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Adaptive Feedback ANC using Multirate Audio-Integrated system

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Abstract

Conventional audio-integrated feedback active noise control (ANC) systems use single rate to process low rate noise signal and high rate audio signal, which degrades the performances of both ANC and audio systems. This paper presents multirate audio-integrated feedback ANC system to admit the different sampling frequency in both ANC and audio system based on interpolation and decimation techniques. The improved performance is verified by real-time experimental results implemented on the audio-integrated ANC headrest system.

Keywords: Digital-to-Analog Converter; Interpolator; Decimator; Algorithm; Audio-integrated feedback

Introduction

In recent years, consumer electronic markets grow enormously because of the rapid development of smart, mobile, and wearable devices. As part of these, audio systems play an important role to deliver good quality of sound for smartphones, headsets, healthcare devices, and automobile electronics To improve the quality of sound in a noisy environment, these audio systems can be integrated with ANC systems to cancel unwanted noise, such as ANC headsets, ANC pillows, ANC helmets, ANC hearing aids, ANC infant incubators, and ANC for automotive audio. By sharing analog components such as mixers, amplifiers, microphones, and loudspeakers, the audio-integrated ANC systems process the high frequency audio signals mixed with the low frequency noise signals without expense of system redundancies. There are two kinds of techniques to mix the high frequency audio signals and the low frequency noise signals, digital-mixing that mixes signal in digital domain and analog-mixing that mixes signals in analog domain. In applications, the audio source is only presented in digital form. Thus, digital-mixing can be used. Analog-mixing can be used as well with additional digital-to-analog converter (DAC).

In digital-mixing technique, the conventional audio- integrated feedback ANC systems use single sampling frequency to mix and deal with both the audio and the noise signals in digital domain. In practical digital applications, typical sampling frequency required by audio signals is much higher compared to the sampling frequency for the noise signals in an audio-integrated ANC system. If the ANC function is forced to work on high sampling frequency, the noise reduction performance will be degraded because of the wide frequency range and the dynamic range. On the other hand, if the system works on low frequency, the performance is not satisfactory for audio signals. Therefore, the use of single sampling frequency (single rate) in digital-mixing technique for audio-integrated feedback ANC system is not applicable.

This paper develops the multirate audio-integrated feedback ANC systems to admit both low sampling frequency for the ANC function and high sampling frequency for the audio function based on interpolation and decimation techniques.

The rest of paper is organized as follows. Section II briefly introduces the conventional audio-integrated feedback ANC system. Section III presents the proposed multirate audio- integrayed feedback ANC system. Section IV presents the real-time experiment using ANC headrest system to demonstrate the satisfactory performance. The evaluation and summary are provided in Section V.

II. Conventional audio-integrated feedback anc system

The conventional audio-integrated feedback ANC system as is shown in (Figure 1). Both ANC and audio subsystems works on single sampling frequency fc.

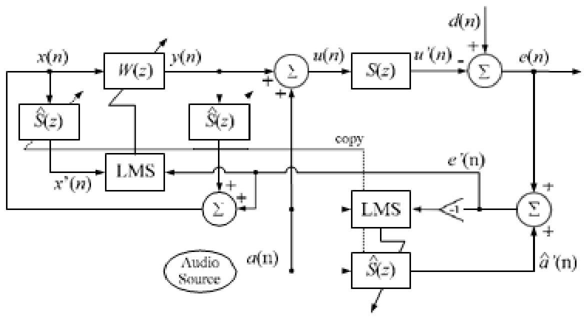

Figure 1: Conventional audio-integrated feedback ANC system.

The adaptive filter W(z) generates anti-noise y(n) using synthesized reference signal x(n) as expressed as

$$y(n)=\sum L-1\ w\,(n)x(n-l)=wT(n)x(n), \quad (1)$$

where wl(n) is the l*th* coefficient of the adaptive finite impulse response (FIR) filter W(z) of length L at time n; w(n)=[wO(n) w1(n) ⋯ wL−1(n)]T ; x(n) is the estimated reference signal at time n; x(n)=[x(n) x(n−1) ⋯ x(n−L+1)]T.

The audio signal a(n) provided by digital audio source is mixed with anti-noise y(n) using summing operation, resulting in driving signal u(n) given by

$$u(n)=y(n)+a(n). \quad (2)$$

After passing the secondary path S(z) , which includes DAC, reconstruction filter, power amplifier, secondary loudspeaker, acoustic-path, error microphone, preamplifier, anti-aliasing filter, and analog-to-digital converter (ADC), the system obtains the error signal given by

$$e(n)=d(n)-u'(n)=d(n)-y'(n)-a'(n), \quad (3)$$

which composes of unwanted noise d(n) ; the anti-noise component y′(n), and audio component a′(n). The residual noise component (d(n) −y′(n)) is necessary for ANC function, but the audio component a′(n) becomes an interference to the FXLMS adaptation. Therefore, the audio cancellation process is necessary to remove the unwanted audio

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component a′(n) from the error signal e(n) . This process applies an audio cancellation filter S^(z) to estimate a′(n) using the digital audio signal a(n) as the reference signal, given by

^′(n)=∑Lc–1 ŝS (n)a(n−l)=s^T(n)a(n),   (4)

where ŝS (n) is the l th coefficient oTf S^(z) of length Lc at time

n,   s^(n) = [s (n) s (n) ··· s   (n)]

O   1   Lc–1;   and

a(n) =[a(n) a(n−1) ··· a(n−Lc+1)]T.

