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A FLEXIBLE SIGNAL PROCESSING SYSTEM WITH APPLICATIONS

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INTRODUCTION

Flexibility in signal procesing is achieved by use of a large range of interrelated and compatible instruments. The use of separate physical instruments for each analysis or class of analysis is clearly not a cost effective proposition. The modern method is of course to use a microprocessor based system, in fact it is almost impossible to obtain any reputable instrument which does not utilise a microprocessor. The degree of flexibility depends on the way in which the microprocesor can conveniently be programmed.

In a dedicated instrument the programming consists largely of setting individual values to parameters, normally by means of knobs or switches or even by keyboard entry of alpha-numeric control sequences. For example, a modern transient recorder offers a considerable degree of programmability over trigger levels, delay periods and the like. It remains a transient recorder however.

Flexibility is not required merely for analysis. The method of data capture must also provide flexibility, as must the method of presentation of the results of any analysis. Additionally there is often a requirement for comparisons between the results of signal processing activities and other experiments, or theoretical calculations. It is only possible to achieve all these objectives using a computer based system. The hardware aspects of the computer system are reasonably self-evident, although cost effective means of achieving the required system are not necessarily straightforward. A key factor in dictation of the flexibility and efectiveness of the system is the control software. At one extreme a single pushbutton could start an entire sequence of data capture, analysis and result presentation. At the other extreme it may be necessary, even using subroutine libraries, to become involved in extensive programming. Whilst the latter activity may be intellectually stimulating it is essential to distinguish between progress and activity.

In a short paper of this nature it is not possible to treat all the aspects involved. A short presentation of a flexible approach to a signal processing system is made following a brief discussion of some aspects of selection of analogue to digital converter subsystems. Finally an example of use of such a system in the design and development of a high quality loudspeaker is presented. Numerous examples involving mechanical as well as acoustic vibration are available, as well as the more obvious examples of speech processing and filtering.

ANALOGUE TO DIGITAL CONVERSION

As noted earlier the type of data to be acquired is a major factor in selection and configuration of a system. Systems can be configured to acquire data at extremely high rates, for short (snapshot) acquisitions, particularly if data continuity is not critical.

Alternatively, for continuous data acquisitions, particularly on a number of parallel channels, constraints are imposed by the type of storage medium to be used. Further constraints are imposed by the different methods used to transfer data from the ADCs to the storage location.

A variety of methods are employed for continuous acquisition systems. Generally they are based on combinations of the following:

(a) Direct Memory Access (DMA) auto switching using a dual buffer system.

(b) FIFO buffers.

(c) Dual Port Memory.

A typical hardware configuration for an acquisition system consists of:

(a) ADC and Sample and Hold amplifier, to provide parallel acquisition on a number of channels.

(b) Communication link with computer by means of a dual buffer system using direct memory access.

(c) Real time clock with uniform time increment to provide clock pulse to initiate a data acquisition.

(d) Disc unit for data storage.

(e) Interrupt Driver to initiate role swap for the two buffers.

Operation of this system is straightforward. Data from the ADCis written into one of the buffers while the content of the alternative buffer is written out to disc. When the data reception buffer is full the interrupt driver initiates a role exchange.

The buffer that was being written out to disc becomes the input buffer and the content of the full buffer is written out to disc. It is important to note that the data, in the form that it is initially written to disc, cannot be used for analysis as all the channel information is in multiplexed (interleaved) form.

Obviously there is a range of data capture requirements for a well balanced system. External control of the starting point through a reliable Schmitt trigger circuit, pre-start sampling, multiple chanel acquisition facilities, and external or internal clock control are all necessary features.

ELEMENTS OF A SIGNAL PROCESSING SYSTEM

A signal processing system usually consists of:

(a) Analogue to Digital Conversion (ADC) Subsystem.

(b) Digital Computer with Memory.

(c) Disc Unit and/or other Storage Medium.

(d) Some Graphics Units for result presentation.

