Institute of Acoustics

A MULTIPLE INPUT MULTIPLE OUTPUT SIGNAL ANALYZING SYSTEM

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ABSTRACT

To observe the interaction between a sound reproducing system and its environment, a number of reference signals and their responses have to be generated and measured simultaneously. A flexible multiple input multiple output signal analyzer combines much of the essential laboratory instrumentation.
This paper discusses the combination of a low cost input channel architecture, high speed multiplexing and analog to digital conversion, and multiple signal generation with high speed digital to analog converters to build a cost effective signal analyzing system with real time processing of the data.

INTRODUCTION

To measure the near field radiation pattern of for instance a loudspeaker box, the tool of acoustic intensity can be used.
The acoustic intensity (I) of a sound wave is defined as the average power per unit area in the direction (r) of the wave propagation (u) and is equal to the average of the product of pressure (p) and the air particle velocity (u) over a period T:

$$I = \frac{1}{T}\int_0^T p(t)\,u(t)\,dt \tag{1}$$

The acoustic intensity measurement is based on the estimation of the pressure gradient using two closely spaced microphones. Applying Newtons second law to a small volume of air yields an equation for the air particle velocity as a function of the pressure gradient:

$$-\operatorname{grad} p = \rho \frac{du}{dt} \tag{2}$$

with only one direction, the function is reduced to:

$$-\frac{dp}{dr} = \rho \frac{du_r}{dt} \tag{3}$$

leading to:

$$u_r = -\frac{1}{\rho}\int \frac{dp}{dr}\,dt \tag{4}$$

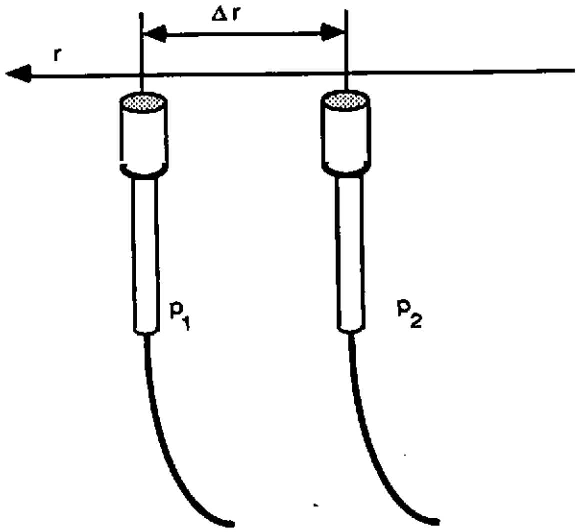

Fig. 1 Measuring probe with two closely spaced microphones.

Using a measurement probe with two microphones (fig. 1) the pressure gradient can be approximated by:

$$\frac{dp}{dr} = \frac{p_1 - p_2}{\Delta r} \tag{5}$$

The pressure at the point midway between the microphones can be approximated by:

$$p = \frac{p_1 + p_2}{2} \tag{6}$$

Substituting (6), (5), and (4) in (1) yields the following equation for the acoustic intensity in the direction r:

$$I_r = \frac{1}{T}\int_0^T \left\{ \frac{p_1 + p_2}{2} \left(\frac{1}{\rho \Delta r}\right) \int (p_2 - p_1)\, dt \right\} dt \tag{7}$$

Transforming to the frequency domain, this leads to the real part of the acoustic intensity as follows:

$$I_{r\,(real)} = \frac{1}{4\pi\rho\Delta r}\int_{-\infty}^{\infty} \frac{Im(P_1 P_2^*)}{f}\, df, \text{ where}$$

$Im(P_1 P_2^*)$ is the imaginary part of the cross spectrum between the two pressure signals.

The acoustic intensity is only measured in one direction. By increasing the number of microphones to four, a measurement in two directions can be performed.

