Active Noise Control Systems with the TMS320 Family

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Abstract:

Active noise control (ANC) is achieved by introducing a secondary "antinoise" through an appropriate array of secondary sources. ANC has application to a wide variety of problems in manufacturing, industrial operations, and consumer products. In this paper, DSP algorithms for broadband feedforward, narrowband feedforward, and adaptive feedback control are implemented on TMS320C25, TMS320C3x, and TMS320C5x for real-time applications.

1. Introduction

Acoustic noise problems become more serious as increased numbers of industrial equipment such as engines, blowers, fans, transformers, and compressors are in use. The traditional passive silencers are valued for their high attenuation over a broad frequency range; however, they are relatively large, costly, and ineffective at low frequencies. Active noise control [1-3] involves an electroacoustic or electromechanical system that cancels the primary (unwanted) noise based on the principle of superposition; specifically, an antinoise of equal amplitude and opposite phase is generated and combined with the primary noise, thus resulting in the cancellation of both noises. The ANC system efficiently attenuates low frequency noise where passive methods are either ineffective or tend to be very expensive or bulky. ANC is developing rapidly because it permits improvements in noise control, often with potential benefits in size, weight, volume, and cost.

Since the characteristics of the acoustic noise source and the environment are time-varying, the frequency content, amplitude, phase, and sound velocity of the undesired noise are nonstationary. An ANC system must therefore be adaptive in order to cope with these variations.

It is also desirable for the noise canceler to be digital, where signals from electroacoustic or electromechanical transducers are sampled and processed in real time using digital signal processing (DSP) systems. In the 1980s, development of DSP chips enabled low-cost implementation of powerful adaptive algorithms [4] and encouraged widespread development and application of ANC systems. The continuous progress of ANC involves the development of improved adaptive signal processing algorithms, transducers, and DSP hardware. More sophisticated algorithms allow faster convergence and greater noise attenuation and are more robust to interference. The development of improved DSP hardware allows these more sophisticated algorithms to be implemented in real time to improve system performance.

ANC is an attractive means to achieve large amounts of noise reduction in a small package, particularly at low frequencies. Many applications of ANC involving real and simulated experiments are introduced in [1]. Current applications for ANC include attenuation of unavoidable noise in the following end equipment:

Automotive. Including electronic mufflers for exhaust and induction systems, noise attenuation inside vehicle passenger compartments, active engine mounts, and so on.

Appliances. Including air-conditioning ducts, air conditioners, refrigerators, kitchen exhaust fans, washing machines, furnaces, dehumidifiers, lawn mowers, vacuum cleaners, headboards, room isolation, and so on.

Industrial. Fans, air ducts, chimneys, transformers, power generators, blowers, compressors, pumps, chain saws, wind tunnels, noisy plants (at noise sources or many local quiet zones), public phone booths, office cubicle partitions, ear protectors, headphones, and so on.

Transportation. Airplanes, ships, boats, pleasure motorboats, helicopters, snowmobiles, motorcycles, diesel locomotives, and so on.

II. Broadband Feedforward Active Noise Control

Broadband feedforward ANC systems that have a single reference sensor, single secondary source, and single error sensor. These systems will be exemplified by the single-channel duct-acoustic ANC system, where the reference input signal $x(n)$ is picked up by a microphone. The reference signal is processed by the ANC system to generate the control signal $y(n)$ to drive a loudspeaker. The error microphone is used to monitor the performance of the

ANC system by sensing the residual error signal $e(n)$. The objective of the controller is to minimize the measured residual noise.

The use of the adaptive filter for ANC application is complicated by the fact that the summing junction represents acoustic superposition in the space from the canceling loudspeaker to the error microphone, where the primary noise is combined with the output of the adaptive filter. Therefore, it is necessary to compensate for the secondary-path transfer function $S(z)$ from $y(n)$ to $e(n)$, which includes the D/A (digital-to-analog) converter, reconstruction filter, power amplifier, loudspeaker, acoustic path from loudspeaker to error microphone, error microphone, preamplifier, antialiasing filter, and A/D (analog-to-digital) converter.

The introduction of the secondary-path transfer function into a controller using the standard LMS algorithm will generally cause instability. This is because the error signal is not correctly "aligned" in time with the reference signal, due to the presence of $S(z)$. There are a number of possible schemes that can be used to compensate for the effect of $S(z)$. The filtered-X LMS (FXLMS) algorithm, which places an identical filter in the reference signal path to the weight update of the LMS algorithm, is generally the most effective approach.