The hardware generally dictates the performance limits, whilst flexibility of use and the operating environment are dependent on the control software. For example almost all investigations into acoustic phenomena will involve a frequency analysis of some form. A popular basis for comparison of similar systems is the speed of execution of the Fast Fourier Transform (FFT). Typical execution of a pure software algorithm for a thousand point transform requires 500 to 1500 milliseconds. Enhanced performance in the 1 to 50 millisecond range is readily attainable by addition of appropriate array processor hardware. The cost penalty for such enhancement ranges between £2000 and £100000. Alternatively, if the requirement is only for third octave data, use of one of the many third octave analysers which are available with the capability for computer setup and readout will be very effective.

Achievement of a high level of functionality and flexibility requires that the software uses a modular approach. Consider a simple spectral density analysis to illustrate this point. The stages here are:

(a) Data Acquisition.

(b) Spectral Analysis.

(c) Graphic Display.

The classic software approach to this is to use of a software library, perhaps even with a special language. Typically this approach requires that the analyst become very familiar with the operational details of computing and linking programs, even requiring use of memory overlays, handling of disc filing, detailing graphic output, interactive processing, window clipping, hidden lines, device drivers and so on. Many of these detail aspects can be obviated by use of software modules providing greater functionality than is normal with subroutines, but retaining the flexibility of the subroutine approach. For example in DATS the simple spectral analysis job would be constructed as the sequence overleaf:

```
/ACQUIR    ! Acquire data
/DEMUX     ! Apply calibration factors
/DISPLY    ! Look at time history
/ASD       ! Auto spectral density
/DISPLY    ! Look at frequency function
```

This whole sequence can be combined as one single job and named MYASD for example. A database structure is implicit in this system in order that data sets may be passed from one module to another.

As implemented these data sets are normal computer system memory or disc files incorporating details of the structure of the information, such as the sampling rate, number of values, real or complex form, etc. Thus the system consists of input data streams which are processed by each module to produce new output data streams. Processed data streams also remain available, in their original form unless specifically deleted, to enhance flexibility.

In the above example each module asks the operator for details specific to its requirements, thus the ASD module asks for the frequency resolution. Whilst this interactive procedure provides considerable flexibility it becomes tedious, and thus subject to operator error, when the requirement is for analysis of a number of data sets. In consequence each module incorporates facilities enabling user specification of the degree of interaction. Thus different levels of interaction with the auto spectral density module could be specified:

```
/ASD("IN","OUT",0.5)     ! Asks no questions
/ASD                     ! Requests operator entry of all the details
/ASD("IN","OUT")         ! Requests operator entry of the resolution
/ASD(*,"OUT",7.2)        ! Requests operator entry of the data source
```

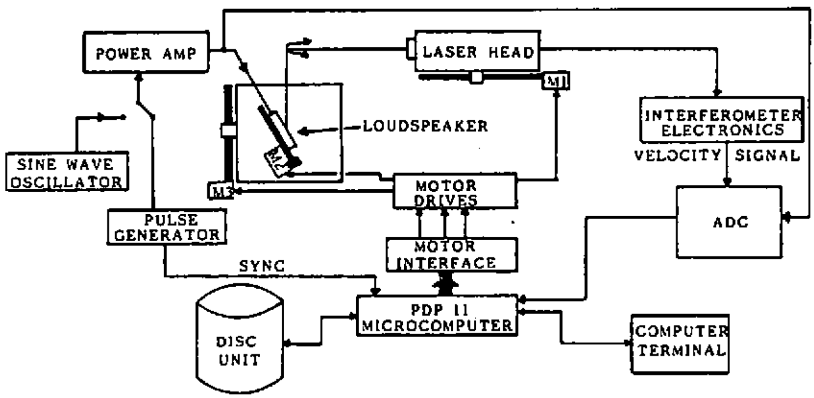
Additionally each module can be embedded in a FORTRAN program, although each module is itself a program. Thus, suppose the acquisition was over 16 data channels and they had been demultiplexedinto data files 1 to 16. In this case the example given below would individually analyse each channel and form a three-dimensional display of the entire set.

```
DO 36 K = 1 , 16
KJ = K + 210
/DISPLY(1,K,0)              ! Look at time history
/ASD(K,KJ,0.5)              ! ASD at 0.5 Hz
/DISPLY(1,KJ,0)             ! Look at frequency data
36    CONTINUE
/CONCAT("TOT", -210, 16)    ! Concatenate set
/ISM("TOT")                 ! 3D display
```

A full signal processing system requires a comprehensive range of signal processing, data acquisition, graphic and data manipulation facilities. Finally one further requirement is the obvious need for a system that enables users to develop their own modules for highly specialised applications.

