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# Design and implementation of self–tuning active noise control systems

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## Abstract

A design method for active noise control (ANC) systems, in a three-dimensional non-dispersive (linear) propagation medium with compact broadband noise sources, is presented. The design and implementation of the controller as a digital filter with a fixed transfer function is presented and verified through a set of practical experiments. The concept of self-tuning control as a combination of system identification and control is developed within the ANC system framework. Moreover, as required by the application, a supervisory level control based on a performance criterion is added that automatically activates self-tuning control. The algorithm is implemented on a high-speed digital signal processor and tested on the exhaust noise of a 100 cc motorcycle. The experimental results are presented and compared with those of the system employing a fixed digital filter controller.

## 1 Introduction

Active noise control uses the intentional superposition of acoustic waves to create a destructive interference pattern such that a reduction of the unwanted sound occurs. This is realised by a single or a number of secondary (cancelling) sources of sound, driven by electrical signals derived from the primary (unwanted) noise through detection sensors and then passed through some form of electronic controller, so that when the secondary wave is superimposed on the primary wave the two waves destructively interfere and reduction of the noise occurs. In this manner, the generation of a stable destructive interference pattern is only possible for low frequencies, say less than 500 Hz, and for within limited areas. This restriction of frequency range is complementary to the limitations of passive methods; any practical solution would require a combination of passive techniques for higher frequencies (greater than 500 Hz, say) and active techniques for low frequencies (less than 500 Hz, say).

Active noise control was one of the earliest applications of electronics to the control of physical systems. Lueg filed for a patent in Germany in 1933 and in the USA in 1934 and was granted US Patent No. 2,043,416 in 1936 [1]. Although since Lueg's first theoretical proposals there has been considerable effort devoted to the theoretical and practical development of ANC systems the area is still under development, and apart from a few special cases practically successful ANC systems have still to find wide practical applications.

Active noise control is realised by using either one or both of the basic principles of interference and absorption. The interference principle consists of the mixing of acoustic waves resulting in constructive and destructive interference which, in turn, causes intensification and weakening of the sound field, respectively. The absorption principle on the other hand, results by synchronising the secondary source, say a loudspeaker's diaphram, in antiphase to the unwanted noise so that the noise energy is absorbed by the loudspeaker.

It was realised rather early by both Jessel and Kido that the primary advantage of ANC systems is in their ability to attenuate low-frequency noise [2, 3]. This is an area of considerable interest because of the pervasiveness of low-frequency sources and the high cost, large bulk, and relative inefficiency of current passive hardware in low-frequency applications [4, 5]. Besides this, an advantage in the control of the one-dimensional propagation of duct noise lies in the fact that active duct noise silencers produce no back-pressure.

Jessel, and others, also discovered some of the problems associated with reducing duct noise. Longitudinal duct modes leading to acoustic feedback, due to reflected waves, tend to confuse the controllers as to the exact level of the noise itself. This leads to system instability and/or no noise reduction in some frequency bands. To solve the longitudinal mode problem, so that the detector microphone detects the unwanted noise only, attempts have been made to use loudspeaker/microphone arrays. The acoustic tripole and acoustic dipole have been developed by Jessel and his co-workers and Swinbanks, respectively [2, 6]. They attempt to provide a canceling signal in the duct that propagates only in the downstream direction. The Chelsea System, developed by Leventhal, is formed by two secondary sources with the detector midway between them [7]. The controller is set to null the resultant of the secondary sources' waves at the detector location, thus isolating the detector from secondary source radiation. The performance of these systems shows that they provide noise cancellation of up to 20-25 dB over a narrow band of less than an octave; the cancellation provided is optimum at only one frequency that is related to the physical spacing of secondary dipole sources [7-9]. These systems have obvious geometry-related limitations. The control problem is also much more complex in such systems. Their third limitation is the so called 'tuning effect' due to the physical spacings of the microphone and loudspeakers relative to each other. By altering these spacings the system is tuned to a different centre frequency, with no significant improvement in the bandwidth of attenuation.

The main part of an ANC system which requires greater attention is a design procedure for the controller. Theoretical and practical investigations have shown that the characteristic feature of cancellation of a broadband noise achieved with an ANC system employing a constant-gain controller, such as an amplifier, is that cancellation occurs at a narrow band around a single frequency [4, 7, 10-12]. Here the controller is equivalent to a constant-length transmission line and hence can delay only a single tone of the noise by the right amount (corresponding to $180^\circ$ phase shift) so that maximum cancellation is achieved at that frequency only, with cancellation decreasing on either side of this frequency and eventually reinforcing the noise. Because of the broadband nature of many acoustic sources, the controller is required to have frequency-dependent characteristics so that it is capable of producing the mirror image of every detected frequency component of the unwanted noise so that attenuation over a broad frequency range is achieved. The dependence of controller characteristics on a number of frequency-dependent factors within the system, such as the characteristics of the sources, transducers and other electronic equipment, and the propagation medium, makes it possible to measure/identify and realise the required characteristics either as an analogue or digital or hybrid controller [12, 13]. The current development in microprocessor technology provides the opportunity to implement such a controller using digital techniques. In particular, there exist powerful digital processors, specially designed for signal processing and digital filtering applications. These have the capability for the real-time processing of signals within the frequency range suitable for ANC, and are cheap enough to allow dedication to one system.

The characteristics of many practical sources drift and vary due to a number of factors, such as operating conditions, and hence, leading to time-varying characteristics. This is also the case with the many of the transducers and other electronic equipment used in an ANC system. The geometric arrangement of an ANC system on the other hand has a direct effect on the performance of the system because the controller transfer function is found to be directly related to the system geometry. The spectrum of an acoustic noise emitted into a propagation medium is affected by the response characteristics of the medium such that each component frequency of the noise will undergo an amplitude and a phase change from the point of emission to the point of detection. Moreover, acoustic feedback and reflected waves which are closely related to the acoustic response of an enclosure into which waves are emitted have a great effect on the performance of an ANC system and can cause instability in the system. Therefore, there are many situations in practice that an ANC system with a fixed controller will not be adequate to result in large amounts of cancellation of an unwanted noise. In such cases an alternative is to design an adaptive ANC system capable of changing the characteristics of the controller in accordance to variations in the system. Through his experiments of reducing transformer noise Conover was the first to realise the need for a 'black box' controller that would adjust the canceling signal in accordance with information gathered at a remote distance from the transformer, as the performance of his ANC system was deteriorating from time to time due to the time-varying nature of the transformer noise [14]. Later it has been realised by numerous authors that an essential requirement for an ANC system to be practically successful is to be adaptive [12, 15-21].

