X(k)e^{j2\pi n/N} \tag{3.16}$$

It is important to realise that even though the original sequence $x(n\Delta)$ may have been zero for n outside the range [0, N-1], the act of 'sampling in frequency' has imposed a periodic structure to the sequence, i.e. it follows from equation (3.16) that $x(n + N) = x(n)$.

The evaluation of the DFT can be accomplished efficiently by a set of algorithms known as the Fast Fourier Transform (FFT) when N is a power of two. The FFT algorithm essentially exploits the periodicity and symmetry properties the complex exponential term in the DFT summation.

The ubiquity of this signal processing tool has dangers associated with it. It has become so familiar and it can be applied so straightforwardly that the requirements for its correct use can easily be forgotten. To appreciate the problems associated with the use of the FFT it is useful to return to the definition of the Fourier Transform integral itself. Firstly, the integral is that of a continuous function defined over the range $\pm\infty$.

$$\begin{aligned} H(\omega) &= a_1 \int_{-\infty}^{+\infty} h(t)\, e^{-j\omega t}\, dt \\ h(t) &= a_2 \int_{-\infty}^{+\infty} H(\omega)\, e^{j\omega t} d\omega \end{aligned} \tag{3.17}$$

Note that in the case of the Fourier Transform pair (the transform and its inverse) the two a_i coefficients take different values depending on the conventions adopted by their user. The convention in the Fourier integrals previously adopted in equations (3.6) and (3.7) was $a_1 = 1$ and $a_2 = 1/2\pi$, and the frequency-related variable will be changed to $f = \omega/2\pi$, since these conventions are commonly adopted.

When one is concerned with digitally sampled signals, two problems arise. One samples the continuous function at discrete points. Conventionally, this process is regarded as multiplying the continuous function by a set of discrete spikes or delta functions of unit amplitude. Secondly, one has a finite data set, rather than a signal of infinite duration. Thus one has again multiplied the continuous function by some kind of windowing function, the simplest being a rectangular function of unit amplitude. As described above, convolution theory tells us that multiplication in the time or space domain is equivalent to convolution in the frequency domain, so that one now has a convolution of three spectra: the underlying function, the sampling function and the windowing function. In addition, FFT algorithms regard the windowed data set as representing a data set which is repeated throughout time or space, i.e., the function of infinite duration can be exactly represented by repeats of the windowed data set. Finally, FFT algorithms produce data only up to the cut-off frequency, i.e. $N/2*1/(N*\Delta)$, where N is the number of points in the data set and Δ is the sampling interval. The frequency resolution of an FFT is given by $1/(N*\Delta)$.

Failure to appreciate the subtleties summarised above can lead to erroneous or misleading results. Metrologists need to have a basic understanding at least of sampling and windowing theory if they are to ensure that their use of FFT algorithms produces reliable output.

Recommendation:

- Metrologists should be able to relate the DFT and FFT algorithms that they routinely use to the expression of the Fourier integral in equation (3.7). Although the algorithms for the DFT and the inverse DFT are almost always those in equations (3.15) and (3.16), there may be some subtle variations: some software, such as LabView®, can give the one-sided spectrum, hence ignoring points above the Nyquist frequency. In this case, when N is even, the $N/2+1$ point is ignored. Hence, when performing the inverse DFT operation, it is not possible to retrieve the time-domain waveform according to the algorithm in equation (3.16). It is therefore important that the user knows which algorithm to use depending on the application.

3.4 RANDOM DATA

3.4.1 Introduction

In this section, the signals considered are random in nature and estimation methods for such signals will be discussed based on a single recording of the process. Random signals may be encountered in metrology when dealing with the concept of diffuse fields, particularly in acoustics or electromagnetics.

3.4.2 Estimator for stochastic processes

The problem of the study of estimators for stochastic processes is a wide one, and some idea of the extent of the field may be obtained from [7] (Jenkins and Watts). The autocorrelation function and power spectral density will be chosen as examples. Some of the features that should be considered when analysing a sample time history will be indicated. Spectra and system identification will then briefly be discussed. Unless otherwise stated, it is assumed that processes are continuous, stationary and random. The data is sampled and is therefore subject to aliasing problems. It is assumed that the sampling rates are 'sufficiently high'.

A. The autocorrelation function

If a signal is defined for $0 \le t \le T$, there are two commonly used estimators of the autocorrelation function $R_{xx}(\tau)$. Assuming zero mean processes, they are written as [7]:

$$\begin{aligned} \hat{R}_{xx}(\tau) &= \frac{1}{T}\int_0^{T-|\tau|} X(t)X(t+|\tau|)dt \qquad && 0 \le |\tau| \le T \\ \hat{R}_{xx}(\tau) &= 0 && |\tau| > T \end{aligned} \tag{3.18}$$

and

$$\begin{aligned} \hat{R}_{xx}'(\tau) &= \frac{1}{T-|\tau|}\int_0^{T-|\tau|} X(t)X(t+|\tau|)dt \qquad && 0 \le |\tau| \le T \\ \hat{R}_{xx}'(\tau) &= 0 && |\tau| > T \end{aligned} \tag{3.19}$$

Often, the expression in equation (3.19) is used, since it is unbiased.

Computation of equation (3.18) for sampled data is achieved using

$$\hat{R}_{xx}(m) = \frac{1}{N}\sum_{n=0}^{N-|m|-1} x(n)x(n+|m|) \tag{3.20}$$

The divisor is N-$|m|$ for $\hat{R}_{xx}'(m)$. m is the lag and may take values $0 \le m \le M$-1, where the maximum allowable value for M is N.

B. Power spectral density

The standard estimation methods for the power spectral density are as follows:

- The Fourier transform of the autocorrelation function
- The direct method of averaging periodograms or segment averaging
- Filtering, either bandpass or complex demodulation (heterodyning) and low-pass filtering

Of the three, the most widely used is the direct method of averaging periodograms, and this method will be outlined. [8] (Bingham et al) considers various aspects of spectral estimation.

The method of segment averaging for spectral estimation has become popular owing mainly to its speed of computation. In this method, a data length T is split up into q segments of length T_r (in this case non-overlapping) and the raw periodogram formed for each segment, i.e.,

$$\hat{S}_{xx_i}(f) = \frac{1}{T_r}\left|X_{T_{r_i}}(f)\right|^2 \text{ for } i = 1, 2, \ldots, q \tag{3.21}$$

and to reduce fluctuations, the average is formed

$$\tilde{S}_{xx}(f) = \frac{1}{q}\sum_{i=1}^{q}\hat{S}_{xx_i}(f) \tag{3.22}$$

Whilst this summarises the essential features of the method, [8] gives more elaborate procedures and insight. An important aspect is the use of data windows that are assigned to each segment. Furthermore, segments may be chosen to overlap each other. A criticism of this method is that linear windowing ignores some data because of the tapered shape of the window. Intuitively, the overlapping of segments compensates for this in some way [9] (Welch). The use of a data window on a segment before transformation reduces leakage, but has the disadvantage of also reducing the total energy of that segment, and this must be compensated in the calculation of modified periodograms of the form:

$$\hat{S}_{xx\,i}(f) = \frac{\frac{1}{T_r}\left|\int_{i^{th}\text{interval}} x(t)w(t)e^{-j2\pi ft}\,dt\right|^2}{\frac{1}{T_r}\int_{-\frac{T_r}{2}}^{\frac{T_r}{2}} w^2(t)\,dt} \tag{3.23}$$

where the denominator compensates for the ‘energy reduction’.

As far as computation is concerned, the discrete Fourier transform is used to compute the required modified periodograms and the appropriate summations performed.