A typical application of this type of package is used by B & W Loudspeakers in Worthing. They have installed a test rig configured as illustrated in Fig. 1.

Fig. 1 System Configuration



Loudspeaker surface vibration is monitored by use of a laser interferometer system. A polarised light beam from a low power helium-neon laser is passed through an electro-optic frequency shifting device, and a lens, onto the point of interest on the surface of the loudspeaker cone. Scattered light from the e surface is collected by the same lens and is mixed with a reference light derived from the main light beam. The frequency shift provides an accurate indication of the velocity of the reflective surface and the constant shift in the reference beam enables the interferometer to detect the direction of travel. The ability to sense the direction of surface movement also enables extrapolation of surface displacement. This is given by the difference between the number of beat frequency and shift frequency cycles.

The limit of resolution using this type of system is appproximately 0.1 um. The loudspeaker driver and the laser head are mounted on a common base to optimise resolution. Precision lead screw slide assemblies are used to mount both the speaker driver and the laser. These slides are driven by stepper motors to allow adjustment of the laser head to cone surface distance and the radial position of the point of illumination on the cone surface. The loudspeaker driver is mounted on a rotary positioning table which in turn is mounted on the slide to enable adjustment of the circumferential position of the light spot. This rotary positioning table is also driven by a stepper motor.

A thorough investigation of speaker cone behaviour necessitates measuring and recording of the motion at a number of discrete frequencies for each of a number of positions. For example a 16 cm diameter cone would require aproximately 300 measurement positions for frequencies up to 3 kHz. The need to repeat this number of readings for each discrete frequency under investigation means that the amount of data to be recorded soon becomes quite substantial.

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This is where another advantage of the computer becomes apparent. Connecting the stepper motors to the computer allows automatic software control of the orientation of the speaker cone relative to the laser head. The software to perform this type of task can be written easily using a FORTRAN callable subroutine library such as ADIOM2. Although the speaker voltage is not currently controlled by the system, software to implement this function could also be generated.

These control outputs are software generated and output through Digital to Analogue Converters (DACs), facilitating a wide variety of control functions which can be performed in parallel with data acquisition. In this manner a computer can be used to control a complete test rig, whilst simultaneously monitoring test effects. B & W have limited the amount of data that they acquire by adopting an impulse response analysis system, which, when converted to the frequency domain, provides information on all frequencies.

The voltage from the interferometer is applied to an ADC which feeds a dual buffer system. Data is written into one buffer whilst the other is being written out to disc, as detailed earlier and illustrated in Fig. 1. The size of these buffers is tailored to divide equally into the disc sector size, enhancing the speed at which data can be transferred to disc.

The rate at which data can be acquired is hardware dependent, with the maximum rates currently attainable, to disc, falling in the region of 100 000 samples per second. This particular system acquires data at speeds of up to 50 000 samples per second.

Having successfully translated the interferometer signal into data the use of a computer maximises the advantages to be gained from the information.

The ultimate benefits available through use of a computer in this manner are exemplified by the use B & W make of their system. The speaker is excited with a 10 to 25 us rectangular input voltage pulse and both the input pulse and the velocity output pulse of the measuring system are sampled and stored by the computer. The transfer function between the input voltage to the loudspeaker and the cone surface velocity is then found by transforming the pulses to the frequency domain using Fast Fourier Transform (FFT), a DATS module. This performs a division of the output pulse spectrum by the input spectrum.

