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DIGITAL ASSESSMENT OF LOUDSPEAKER PERFORMANCE

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Introduction

In the past few years there has been an increasing use of digital signal processing methods in acoustics (1,2,3). The area of application dealt with in this paper is the use of these digital techniques for the analysis of loudspeaker performance. The implementation of these techniques in a general-purpose computer means that not only can conventional measurements be made and displayed in many forms but also "non-standard" measurements, which may be difficult or impossible to perform on analogue equipment, can be made just as easily. Only one such application is considered here, and this is the use of homomorphic filtering (4) to determine the minimum and maximum phase components of the loudspeaker's amplitude response.

Experimental Techniques

An estimate of the frequency response (both magnitude and phase) of a loudspeaker system is made as the ratio of the discrete Fourier transform of its sampled output and input signals. The motivation behind this method is as follows:

It is well known that in the time domain, the output, $y(t)$, of a linear time-variant system to an input $x(t)$ is given by the convolution integral:

$$y(t) = \int_0^\infty h(\tau)\ x(t - \tau)d\tau \qquad (1)$$

where $h(\tau)$ is the impulse response of the system, i.e. the response of the system to an input of a unit impulse applied at a time τ before. By taking the Fourier transform of both sides of equation (1) we obtain:

$$Y(f) = H(f)\ X(f) \qquad (2)$$

where $X(f)$, $Y(f)$ are the Fourier transforms of $x(t)$, $y(t)$ respectively, and $H(f)$ is the frequency response, defined as the Fourier transform of the impulse response $h(t)$.

Now by dividing equation (2) by $X(f)$ we can obtain a formula for the frequency response as:

$$H(f) = Y(f)/X(f) \qquad (3)$$

If X(f), Y(f) are now the discrete Fourier transforms of x(t), y(t) respectively then we can obtain an estimate of H(f) given by equation (3) at each value of f for which $X(f) \neq 0$. If H(f) is evaluated from d.c. to WHz and x(t) and y(t) are measured for T seconds, there will be WT estimates of H(f) available at frequency intervals of 1/T.

The input signal x(t) was chosen to be a pseudorandom binary signal (5) (PRBS) which has three advantages in this type of application. First, the magnitude of its spectral components $|X(f)|$ are constant over the frequency range of interest and secondly, the transform of these signals and, in the absence of noise, the transforms of the corresponding system responses are deterministic. That is, statistical errors from this source are eliminated as are the need for large sample sizes and spectrum-smoothing techniques usually associated with the use of random signals. The third advantage is that these signals are easy to generate and can be made to be repeatable as opposed to the true randomness of genuine white noise.

The apparatus used consists of a Computer Associates Alpha minicomputer with 32k of store. It has a disc handler for two floppy discs, on which the programs are developed and held. Within the computer are the digital-to-analogue (D/A) and analogue-to-digital (A/D) converters together with sample/hold and multiplexing arrangements. The programs are written in Fortran except where high-speed processing is required, when assembler is used. The suite of programs is interactive, giving the user the ability to make various decisions, for example, to select the sampling frequency or to decide which displays to use. The requests for information appear on a Tektronix 613 VDU which also displays the various output graphs. User input is by means of a Keyboard and an optional X-Y plotter is available to produce a hard-copy output of any of the graphs.

The loudspeaker output is normally picked up by a ½" B. & K. microphone and amplifier and fed via an antialiasing filter to the A/D converter in the computer. Figure 1 shows the arrangement of the equipment. The PRBS is generated within the computer and sent via the D/A converter and amplifier to the speaker under test. The program samples the microphone output signal for one complete period of the PRBS, and many of these signals can be added together and averaged to improve the signal-to-noise ratio. The input signal is also sampled in a similar way and the ratio of the discrete Fourier transform of these two signals gives the complex frequency response according to equation (3).

Once the frequency response has been obtained in this way, its Fourier transform gives the impulse response of the system, from which room reflections and the time delay between the microphone and loudspeaker can be easily removed.

This impulse response is now available for any number of operations to be performed on it, both "standard" and "non-standard". For example, it can be transformed back into the frequency domain to give the familiar frequency and phase responses, but now without the time delay. An example of a "non-standard" technique, which can only be performed digitally, is that of homomorphic filtering, or cepstrum techniques, which will now be elaborated upon.

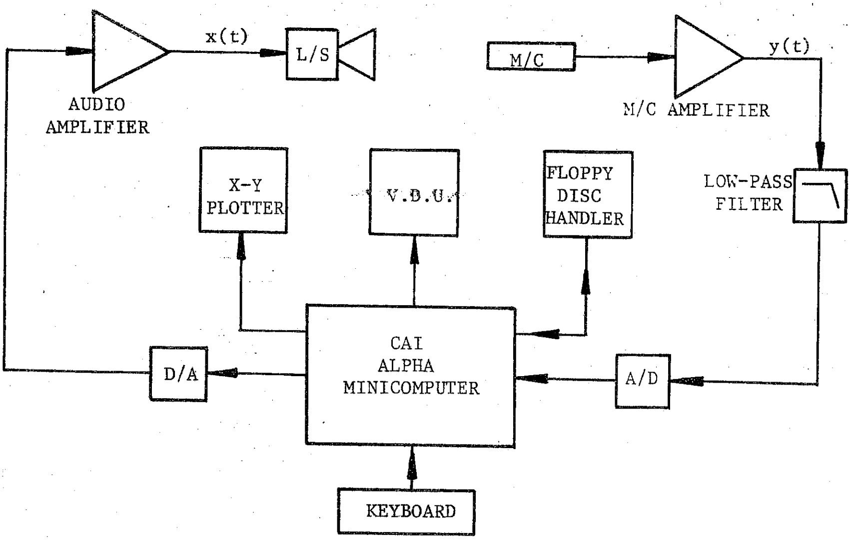

Figure 1 Computer-based test system

Lately there has been an upsurge of interest in minimum-phase systems for which there exists a unique relationship between the magnitude and phase responses, so that the measurement of either magnitude or phase is sufficient to determine the frequency response completely, and it is here that the complex cepstrum can be used to determine the minimum-phase portions of the loudspeaker's response. The complex cepstrum is the inverse Fourier transform of the complex logarithm of the Fourier transform of the time sequence and one of its properties is that the component for negative time corresponds to the non-minimum (maximum) phase part of the response, while the positive time component corresponds to the minimum phase portion of the response. Thus by obtaining the complex cepstrum and then transforming the negative and positive time portions individually, the frequency responses corresponding to the minimum and maximum phase components of the speaker can be found, and kept for further processing if necessary.

Results and Conclusions

The frequency responses obtained by digital means compare favourably with conventional analogue results and thus give confidence in the methods used. However, with the newer, purely digital techniques used, there are no equivalent analogue results for comparison and so greater care must be taken in the interpretation and use of these results.

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The digital techniques used provide an extremely flexible and useful tool in evaluating loudspeaker performance. Not only can conventional measurements be made but also the use of digital signal processing methods can provide new measurements which can give a new insight into loudspeaker performance, thereby leading to new understanding and improvement in loudspeaker design.

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