ACQUISITION

A multi channel acquisition system can be built up in two ways:
1 an analog to digital converter per channel, followed by a digital multiplexer (fig. 2)
2 a sample and hold amplifier per channel, followed by an analog multiplexer and one analog to digital converter (fig. 3)

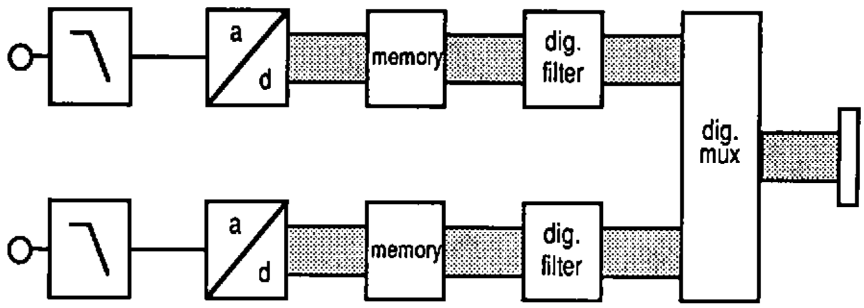

Fig. 2 Data acquisition system with ADC per channel

In the first data acquisition system each measuring channel has its own analog to digital converter, followed by a double buffer memory for intermediate storage. In this configuration a digital filter can be used for low pass and band translated filtering. However in any case an analog filter is required to prevent the aliasing effects. After the conversion and filtering, the data is digitally multiplexed before transfer to the computer.
In the data acquisition system with an analog multiplexer, only one analog to digital converter is used. Here a sample and hold amplifier in each channel ensures that the data for each individual channel is acquired at the same time to prevent time skew between the different channels due to the scanning process.

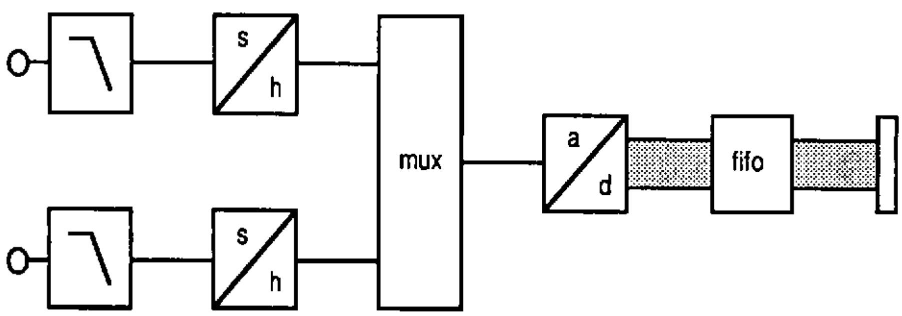

Fig. 3 Data acquisition system with multiplexer.

After the ADC a FIFO buffer is used to allow easy synchronisation between the data acquisition system and the computer. Band translated filtering for zoomed FFT is not possible in this system; in practice this is not a severe disadvantage as this can be easily accomplished in the computer.
In multi channel systems, the price per channel is a very important factor. In both systems an analog filter is required. The additional costs to make the cut off frequency of the filter programmable is only about 25 %. Modern opamps, thin film resistor networks and artificially aged capacitors ensure excellent

long term stability. A well designed analog filter can have slopes up to 150 dB/oct, eliminating the need for a digital filter. Having only one ADC in the system will reduce the price considerably. Modern ADC's are capable to acquire data with a sampling speed over 800 kHz. This means that in a 16 channel system each channel has a measuring bandwidth of about 20 kHz, which is sufficient for the majority of acoustic applications. In addition, increasing the number of acquisition channels is a relatively low cost investment. Additional flexibility is added as the speed per channel can be increased in case only a few channels are required.
For the reasons indicated above, the multiplexed data acquisition system will be the best choice for a multi channel acquisition system.

Scadas

Data acquisition
The Scadassystem is built up around a high speed analog to digital converter and multiplexer. The system can have up to 64 input channels; synchronisation between different mainframes combined with parallel interface chaining allows expansion up to 256 channels.