It should be noted that these basic considerations also apply to cross-spectral estimations. An estimate for the coherence function can be obtained from estimates of smoothed power and cross-spectra and can serve as a measure of 'degree of linearity' between the input and output of a system:

$$\tilde{\gamma}_{xy}^2 = \frac{\left|\tilde{S}_{xy}(f)\right|^2}{\tilde{S}_{xx}(f)\tilde{S}_{yy}(f)} \tag{3.24}$$

Frequency response functions can be estimated using smoothed spectra. Results for errors and confidence can be found in [10] (Otnes and Enochson). Different types of transfer function estimators exist, depending, for example, on whether the input or output measurement is contaminated by noise.

Recommendations:

- Auto and cross spectra, transfer functions and coherence functions are estimated from finite lengths of data. These estimates suffer bias and variability errors. Bias errors are controlled by resolution: the finer the resolution, the less the bias. Variability errors depend on the amount of averaging employed. When using a transfer function estimator, ensure that it is tailored to the experimental conditions, e.g. whether it is biased or unbiased in the presence of input or output noise. Also note that the presence of feedback and other noise can cause further problems.

- A good understanding of the theory behind random data and its analysis should be gained before any of those techniques are used, unless adequate supervision is provided.

3.5 DIGITAL FILTERING

Computers offer great flexibility in the processing of data, and the vast array of procedures that may be carried out on a sequence of numbers is referred to as 'digital filtering'. This short section is only meant as a guide toward some of the methods that may prove to be relevant. The discussion will be restricted to linear operations on input data with time invariant filters.

If $x(n)$ denotes an input sequence (the time increment Δ is dropped for convenience) and $y(n)$ denotes the output sequence, a difference equation relating the sequences can be written

$$y(n) = -b_1 y(n-1) - b_2 y(n-2) \cdots - b_N y(n-N) + a_0 x(n) + a_1 x(n-1) + \cdots + a_M x(n-M) \tag{3.25}$$

This is a general filter that can be programmed to produce an output sequence, given an input sequence and some starting conditions. The z-transform may be used to solve this equation and this leads to the definition of a transfer function. The z-transform of a sequence of numbers $x(n)$ is given by:

$$X(z) = \sum_{n=-\infty}^{\infty} x(n) z^{-n} \tag{3.26}$$

where z is a complex variable.

By suitable selection of coefficients b_i and a_i and filter orders N and M, the transfer characteristics can be adjusted to approximate some desired form. It is important to note that finite word length effects mean that coefficients cannot be represented exactly, and arithmetic round-off in the multiply operations constitute another source of error.

In equation (3.25), if at least one coefficient b_i is not zero, the filter is recursive, whilst if all the b_i coefficients are zero, then the filter is non-recursive. If the filter has a finite memory, (i.e. if its impulse response function is zero after a certain time), it is then called a Finite Impulse Response or FIR filter, in contrast to Infinite Impulse Response or IIR filters, which have infinite memory. It should be noted that the terms recursive and non-recursive describe how a filter is realised and not whether an impulse response is finite or infinite, though FIR filters are usually non-recursive and IIR filters are usually recursive when implemented.

Recommendation:

- FIR filters have certain advantages over IIR filters. For example, they are always stable and can have linear phase characteristics. Their major disadvantage is that they must be of high enough order to achieve adequate cut-off.

3.6 CEPSTRAL ANALYSIS

Sometimes called homomorphic filtering, Cepstral Analysis is essentially aimed at separating convolved signals. The name arises since it is a spectrum of a (log) spectrum, and some sort of new name seems in order to avoid confusing the spectra [11] (Bogert et al). Applications include echo detection, system identification and pattern recognition and the procedures are sufficiently well developed to be used regularly in speech processing and seismology.

In the calibration of an underwater electroacoustic transducer, reflections from the tank boundaries often arrive before the steady-state response of the transducer can be observed directly. By use of Cepstral analysis techniques, attempts have been made to try and predict the steady-state response from the initial transient-dominated part of the waveform [12] (Robinson and Harris).

4. THE STATE OF THE ART IN DSP

4.1 UNDERSTANDING NOISE PROCESSES AND ADAPTIVE FILTERING

Signals are often contaminated by unwanted noise or interference. Where the signal and noise are located in separate and fixed frequency ranges simple filtering may be sufficient to extract the signal of interest. The FFT itself can be regarded as a form of filter in that it identifies the specific frequency content of a signal so that the components of interest and their amplitudes and phases may be extracted directly. If one requires real-time filtering in the time domain in such cases then analogue filters of the appropriate bandwidth may be useful for noise removal, as may simple linear digital filters. However, for many signals of interest there is overlap with the noise frequencies, and often the frequency band occupied by the noise may change with time or it may not, in fact, be known. In such cases, adaptive filtering is appropriate.

An adaptive filter has two parts: a digital filter with adjustable coefficients and an algorithm to adjust the coefficients of the filter [13] (Ifeachor and Jervis, p 543). The method of filtering is as follows:

- Two signals are applied simultaneously to the filter.
- The first is the contaminated signal, which includes both the signal of interest and the noise, which are assumed to be uncorrelated.
- The second signal is a measure of the contaminating noise signal, which is correlated with the noise signal.
- This second signal is processed by the filter to produce an estimate of the noise signal.
- The estimated noise signal is subtracted from the contaminated signal to estimate the desired signal.
- A feedback loop ensures that the filter coefficients are adjusted to minimise the noise in the estimated signal.
- The filter output signal has two purposes: it provides an estimate of the desired signal, and by subtracting it from the original contaminated signal one obtains an error signal, which is used to adjust the coefficients.

Algorithms for adaptive filtering adjust the filter coefficients so that the error signal is minimized against a mathematical criterion, such as least squares. Examples of algorithms that are commonly used include least mean squares (LMS), recursive least squares (RLS) and the Kalman filter. The algorithms all possess advantages and disadvantages associated with questions such as numerical instability, rate of convergence and difficulties in estimating nonstationary signals. Metrologists who are contemplating the use of adaptive filtering in the work need guidance both on which algorithms might be appropriate for their application and also on how to estimate the contribution to the experimental uncertainty budget that arises from their choice of filter.

If one is not concerned with processing signals in real time, simpler approaches are possible, even when the spectral content of the signal of interest and the noise overlap. One such method is the Wiener filter. In this case two signals are used as simultaneous input to the filter. The contaminated signal is regarded as possessing one component that is correlated with the noise signal (which is also applied to the filter) and another that is uncorrelated with the noise. The filter produces an optimal estimate of the part of the contaminated signal that is correlated with the noise signal, which can then be subtracted to give the error signal. In real-time processing the Wiener filter is often impractical because its calculation requires the autocorrelation matrix and the cross-correlation vector to be known *a priori.* In addition, the calculation requires a matrix inversion process, which can be time consuming. Finally, for nonstationary signals the optimal value of the filter vector has to be continually recalculated. However, for off-line calculations it may allow the metrologist to make a better evaluation of uncertainties arising from noise components.

Recommendation:

- Metrologists who are contemplating the use of adaptive filtering in their work need guidance both on which algorithms might be appropriate for their application and also on how to estimate the contribution to the experimental uncertainty budget that arises from their choice of filter.

4.2 NONSTATIONARY SIGNALS AND TIME-FREQUENCY ANALYSIS

Classically, digital signal processing has been based on models that assume stationary signals (i.e., their properties do not change with time), linearity, and Gaussian noise processes. In cases in which systems are linear and stationary, it is possible to calculate their unit impulse response, which can then be used to predict the response of the system to any excitation of interest.