The capability of todays electronic technology provides the opportunity to develop sophisticated ANC systems that can 'self-tune' to the required controller characteristics, thereby minimising the need for extensive analysis. Such systems will also be able to allow to 'track' variations in spectral contents of real sources so that stable cancellation is maintained over a range of operating conditions. This adaptive capability is essential to the realisation of successful ANC systems for real industrial applications.

## 2 Active noise control structure

A schematic diagram of the geometric arrangement of a general ANC structure, the so called feedforward control structure (FFCS), is shown in Fig. 1. The secondary source located at a distance $d$ relative to the primary source is artificially generated through a processes of detection and control so that to achieve cancellation of the unwanted primary noise at the observer location. A block diagram of Fig. 1 is shown in the complex frequency $s$ domain in Fig. 2 where

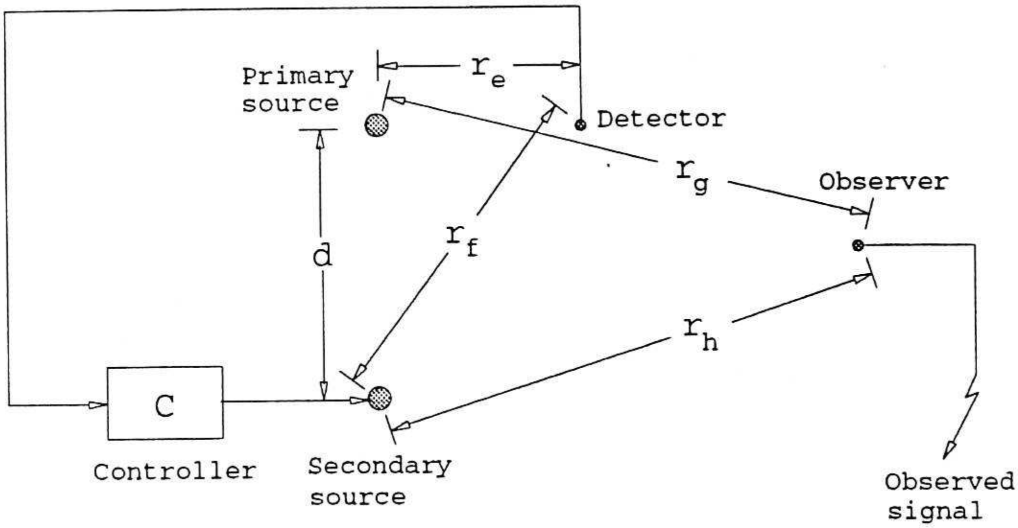



Fig. 1: Schematic diagram of the ANC structure.

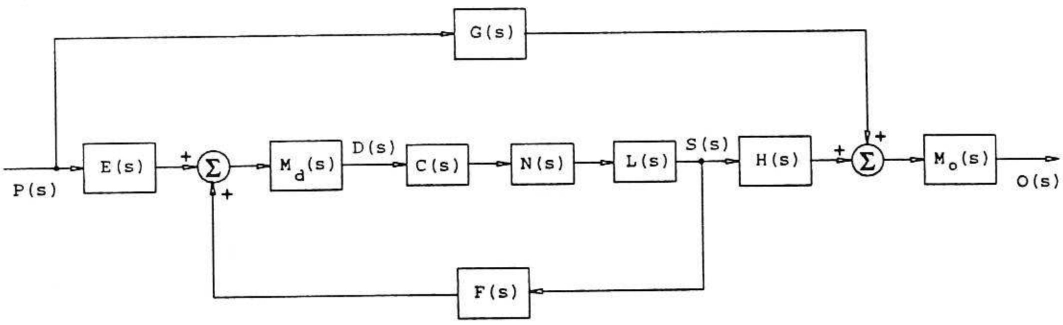



Fig. 2: Block diagram of the ANC system.

$P(s)$ = primary source signal at the source,

$D(s)$ = signal measured by the detector,

$S(s)$ = secondary source signal at the source,

$O(s)$ = signal measured by the observer,

$N(s)$ = equivalent transfer function of amplifier, processor A/D and D/A etc.,

$C(s)$ = transfer function of the controller,

$L(s)$ = transfer function of secondary source,

$M_d(s)$ = transfer function of the detector,

$M_o(s)$ = transfer function of the observer,

$E(s)$ = transfer function of the acoustic path through $r_e$,

$F(s)$ = transfer function of the acoustic path through $r_f$,

$G(s)$ = transfer function of the acoustic path through $r_g$, and

$H(s)$ = transfer function of the acoustic path through $r_h$,

It follows from the block diagram in Fig. 2 that the secondary path, through $C(s)$ to the observation point, attempts to compensate for the primary path, through $G(s)$ to the observation point, such that the superposition of the primary and secondary waves results in cancellation at the observation point.

## 2.1 *Design* and *implementation of a fixed controller*

The detector and observer signals; namely $D(s)$ and $O(s)$ in Fig. 2, can each be thought of as composed of the sum of two signals one due to the primary source and one due to the secondary source. From the block diagram in Fig. 2 this can be expressed as

$$D(s) = M_d(s)E(s)P(s) + M_d(s)F(s)S(s)$$
$$O(s) = M_o(s)G(s)P(s) + M_o(s)H(s)S(s) \tag{1}$$

The secondary source signal $S(s)$, on the other hand, can be expressed, through the controller path, as

$$S(s) = C(s)N(s)L(s)D(s) \tag{2}$$

For complete cancellation of the noise to be achieved at the observation point the condition

$$O(s) = 0 \tag{3}$$

must be satisfied. This is the performance criterion, corresponding to a minimum variance criterion in a stochastic environment, on which the design of the controller is based. Applying this criterion to the system through manipulation of equations (1) and (2) yields the required transfer function for the controller as

$$C(s) = \frac{G(s)}{M_d(s)N(s)L(s)\Delta(s)} \tag{4}$$

where

$$\Delta(s) = F(s)G(s) - E(s)H(s)$$

Equation (4) is the required controller transfer function that satisfies the performance criterion given in equation (3). It follows from this equation that for a particular detector and secondary source with necessary electronic components, the controller characteristics required for optimum cancellation at an observation point is dependent on the geometric set-up of the system; i.e. locations of sources, detector and observer (sensors) relative to one another. Any change in the geometry requires a particular controller characteristic. In particular, if the change is such that $\Delta(s)$ becomes zero then the critical situation of infinite gain controller requirement arises. The locus of such points in the medium (as a practical limitation in the design of the controller) is therefore of crucial interest. This locus can easily be obtained by an analysis of the system [22, 23]. As noted in Fig. 2 the loop formed by the detector, controller, secondary source and the acoustic path between the secondary source and the detector is a feedback loop that may cause instability in the system response. This is due to the acoustic feedback from the secondary source radiation towards the detector. Therefore, an analysis of this loop from a stability point of view is important in the design stage.

