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THE IMPLEMENTATION OF ACTIVE NOISE CONTROL SYSTEMS USING DIGITAL SIGNAL PROCESSING TECHNIQUES

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INTRODUCTION

The development of active noise control (ANC) systems has tended to focus on improving the performance of practical systems by developing complex geometrical structures of multiple-detector and/or multiple-loudspeaker configurations. Only limited success with these schemes has been reported [1]. However, recent advances in semiconductor technology have produced extremely fast, dedicated digital signal processing devices capable of implementing, in real-time, controllers of significant complexity. The opportunity now exists to reduce the geometric complexity whilst increasing controller complexity, resulting in a number of significant advantages. This approach requires a proper analysis of ANC systems from a control engineering perspective, and a clear classification of the characteristics of the sources [1,2] leading to a suitable design methodology. This coupled with the capabilities provided by the current microprocessor technology provides the opportunity to overcome the problems associated with previous attempts and to develop practical ANC systems. This paper presents a methodology for the design of ANC systems with compact broadband sources, in a non dispersive (linear) propagation medium, with the motivation of generating a basis for implementing practical ANC systems using advanced digital systems technology.

From a practical point of view, the process of actively attenuating (or cancelling) an unwanted sound wave, through wave interference, consists of three main steps; detection, inversion, and superposition. In the process of detection the aim is to obtain a signal coherent with the unwanted noise. The device to be used as a detector is required to have such a response characteristic over the frequency range of interest to provide the necessary information about the unwanted noise. Microphones which have reasonably flat amplitude and linear phase characteristics, although relatively expensive, are commonly used as detectors. Here the detector is placed at a fixed distance relative to the source of noise (primary source). The acoustic properties of the path between the primary source and the detector as well as the response characteristics of the detector will affect the amplitude and phase of the noise being detected.

There are many sources of noise which vibrate continuously when in operation and the acoustic waves they produce are coherent with their vibration. In these circumstances vibration sensitive devices can be used as detectors. This method eliminates the effects of the acoustic properties of the transmission path, in the process of detection, on the unwanted noise wave and can give good results.

The most important part, and the main body of an ANC system, is the inversion of the signal. The device performing this task (referred to as the controller) should be capable of not only shifting the phase of each frequency component of the detected signal by 180 degrees but also adjusting the amplitude of each component. The controller is, therefore, required to have a continuous transfer function representing the desired amplitude and phase characteristics. Thus, when the wave of the artificially generated (secondary) source is superimposed on that of the primary source destructive interference results at a specified observation point.

After the detected signal has been processed by the controller, the output of the controller is used to drive the secondary source producing an acoustic wave (secondary wave) that interacts with the acoustic wave from the primary source (primary wave). Here a loudspeaker placed at a set distance from the primary source can be used as a secondary source. The result of superimposing the acoustic waves can be observed by using an observer microphone.

From the above discussion it follows that processes of detection and superposition are relatively straightforward whereas the main task in developing an ANC system is the design of a suitable controller. This requires first a consideration of the conditions of cancellation in terms of the amplitude and phase parameters of the primary and secondary waves upon which the design of ANC systems can be based. Such a basis has already been established [2] and is used in the development of the design method presented in this paper.

ACTIVE NOISE CONTROL STRUCTURE

The schematic diagram of the ANC system in the form of a feedforward control structure (FFCS) is shown in Fig. 1a. The (unwanted) primary wave is detected by the detector and transferred to the controller. After the detected signal has been adjusted in phase as well as amplitude by the controller it is emitted by the secondary source to be superimposed on the unwanted noise. The objective is to reduce the sound pressure level at an observer location (observation point). Fig. 1b shows the equivalent block diagram in the complex frequency, s, domain where $X_1(s)$, $X_2(s)$, $X_3(s)$, and $X_4(s)$ respectively represent the transfer functions of the paths r_1, r_2, r_3 and r_4. M(s), C(s), and L(s) respectively are the transfer functions of the detector, controller, and secondary source. The signals P(s) and S(s) represent the primary and secondary waves, respectively, before propagation whereas $P_o(s)$ and $S_o(s)$ respectively represent the primary and secondary waves at the observation point. O(s) is the combined (observed) signal.

