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CONSTRAINTS ON ACOUSTIC DETECTION IN VERY NOISY ENVIRONMENTS

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INTRODUCTION

The detection of low level signals in a noisy environment has been extensively studied as it has applications in many areas e.g. SONAR, acoustic fault detection etc. The signal to noise ratio is increased by making many simultaneous measurements of the acoustic field using an array of sensors. In the early days the array was simply steered to "look" in a particular direction by adding up the outputs of the sensors with appropriate delays so as to add the sound from that direction coherently. This technique of Linear Beamforming is relatively easy to implement and requires a limited amount of computer processing of the raw signals. The gain that is achieved in rejecting noise coming from another direction depends upon the ratio of the sensitivity in the two directions. Large rejections can generally only be achieved by using a very large number of sensors. More recently Adaptive Beamforming has been used to increase greatly the rejection of noise coming from directions away from the look direction.

The crux of this technique is to sum the sensor outputs in such a way as not only to add the sound from the look directions coherently but also to ensure that the coherent signal from a noise source is set to zero. By effectively steering a "null" of the array in the noise source direction the rejection of this noise can be substantially increased. The cost of the increased performance for the same number of sensors is a substantial processing requirement. In the past this has been prohibitive in all but the most essential applications. The continual fall in the cost of high speed computation is now changing the balance so that the cost of sensors is now becoming dominant. Therefore the use of Adaptive Beamforming is likely to increase rapidly in the future.

This paper will outline the factors which need to be considered in applying Adaptive Beamforming. In particular the paper will discuss the way in which the optimum performance is limited by the properties of the signal and noise environment and what practical constraints are imposed by a realistic microprocessor implementation.

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OPTIMUM ARRAY PERFORMANCE

The ability of an array to increase the signal to noise ratio is measured by the array gain. This is the ratio of the noise output of one sensor to the noise output of the array for the same sensitivity to a signal in the look direction.

The noise field incident on the array will be composed of a part which is coherent across the array and is due to propagating waves from distinct sources of sound. The rest of the noise field will be incoherent from sensor to sensor and will be due to a combination of isotropic incident sound, wind generated noise and electronic and sensor self-noise on each channel. Perfect Adaptive Beamforming will eliminate the coherent component of the noise leaving only the incoherent part. The incoherent noise power will be rejected, in the same manner as for Linear Beamforming, by a factor of the square root of the number of sensors. Thus the total array gain, G, in decibels is

$$G = G_a + 10 \log K$$

where G_a is the gain due to adaptive processing and is related to the ratio of coherent to incoherent noise received at the array and K is the number of sensors.

The extent to which the noise field incident upon the array is coherent can not be predicted except in model environments. To predict the performance of an adaptive array, there is no substitute for measurement of the coherence properties of the physical environment. However, it is possible to make some general comments.

DISCRIMINATION OF SIGNAL AND NOISE

It is most important to ensure that the design of the array geometry is chosen so as to maximise the difference in the character of the cross-correlations of the transducer voltages caused by the signal and by the noise. This will ensure that the signal can be best distinguished from the noise and will mean that there are no array nulls "close" to the look direction. The sensitivity of the array performance to errors in estimation of the noise environment will then be substantially reduced.

If the objective of the array is purely to alert the presence of a signal, and its direction is unimportant then it is also important to minimise the variation of the character of the cross-correlations of the transducer voltages over the range of all possible signal directions. This will minimise the number of orthogonal directions in which the array must look and thus reduce the adaptive processing requirement.

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PRACTICAL CONSTRAINTS IMPOSED BY THE MICROPROCESSOR ENVIRONMENT

The amount of processing needed for a particular detection system design is related in a variety of ways to the various operational requirements. If these relationships are clearly understood in the way that they interact to affect the amount of processing then designs may be chosen to meet the desired specification for minimum cost.

Figure 1 represents the interaction of the various factors which are described below.