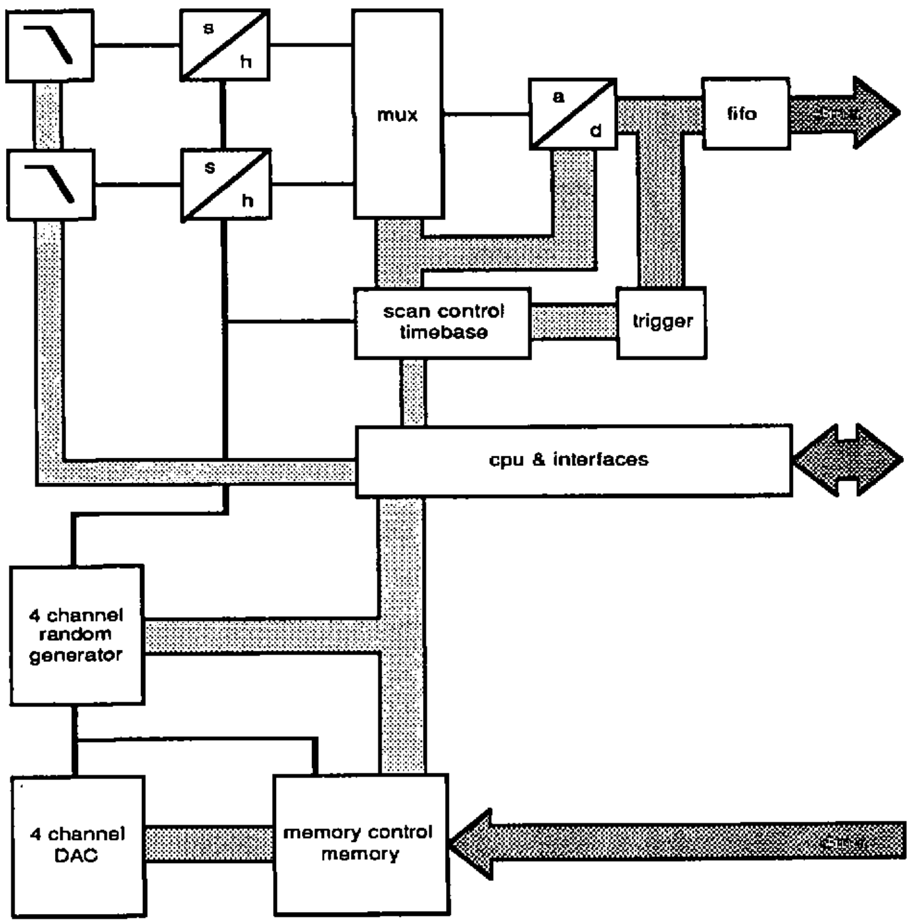

Fig. 4 Block diagram of *Scadas*

Each input is equipped with a dual analog filter module; a wide selection can be made between different filter characteristics to select the best filter for the application. Four filter types are available: a Butterworth filter with a slope of 48 dB/oct., an Equal Time Delay filter with 60 dB/oct. without overshoot, an Inverted Chebyshev filter of 80 dB/oct and an Elliptic filter of 150 dB/oct with 0.3 dB pass band ripple. The last filter allows the sample rate to be close to the theoretical limit. The filter modules incorporate a range amplifier and overload detector. Phase difference between the different channels is below 0.2 degrees.
In front of these filters, signal conditioning modules as differential bridge amplifiers or charge amplifiers can be used to condition any kind of vibration transducer.
The Scadassystem includes a programmable clock, scan control logic, trigger detector etc. to free the computer from overhead during the actual acquisition. The triggerdetector only allows data to be sent to the computer after a trigger condition has been met. A 32 K word FIFO buffer simplifies synchronisation between *Scadas* and the computer and enables a high throughput to disk in continuous acquisition. For demanding applications, where e.g. the use of more than 64 channels would be restricted by the DMA throughput of the computer, the FIFO buffer can be extended to more 256 K words.