The objective is to reduce the observed signal to zero. This requires that the observed primary and secondary signals should be equal in amplitude and opposite in phase; i.e.

$$P_o(s) = -S_o(s) \tag{1}$$

Obtaining $P_o(s)$ and $S_o(s)$ from Fig. 1b in terms of P(s) and substituting in Eq. (1), after simplification, yields the following expression for C(s)

$$C(s) = \frac{X_3(s)}{M(s)L(s)\left[X_2(s)X_3(s) - X_1(s)X_4(s)\right]} \tag{2}$$

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This represents the desired controller transfer function for obtaining cancellation over the frequency range of interest. In designing such a controller care must be taken that the stability conditions are satisfied. Specifically, the closed feedback loop present in the structure, due to the secondary wave reaching the detector, which can make the system unstable, requires a careful attention so that with the designed controller the system satisfies the stability conditions. These conditions can be verified through the use of Nyquist stability criterion [4].

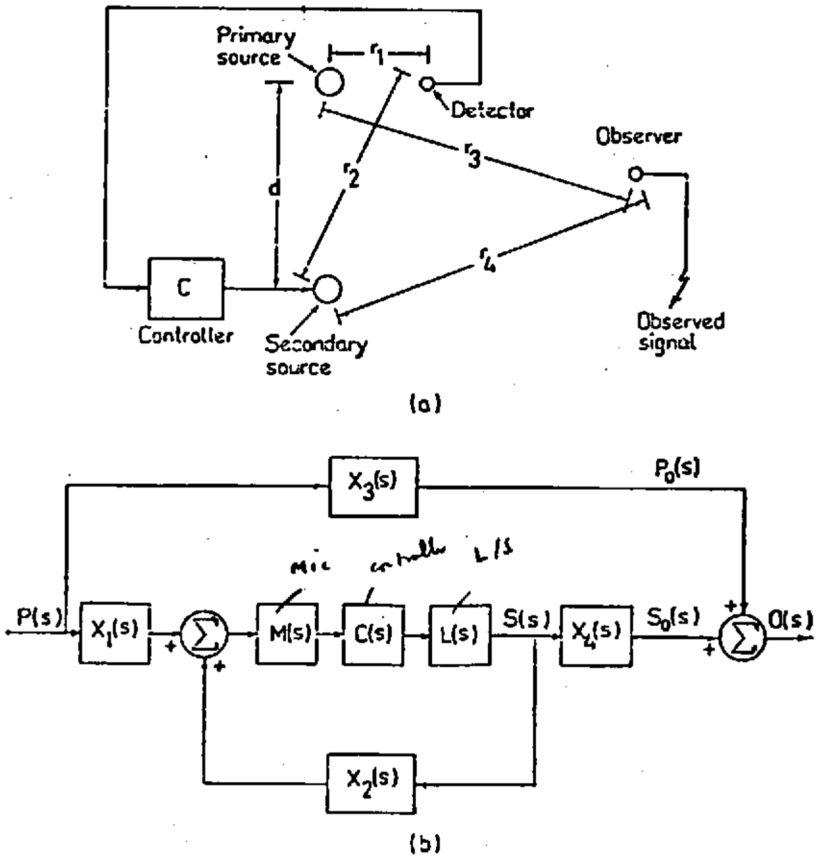

Fig.1 Active noise control system structure
(a) schematic diagram, (b) block diagram.

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DIGITAL IMPLEMENTATION OF THE CONTROLLER

The desired transfer function of the controller for the FFCS (Eq. (2)) is dependent on the transfer characteristics of the secondary source, detector, acoustic paths from the detector to the primary and secondary sources, and the acoustic paths from the observer to the primary and secondary sources. If the observer location is restricted to such points that the distance difference $|r_3 - r_4|$ is much smaller than (say, less than a tenth of) the smallest of r_3 and r_4 then the transfer functions $X_3(s)$ and $X_4(s)$ will approximately be equal for periodic signals ($s = j\omega$) over the low frequency range (say 0 - 500 Hz) of concern in the design of ANC systems. Using this assumption the required controller characteristic given in Eq. (2) is simplified to yield, for $s = j\omega$,

$$C(j\omega) = \frac{1}{L(j\omega)M(j\omega)[X_2(j\omega) - X_1(j\omega)]} \; ; \; X_3(j\omega) \equiv X_4(j\omega) \tag{3}$$