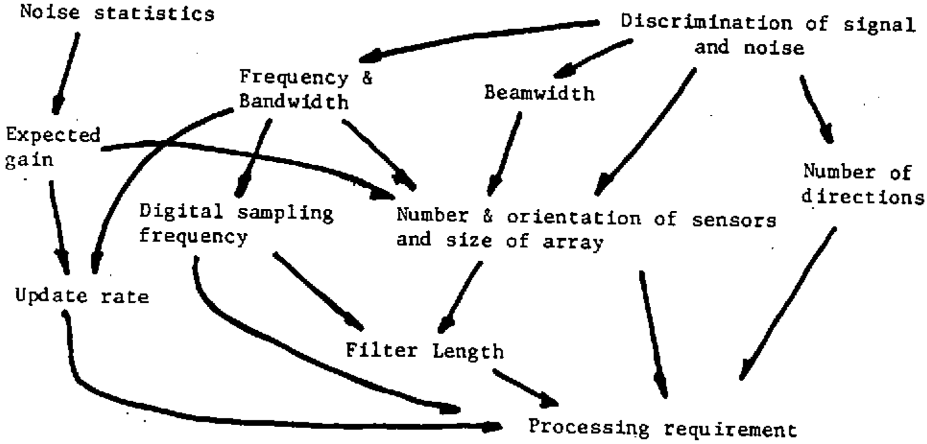
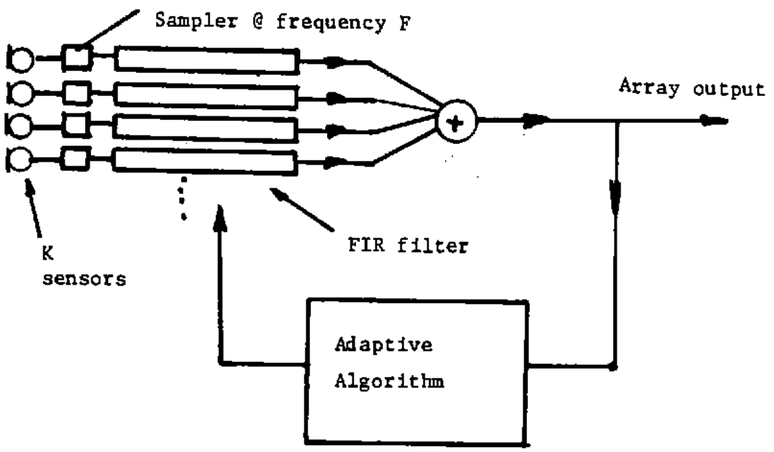


Figure 2 shows the Array processor



Digital Sampling Frequency
As the processing requirement is at least proportional to the sampling frequency, F, this should be kept to a minimum The best practical anti-aliasing filters allow F to be as little as three times the upper operational frequency.

Size of Array and Number of Elements
The linear dimension of the array, D, is primarily determined by the required width of the beam, θ, in the look direction. If the array is required to locate the direction of a signal source then the accuracy of angular location will fix the beamwidth. The size of the array is proportional to λ/θ, where λ is the wavelength of the lowest frequency that the array is to receive. The number of elements is controlled by the non-adaptive array gain as described before. Additionally there should be at least one adaptive element for each significant noise source at any frequency in the band so as to ensure that there are sufficient "nulls" to cancel them all.

Filter Length
The digital filters operating on the transducer outputs must have an impulse response long enough to match the maximum acoustic delay between transducers. The filter processing rate will be proportional to the product of the number of coefficients in the filter and the rate at which data is received., i.e. proportional to $\frac{DF^2}{C}$ where C is the speed of sound.

Adaption Rate
The rate of change of the noise statistics determines the maximum achievable adaptive gain and in order to attain this maximum the adaption rate (or equivalently the measurement time for each update) needs to be carefully matched to this rate of change. Too short a measurement time will produce an inaccurate estimate of the noise resulting in a reduction in gain. Too long a measurement time will mean that the updates from the adaptive processor are applied too late.

Adaptive Algorithm
Different algorithms make different demands on the processing power and the way the processing requirements relate to the remaining system variables. A full discussion of this is beyond the scope of this paper.

CONCLUSIONS

The operational requirements need to be traded-off against each other by appropriate selection of the system variables described above in order to obtain an optimum performance for minimum processing power. Another factor which needs to be included in this complicated equation is the balancing of the requirements for computational speed and memory.

The choice of the system variables can be optimised to allow a sophisticated acoustic detection system to be implemented by modern microprocessor technology at reasonable cost.