For practical systems a measure of absolute stability is not useful; a system that has an extremely long and oscillatory transient is unlikely to be accepted. In this respect a measure of the relative stability can provide a more acceptable design criterion. These can be provided from frequency response plots in terms of the gain and phase margins [24].

The practical realisation of the controller transfer function obtained from the design procedure outlined above consists of measuring the various transfer functions that appear in the controller design equation using a suitable measurement technique, calculating the controller transfer function using these measurements and finding an appropriate transfer function that matches the controller transfer function to within practical limits. This will result in a transfer function in the continuous-time domain. The controller transfer function thus obtained can be realised either as an analogue filter, or discretised and realised as a digital filter controller (DFC) [12].

## 3 Adaptive active noise control

An ANC system employing a controller with a fixed transfer function, such as the DFC, will result in significant amounts of cancellation over a broad frequency range of the noise provided (*a*) the characteristics of the various components in the system, such as the source, transducers and other electronic equipment used, are time-invariant,

(*b*) the geometric arrangement of the system is not altered once designed, and

(*c*) the acoustic response of the medium stays linear and the effects of any possible acoustic reflections are minimised.

In practice, however, all these conditions can not be met, as a result of which the performance of the system deteriorates and the system may even exhibit instability problems [12]. Therefore, to compensate for these problems, an alternative is to design an ANC system capable of adapting the characteristics of the controller in accordance with the variations in the system.

The design of an adaptive controller for use in an ANC system involves two processes. Firstly, of finding a model for the system between its input and output using a system identification algorithm, say the RLS algorithm and, secondly, of controller design computation based on this model and the design objectives. Modelling of the system in terms of the general ANC structure of Fig. 1, will mean measuring the primary source signal (as system's input), at the source, and the observed signal (as system's output).

Measurement of the observed signal in Fig. 1 can easily be achieved by using an acoustic sensor, such as a microphone, at the observation point. Measurement of the primary source signal at the point of emission, however, can not be achieved easily because no direct access to this signal will be available in practice. Therefore, to obtain the required information on the system input, measurement of a signal coherent with the emitted noise is required. This can easily be achieved by one of two methods

(*i*) a direct measurement of the acoustic signal at a distance from the source; e.g. using a, detector, microphone, as in Fig. 1. Here, the design of the controller will be based on a model obtained from measurements of detected signal (as system input) and observed signal (as system output).

(*ii*) an indirect measurement of the signal; e.g. by a vibration sensor. In numerous practical situations the unwanted acoustic noise can be detected through a measurement of non-acoustic signals that are coherent with the noise. For instance, many sources of noise vibrate while in operation constituting spectra that are coherent with the noise spectrum. Flame noise is another example where it has been found that the correlation coefficient between sound and light can be as much as 0.99 [25, 26]. In such cases sensors, suitable for the application, can be used to detect the corresponding signal that is coherent with the primary source signal. Here, the controller design will be based on a model describing the system between the detected and the observed signals. An advantage of indirect detection over direct detection is that the acoustic feedback from the secondary source to the detector, which can cause instability problems in the system, is avoided. However, care must be taken to ensure that full information about the primary source noise spectrum is obtained.

### 3.1 *Controller design*

An ANC system can, in general, be considered as having two signal generating, or source, points; namely, the primary and secondary sources. The primary source is an independent source of noise whereas the secondary source is derived from the primary source and hence is a dependent source. Therefore, the following two conditions are possible.

(*i*) The secondary source being off, which in terms of Fig. 2 is characterised by $S(s) = 0$. This is achieved by disconnecting the link between the detector and the secondary source.

(*ii*) The secondary source being on, which in terms of Fig. 2 is characterised by $S(s) \neq 0$. This is achieved by linking the path between the detector and secondary source through either replacing $C$ in Fig. 1 by a wire ($C(s) = 1$ in Fig. 2) or by a suitable choice of a transfer function, the effect of which can be accounted for within $N(s)$ in Fig. 2.

This sub-divides the ANC system into two equivalent sub-systems. Therefore, the design of an adaptive controller will invariably be based on models of these sub-systems. Let the system transfer functions under the two conditions be $X_0(s)$ and $X_1(s)$;

$$X_0(s) = \frac{O(s)}{D(s)} \Big|_{S(s) = 0} \quad , \quad X_1(s) = \frac{O(s)}{D(s)} \Big|_{S(s) \neq 0}$$

Substituting zero for $S(s)$ into equation (1) and simplifying yields the system transfer function as

$$X_0(s) = \frac{M_o(s)}{M_d(s)} \left[ \frac{G(s)}{E(s)} \right] \tag{5}$$

Substituting unity for $C(s)$ into equation (2), corresponding to $S(s) \neq 0$, and using equation (1) yields the system transfer function as

$$X_1(s) = \frac{M_o(s)}{M_d(s)} \left[ \frac{G(s)}{E(s)} \right] - M_o(s)N(s)L(s) \left[ \frac{G(s)F(s) - E(s)H(s)}{E(s)} \right] \tag{6}$$

Manipulating equations (5) and (6), and using equation (4) yields

$$C(s) = \left[ 1 - \frac{X_1(s)}{X_0(s)} \right]^{-1} \tag{7}$$

Equation (7) is the required controller transfer function for optimum noise cancellation at the observation point. Using a similar procedure as above and equivalent discrete-time transfer functions yield the equivalent controller transfer function $C(z)$;

$$C(z) = \left[ 1 - \frac{X_1(z)}{X_0(z)} \right]^{-1} \tag{8}$$

where

$$X_0(z) = \frac{M_o(z)}{M_d(z)} \left[ \frac{G(z)}{E(z)} \right]$$

$$X_1(z) = \frac{M_o(z)}{M_d(z)} \left[ \frac{G(z)}{E(z)} \right] - M_o(z)N(z)L(z) \left[ \frac{G(z)F(z) - E(z)H(z)}{E(z)} \right]$$

Equation (8) gives the required controller transfer function in terms of the system transfer functions $X_0(z)$ and $X_1(z)$.