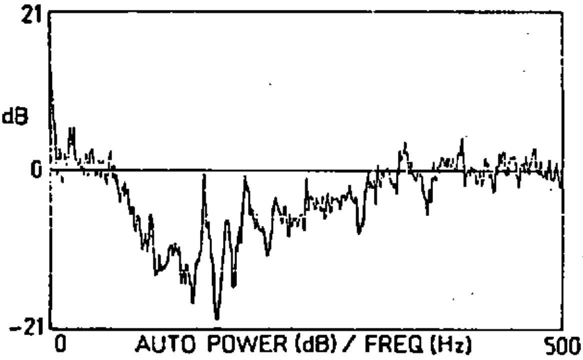
For a specific detector location relative to the primary and secondary sources, the transfer functions $LMX_1(j\omega)$ and $LMX_2(j\omega)$ can be measured over a specified frequency range and substituted in Eq. (3) to yield the required controller transfer function for that frequency range. The controller characteristic has thus been obtained using a condenser type microphone as a detector, at a negligible distance from the face of the primary source (a loudspeaker), and about 34 cm away from the centre of the secondary source face (a loudspeaker, having characteristic $L(j\omega)$). To implement the controller as a digital filter a suitable transfer function in the complex frequency s, $H(s)$, is obtained that approximates the characteristics of the controller (Eq. (3)). In obtaining $H(s)$ care has been taken to ensure system's causality and stability: the number of zeros must be less than or equal to the number of poles and the poles must have negative definite real parts. Also, the practical limitations of the signal processor on which it was to be implemented (Intel-2920) and the satisfaction of the amplitude and phase conditions of cancellation [2] over the desired frequency range have been observed.

Having obtained a continuous-time approximate prototype a discrete-time equivalent of the controller transfer function was obtained by transforming $H(s)$, via the bilinear z-transformation [3], to a discrete-time transfer function. This was implemented as a digital filter on the Intel-2920 signal processor using the full program memory of the device, giving a sampling frequency of 6.5 KHz. The characteristics of the digital filter thus obtained approximate the continuous-time transfer function, $H(s)$, closely. This is referred to as the digital filter controller (DFC).

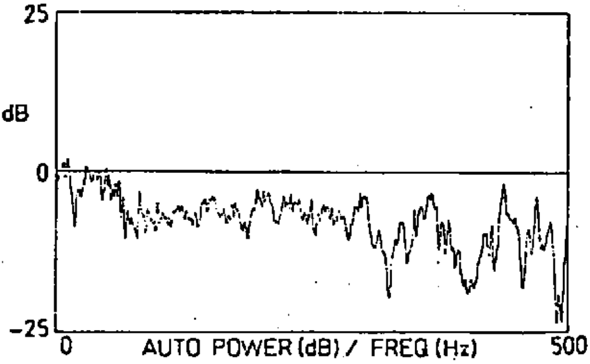
EXPERIMENTAL RESULTS

For the purposes of comparison, a second Intel-2920 signal processor was programmed to give a constant characteristic of unity gain and linear phase over the frequency range of 0 - 500 Hz. This is referred to as the constant-gain controller (CGC).

The performance of the ANC system with the CGC and the DFC was first investigated using a loudspeaker as the primary source. This was driven by a PRBS signal of bandwidth 0 - 500 Hz, simulating a broadband white noise source; the noise was measured at about 1.5 m from the primary and secondary sources. Using the CGC with the gain adjusted (using an amplifier) to give maximum cancellation the system performance is as shown in Fig. 2a. The 0 dB line partitions the regions of cancellation (below the line) and reinforcement (above the line). The system provides cancellation from 62 Hz to about 320 Hz. The overall average cancellation is 3 dB with the maximum peak cancellation being 19.6 dB at 163 Hz. Here the secondary (cancelling) path of the system is equivalent to a constant-length transmission line. Thus the condition for maximum cancellation is satisfied at only one frequency (163 Hz).