noise generation

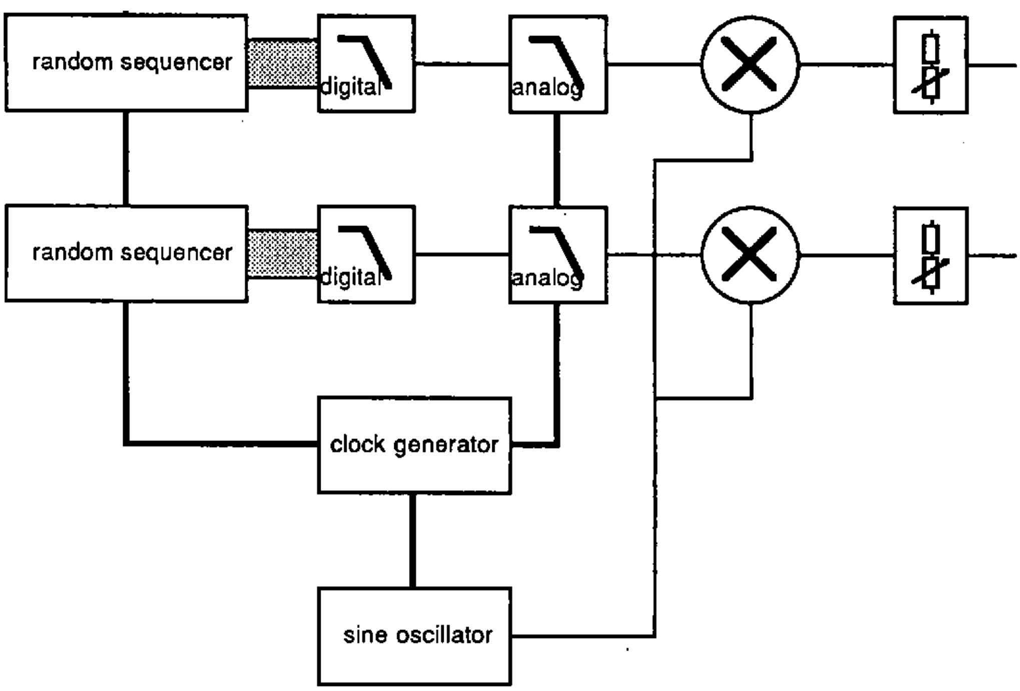

Fig. 5 Four channel noise generator

*Scadas*has a 4 channel random noise generator, which produces 4 uncorrelated random noise signals. The power bandwidth of the noise signals is automatically adjusted to the measuring bandwidth using analog and digital filters, so that almost no power will be generated outside the band of interest. It is well known that the accuracy of the measured spectra depends on how well one controls a digital signal processing error known as leakage. Leakage is caused by the violation of the fundamental assumption of the FFT that the measured spectra should be periodic in the time frame. This leakage can be minimized by using special excitation signals, such as burst random . The transient nature of burst excitation signals eliminates the pre- excitation and post-response errors. Since the excitation gets turned off in each time frame, there should be no response from the structure at the start of a new time frame. Additionally in the current time frame no force will be measured that would have responses in the next time frame. Hence no response of excitation in the previous time frame nor excitations in the next time frame are measured. This will make each time frame unique to its forces and responses. As the response will be zero at the beginning and the end of the time frame, the measured spectra will be periodic. Therefore no weighting functions or windows need to be employed to eliminate the leakage problem. The noise generator supports the burst random excitation method as well as others, such as periodic random or band translated noise. The output signals can be attenuated up to - 80 dB. All parameters for the random noise generator can be downloaded before the actual acquisition starts.

sine generation
When using stepped sine testing, the cross spectra data are acquired frequency by frequency; at each frequency point, a pure sinusoidal excitation is measured. By stepping the sine frequency through the band of interest cross spectra can be evaluated for a number of discrete frequency points.