## 3.2 *Controller identification*

It follows from the previous section that on-line identification of the controller requires estimating the system transfer functions $X_0(z)$ and $X_1(z)$ and then using equation (8) as the controller design rule to calculate the required controller transfer function.

Unlike commonly occurring control problems, the controller in an ANC system is contained within the path between the input and output of the plant. Isolation of the controller in the process of identification is achieved by replacing it by an automatic switch. The switch is implemented in software as part of identification algorithm. During the identification process samples of the plant input, $D$, and output, $O$, are measured via two independent input channels of the processor, based on which the plant model parameters are estimated, and at the same time the switch is implemented between the input channel taking $D$ and output channel of the processor. In the process of identifying $X_0(z)$ the output channel of the processor is kept at zero signal level, implementing an open switch, resulting $S = 0$, whereas during the identification of $X_1(z)$ the processor outputs the plant input samples as received, thereby implementing a closed switch, resulting $S \neq 0$.

The identification procedure is based on a RLS algorithm which gives estimates of the system model, of known structure in parametric form, based on measurements of system's input and output samples at each sampling period. The model structure chosen for the system is in the general form of an $m$th order IIR filter with $D(nT)$ and $O(nT)$ as samples of the detected and observed signals, respectively, at sample time $nT$, where $n$ is an integer real number. The identifier gives estimates of the parameters of the plant model for known model order $m$ and suitable sampling period $T$ under the two conditions of $S = 0$ and $S \neq 0$.

It is important to note that the identification process will proceed in two phases, once for the estimation of a model for $X_0(z)$ and then for the estimation of a model for $X_1(z)$, resulting in the corresponding estimated transfer functions $\hat{X}_0(z)$ and $\hat{X}_1(z)$. The switching of identification from one phase to the other is performed automatically within the process and results of the two phases are stored and viewed together at the end of the identification exercise. Note that since the results of the two identification processes are subsequently used in the design calculation of the controller transfer function care must be taken to ensure that the sampling periods are consistent in both phases otherwise the resulting controller transfer function will be incorrect.

The model order is chosen reasonably on the basis of practical knowledge about the plant. If the model order is over estimated, i.e. $\hat{m} > m$, there will be unnecessary redundant parameters in the model which will limit the computational speed and available data storage and memory space of the processor. On the other hand, if the model order is under estimated, i.e. $\hat{m} < m$, the identifier will converge to a wrong model which is potentially more severe than that when $\hat{m} > m$.

The choice of initial values of elements of the covariance matrix and accuracy of computation are influential factors affecting the speed and accuracy of convergence. The initial covariance matrix is chosen as $\rho I$, where $I$ is the identity matrix and $\rho$ a positive real number. The size of $\rho$ gives a measure of the degree of confidence with initial parameter values and speed of convergence. Although a large value, in the order of thousands, may be desired for $\rho$, however, in practice, the size of $\rho$ is limited by the processor word-length and dynamic range of computation. A value in the order of even 2000 or more may cause overflow and subsequent estimation errors. Moreover, since the elements of the covariance matrix change over a large range of integer to fractional values a large dynamic range of computation is required. These requirements, with a fixed-point processor, are met by either implementing floating-point arithmetic or splitting variables into integral and decimal parts and storing each part in separate data memory locations. Both these methods trade off for the accuracy of convergence at the expense of additional computation which result a reduction in processing speed and hence increase the sampling time.

The ability to track variations in the plant dynamics during the identification process is achieved, by not allowing elements of the covariance matrix to become too small in magnitude, through the use of a forgetting factor to discount old data. Other techniques of discounting old data include the random walk method which is application dependent and its use needs caution. Due to its ease of implementation the method of forgetting factors is popularly used in self-tuning control applications [27-31].

After obtaining the estimated plant model transfer functions $\hat{X}_0(z)$ and $\hat{X}_1(z)$ the controller design rule, equation (8), is used, with the plant model transfer functions replaced by their estimates. The resulting controller transfer function thus obtained is then implemented as a digital filter on the digital signal processor. As is common to the implementation of digital filters this will result in an additional linear delay in the phase of the controller transfer function due to the A/D and D/A conversion circuitry of the processor. To compensate for the error in phase characteristics of the controller thus introduced a compensating network is required to be connected, in cascade, with the process. This can be achieved by either using an analogue network of a proper transfer function connected (in series with A/D and D/A circuitry) externally, or implementing a discrete-time transfer function in cascade with $C(z)$ within the processor. The later option is more adequate due to its ease of implementation.

### 3.3 *Self–tuning control*

The self-tuning ANC algorithm as a combined identification and control is merely the on-line implementation of the controller design procedure outlined in the previous section. Note that, here the controller design calculation is to follow after the completion of the two phases of plant model estimation. Therefore, to achieve on-line adaptation of the controller parameters whenever a change in the system is sensed the addition of a supervisory level control is required. The supervisor is designed to monitor system performance and, based on a pre-specified quantitative measure of cancellation, initiate self-tuning control. The actual cancellation achieved at the observation point is measured at each sample time, if this is within the specified limit then the controller continues processing the detected signal, generating and outputting the cancelling signal, however, if the cancellation is outside the specified limit then the supervisor de-activates the controller and re-initiates the process at the level of plant model identification. The cancellation can be calculated on the basis of either measuring the average power of the signal or the average level of noise, by measuring and averaging the absolute value of the level of signal, received at the observation point. Fig. 3 shows the self-tuning ANC structure realising the following algorithm

*Algorithm*

(*i*) Set the controller transfer function as $C(z) = 0$.

(*ii*) Estimate the plant model transfer function, $\hat{X}_0(z)$, and store the result.

(*iii*) Set the controller transfer function as $C(z) = 1$.

(*iv*) Estimate the plant model transfer function, $\hat{X}_1(z)$, and store the result.

(*v*) Use the estimated transfer functions $\hat{X}_0(z)$ and $\hat{X}_1(z)$ to calculate the controller transfer function $\hat{C}(z)$ using

$$\hat{C}(z) = \left[ 1 - \frac{\hat{X}_1(z)}{\hat{X}_0(z)} \right]^{-1}$$

(*vi*) Process detected signal by the controller and output cancelling signal.