However, many signals do not meet these classical criteria and recent theoretical advances have allowed such cases to be investigated and understood. The Fourier Transform assumes that the signal on which it acts is stationary over the range in which it exists. One popular approach to the study of signals whose frequency content changes over time is the Short Time Fourier Transform (STFT), which is defined as follows:

$$STFT(t,f) = \int_{-\infty}^{+\infty} x(t+\tau)g(\tau)e^{-j2\pi f\tau}\,d\tau \tag{4.1}$$

The sliding window $g(t)$ thus selects the local frequency properties of the signal. If one takes the squared modulus of the STFT one obtains what is commonly referred to as the spectrogram of the signal. The STFT is thus a method of splitting a nonstationary signal into short-time sliding windows and obtaining the Fourier spectrum in each window. Its limitations are that the window length should be short enough for the signal to be considered to be stationary so that frequency components are located in time with good resolution. However, the window must be long enough to guarantee the required frequency resolution. In addition, as one is using the same window type and length across the whole range of the original data set one must be aware of the artefacts which may arise from inappropriate windowing. NPL metrologists have studied the application of the STFT to the calibration of power frequency harmonic analysers [14] (Wright). As Wright points out, if the window function is rectangular the window must be carefully synchronized to avoid Gibb's effects. It is also not possible to increase indefinitely the number of windows as the product of time resolution and frequency resolution remains constant.

An alternative means of investigating nonstationary signals, the Wigner-Ville distribution, has also been studied at NPL [14]. The Wigner distribution is obtained by taking autocorrelations of the signal as a time lag is varied, followed by the Fourier Transform of the result with respect to the lag variable. To avoid unwanted interference from negative frequency components of the sample sequence, the signal under test is converted to its analytic form by means of the Hilbert Transform. The result is the Wigner-Ville distribution, due to Ville, who first proposed this technique. Wright found that the Wigner-Ville distribution had some advantages over the STFT, but that the distribution contained interference terms that caused large errors at intermediate frequencies.

Other transforms, such as those using Wavelet methods provide a means of studying time-frequency variation. However, they do not provide results in the form of sine wave basis functions, which are often what is required in metrology and calibration operations. NPL mathematicians have made some studies of the potential benefits of the use of wavelets in metrology [15] (Lord et al) [16] (Harris et al) and concluded that they provide an effective basis for time-frequency analysis. They cannot be regarded as a stand-alone tool but as complementing and supplementing well-established techniques.

Wavelets are being used in a harmonic burst detection system applied to the calibration of harmonic analysers under fluctuating harmonic conditions. This is part of project D4.3 in the NMS 2000-2003 Electrical Metrology programme. Wavelets do not readily lend themselves to harmonic analysis because the results from the transforms give rise to coefficients of wavelet basis functions rather than the required sinusoidal basis functions. However, they have found use in this project as a method of detecting discontinuities in various harmonics. Having identified the position of a discontinuity, it is possible under some conditions to adapt a short time Fourier transform to the signal in order to measure the harmonic content of the waveform. The system uses the information from the wavelet to adjust the window positions and sizes to reduce leakage errors.

Recommendation:

- Existing good practice developed in the NMS Electrical Metrology programme and the results of studies on wavelets in metrology should be disseminated more widely to metrologists who are concerned with the analysis of nonstationary signals. Of particular concern is the identification and quantification of the contributions to calibration uncertainty budgets that arise from the choice of signal processing method for these kinds of signals and the limitations of the respective techniques.

5. ISSUES TO BE ADDRESSED BY SS*f*M

5.1 HARDWARE

In many modern measurement systems, metrologists make use of commercial apparatus, the performance of which may not be fully understood, or which contains embedded signal processing facilities that appear either in the form of DSP chips or in firmware or software. Not only do metrologists need to know that they are using the apparatus in question appropriately (within its performance bandwidth, say) but they need to be able to quantify the uncertainties that the instrumentation introduces into the measurement process. Here, it is often insufficient to rely on data provided in manufacturer's catalogues and the metrologist may need to carry out extensive tests to evaluate type B uncertainties.

Clearly, from what has been discussed in Section 3, there is the potential for DSP-related uncertainties to make a contribution to both type A and type B uncertainties in a measurement. This uncertainty can often be minimised by appropriate signal conditioning before digitisation, good quality ADC's, use of appropriate sampling frequency, awareness of limitations of windowing etc. There is also a need to characterise ADC's for applications where accuracy is important.

At present, it appears that many metrologists employ a 'black-box' type approach with their DSP hardware and are not aware of the details of, for example, how the analogue-to-digital conversion process may affect the signals they acquire, whether or not it is important to use an anti-aliasing filter on the signal prior to digitisation and, if so, what type of filter should be used. This is particularly true of in the NMS Acoustics programme and in the NMS Optical programme. There is a clear need for the metrologist to understand the basic principles of operation of the DSP hardware used and how uncertainties may arise due to the limitations of this hardware. This may involve the following:

a) Devising a test plan for the hardware;
b) Obtaining detailed specifications of the hardware from the manufacturer;
c) Using custom-built hardware, so that it can be tailored to the purpose of the experiment and so that there is a complete understanding of all the operations it performs.

Note that these points are not mutually exclusive. Although point c) represents the most difficult approach, it should be noted that it is the most flexible in terms of traceability requirements. From the replies to the questionnaire in Appendix A, it appears that many organisations claim to be benefiting from using purpose-built hardware.

It may be worth establishing a document (Good Practice Guide) to guide the metrologist in the purchasing of hardware that is best suited to the measurement to be carried out. Particular attention should be paid to the following:

- What type of signal is the DSP hardware going to be acquiring (nature of signal, frequency content, dynamic range required);

- Does the hardware have an anti-aliasing filter and if so, what are its characteristics in terms pass-band ripple, cut-off frequency, attenuation after cut-off and phase response. How does the filter affect the signal being acquired?
- Errors in sample and hold devices in analogue-to-digital converters;
- Quantisation errors in analogue-to-digital converters;
- How does one derive type B uncertainties for digital hardware?

Algorithms that are implemented in the device will be dealt with in the following Section.

5.2 SOFTWARE

DSP algorithms are themselves sources of both type A and type B uncertainties. The outputs of spectral analysis routines will all depend on exactly how the metrologist windows the data, the window position and shape, the window length and on the manner in which the windowing truncates specific frequency components. The results of filtering processes will vary depending on the type of filter, the number of coefficients the filter algorithm contains, and the filter's phase response. If metrologists are processing large amounts of data they may also wish to automate the windowing and filtering process. Can they define algorithms that ensure that the software always makes the best choice of window length and position? If the process is not automated but the metrologist is allowed to choose window and filter parameters for specific signals of interest, can we ensure that consistent practices are adopted?

A test plan should be formulated to evaluate the performance of DSP routines in commercially available software and hardware. Where possible, the algorithms used should be traceable and their correct implementation verified.

Some algorithms may require input parameters other than the sequence of numbers to be operated on. For example, the algorithm that provides a Power Spectral Density estimate of a discrete-time signal vector using Welch's averaged, modified periodogram method, may provide the user with a choice of windows.

In-house developed DSP algorithms would probably not require treatment different than other in-house generated software/code.

It would be interesting to send a signal generator to a few laboratories and get them to carry out some signal analysis on its output using their equipment and algorithms and send the results back to get a feel for the extent of the discrepancies involved. In 1999, the EMC Testing Laboratories Association (EMCTLA) conducted an inter-comparison study to assess the performance and operation of various commercial harmonic analysers as used for mains harmonic measurements in support of international standard IEC 61000-3-2. The associated analyser designs were specified in the international standard, which included details on window size and synchronisation, filtering and other aspects. This inter-comparison was conducted by use of a variable electrical load that could induce an harmonically rich current to be drawn from a mains level supply. Each participating laboratory measured the harmonic currents drawn by the loads using their own harmonic analysis equipment and results were compared. The results revealed some significant discrepancies between the laboratories, in particular in the application of the analysis equipment to the assessment of EMC compliance limits. It is interesting to note that even when using "high end" commercial analysis equipment designed as prescribed by an international standard, there are still significant departures in the results obtained in the inter-comparison. It would be of value to repeat a similar exercise where different DSP users measure a travelling waveform source using their own analysis equipment to compare results. Such an exercise could build in some of the pitfalls identified in this report, for example high frequency tones likely to cause aliasing or inter-harmonics components likely to cause "picket fencing".