(a)



(b)

Fig.2 Cancellation of synthetic source noise (a) using the CGC, (b) using the DFC.

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Noise components at frequencies smaller or larger than 163 Hz are less attenuated, and eventually reinforced. Ideally the curve in Fig. 2a should be symmetrical about a vertical line at 163 Hz. However, due to the response characteristics of the secondary source loudspeaker the cancellation is not symmetrical.

Using the DFC the performance shown in Fig. 2b is obtained. Here cancellation starts before 50 Hz and extends to the maximum frequency of the source (500 Hz). Besides broadening the frequency range of cancellation from less than 2.5 octaves to more than a decade the DFC provides a greater amount of cancellation; the average cancellation is more than 8 dB, with a maximum peak cancellation of 23.5 dB occurring at 484 Hz. Here the secondary path of the system is no longer equivalent to a constant-length transmission line but has frequency-dependent characteristics that alter the amplitude as well as the phase of each detected frequency component so that a reduction in each frequency component of the noise is obtained. The relatively smaller amount of cancellation at frequencies less than 300 Hz in Fig. 2b is a consequence of relatively larger approximation error in the amplitude characteristic of the DFC relative to the desired controller characteristics in that range. However, the performance of the DFC shows a considerable improvement over the CGC. Essentially, this is achieved by incorporating a model of the secondary source (loudspeaker), detector (microphone), and the acoustic transmission paths in the design of the controller. The performance is currently being further enhanced by using a more powerful signal processor, such as the Texas Instruments TMS320 device. This will allow the implementation of higher order filters thereby reducing, further, the approximation errors.

To further verify the design procedure, the performance of the ANC system was investigated with a real source. The exhaust noise of a motorcycle (Yamaha 125 cc) was chosen as this is a compact, low-frequency, source. In addition, the frequency content of the noise can be adjusted by altering the throttle setting, thereby, allowing the performance under time-varying condition to be assessed. Using the CGC the performance of the system is as shown in Fig. 3a. A significant amount of cancellation occurs in the range 72 - 240 Hz, with a maximum cancellation of 19. 9 dB at 140 Hz. The reinforcement in the noise level increases with frequency after 240 Hz making the overall average cancellation slightly more than 0 dB. Compared with Fig. 2a, the performance has deteriorated from the results obtained with the synthetic source. This is because of the difference in the location(s) of the detector and/or observer relative to the primary and secondary sources as compared with the situation corresponding to Fig. 2a.

Using the DFC the performance of the system, as shown in Fig. 3b, is significantly improved. Cancellation is obtained from about 62 Hz to about 373 Hz. The overall average cancellation is more than 4 dB with a maximum peak cancellation of 19.9 dB occurring at 284 Hz. In this experiment the DFC significantly increased the amount of cancellation over a broader frequency range. However, as compared with the previous results (synthetic source), reinforcement occurred at frequencies greater than 373 Hz. This is because of the difficulty in locating the detector relative to the primary and secondary sources to be identical to that used in defining the required controller characteristics. At very low frequencies the effect of the detector location will not be significant; i.e. the additional phase shift introduced due to the error in locating the detector is negligible at very low frequencies, but as the frequency increases this additional phase shift becomes more significant so that it eventually produces reinforcement.

The above experimental results are encouraging. Subjectively, a significant reduction with low frequency noise was observed, although an increase in the noise level at frequencies greater than 400 Hz (Fig. 3b) could be perceived. These frequency components could be removed by further filtering of the secondary signal, taking care not to disturb the desired controller transfer function within the range of cancellation. Alternatively, these higher frequencies can be reduced by conventional passive noise reduction techniques.