As illustrated in Figure 1, the secondary signal is generated as

$$y(n) = \mathbf{w}^T(n)\mathbf{x}(n), \tag{1}$$

where $\mathbf{w}(n) = [w_0(n)\ \ w_1(n)\ \ \ldots\ \ w_{L-1}(n)]^T$ and $\mathbf{x}(n) = [x(n)\ \ x(n-1)\ \ \ldots\ \ x(n-L+1)]^T$ are the coefficient and signal vectors of $W(z)$, respectively, and L is the filter order. The filter $W(z)$ must be of sufficient order to accurately model the response of the physical system. The adaptive filter minimizes the instantaneous squared error using the FXLMS algorithm

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \mu\,\mathbf{x}'(n)e(n), \tag{2}$$

where

$$\mathbf{x}'(n) = \hat{s}(n) * \mathbf{x}(n), \tag{3}$$

and $\hat{s}(n)$ is the estimated impulse response of the secondary-path filter, $\hat{S}(z)$.

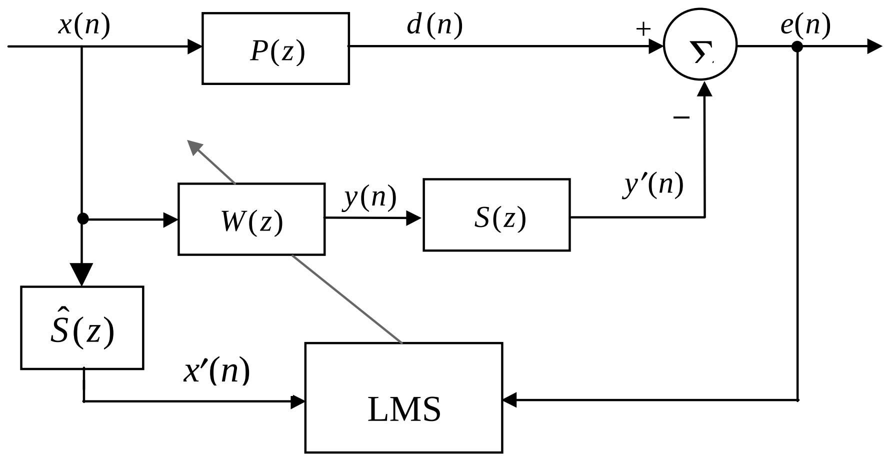

Figure 1 Block diagram of single-channel broadband feedforward ANC system.

The FXLMS algorithm for ANC applications is implemented on TMS320C25, TMS320C3x, and TMS320C5x. The assembly programs called "FXLMS.ASM" for the TMS320C25 and "FIRANC.ASM" for the TMS320C3x are available in the attached disk of book [1]. The FIRANC.ASM has options of using normalized and leaky FXLMS algorithms expresses as

$$\mathbf{w}(n+1) = v\,\mathbf{w}(n) + \mu\,\mathbf{x}'(n)e(n)\,, \tag{4}$$

where $v = 1 - \mu\gamma$ is the leakage factor and $0 < v < 1$. This leaky FXLMS algorithm also reduces numeric error in the finite-precision implementation. The introduction of a leakage factor has a considerable stabilizing effect on the adaptive algorithm, especially when very large source strengths are used.

The acoustic ANC system uses a reference microphone to pick up the reference noise and processes this input with an adaptive filter to generate an antisound $y(n)$ to cancel primary noise acoustically in the duct. Unfortunately, the antisound output to the loudspeaker also radiates upstream to the reference microphone, resulting in a corrupted reference signal $x(n)$. The coupling of the acoustic wave from the canceling loudspeaker to the reference microphone is called *acoustic feedback*.

The simplest approach to solving the feedback problem is to use a separate feedback cancellation, or "neutralization," filter within the controller, which is exactly the same technique as used in acoustic echo cancellation. This electrical model of the feedback path is driven by the secondary signal, and its output is subtracted from the reference sensor signal. The feedback component of the reference microphone signal is canceled electronically using a feedback neutralization filter $\hat{F}(z)$, which models the feedback path $F(z)$. Thus, the input signal $x(n)$ is computed as

$$x(n) = u(n) - \sum_{m=0}^{M-1} \hat{f}_m y(n-m-1), \tag{5}$$

where $u(n)$ is the signal picked up by the reference microphone and $\hat{f}_m$ are the coefficients of the Mth order of feedback neutralization filter $\hat{F}(z)$.

Since the primary noise is highly correlated with the antinoise, the adaptation of the feedback neutralization filter must be inhibited when the ANC system is in operation, similar to adaptive echo cancellation during periods of double-talk. Thus, feedback neutralization is achieved, in effect, by using an off-line adaptive method for determining the transfer function of the feedback path. The models $\hat{S}(z)$ and $\hat{F}(z)$ can be estimated simultaneously by using the off-line modeling technique [1].