5.3 TRAINING AND BEST PRACTICE

Although most of the people who responded to the questionnaire in Appendix A appeared to have a good grasp of the theory behind DSP, it appears that this is not always the case amongst metrologists. Measurement scientists may often use DSP software and hardware without a good appreciation of the potential and limitations of these tools. Discussions with a number of metrologists working on various NMS programmes have demonstrated that best practice is not always adopted. Crucial questions to be addressed by SS*f*M about DSP are:

- What do metrologists already know?
- What do they need to know?
- Have they some appreciation of the limits of their own knowledge?
- How can the SS*f*M programme help?

Although it is unrealistic to expect all measurement scientists and technicians to be DSP experts, these need to be able to have an appreciation, or be in a position to exchange information with someone who has an appreciation, of DSP-related issues that arise in metrology. Individual NMS programme teams must also take into account their own DSP requirements and provide their staff with adequate training if necessary, but progress in assessing DSP-related uncertainties in teams where knowledge of DSP is poor is unlikely to occur unless some degree of awareness and external guidance is provided.

A DSP Good Practice Guide in which basic DSP topics (e.g. sampling, analogue-to-digital conversion, basic concepts behind Fourier series and transforms, signal truncation, etc.) are covered may prove to be useful for some metrologists, but this should not be a substitute for consultation with someone with good knowledge of DSP to ensure that DSP-related uncertainties are minimised during the course of the measurement and during eventual post-processing of the measured data. This Good Practice Guide would include an in-depth discussion of the topics discussed in sections 3 and 4. It should be noted that it will be challenging to write such a guide without consideration of the mathematics behind DSP, and a balance will have to be struck so that the guide is accessible to a majority of its intended audience.

As part of the issues to be addressed by the next SS*f*M programme, it may be worthwhile investigating some case studies to find out if DSP-related issues cause a substantial underestimate of the overall measurement uncertainties.

In order to complement the information provided by the DSP Good Practice Guide, it would be worthwhile to create a DSP Support Group. The group would consist of individuals highly knowledgeable in various areas of DSP, which would be given funding to address DSP-related queries. Instead of being part of a specific team, this support group should be set up as an NMS virtual centre, similar in concept to NPL's virtual mathematical modelling centre.

Finally, in order to have complete control over a measurement, it may be worth considering having a dedicated DSP engineering capability, which would be able to assemble hardware depending on specifications provided by the metrologist. This would allow control over components such as analogue-to-digital converters, anti-aliasing filters, etc. One would then no longer have to rely on the manufacturer's specifications for a given piece of hardware, since the source of all components used would be traceable.

5.4 RELATIONSHIP TO CORE SS*f*M THEMES

Within the SS*f*M programme, there are core themes or topic areas. The following ones are relevant to DSP:

- Modelling techniques, including model validation
- Modelling tools, including numerical algorithms and software
- Uncertainty evaluation and statistical techniques
- Numerical software and algorithm testing

It is recommended that the next SS*f*M programme should consider including under each of these themes the following DSP-related projects:

- Guidance and case studies on modelling techniques to be used for signal processing, including when to use analogue and when to use digital techniques, and guidance on the validation of the techniques chosen
- Guidance and case studies on algorithms and software to be used for signal processing, with particular emphasis on DSP, and where necessary the development of improved algorithms with better properties for metrological applications
- Guidance and case studies on uncertainty evaluation related to signal processing, particularly to the main techniques used in DSP
- Numerical software and algorithm testing applied to the most commonly used DSP algorithms and software implementations of the same to check their numerical adequacy, where possible providing web-enabled data generators to be used in such testing

Clearly, all the guidance developed could be gathered together into a Good Practice Guide. It is probably premature to attempt to produce a Best Practice Guide.

6. SUMMARY AND CONCLUSIONS

This report has attempted to identify the signal processing issues that are of most concern to metrologists, and to make recommendations for the Software Support for Metrology programme, which will lead to work that will assist metrologists in applying best practice in their DSP-related measurement activities.

The report summarises the consultation with the DSP user community, the metrological requirements to be met by DSP and the state of the art in DSP, and identifies best practices within the DSP field that might need dissemination within metrology.

The successful implementation of these recommendations relies on the creation of a DSP quality system in which the following points will be addressed:

- Test plan for DSP software, algorithms and hardware (avoid relying on manufacturer's specifications)
- Evaluation of how DSP software, algorithms and hardware can contribute to type A and type B uncertainties

Metrologists need to have an appreciation of this quality system, which may be achieved through training and the creation of a DSP Good Practice Guide and consultation with members of staff who possess a good knowledge of DSP. These DSP experts may be part of an NMS Virtual DSP Centre, which would receive funding for these consultations.

Also, it may be worth progressively switching to in-house and/or custom-built hardware and software, where all aspects in which the signal is processed are known, so that more control over uncertainties is achieved.

Metrology often has to adopt a conservative approach to the choice of technologies for particular applications, to ensure that the measurement process can be shown to be on a firm footing. It can take some time for the leading edge of research in the academic community to be adopted as routine in measurement laboratories. This process is an inevitable result of the metrologist's need to understand fully the sources of uncertainty in the measurement process.

7. ACKNOWLEDGEMENTS

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APPENDIX A – DSP QUESTIONNAIRE

SS*f*M Digital Signal Processing requirements study

As part of the National Measurement System (NMS) Software Support for Metrology (SS*f*M) New Directions Project, NPL is conducting a study to review the state of the art and metrological requirements of Digital Signal Processing (DSP) in order to identify key issues for the NMS that will need to be tackled in the next SS*f*M programme. Particular attention is to be given to error modelling, uncertainty evaluation, validation and traceability requirements.

In this study, NPL is contacting a number of recognised experts in the area of DSP and requesting their views on the use of DSP in the context of metrology. The response to this questionnaire will only be published in summary form, not attributed to specific respondents, although we would like your permission to include your organisation name in a list of organisations consulted.

Please e-mail your response to pierre.gelat@npl.co.uk or fax it to 020 8943 6161 marked "For the attention of Pierre Gélat". Responses are requested by the end of October 2002. Many thanks, in advance, for providing this response, which will help to secure DTI funding for relevant DSP related projects in the future.

Name:
Organisation:
Address:

1. What applications do you have/expect to have in the future for digital signal processing in your current business/project/research?

2. How do you/your organisation/research team define digital signal processing?

3. What hardware do you use for signal acquisition? Is it purpose-built or do you just combine existing proprietary equipment to do what you want? Who typically designs and implements DSP systems in our organisation?

4. Do you design your own transforms or DSP algorithms? Do you use modelling software and if so, which? Do you test algorithms by modelling? Do you test transform algorithm/code by repeated inversion to look for errors?

5. Do you do your signal processing in real time or on stored data after the acquisition has been completed? Do you process mainly in the time domain or the frequency domain?

6. Do you have staff who have specific skills/training in DSP? Do you think there is any benefit in having specifically trained staff or do you expect that DSP should be part of the range of skills needed by all physicists and engineers?

7. What do you think are the main pitfalls facing inexperienced DSP users? Can you recount any specific problems/mistakes etc. from your own/your organisation's/team's experience?

8. Can you identify any specific training needs for yourself or others within your team, which you think would be of benefit to you and your colleagues?

9. Do you consider the effect that the measurement system has on your measurements, e.g. do you deconvolve instrumentation/transducer responses and so on? If so, how?