The signal to noise ratio of this measurement technique can be improved by repeating the pulse at about 1 Hz storing the resultas in the computer and then averaging the results obtained. If n pulses are averaged in this mannner then the noise present in the pulse signal is reduced by a factor of n relative to the main signal content.

FFT produces frequency response data at a number of discrete frequencies determined by the number of samples of the input and output pulses which are stored and then transformed to the frequency domain. The amount of time saved by utilisation of the computer in this manner is exemplified by Figs. 2 and 3. Fig. 2 shows the averaged input voltage and output voltage pulses of the measurement system taken for one position on a loudspeaker cone.

The pulses were sampled at a rate of 50 000 per second, for 1024 samples, and were repeated at a rate of approximately 1 per second for just over 30 seconds. During the following minute the transforms were performed, the division carried out to determine the transfer function and the amplitude and phase of the cone surface velocity calculated and displayed for over 500 discrete frequencies. Fig. 3 shows the calculated amplitude and phase responses at different frequencies that are obtained from the pulses shown in Fig. 2. The values of the amplitude and phase are stored on disc, facilitating easy recall for further analyses.

The value of the DATS package now becomes apparent. Analysis tasks can be implemented rapidly, using the analysis modules. On completion of any analysis task the results can be displayed using a variety of display functions incorporated in the system. These display facilities include three dimensional graphic capabilities as demonstrated in Fig. 4. It is this flexibility of result presentation which provides true benefits by cutting out a lot of the tedious legwork required previously. One person can accomplish more on his own with this type of system than could previously be coped with by a whole team of people.

The applications to which this type of flexible signal processing system can be applied are myriad, with the flexibility of the modular approach enabling rapid generation of analysis jobs for specific applications. Models of modification effects can be generated by modification of the data files.

This field is becoming more important with applications of acoustic monitoring systems proliferating.

It is important to note that, although there is no requirement for a knowledge of computers to use this type of system, a proper knowledge and understanding of the research field is vital to validate conclusions drawn from any analysis. Packages such as DATS provide a powerful flexible analysis tool allowing rapid processing of data, but it should always be remembered that they are tools and as such the skill of the user, in his particular application field, dictates the true value to be obtained from them.

Fig. 2.

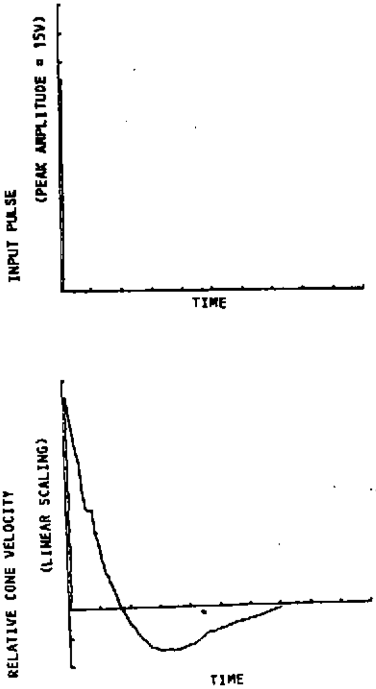

Fig. 3.

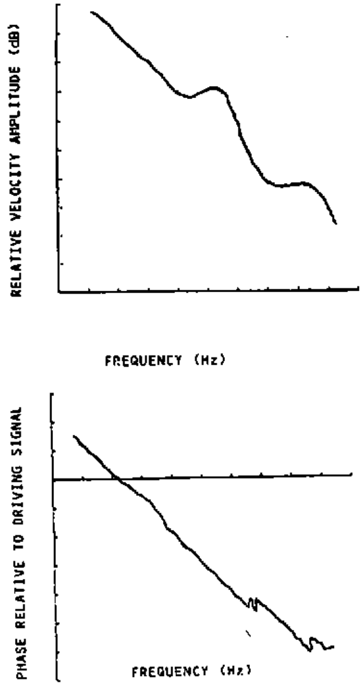

Fig. 4. TYPICAL LASER VIBRATION INTERFEROMETER SCAN