The FXLMS algorithm with feedback neutralization for ANC applications is implemented on TMS320C25, TMS320C3x, and TMS320C5x. The assembly programs called "FXLMS-FN.ASM" for the TMS320C25 and "FIR-FN.ASM" for the TMS320C3x are available in the attached disk of book [1].

The output signal of the IIR filter $y(n)$ is computed as

$$y(n) = \mathbf{a}^T(n)\mathbf{x}(n) + \mathbf{b}^T(n)\mathbf{y}(n-1), \tag{6}$$

where $\mathbf{a}(n) \equiv [a_0(n) \;\; a_1(n) \;\; \ldots \;\; a_{L-1}(n)]^T$ is the weight vector of $A(z)$, $\mathbf{x}(n)$ is the reference signal vector, $\mathbf{b}(n) \equiv [b_1(n) \;\; b_2(n) \;\; \ldots \;\; b_M(n)]^T$ is the weight vector of $B(z)$, and $\mathbf{y}(n-1)$ is the output signal vector delayed by one sample. Many algorithms can be employed to find the optimal set of coefficients a_l and b_m to minimize the error signal $e(n)$. The "filtered-U recursive LMS algorithm" for ANC is derived as [1]

$$\mathbf{a}(n+1) = \mathbf{a}(n) + \mu\, \mathbf{x}'(n)e(n) \tag{7}$$

$$\mathbf{b}(n+1) = \mathbf{b}(n) + \mu\, \mathbf{y}'(n-1)e(n)\,, \tag{8}$$

where $\mathbf{y}'(n-1) = \hat{s}(n) * \mathbf{y}(n-1)$ is the filtered version of the canceling signal vector at time $n-1$.

The filtered-U recursive LMS algorithm for ANC applications is implemented on TMS320C25, TMS320C3x, and TMS320C5x. The assembly programs called "FURLMS.ASM" for the TMS320C25 and "IIRANC.ASM" for the TMS320C3x are available in the attached disk of book [1].

III. Narrowband Feedforward Active Noise Control

Many noises are periodic, such as those generated by engines, compressors, motors, fans, and propellers. Direct observation of the mechanical motion of such sources is generally possible by using an appropriate sensor, which provides an electrical reference signal that contains the fundamental frequency and all the harmonics of the primary noise. However, this technique is only effective for periodic noise because the fundamental driving frequency is the only reference information available.

A basic block diagram of narrowband ANC for reducing periodic acoustic noise in a duct is illustrated in Fig. 2. This system controls harmonic sources by adaptively filtering a synthesized reference signal $x(n)$ internally generated by the ANC system. This technique has the following advantages: (1) undesired acoustic feedback from the canceling loudspeaker back to the reference microphone is avoided, (2) nonlinearities and aging problems associated with the reference microphone are avoided, (3) the periodicity of the noise removes the causality constraint, (4) the use of an internally generated reference signal results in the ability to control each harmonic independently, and (5) it is only necessary to model the acoustic plant transfer function over frequencies in the vicinity of the harmonic tones; thus, an FIR filter with substantially lower order may be used.

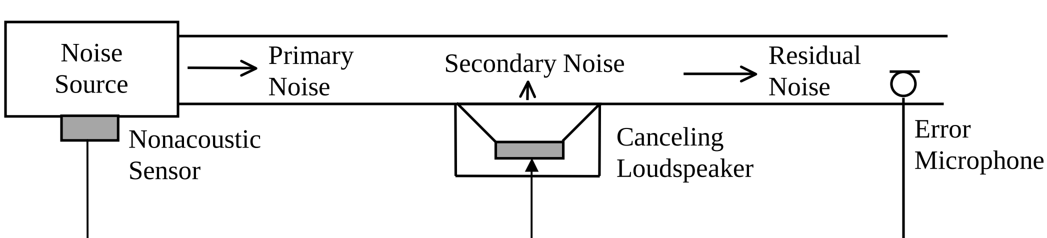

Figure 2 Narrowband ANC system with the FXLMS algorithm for acoustic duct application.

The reference signal generator is triggered by a synchronization pulse from a nonacoustic sensor, such as a tachometer signal from an automotive engine. In general, two types of reference signals are commonly used in narrowband ANC systems: (1) an impulse train with a period equal to the inverse of the fundamental frequency of the periodic noise and (2) sinewaves that have the same frequencies as the corresponding harmonic tones to be canceled. The first technique is called the waveform synthesis method. The second technique embodies the adaptive notch filter, which was originally developed for the cancellation of tonal interference and applied to periodic ANC.