10. Do you make relative measurements or absolute measurements? Are traceability issues important to you?

11. Does your organisation have a quality system that governs the development of DSP systems (both hardware and software)?

12. Are you willing for your organisation to be named in the list of contributing organisations?

APPENDIX B – DSP QUESTIONNAIRE REPLIES

Name: Alan A. Smith
Organisation: Qualcomm UK Ltd

1. What applications do you have/expect to have in the future for digital signal processing in your current business/project/research?

   We use DSP's in all communications chipsets to perform front-end signal recovery/demodulation, message decoding, message encoding and modulation.

   We use another identical DSP to perform voice coding/decoding.

2. How do you/your organisation/research team define digital signal processing?

   The processing of real-time signals in the digital domain.

3. What hardware do you use for signal acquisition? Is it purpose-built or do you just combine existing proprietary equipment to do what you want? Who typically designs and implements DSP systems in our organisation?

   We use our own proprietary, purpose built, chipsets for signal acquisition. The specification for these is partly determined by the architecture and interface requirements of the DSP. The DSP cores are design by the ASIC DSP group.

4. Do you design your own transforms or DSP algorithms? Do you use modelling software and if so, which? Do you test algorithms by modelling? Do you test transform algorithm/code by repeated inversion to look for errors?

   We design all of our own algorithms. We first model the algorithm in a simulation written by the systems group in C++. Once proven, the DSP group then code the algorithm in DSP firmware to be bit-exact with the system simulation.

5. Do you do your signal processing in real time or on stored data after the acquisition has been completed? Do you process mainly in the time domain or the frequency domain?

   We acquire the receive symbol samples into a circular RAM buffer using DMA. Once the DMA has finished, we trigger an interrupt to initiate processing.

6. Do you have staff who have specific skills/training in DSP? Do you think there is any benefit in having specifically trained staff or do you expect that DSP should be part of the range of skills needed by all physicists and engineers?

   The DSP group contains systems engineers, VHDL ASIC designers, a wide variety. It is essential to have broad skills in the DSP group as issues frequently have multiple effects.

7. What do you think are the main pitfalls facing inexperienced DSP users? Can you recount any specific problems/mistakes etc. from your own/your organisation's/team's experience?

   There is a steep learning curve to DSP. The user needs to appreciate the subtleties of the DSP architecture to truly get the best from the DSP.

8. Can you identify any specific training needs for yourself or others within your team, which you think would be of benefit to you and your colleagues?

   As we design our own DSP cores, we also provide internal training on them. Using them is the best way to learn.

9. Do you consider the effect that the measurement system has on your measurements, e.g. do you deconvolve instrumentation/transducer responses and so on? If so, how?

   Absolutely, we fight for every dB of sensitivity we can get !

10. Do you make relative measurements or absolute measurements? Are traceability issues important to you?

    We take both absolute and relative measurements. Traceability and testability are crucial.

11. Does your organisation have a quality system that governs the development of DSP systems (both hardware and software)?

    We have a defined coding standard that all DSP engineers follow. In addition, we draft Design Documents and review them as a group prior to implementation.

12. Are you willing for your organisation to be named in the list of contributing organisations?

    Yes.

Name: Andrew Hurrell
Organisation: Precision Acoustics Ltd

1. What applications do you have/expect to have in the future for digital signal processing in your current business/project/research?

   DSP used extensively in both hydrophone calibration (FFT to obtain frequency spectra) and in hydrophone deconvolution. This is expected to increase when phase calibration data is available from NPL

2. How do you/your organisation/research team define digital signal processing?

   We define DSP as the processing any of signals that have been acquired through a digital sampling operation.

3. What hardware do you use for signal acquisition? Is it purpose-built or do you just combine existing proprietary equipment to do what you want? Who typically designs and implements DSP systems in our organisation?

   Either LeCroy 9310AM DSO or proprietary data acquisition card acquired from SI Transducers. Implementation typically by myself or David Bell. DSP sometimes accomplished on dedicated hardware and sometimes in software.

4. Do you design your own transforms or DSP algorithms? Do you use modelling software and if so, which? Do you test algorithms by modelling? Do you test transform algorithm/code by repeated inversion to look for errors?

   DSP algorithms are designed in house, often with aid of Labview, but sometimes with MathCAD, Matlab (V2) or in C. Algorithms always tested with modelled / test data prior to release. Repeated inversion only useful when transform is unitary; non-unitary transforms cannot be tested in this way

5. Do you do your signal processing in real time or on stored data after the acquisition has been completed? Do you process mainly in the time domain or the frequency domain?

   For routine calibration DSP is conducted real-time, as is hydrophone de-convolution. For more experimental work, DSP is normally a post-processing operation. Processing is a 70/30 split frequency/time domain

6. Do you have staff who have specific skills/training in DSP? Do you think there is any benefit in having specifically trained staff or do you expect that DSP should be part of the range of skills needed by all physicists and engineers?

   All physicists and most engineers should be reasonably capable in the basic aspects of DSP. Should additional capability be required, these staff should be willing and able to learn the necessary new skills

7. What do you think are the main pitfalls facing inexperienced DSP users? Can you recount any specific problems/mistakes etc. from your own/your organisation's/team's experience?

   - Aliasing, and other assumed periodicity errors
   - FFT double sided to singled sided transformation in that it is not sufficient to just consider the first N/2 data points (as many common packages including LabVIEW do). This approach only produces frequencies from 0 -> N/2 –1, and neglects the N/2th point (i.e. the Nyquist frequency)

8. Can you identify any specific training needs for yourself or others within your team, which you think would be of benefit to you and your colleagues?

   No, we have sufficient expertises in-house

9. Do you consider the effect that the measurement system has on your measurements, e.g. do you deconvolve instrumentation/transducer responses and so on? If so, how?

Yes FFT deconvolution of hydrophone response is often conducted, but the limit to the problem is obtaining the phase response of the hydrophone. Magnitude response is readily available from NPL, but until the NPL phase calibration facility comes on-line the only option is to model the phase response of a hydrophone. De-convolution conducted in frequency domain by dividing the measured spectrum, by the response spectrum and then inverse Fourier transforming back to time domain.

10. Do you make relative measurements or absolute measurements? Are traceability issues important to you?

Both. For relative measurements traceability is not important, but for absolute measurements we use hydrophones calibrated against national standards.

11. Does your organisation have a quality system that governs the development of DSP systems (both hardware and software)?

Not formally, although at all points we endeavour to follow rigorous and precise measurement practice, with all procedures being thoroughly checked before "going live".

12. Are you willing for your organisation to be named in the list of contributing organisations?

Yes.

Name: Did not wish to be named

Organisation: N/A

1. What applications do you have/expect to have in the future for digital signal processing in your current business/project/research?

   I have DSP to achieve several measurements and mathematics operations with a high sampling rate (20MHz)

2. How do you/your organisation/research team define digital signal processing?

3. What hardware do you use for signal acquisition? Is it purpose-built or do you just combine existing proprietary equipment to do what you want? Who typically designs and implements DSP systems in our organisation?

   The use of DSP was purpose-built. The design and implemented were achieved by Ultra Electronics Weymouth.

4. Do you design your own transforms or DSP algorithms? Do you use modelling software and if so, which? Do you test algorithms by modelling? Do you test transform algorithm/code by repeated inversion to look for errors?

   No.

5. Do you do your signal processing in real time or on stored data after the acquisition has been completed? Do you process mainly in the time domain or the frequency domain?

   Real time in Time domain.

6. Do you have staff who have specific skills/training in DSP? Do you think there is any benefit in having specifically trained staff or do you expect that DSP should be part of the range of skills needed by all physicists and engineers?

   No. DSP is part of electronics skills, and therefore I would contract an electronics company to provide me with DSP algorithms and programs.

7. What do you think are the main pitfalls facing inexperienced DSP users? Can you recount any specific problems/mistakes etc. from your own/your organisation's/team's experience?