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DESKTOP COMPUTERS FOR ACOUSTIC TEST HOUSES, STUDIO TUNER SELECTION, AND MECHANICAL SERVICES ATTENUATOR SELECTION

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Both Sound Research Laboratories, who operate a general acoustic test house facility, and Sound Attenuators Ltd., who use their acoustic facility for product development, now employ a micro-computer system consisting of

NCR Decision Mate 16/8bt Desktop Computer with 256k of memory	£2500
A Printer, type 6411, of the Dot Matrix type from C'ITOH	£ 700
A Plotter from Mannesmann-Tally called Pixy 3 employing 3 pens	£ 700
General Radio Analyser	£1200

The operating system is MS-DOS and GW basic language. The communication to the other three items is through three output/input ports, but in particular the plotter is buffered by a 32k memory allowing the rest of the system to be set back into action whilst the inexpensive plotter slowly carries out its tasks.

It is worth mentioning that the Decision Mate is a good industrial computer, as since its introduction primarily to be dedicated to the laboratory facility, the use of the 4 packages

Word Star
Report Star
Data Star
Multiplan

have proved incredibly useful. In fact, Multiplan has proved so useful for table work that it has enabled the origination of alternative technical presentation techniques to be revolutionised, which will subsequently feed through into catalogues. The lecture itself will be profusely illustrated, bringing the somewhat dry commentary below to life.

The two main acoustic measurement applications are both based on reverberation chamber techniques (no anechoic facilities are operated) for the determination of sound power level and sound reduction index.

Sound Power Level

The multi-microphone technique is employed, the output from each microphone being separately processed allowing a standard deviation between the microphone locations to be originated. This is considered a most significant feature of the analysis process, as it is indicative of the test accuracy during each and every test and is considered superior to the ISO principle of accrediting the reverberation chamber as suitable for measurements to a given accuracy. Our own experience has shown that the self-same room behaves with various measured standard deviations, depending primarily on the size of the source, the presence of tonal characteristics (but not pure tones) and the directivity of the source. The rooms are calibrated for conversion from average sound pressure levels to absolute sound power levels by means of separate reverberation time measurements which are not as yet part of a micro-processor technology. On occasions a Bruel

& Kjaer spinning fan impeller calibrated noise source is used for convenience, but is never employed as an absolute calibration. Analysis always takes place in ⅓ octave band widths, but is often collapsed to whole octaves for presentation convenience. The output is usually presented as an alpha numerical printout on the printer of the ⅓ octave pressure and power readings, together with their corresponding standard deviations. The printer may be selected to print bar charts, or more usually line graphs in octaves or ⅓ octave format.

Sound Reduction Index

Generally, the same techniques are employed, except that multi-microphone techniques are employed in the two adjacent rooms separated by the panel under test, and in this case emphasis is on ⅓ octaves. Again, standard deviation between the separate microphone sound pressure levels is ascertained in each measuring room and the difference of these is processed in conjunction with the area of the test specimen, to yield sound reduction index. As an alternative, this can be normalised to half a second, or 10sq.m. of absorption in the receiving room. For guidance, the maximum average sound reduction index between the rooms is considered to be just in excess of 60dB, and generally most tests do not require correction for this direct or flanked transmission weakness. When necessary, it can be included in the calculation procedure. Again, an alpha numerical printout is produced of the sound reduction index. Included is the sound pressure level in the receiving room to ensure that it is 10dB above previously monitored background level. With the better panels, it is amazing how difficult this is to achieve at higher frequencies. Also printed out is the classic average sound reduction index, the American STC classification, and the more recent weighted sound insulation index, Rw. Attenuator insertion losses are established with much the same program, except that the empty duct reference level has to be retained between tests and assembly changes. For panel sound reduction index the source room average sound pressure level is separately obtained on each and every occasion in case the panel modifications change the room characteristics of the source room.

Comment

Whilst obviously the computerisation of the above techniques adds speed to the measurement even with 30 seconds integration time on each microphone, probably the most important contribution is the elimination of technician error from the recording of meter indications - such a profuse but tedious numerical collation. The adoption of the computer itself does not add basic accuracy to the acoustic techniques, which is a falsehood often associated with such prestigeous hard-copy outputs. Secondly, I have found that the comparison of results from multi-lined plotting has been a tremendous aid; all sets of results are conveniently stored on the floppy disk and it is a matter of half a minute to compare result A with result B-Z at the touch of a button, and this has been of most value in product development when many variables are being interplayed.