A single-channel narrowband feedforward ANC system using the adaptive notch filter with the leaky FXLMS algorithm is implemented on the TMS320C25. The assembly programs called “ANFANC.ASM” for the TMS320C25 is available in the attached disk of book [1].

IV. Feedback Active Noise Control

The basic idea of an adaptive feedback ANC is to estimate the primary noise and use it as a reference signal $x(n)$ for the ANC filter. In Fig. 1, the primary noise is expressed in z-domain as $D(z) = E(z) + S(z)Y(z)$ where $E(z)$ is the signal obtained from the error sensor and $Y(z)$ is the secondary signal generated by the adaptive filter. If $\hat{S}(z) \approx S(z)$, we can estimate the primary noise $d(n)$ and use this as a synthesized reference signal $x(n)$. That is,

$$X(z) \equiv \hat{D}(z) = E(z) + \hat{S}(z)Y(z). \tag{9}$$

This is the reference signal synthesis (regeneration) technique, whereby the secondary signal $y(n)$ is filtered by the secondary-path estimate $\hat{S}(z)$ and then combined with $e(n)$ to regenerate the primary noise.

The complete single-channel adaptive feedback ANC system using the FXLMS algorithm is illustrated in Fig. 3, where $\hat{S}(z)$ is also required to compensate for the secondary path. The reference signal $x(n)$ is synthesized as

$$x(n) \equiv \hat{d}(n) = e(n) + \sum_{m=0}^{M-1} \hat{s}_m y(n-m), \tag{10}$$

where $\hat{s}_m$, $m = 0, 1, \ldots, M-1$ are the coefficients of the Mth order FIR filter $\hat{S}(z)$ used to estimate the secondary path.

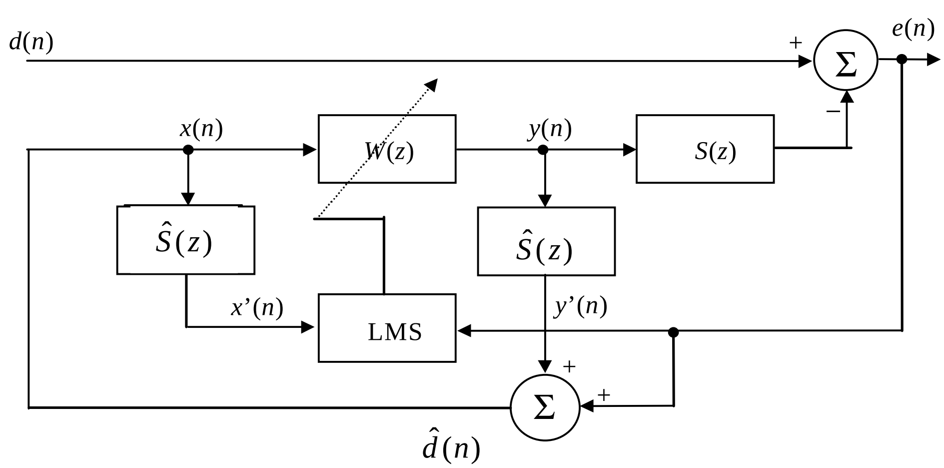

Figure 3 Adaptive feedback ANC using the FXLMS algorithm

The single-channel adaptive feedback ANC system using the FXLMS algorithm is implemented on TMS320C25, TMS320C3x, and TMS320C5x. The assembly programs called

"AFANC.ASM" for the TMS320C25 and "AFANC.ASM" for the TMS320C3x are available in the attached disk of book [1].

V. Multiple-Channel ANC

Since the noise field in an enclosure or a large-dimension duct is more complicated than in a narrow duct, it is generally necessary to use a multiple-channel ANC system with several secondary sources, error sensors, and perhaps even several reference sensors. Some of the best-known applications are the control of exhaust "boom" noise in automobiles, earth-moving machines, and the control of propeller-induced noise in flight cabin interiors. Other ANC applications such as vibration control in complex mechanical structures also require multiple channels.

Detailed multiple-channel ANC algorithms are presented in [1]. Multiple-channel ANC system (1x2x2 example) using the FXLMS algorithms with optional leakage and/or normalization is implemented on TMS320C3x. The assembly program called "MC-FIR.ASM" is available in the attached disk of book [1]. The assembly program called "MCFIR-FN.ASM" in the attached disk of book [1] with feedback neutralization on TMS320C3x. In addition, MC-FANC.ASM is the TMS320C3x assembly program that implements multiple-channel adaptive feedback ANC system (2x1 example) using the FXLMS algorithm with optional leakage and/or normalization.

References:

[1] S. M. Kuo and D. R. Morgan, *Active Noise Control Systems - Algorithms and DSP Implementations,* New York: Wiley, 1996.

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