8. Can you identify any specific training needs for yourself or others within your team, which you think would be of benefit to you and your colleagues?

9. Do you consider the effect that the measurement system has on your measurements, e.g. do you deconvolve instrumentation/transducer responses and so on? If so, how?

   No.

10. Do you make relative measurements or absolute measurements? Are traceability issues important to you?

    I do only absolute measurements. Traceability is an important issue for me.

11. Does your organisation have a quality system that governs the development of DSP systems (both hardware and software)?

    No.

12. Are you willing for your organisation to be named in the list of contributing organisations?

    No, since I have not expertise in DSP

Name: Dr. Richter, Dieter and Dr. Link, Alfred
Organisation: PTB, Department of Metrological Information Technology

1. What applications do you have/expect to have in the future for digital signal processing in your current business/project/research?

   Analysis of biomedical signals and development of DSP procedures for interferometric measurements of mechanical quantities.

2. How do you/your organisation/research team define digital signal processing?

   Extraction of information from discrete-time signals.

3. What hardware do you use for signal acquisition? Is it purpose-built or do you just combine existing proprietary equipment to do what you want? Who typically designs and implements DSP systems in our organisation?

   Purpose-built and existing propriety equipment.

4. Do you design your own transforms or DSP algorithms? Do you use modelling software and if so, which? Do you test algorithms by modelling? Do you test transform algorithm/code by repeated inversion to look for errors?

   Our design of DSP algorithms includes tests by modelling.

5. Do you do your signal processing in real time or on stored data after the acquisition has been completed? Do you process mainly in the time domain or the frequency domain?

   We design DSP procedures for offline processing, mainly in the time domain.

6. Do you have staff who have specific skills/training in DSP? Do you think there is any benefit in having specifically trained staff or do you expect that DSP should be part of the range of skills needed by all physicists and engineers?

   We have specially trained staff and think it is beneficial to have it.

7. What do you think are the main pitfalls facing inexperienced DSP users? Can you recount any specific problems/mistakes etc. from your own/your organisation's/team's experience?

   One of the pitfalls may be to give a real estimate of the random effects on the output signals.

8. Can you identify any specific training needs for yourself or others within your team, which you think would be of benefit to you and your colleagues?

   No.

9. Do you consider the effect that the measurement system has on your measurements, e.g. do you deconvolve instrumentation/transducer responses and so on? If so, how?

   No.

10. Do you make relative measurements or absolute measurements? Are traceability issues important to you?

    Relative and absolute. Yes.

11. Does your organisation have a quality system that governs the development of DSP systems (both hardware and software)?

    A generally quality system exists, but not tailored for the development of DSP systems.

12. Are you willing for your organisation to be named in the list of contributing organisations?

    Yes.

Name: Dr Malcolm Woolfson
Organisation: Did not get permission for it to be named

1. What applications do you have/expect to have in the future for digital signal processing in your current business/project/research?

   Biomedical Engineering, Communications, Optical, Ultrasound

2. How do you/your organisation/research team define digital signal processing?

   The application of digital hardware and computer methods to the filtering and analysis of signals. Some colleagues lump DSP together with Communications.

3. What hardware do you use for signal acquisition? Is it purpose-built or do you just combine existing proprietary equipment to do what you want? Who typically designs and implements DSP systems in our organisation?

   DSP boards usually. Sometimes purpose-built systems. Processing tends to be done on a computer, although VLSI implementation is also carried out.

4.1 Do you design your own transforms or DSP algorithms?

Yes.

4.2 Do you use modelling software and if so, which?

Most of my colleagues working in this area use MATLAB. I still use Turbo Pascal.

4.3 Do you test algorithms by modelling?

Yes, extensively.

4.4 Do you test transform algorithm/code by repeated inversion to look for errors?

Do not know the term "repeated inversion"

5.1 Do you do your signal processing in real time or on stored data after the acquisition has been completed?

Mainly off-line, although algorithms are designed with on-line applications in mind.

5.2 Do you process mainly in the time domain or the frequency domain?

Both domains equally.

6.1 Do you have staff who have specific skills/training in DSP?

Yes.

6.2 Do you think there is any benefit in having specifically trained staff or do you expect that DSP should be part of the range of skills needed by all physicists and engineers?

There is benefit by having specially trained staff who have a specialist knowledge the field of signal processing.

All electrical, electronic and mechanical engineers and also applied physicists should have a basic knowledge of signal processing techniques. This should not be a requirement for pure physicists.

7.1 What do you think are the main pitfalls facing inexperienced DSP users?

Applying algorithms blindly without knowledge of the limitations of the method used. For example not knowing about leakage effects in DFT's. A good grounding in the theory of a method should be obtained before one implements it in practice.

7.2 Can you recount any specific problems/mistakes etc. from your own/your organisation's/team's experience?

From my own experience I applied Maximum Entropy analysis to the analysis of some data and found one peak in the spectrum for a particular model order. I showed this to a colleague, who suggested that I tried a range of model orders and I then found another peak! I now carry out such calculations for a variety of model orders, which of course is accepted practice.

8. Can you identify any specific training needs for yourself or others within your team, which you think would be of benefit to you and your colleagues?

Overview of new techniques in signal processing, developed for a wide range of applications. I find the IEEE Signal Processing magazine particularly useful.

9. Do you consider the effect that the measurement system has on your measurements, e.g. do you deconvolve instrumentation/transducer responses and so on? If so, how?

I do not, but I know of colleagues working in optical and ultrasound signal processing who do.

10. Do you make relative measurements or absolute measurements? Are traceability issues important to you?

We process both types of data

11. Does your organisation have a quality system that governs the development of DSP systems (both hardware and software)?

Not an official one, although myself and colleagues have our own informal systems in place.

12. Are you willing for your organisation to be named in the list of contributing organisations?

I am replying on an individual basis, not on behalf of the University or School, so I suggest that you contact my Head of School for permission.

Name: Optical Radiation Measurement Group
Organisation: NPL

1. What applications do you have/expect to have in the future for digital signal processing in your current business/project/research?

   We use digital signals in many of our optical radiation measurement systems. For example, we use imaging systems such as CCD arrays to collect information on how the incident optical radiation varies with time or position, and manipulate the digital signals to obtain information on colour or intensity. We are also seeking to link these measurements to 'visual' effects such as gloss and texture.

2. How do you/your organisation/research team define digital signal processing?

   The manipulation of digital data from a measurement system to obtain a measurement result which is expressed in non-digital terms.

3. What hardware do you use for signal acquisition? Is it purpose-built or do you just combine existing proprietary equipment to do what you want? Who typically designs and implements DSP systems in our organisation?

   We generally use proprietary equipment. Any modifications required are usually carried out by members of the project team, although we sometimes sub-contract outside.

4. Do you design your own transforms or DSP algorithms? Do you use modelling software and if so, which? Do you test algorithms by modelling? Do you test transform algorithm/code by repeated inversion to look for errors?

   What we have done so far is relatively straightforward manipulation of digital data, which hasn't required the development of algorithms. We may need to do this into the future, however.

5. Do you do your signal processing in real time or on stored data after the acquisition has been completed? Do you process mainly in the time domain or the frequency domain?

   Using stored data.

6. Do you have staff who have specific skills/training in DSP? Do you think there is any benefit in having specifically trained staff or do you expect that DSP should be part of the range of skills needed by all physicists and engineers?

   For simple processing we have staff in the team with suitable skills. For more complex tasks we would prefer to sub-contract to specialists.

7. What do you think are the main pitfalls facing inexperienced DSP users? Can you recount any specific problems/mistakes etc. from your own/your organisation's/team's experience?

   Don't know!

8. Can you identify any specific training needs for yourself or others within your team, which you think would be of benefit to you and your colleagues?