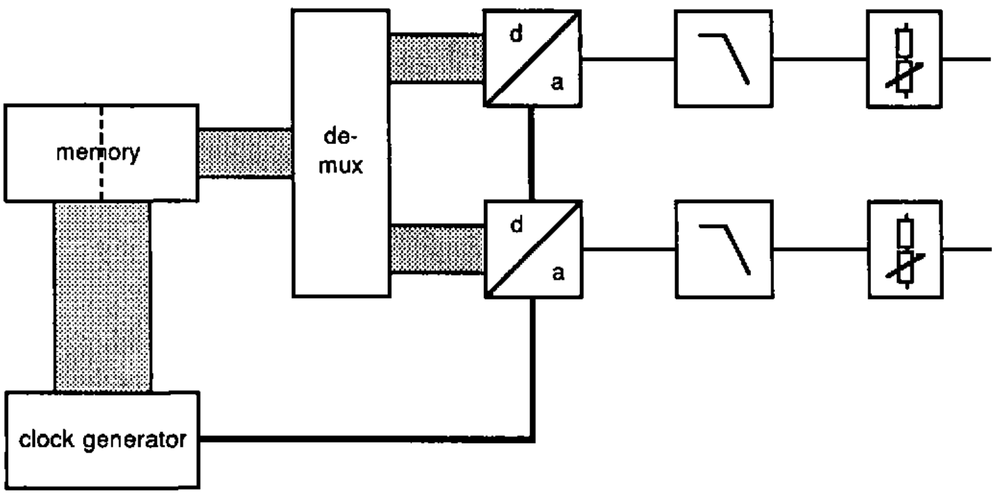

Fig. 6 Block diagram of DAC module

To support these requirements, the memory block of the DAC module consists of 4 dual buffers, which allows simultaneous access from the DMA interfaces and the DAC's. This means that new sine waves can be downloaded via the DMA interface in one part of the memory, while at the same time signals are generated from another part of the memory. In this way signal generation and processing can take place simultaneously. The clock generator handles the memory addressing scheme and provides an accurate crystal controlled clock as well.
The DAC module features 16 bit digital to analog convertors to provide the necessary dynamic range. The signals from the DAC's are smoothed by analog filters and can be attenuated up to - 80 dB. Of course applications for the DAC module are not limited to sine wave excitation. Any kind of signal, from random noise to burst chirps, can be generated in the computer and downloaded to the DAC module.

MULTIPLE INPUT MULTIPLE OUTPUT SPECTRUM ANALYZER

The only element to expand *Scadas* to a multiple input multiple output signal analyzer is the addition of a desk top computer or a PC, optionally completed with a hardware digital signal processing board to perform real time FFT processing.

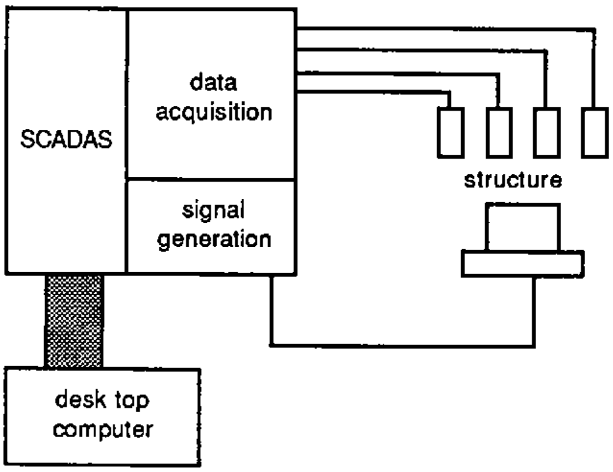

Fig.7 Multiple Input Multiple Output Signal Analyzer

The system is capable to acquire data at a throughput rate of up to 800 kHz and to calculate a complete FFT over 1024 points in less than 150 ms in software and 20 ms using a hardware FFT card. With this combination real time rates can be achieved up to more than 20 kHz for 1 channel outperforming dedicated FFT analyzers.