(*vii*) Measure the observer signal and calculate the cancellation achieved.

(*viii*) If cancellation is within the pre-specified limit then go to step (*vi*) otherwise go to step (*i*).

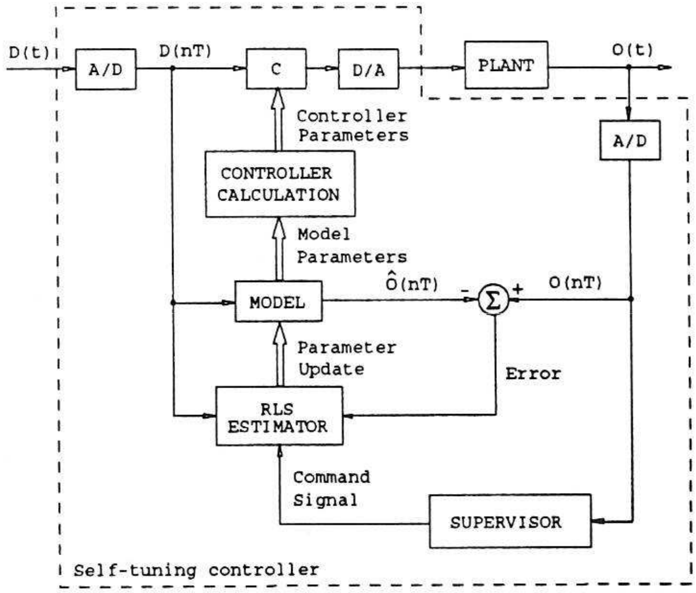



Fig. 3: Self-tuning controller.

### 3.4 *Implementation of the self–tuning ANC algorithm*

The self-tuning ANC algorithm outlined in the previous section has been presented in a way that makes it easy to implement on a dedicated signal processor such as the TMS32010 [32-33]. This, however, requires a careful consideration of the practical issues that affect the implementation process and hence the performance of the system. These, that are also common to the implementation of almost all digital controllers, include factors such as properties of the input signal, robustness of the estimation process, initialisation of parameters at the start, sampling time etc. and processor related factors such as signal quantisation, computational power, speed of computation, programming support, data format etc.

In digital control applications it is important to have a proper conditioning of the signals. Due to the aliasing problem connected with the sampling procedure it is necessary to remove all frequencies above the Nyquist frequency before sampling the signals. The filtering of the signals also has the effect that the system will be excited by frequencies where it is important to have good process models. In the current VLSI signal processors, such as the TMS32010 signal processor, the aliasing problem is eliminated by a low-pass (anti-aliasing) filter at the analogue input channel that band-limits the input signal before sampling [34]. The anti-aliasing filter has a flexible cutoff frequency which can be altered so that the in-coming analogue signal is band-limited to the required maximum frequency before sampling. In ANC applications a disadvantage of this filter, however, is the introduction of an additional phase lag in the path of the controller that should be compensated for by the controller transfer function.

In an ANC system the excitation is provided by the primary source. In simulation experiments where the primary source can be chosen properly to suit the robustness requirements of the control algorithm the problems due to the properties of the input signal will not arise. In practice, however, where there is no control on the primary source signal, care must be taken to condition the input signal properly before sampling. The process model order is a second important factor influencing the accuracy of the model and hence the robustness of the control algorithm. If the model order is under-estimated the resulting estimates will not be accurate. This will consequently result in an unsatisfactory performance. If the model order is over-estimated, however, an estimated transfer function with redundant, approximately cancelling, poles and zeros will be obtained. This can have two consequences: (1) the redundant parameters will take more of the computation time and hence increase the sampling time and (2) at high sampling frequencies the difference between the redundant poles and zeros, that are numerically close to each other, could be significant and result in a wrong estimated model transfer function.

The key property of an adaptive controller is the ability to track variations in the process dynamics and to do so it is necessary to discount old data. The speed at which data is discounted requires to be properly chosen. If the parameters are constant and the data is discounted too fast the estimates will become uncertain. Too slow discounting of data, on the other hand, will make it impossible to track rapid parameter variations. The method of exponential forgetting factor is one way to discard old data and works well only if the process is properly excited at all times. Several problems associated with the use of exponential forgetting and a number of ways of avoiding these problems are cited in the literature [35].

There are basically two categories of parameters to be chosen in a self-tuning ANC application, process related and performance related. The process related parameters are typically the order of the plant model and possibly the time-delay in the controlled process. The performance related parameters are the performance indices and initial values for the estimation routine. The process related parameters are usually easy to determine. The performance related parameters, however, require some understanding of the system behaviour and can be chosen such that the performance indices are insensitive to the initial values of the estimator.

One parameter that can greatly influence the behaviour of the algorithm is the sampling period. The choice of the sampling interval is not specific to adaptive controllers but is an important design parameter for all sampled data design methods. The controller in a self-tuning ANC system requires estimates of the transfer functions $X_0(z)$ and $X_1(z)$ to be accurate and consistent with the same sampling frequency. This can be achieved by properly choosing the orders of $X_0(z)$ and $X_1(z)$ and making sure the sampling frequency remains constant throughout the processes of identification and control. The model orders have a significant effect on the sampling time. An increased order taking more processor computational time and data memory space will reduce the sampling frequency significantly. This will consequently limit the maximum frequency of the input signal and can even make the controller impractical to use.

In ANC applications, where the accuracy requirements of the phase characteristics of the digital controller are crucial, the phase delay introduced by the ZOH can have a significant effect on the performance of the system. This requires the use of a compensating transfer function in cascade with the controller. The introduction of a delay compensation, however, will increase high frequency controller gain and can cause instability in the system. Although, after choosing a proper compensating transfer function, the instability problem at the high frequencies can easily be avoided by band-limiting the controller output, this problem, however, is not crucial in ANC because the system is not intended for high frequency applications.

The accuracy requirements of the estimated models and consequently of the controller in implementing the self-tuning ANC algorithm requires a large dynamic range.

One way to achieve this, with the TMS32010 signal processor, is to use floating-point arithmetic. However, previous applications have shown that this can not be achieved easily with the TMS32010 signal processor. An alternative, which at the same time avoids overflow and minimises quantisation effects, is that variables are separated into integral and decimal parts, scaled and each stored in 16-bit data memory locations. The implementation of the algorithm is then achieved by designing routines for addition/ subtraction and multiplication/ division of numbers. The disadvantages of this technique are limiting the available data memory space, specially as the order of the plant model increases and decreasing the sampling frequency.