   No.

9. Do you consider the effect that the measurement system has on your measurements, e.g. do you deconvolve instrumentation/transducer responses and so on? If so, how?

   We consider this as part of the uncertainty analysis.

10. Do you make relative measurements or absolute measurements? Are traceability issues important to you?

    Both. Yes, traceability is essential.

11. Does your organisation have a quality system that governs the development of DSP systems (both hardware and software)?

    Yes.

12. Are you willing for your organisation to be named in the list of contributing organisations?

    Yes.

Name: R. J. Chaney
Organisation: Renishaw plc, Wotton-Under-Edge

1. What applications do you have/expect to have in the future for digital signal processing in your current business/project/research?

   We use digital signal processing in metrology and spectroscopic applications in the past and in the future. Two examples of where we use DSP are converting a reading from a sensor and converting it into metrology units, and in motion control.

2. How do you/your organisation/research team define digital signal processing?

   Depends who you talk to within the organisation. Some refer to DSP as nearly being synonymous with the term firmware, to that which is used in a DSP chip.

3. What hardware do you use for signal acquisition? Is it purpose-built or do you just combine existing proprietary equipment to do what you want? Who typically designs and implements DSP systems in our organisation?

   That which is deemed to be cost effective at the time. We design our own circuits using commercially available chips, to using commercially complete DSP boards, and hybrids between. We are willing to use a single chip with analogue inputs with digital outputs containing a DSP.

4. Do you design your own transforms or DSP algorithms? Do you use modelling software and if so, which? Do you test algorithms by modelling? Do you test transform algorithm/code by repeated inversion to look for errors?

   Preferably we like to use already existing code. Typical sources consulted include: Numerical Recipes and other textbooks, Manufacturers of DSP chips code, previous Renishaw code.

   We use many methods of checking code including modelling, repeated inversion, checking against other mathematical packages including Mathematica, MatLab and MathCad.

5. Do you do your signal processing in real time or on stored data after the acquisition has been completed? Do you process mainly in the time domain or the frequency domain?

   DSP is usually used for "real time" signal processing. We tend to use PCs for manipulating stored data. Although some people within the company would refer to this as DSP. Most of our applications manipulate data in the time domain.

6. Do you have staff who have specific skills/training in DSP? Do you think there is any benefit in having specifically trained staff or do you expect that DSP should be part of the range of skills needed by all physicists and engineers?

   Our staff who write DSP code tend to be trained software or electrical engineers, and tend to be self taught, i.e. learn DSP whilst doing the project because it is the solution to the problem. At this point in time we do not have the need for all our Engineers to be DSP trained, only some.

7. What do you think are the main pitfalls facing inexperienced DSP users? Can you recount any specific problems/mistakes etc. from your own/your organisation's/team's experience?

   The main pitfalls I can think of at the moment include: unreliable code (it works most of the time but not all of the time), incomplete or even false documentation so it is difficult to maintain the code. Maintaining a version log. Using inefficient code so that the duty cycle is considerably longer than necessary. Etc.

8. Can you identify any specific training needs for yourself or others within your team, which you think would be of benefit to you and your colleagues?

   Not at the moment.

9. Do you consider the effect that the measurement system has on your measurements, e.g. do you deconvolve instrumentation/transducer responses and so on? If so, how?

   Yes, and in some instances with great difficulty. The methods that I have used to date include mathematically modelling the code to deduce the code errors. Using accurate instrumentation to measure the transducer's performance. Investigating the final result to check that the final result is an appropriate sum of the parts. For transient responses FFT analysis has been used. Etc.

10. Do you make relative measurements or absolute measurements? Are traceability issues important to you?

    Yes, for our calibration instruments and instruments requiring calibration certificates getting results that are traceable back to the National Lab is very important, and is key to our marketing success.

11. Does your organisation have a quality system that governs the development of DSP systems (both hardware and software)?

    Yes.

12. Are you willing for your organisation to be named in the list of contributing organisations?

    Yes.

Footnote:

To answer this questionnaire in time I have not consulted all relevant parties within the company so the views above are my own and cannot be attributed to the company.

Name: Robert Jones
Organisation: Renishaw plc, Edinburgh

1. What applications do you have/expect to have in the future for digital signal processing in your current business/project/research?

   Wavelets, de-noising, advances in 3D scanning techniques, sophisticated error mapping techniques, advanced scanning and calibration procedures.

2. How do you/your organisation/research team define digital signal processing?

   Any sampled data system (the design there of).

3. What hardware do you use for signal acquisition? Is it purpose-built or do you just combine existing proprietary equipment to do what you want? Who typically designs and implements DSP systems in our organisation?

   Purpose-built hardware. This is mainly constructed by our electronics engineers.

4. Do you design your own transforms or DSP algorithms? Do you use modelling software and if so, which? Do you test algorithms by modelling? Do you test transform algorithm/code by repeated inversion to look for errors?

   Yes (ish). We use Matlab and Simulink. We often tend to modify existing algorithms found in scientific papers. We tend to look for errors by comparing our algorithms with known standards.

5. Do you do your signal processing in real time or on stored data after the acquisition has been completed? Do you process mainly in the time domain or the frequency domain?

   Our signal processing is done <u>BOTH</u> in real time and after the acquisition has been completed. Processing is done mainly in the time domain but the frequency domain is used on occasion.

6. Do you have staff who have specific skills/training in DSP? Do you think there is any benefit in having specifically trained staff or do you expect that DSP should be part of the range of skills needed by all physicists and engineers?

   Yes, and I think that there is a place for staff who have specific DSP skills, especially as your mathematics knowledge needs to ramp up with the latest developments.

7. What do you think are the main pitfalls facing inexperienced DSP users? Can you recount any specific problems/mistakes etc. from your own/your organisation's/team's experience?

   Yes, inexperienced DSP users can often misuse the data that gets filtered out from what they may consider a ‘black-ox’. A simple example of this was where n inadequate low pass filter was put in line with our ADC leaving us with aliased data “noise” in our software.

8. Can you identify any specific training needs for yourself or others within your team, which you think would be of benefit to you and your colleagues?

   Yes, training to keep up with the latest techniques in measurement; good practice and bad practice; perhaps a course on wavelets.

9. Do you consider the effect that the measurement system has on your measurements, e.g. do you deconvolve instrumentation/transducer responses and so on? If so, how?

   Not in my area.

10. Do you make relative measurements or absolute measurements? Are traceability issues important to you?

    We take absolute special measurements. Traceablility issues are <u>very</u> important to us.

11. Does your organisation have a quality system that governs the development of DSP systems (both hardware and software)?

    Not specifically for DSP, but yes if you include the general remit of hardware and software development.

12. Are you willing for your organisation to be named in the list of contributing organisations?

    Yes.

Name: Stewart Wylie
Organisation: BAE SYSTEMS

1. What applications do you have/expect to have in the future for digital signal processing in your current business/project/research?

   Frequency to time domain transformation of discrete data.

   Time domain filtering of data resulting from the immediately above.

   Time to frequency domain transformation of data resulting from the immediately above.

   Including windowing between domains.

   Used in all free-field (i.e. antenna-based) and increasingly more guided-wave (e.g. Coax & Waveguide) microwave characterisation/measurement of materials.

2. How do you/your organisation/research team define digital signal processing?

   Discretised analogue or discrete data processed to identify and modify and/or extract information.

3. What hardware do you use for signal acquisition? Is it purpose-built or do you just combine existing proprietary equipment to do what you want? Who typically designs and implements DSP systems in our organisation?

   Custom-configured commercially available Vector Network Analysers (VNA's). Within my own group IT, software & interface specialists design & implement non-proprietary DSP.