Obviously, the addition of colour - we use six different colours - enhances the ease of this comparison. As a warning, it's a good idea to arrange that each colour code can be variously solid line, dashed line, dotted line, etc., in the event of conventional two-tone copying.

DESKTOP COMPUTERS FOR ACOUSTIC TEST HOUSES, STUDIO TUNER SELECTION, AND MECHANICAL SERVICES ATTENUATOR SELECTION

Studio Tuner Selection

Over the years studio tuners possessing various absorption chracteristics have been developed and measured under reverberant room conditions to establish their absorption coefficient as a function of frequency. In our case, Sound Attenuators Ltd. have 18 studio tuner options. Whilst not suggesting that the science of predicting reverberation times is precise, it is usually useful to set out a studio design such that a required reverberation time will be achieved. The whole concept of the interchangeable studio tuner then, of course, allows the later adjustment of the studio's reality to give a satisfactory - if not quite the required - reverberation time. However, with 18 options to choose from, this process can be exceedingly tedious, even with the aid of a good calculator.

Manual

For this case we have employed the desktop computer as a powerful and rapid pictorial calculator. We have employed Eyring's formula for the calculation of reverberation time, whilst fully realising that the laboratory's values of absorption coefficient were obtained by the use of reverse Sabine formula. This could be an interesting acoustic discussion point. The input to the computer is fairly straightforward, requiring the room volume, the total surface area, the floor area (hard or carpeted) and the option to pre-select a mid-range reverberation time when the IBA curve will be employed, or the incorporation of a unique ⅓ octave reverberation time requirement. This is the line which appears on the screen as a plot of reverberation time against frequency. For reasons of commercial sponsorship, the absorption units are in surface area blocks of 600mm x 600mm. It is assumed then that the rest of the wall and ceiling surface area is concrete. General performance absorption coefficient such as curtains, wooden panels, etc. can be included either in the main computer bank (which includes the 18 absorbers mentioned above), or as a dedicated entry at the time of calculation. One then just by manual trial and error chooses quantity selections from the absorbent types and a calculated reverberation time appears superimposed on the selected curve. Manual trial and error continues until a satisfactory fit is compromised.

Automatic

However, whilst this has proved fairly useful, we have extended the desk computer to its extreme computational abilities and made this process of curve matching automatic. Again, the required reverberation time is entered and a guessed selection is made from a limited range of 8 types. The program asks how many of each type is required, gives you the option to vary each type or to keep it constant, requires that for variable elements you choose a lower and upper limit to the number of unitary absorbers, and it is required to state the number of absorbers to be changed in each step - usually 10 at a time. After a first calculation, it displays the first guess. It then requests a necessary error limit required for the final reverberation time with respect to the requested reverberation time, and this is stated as an overall deviation on the average and is chosen around 0.1 seconds. The computer then sets to work in an iterative manner, the program for which must not be under-estimated for its complexity or originality. In reality it seems to concentrate on the lower frequency approximation most strongly, although this is not a feature of the design.

DESKTOP COMPUTERS FOR ACOUSTIC TEST HOUSES, STUDIO TUNER SELECTION, AND MECHANICAL SERVICES ATTENUATOR SELECTION

Whilst the design goal and the latest calculations are displayed as bright yellow lines, the previous calculations are retained as a background of red lines. When a best fit has been found, the display clears down to just the yellow lines, which are then printable and plottable. This process can take several hours, indicating that the limited power of the desktop computer is being stretched.

Attenuator Selection

Lastly, an attenuation selection program used in the mechanical services industry will be demonstrated on a brief-case portable Epson computer, where the program is fused in to a chip. The early part of the program is simply a duct calculator computation, which certainly is no more accurate than the data incorporated, and it is only fair to point out that the broad brush approach to this basic sound power and natural attenuation losses is by no means to be considered particularly accurate.

Having calculated the required attenuation, this particular program then proceeds to select a commercial attenuator taking into account the air volume being handled and the allowed pressure loss for the particular application situation.