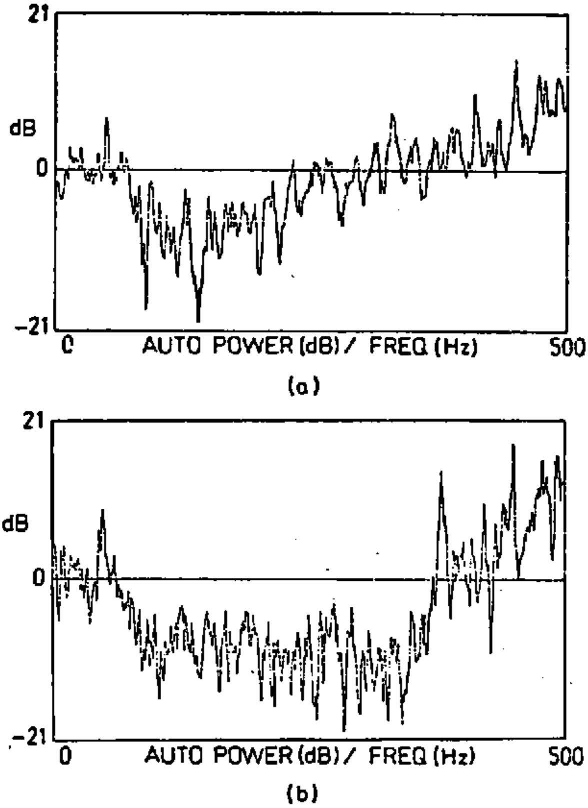

Fig.3 Cancellation of motorcycle exhaust noise (a) using the CGC, (b) using the DFC.

The results presented are for sources with stationary spectral characteristics. In the practical case, the exhaust noise varies with engine speed, and because of the nonlinear characteristics of the components the desired controller transfer function must also be adjusted. This requires an adaptive mechanism to adjust the DFC to maintain maximum attenuation of the noise. Work on an adaptive mechanism has already been reported [4].

CONCLUSIONS

The design of an ANC system requires a proper study of the system's structure and a clear characterisation of the unwanted acoustic noise source. The ANC system in the form of a feedforward control structure has been considered and analysed from a control engineering perspective. In order to utilise the computational power now provided by advanced digital technology, a method for the design of ANC systems with compact broadband sources has been presented.

Three types of acoustic sources have been characterised: (1) broadband, (2) time-varying, and (3) distributed sources. The unwanted noise emitted by practical sources are generally of broadband nature. In attenuating a noise of this nature a (proper) controller, which is the main part of the system, is required to be able to produce the inverse of every frequency component such that when superimposed on the unwanted noise a reduction of the noise level results. This means that the controller is required to have a continuous transfer function. The dependence of the characteristics of the controller on measurable frequency-dependent parameters of the system makes it possible to obtain the required frequency-dependent characteristics of the controller. The affect of such frequency-dependent characteristics was shown to significantly increase the bandwidth of cancellation. In particular, the results for the cancellation of motorcycle exhaust noise are encouraging.

The spectral contents of most practical sources are generally not constant but depend upon external parameters, resulting in a time-varying spectrum. In reducing the noise of a compact source with time-varying characteristic a fixed digital filter controller is not adequate. In such a case the controller is required to be able to adapt its characteristics, in accordance to the source, to maintain maximum cancellation over the required frequency range. Such a system would also have the property of self-tuning the controller to the desired characteristics.

A feature of many practical noise sources is that they can not be considered as compact; the physical size of the source is an appreciable proportion of the wavelength of the emitted noise. The noise emission is distributed over the source. A solution to such a case would be to use a multiple-detector and/or multiple-source configuration incorporating the methodology presented as the design basis. If all the three characteristics, namely, broadband, time-varying, and distributed, are accumulated in one source a solution (a complex one) would be the requirement of having a controller with frequency-dependent characteristics and adaptive capability incorporated with a multiple-detector and/or multiple-source configuration.

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REFERENCES

[1] R.R. Leitch and M.O. Tokhi, 'Review of active noise control systems', (submitted) IEE Proc., Part A, (1985).

[2] R.R. Leitch and M.O. Tokhi, 'General conditions for phase cancellation in active noise control systems', (submitted) IEE Proc., Part A, (1985).

[3] R.R. Leitch and M.O. Tokhi, 'Active noise control systems: structures and implementation', (submitted) J. of Sound and Vib., (1986).

[4] A.M. Alvarez-Tinoco, 'Adaptive algorithms for the active cancellation of acoustic noise', Ph.D. Thesis, Heriot-Watt University, Edinburgh, U.K., (1985).