4. Do you design your own transforms or DSP algorithms? Do you use modelling software and if so, which? Do you test algorithms by modelling? Do you test transform algorithm/code by repeated inversion to look for errors?

   Activity to date within my own group has been restricted to off-line (ie PC-based) replication of VNA in-built transforms & DSP algorithms using MATLAB, the purpose being to move from black box to analysable. Algorithms & transforms are tested numerically and compared with output from validated algorithms/transforms/codes.

5. Do you do your signal processing in real time or on stored data after the acquisition has been completed? Do you process mainly in the time domain or the frequency domain?

   I perform preliminary signal processing in real time and final signal processing on post-acquisition stored data, in time domain only.

6. Do you have staff who have specific skills/training in DSP? Do you think there is any benefit in having specifically trained staff or do you expect that DSP should be part of the range of skills needed by all physicists and engineers?

   I have staff with specific skills/training in DSP. A combination of specialists in the field and a trained user community I believe to be most helpful to my business.

7. What do you think are the main pitfalls facing inexperienced DSP users? Can you recount any specific problems/mistakes etc. from your own/your organisation's/team's experience?

   If I were to single out one pitfall above all others it would be the somewhat subjective nature of application within the user community. The biggest DSP-related mistake my group made until a few years ago was retaining only the processed data. Had we retained the original data re-processing would have been possible.

8. Can you identify any specific training needs for yourself or others within your team, which you think would be of benefit to you and your colleagues?

   Awareness of DSP theory & technologies, best/good practice in respect of development, testing & application and advice wrt implications for metrology.

9. Do you consider the effect that the measurement system has on your measurements, e.g. do you deconvolve instrumentation/transducer responses and so on? If so, how?

   Non-DUT (Device-Under-Test) contributions to measurements are quantified where possible &/or eliminated/suppressed where possible otherwise included in the uncertainties. One example is the application of filtering in the time domain to remove long-path reflections not attributable to a DUT measured in free-field.

10. Do you make relative measurements or absolute measurements? Are traceability issues important to you?

    I make both relative and absolute measurements, though in the interests of cost-effectiveness the latter is kept to a minimum. Traceability issues are rapidly growing in importance to me.

11. Does your organisation have a quality system that governs the development of DSP systems (both hardware and software)?

    Yes.

12. Are you willing for your organisation to be named in the list of contributing organisations?

    Yes.

Name: Thomas Sors
Organisation: Industrial Research Limited, (NZ)

1. What applications do you have/expect to have in the future for digital signal processing in your current business/project/research?

   Implementation of digital control systems

   Analysis of measurements

2. How do you/your organisation/research team define digital signal processing?

   Processing of sampled data.

3. What hardware do you use for signal acquisition? Is it purpose-built or do you just combine existing proprietary equipment to do what you want? Who typically designs and implements DSP systems in our organisation?

   Use a variety or proprietary equipment, from benchtop measurements systems to PC based systems. These systems are chosen and implemented by project leaders.

4. Do you design your own transforms or DSP algorithms? Do you use modelling software and if so, which? Do you test algorithms by modelling? Do you test transform algorithm/code by repeated inversion to look for errors?

   Don't design own transforms but may design algorithms. Use MATLAB for modelling (FE/SEA software also used within team). Algorithms may be tested with modelling.

5. Do you do your signal processing in real time or on stored data after the acquisition has been completed? Do you process mainly in the time domain or the frequency domain?

   Real-time and post-acquisition processing depending on application. Both time domain and frequency domain processing also used depending on application.

6. Do you have staff who have specific skills/training in DSP? Do you think there is any benefit in having specifically trained staff or do you expect that DSP should be part of the range of skills needed by all physicists and engineers?

   Some staff have high level of training in DSP. I think DSP is a specialised subject and not necessarily part of a general set of skills required by physicists and engineers. In light of this, and the current abundance of DSP, specifically trained staff would be a requirement in most science and technology groups.

7. What do you think are the main pitfalls facing inexperienced DSP users? Can you recount any specific problems/mistakes etc. from your own/your organisation's/team's experience?

   Pitfalls include conversion between analogue and digital domains, appropriate signal-conditioning before sampling of signals, and conveying analysis results to those without DSP training.

8. Can you identify any specific training needs for yourself or others within your team, which you think would be of benefit to you and your colleagues?

   Not really. On the other hand, I think it is important when non-trained members of a team have DSP requirements, they communicate with team members who have training in DSP, rather than attempting to use DSP techniques themselves.

9. Do you consider the effect that the measurement system has on your measurements, e.g. do you deconvolve instrumentation/transducer responses and so on? If so, how?

   I am aware of the effect of the measurement system on measurements and may comment on this during reporting when I think it is appropriate.

10. Do you make relative measurements or absolute measurements? Are traceability issues important to you?

    Both relative and absolute measurements, depending on application. Traceability can be important.

11. Does your organisation have a quality system that governs the development of DSP systems (both hardware and software)?

    Not specifically for DSP systems although development may fall under other, more general, quality systems.

12. Are you willing for your organisation to be named in the list of contributing organisations?

    These comments are my own and not necessarily representative of my firm. Otherwise, yes.

Name: Dr. Victor Humphrey
Organisation: University of Bath

1. What applications do you have/expect to have in the future for digital signal processing in your current business/project/research?

   Mainly processing of signals obtained from sonar/ultrasonic imaging systems.

2. How do you/your organisation/research team define digital signal processing?

   I would define it as any signal processing performed on a digital system.

3. What hardware do you use for signal acquisition? Is it purpose-built or do you just combine existing proprietary equipment to do what you want? Who typically designs and implements DSP systems in our organisation?

   Mainly use digital oscilloscopes (LeCroy, Tektronix) as front end (avoid purpose-built wherever possible). Usually will be implemented by Postdoc/Postgraduate responsible for individual project.

4. Do you design your own transforms or DSP algorithms? Do you use modelling software and if so, which? Do you test algorithms by modelling? Do you test transform algorithm/code by repeated inversion to look for errors?

   Mostly, we use transforms and algorithms in Matlab, Labview etc. Modelling software includes PAFEC, ANSYS etc. In general do not specifically test transform algorithm, but do extensive tests of code containing algorithms. Will compare results with results obtainable from analytical solutions whenever possible.

5. Do you do your signal processing in real time or on stored data after the acquisition has been completed? Do you process mainly in the time domain or the frequency domain?

   Usually after acquisition is completed, but this is now after a relatively short time (quasi-real time). Mainly in the frequency domain.

6. Do you have staff who have specific skills/training in DSP? Do you think there is any benefit in having specifically trained staff or do you expect that DSP should be part of the range of skills needed by all physicists and engineers?

   No. I would consider this to be a useful skill for most physicists/engineers.

7. What do you think are the main pitfalls facing inexperienced DSP users? Can you recount any specific problems/mistakes etc. from your own/your organisation's/team's experience?

   The main problems I encounter relate to the inappropriate used of Fourier processing for signal analysis. These especially relate to the use of windows on pulsed waveforms.

8. Can you identify any specific training needs for yourself or others within your team, which you think would be of benefit to you and your colleagues?

   No.

9. Do you consider the effect that the measurement system has on your measurements, e.g. do you deconvolve instrumentation/transducer responses and so on? If so, how?

   Depends on application. Where it is appropriate, we do deconvolve out the transducer response by use of calibration data for the device. Try to avoid whenever possible. Usually using complex response in frequency domain.

10. Do you make relative measurements or absolute measurements? Are traceability issues important to you?

    Make measurements in a relative way whenever possible. In some cases, absolute measurements have to be made. Traceability is not important, although accuracy is.

11. Does your organisation have a quality system that governs the development of DSP systems (both hardware and software)?

    No, although promote good practice and encourage sensible writing of code with adequate testing.

12. Are you willing for your organisation to be named in the list of contributing organisations?

    Yes.