## 4 Experimentation

Presented here are a set of practical experiments to investigate the performance of the system with the DFC and the self-tuning controller (STC) as the required controller. These experiments use the system arrangement shown in Fig. 4 where Amp1 and Amp2 are respectively the detector and observer microphones' amplifiers, Amp3 is a digital amplifier

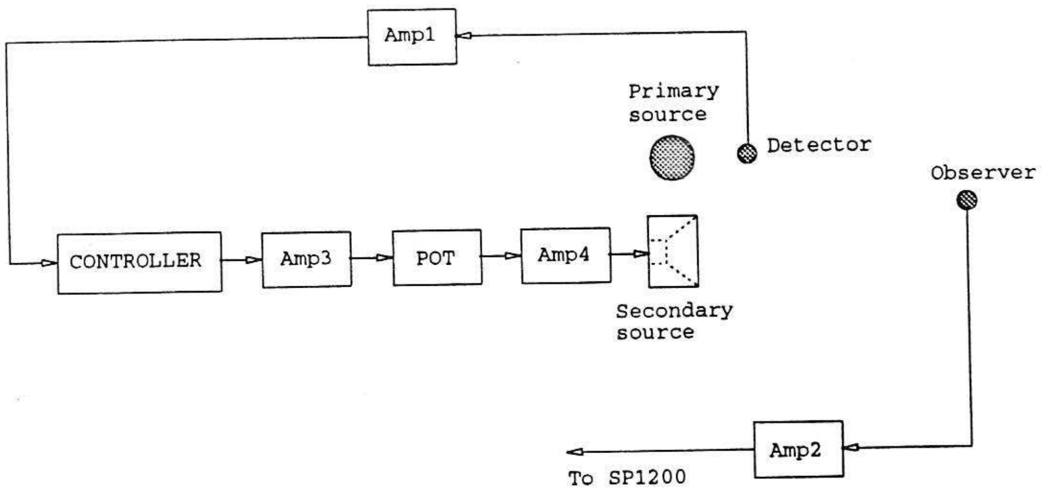


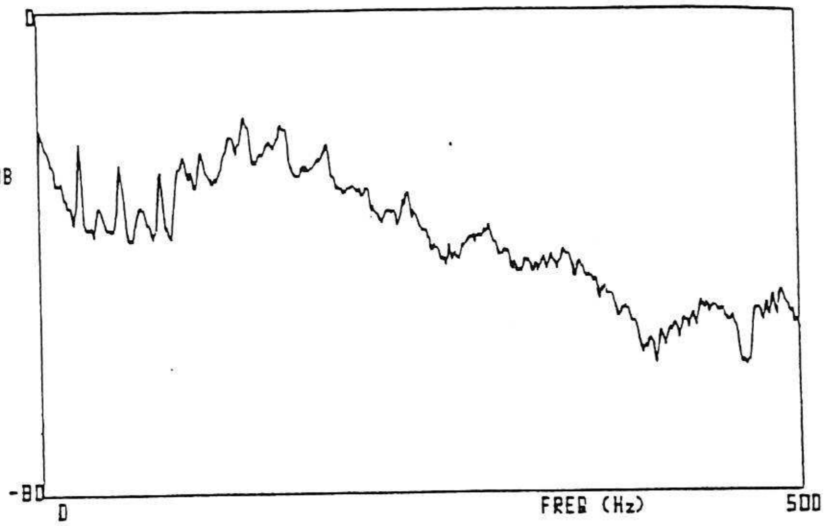

Fig. 4: Arrangement for active noise cancellation.

needed to amplify the filter output, Amp4 is a power amplifier used to drive the secondary source loudspeaker, the POT is a potentiometer used to control the volume of the power amplifier, as this amplifier did not have a volume control of its own, and the SP1200 is the Solartron 1200 FFT based signal processor used for measurement of noise cancellation achieved at the observation point [36].

The DFC is designed and realised as a digital filter through a process of measurement of the required controller transfer function over a frequency range of 0 – 500 Hz [12]. This is implemented on the Intel-2920 digital signal processor [37-38]. To obtain comparative results of the performance of the system a second Intel-2920 was programmed realising fixed transfer characteristics of unity gain and zero degrees phase over the frequency range of 0 – 500 Hz. This is referred to as the constant-gain controller (CGC). The performance of the system was investigated with the exhaust noise of a (Yamaha, 100 cc) motorcycle as the unwanted primary source. This represents a practical, compact, low-frequency source whose noise spectrum changes with operating conditions (throttle) resulting in a time-varying characteristic. Figs. 5 show the exhaust noise spectrum with two engine speeds, namely, idle and medium speed, at a distance of about 1.5 *metres* from the source. The noise is essentially low-frequency and is composed of harmonics of the fundamental (engine firing) frequency. An increase in the engine speed results a decrease in the spacing between successive harmonics and an increase in the level of noise.

Using the CGC as the controller in Fig. 4 and running the motorcycle engine at idle speed the cancellation was measured at a distance of about 1.5 *metres* from both sources. This is shown in Fig. 6a where the 0 dB line partitions the regions of cancellation (below the line) and reinforcement (above the line). A significant amount of cancellation is achieved in the frequency range of 72 Hz to 240 Hz, with a maximum cancellation of

(a)




(b)

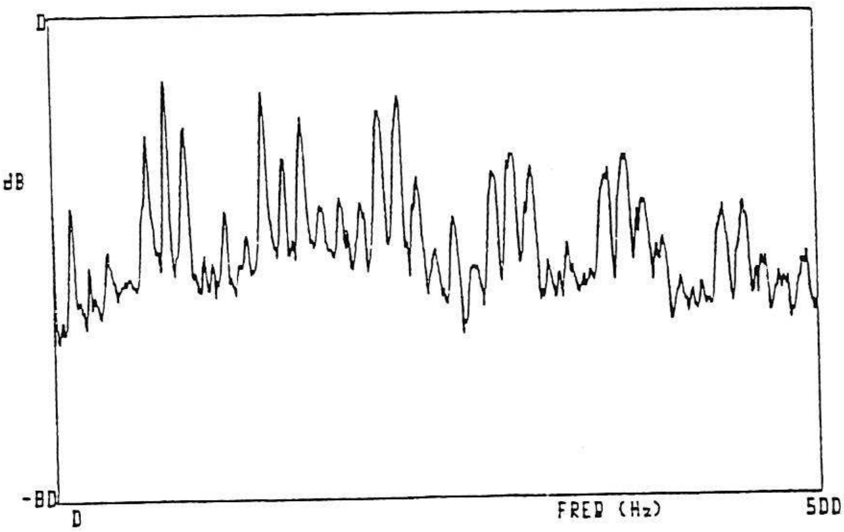



Fig. 5: Noise spectrum of the motorcycle exhaust;
(a) Idle engine speed,
(b) Medium engine speed.

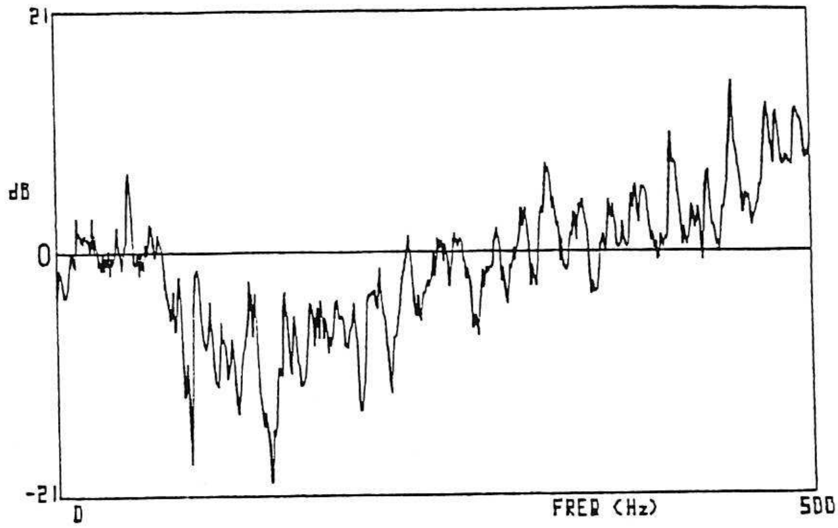
19.9 dB at 140 Hz. However, the noise is reinforced almost linearly for frequencies above 240 Hz making the overall average cancellation slightly more than 0 dB. Cancellation of a broadband noise thus obtained is featured by cancellation at one frequency range and reinforcement in the following and so on. This feature is a characteristic of the CGC as this is equivalent to a constant-length transmission line that satisfies the noise cancellation requirements at only a narrow band of frequencies.

Replacing the CGC with the DFC the result depicted in Fig. 6b is obtained. This shows a significant improvement in cancellation of the noise as compared with Fig. 6a. Noise is cancelled over a broad frequency range of nearly 62 Hz to 373 Hz. The overall average amount of cancellation achieved is more than 4 dB with a maximum peak cancellation of 19.9 dB occurring at 284 Hz.

The cancellation result in Fig. 6b shows a significant increase in the amount as well as frequency band of cancellation. However, the noise has been reinforced for frequencies greater than 373 Hz. This is because of the difficulty in positioning the detector relative to the primary and secondary sources to be the same as in the original geometric arrangement used in measuring the required controller characteristics; the controller here has a fixed transfer function that corresponds only to a specific geometric arrangement of sources and sensors under which the system will result good cancellation. At low frequencies the effect of an error in the geometry will not be significant in that the additional phase shift introduced, and hence demanded of the controller, will be negligible. However, as the frequency increases this additional phase shift becomes more significant so that it eventually leads to reinforcement.

The STC is implemented on the TMS32010 digital signal processor as this is more powerful than the Intel-2920 signal processor and hence is suitable for such applications [32-34]. Nevertheless, as noticed in the experiments to follow, the maximum frequency of

(a)




(b)

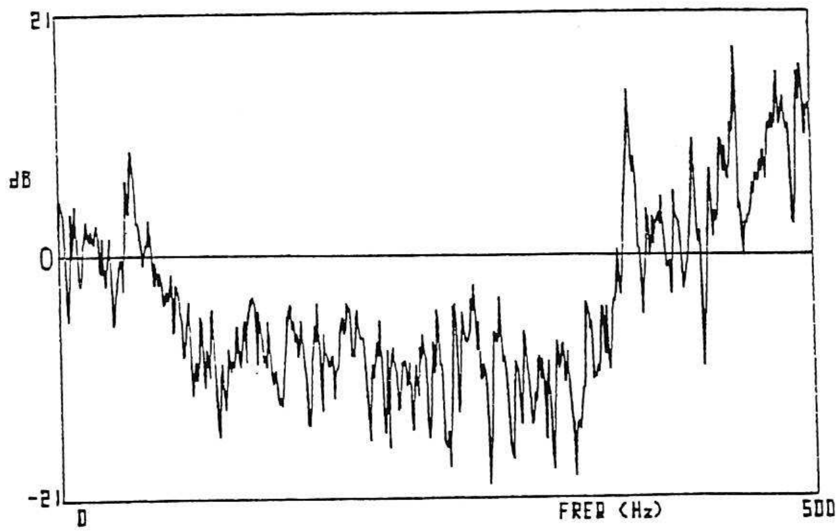



Fig. 6: Noise cancellation of the motorcycle exhaust;
(a) Using the CGC,
(b) Using the DFC.

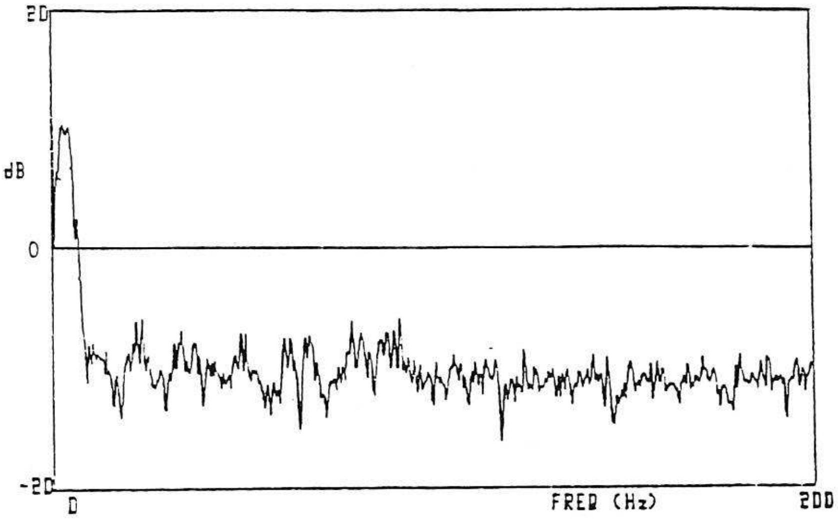
the unwanted noise is limited to less than 200 Hz rather than the 500 Hz used with the fixed controller. This is due to the large number of arithmetic operations required, specially, in the identification process of the algorithm which limits the sampling frequency. Using the system arrangement of Fig. 4 with the STC in Fig. 3 as the controller and the engine running at idle speed the result shown in Fig. 7a is obtained. The cancellation achieved covers a broad frequency range of about 8 Hz to the 200 Hz maximum measurement frequency. The average amount of cancellation achieved is more than 10 dB (pre-specified).

In order to show the difference in performance of the STC at different engine speeds a further experiment was conducted with the same experimental arrangement as above but the engine running at a medium speed. The result of this experiment is shown in Fig. 7b. Here, again, the noise is reduced over a broad frequency range of almost the lowest noise frequency to the 200 Hz maximum measurement frequency. The average amount of cancellation achieved is more than 10 dB (pre-specified).

The cancellation achieved with the STC in each of the above experiments is characterised by a steady and large amount over a broad frequency range. As compared with the fixed controller the STC having the ability to alter the controller characteristics in accordance to changes in the noise spectrum and geometry of the system gives cancellation of the noise from almost the minimum to maximum noise frequency. The significant amount of cancellation achieved in each of the above experiments demonstrates and verifies the practical applicability of the self-tuning ANC algorithm for cancellation of time-varying broadband noise emanating from a compact source in three dimensions.

It follows from the results of the above set of experiments that the self-tuning ANC algorithm as implemented on a dedicated signal processor can perform well and to within practically acceptable requirements. The algorithm is capable of providing the desired

(a)




(b)

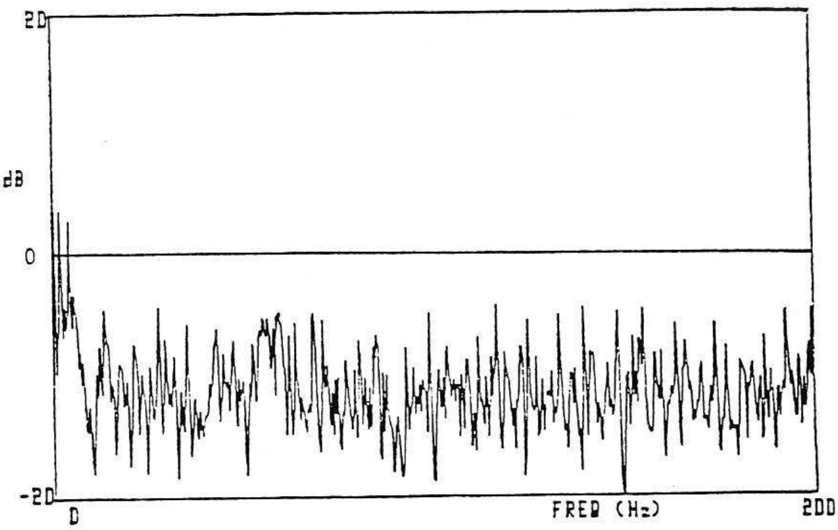



Fig. 7: Noise cancellation of the motorcycle exhaust using the STC;
(a) Idle engine speed,
(b) Medium engine speed.

amounts of cancellation of the unwanted noise under changes in the geometrical arrangement of the system as well as under time-varying conditions. A change in the system, leading to performance degradation, is automatically detected by the supervisory level which in turn de-activates the controller and re-initialises the self-tuning control so that the desired level of performance is maintained. Due to the TMS32010 signal processor's limitations the range of cancellation provided by the self-tuning ANC system presented is limited to a frequency band of 0 to about 250 Hz. However, the frequency range of cancellation can further be increased by using a more powerful digital signal processor that has a longer processor word length and faster instruction execution time and supports floating point arithmetic.

## 5 Conclusion

The design and implementation of a self-tuning ANC algorithm has been presented, discussed, and verified through a set of experiments. The active cancellation of broadband noise requires a careful consideration of the system from the acoustics and control engineering point of view. The results of such a consideration, leading to an accurate and practical design procedure, suggest that the controller satisfying the system's optimum performance criterion; i.e. complete cancellation of the unwanted noise, will have frequency-dependent characteristics that in turn are dependent on frequency-dependent parameters within the system. These include the electronic components, such as amplifiers, transducers, secondary source, etc. and the acoustic paths between the sources and sensors.

Considering the practical limitations in the measurement, design and implementation of the required controller transfer function an ANC system employing a digital filter controller with fixed transfer function performs to an impressive and significant level in case of a compact, broadband source of noise of a time-invariant nature.

Most of the sources of noise encountered in practice have characteristics that vary with operating conditions, thereby, having time-varying characteristics. Moreover, the geometric arrangement of sources and sensors in an ANC system will, generally, not remain constant, leading to variations in the characteristics of the acoustic paths of the system. These factors all together result in ANC systems with time-varying parameters. A controller with the required frequency-dependent characteristics designed under one situation will thus produce a secondary source that results in optimum cancellation of the unwanted noise under only that situation. If any parameter of the system changes the system will not perform as desired. In such a situation a controller whose characteristics are adjusted for full cancellation in accordance to the changes in the system is required.

An ANC system is characterised by two distinct sub-systems' behaviour; namely, when the secondary source is off and when the secondary source is on. The dependence of the required controller characteristics on the two sub-systems' behaviour, on the other hand, makes it possible to formulate an algorithm that meets the time-varying requirements of the system. A proper formulation, design and subsequent implementation of such an algorithm, as a self-tuning adaptive controller, will result a secondary source that cancels the unwanted noise to a desired level over a broad frequency band of the noise. The cancellation achieved with such a controller is characterised by a flat-shaped cancellation as function of frequency ranging from almost the lowest to the maximum noise frequency and, moreover, a significant increase in the amount of cancellation as compared with that achieved with a constant gain controller or a controller with fixed transfer function. Such a significant increase in the amount of broadband cancellation and the ability of the algorithm to track changes in the system and adjust the controller accordingly makes the self-tuning active noise control algorithm practically desirable and superior over an ANC system that employs a controller with fixed characteristics, and proves to be a major step forward in eliminating the numerous practical problems due to low frequency acoustic noise.

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