The coefficients of S^(z) are updated using LMS algorithm given by

s^(n+1)=s^(n)+μ1 a(n)e′(n),   (5).

where μ1 is the step size for this adaptation. The adaptation of audio cancellation filter S^(z) is also functioning as online secondary-path modeling required by FXLMS algorithm. Then, the unwanted audio component a′(n) is subtracted from the error signal e(n) as expresses as

eu(n)=e(n)+a^u(n).   (6)

Substituting (3) into (6) results in

e′(n) = d(n) − [yu(n) + au(n)] + a^u(n).   (7)

After convergence has been achieved such that a^u(n) ≅ a′(n), the system obtains

e′(n)=d(n)−yu(n),   (8)

as the true error signal required for the FXLMS algorithm and reference signal estimation in the ANC function. Then, the estimated reference signal x(n) can be obtained by

x(n)=eu(n)+∑Lc–1 ŝ S(n)y(n−l)

= eu(n)+s^T(n)y(n).   (9)

The coefficients of adaptive ANC filter W(z) with length L is updated by FXLMS algorithm given by

w(n+1)=w(n)+μ2x′(n)eu(n),   (10)

where μ2 is the step size for this adaptation; x′(n)=[xu(n) xu(n−1) ··· x′(n−L+1)]T is the filtered reference signal composed of x(n) ∗ ŝ (n).

Iii. Multi-rate audio-integrated feedback anc systems

In this section, the multirate processing is proposed for audio-integrated feedback ANC system, as shown in (Figure 2). The highlight blue color emphasizes the added part to distinguish with the conventional system. The proposed multirate processing using interpolation and decimation technique solves the problem of different sampling frequency required by ANC and audio systems. The notation “k” is used for the time index at low sampling frequency fanc, and notation “n” is used for the time index at high sampling frequency fs.

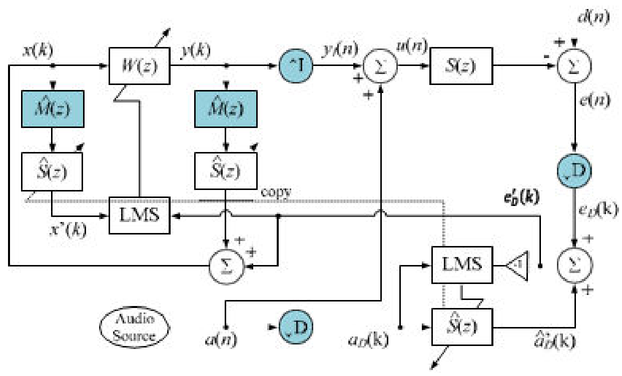

Figure 2: Multirate audio-integrated feedback ANC system.

The ANC filter W(z) uses the reference signal x(k) to generate the anti-noise signal y(k) in sampling frequency fanc, as expresses as

y(k) = ∑L–1 wS(k)x(k − l) = wT(k)x(k),   (11)

where wS(k) is the lth coefficient of the adaptive FIR ANC filter W(z) of length L at time k; w(k) = [wO(k) w1(k) ··· wL–1(k)]T; x(k) is the estimated reference signal at time n; x(k) = [x(k) x(k−1) ··· x(k−L +1)]T.

On the other hand, the digital audio signal a(n) is received either from digital audio source or converted from analog audio source on high sampling frequency fs. To mix/sum both signals in digital domain, the low-rate anti-noise signal y(k) is converted to the high-rate anti-noise signal yI(k) with sampling frequency fs using the interpolation technique by factor I=fc⁄fanc. The interpolator increases the sampling frequency of y(k) by inserting (I−1) zero samples between each adjacent pair of the real samples according to the relation

y(n⁄I),   n= 0, ± I, ± 2I,…

yI(n) ={   0,   otherwise   (12)

where I is the interpolation factor and n is the time index of the interpolated signal yI(n), and then filtering it by a lowpass filter with the cutoff frequency fc given as

fc = fanc/2.   (13)

Finally, the high-rate anti-noise signal yI(n) can be mixed with audio signal a(n) , producing single driving signal u(n) as expressed as

u(n)=yI(n)+a(n).   (14)

We obtain error signal e(n) after passing through the secondary-path S(z), which generally includes reconstruction filter, power amplifier, secondary loudspeaker, acoustic-path, used in ANC adaptation process, the high-rate error signal e(n) is required to be converted to the low-rate signal with sampling frequency fanc using the decimation technique. Decimation of high-rate signal with sampling frequency fs by the factor D=fc⁄fanc results in the lower rate sampling frequency fanc=fs/D. The decimator output can be expressed as

eD(k) = e(kD),   (15)

where k is used for the decimated sequences eD(k). Aliasing will occur if the original high-rate signal has frequency components outside the new narrower bandwidth. Therefore, the lowpass filtering of the high-rate error signal e(n) prior to the decimation process is required to solve the aliasing problem. The cutoff frequency of the lowpass filter is given as equation (13).

In fact, the low-rate error signal eD(k) contains the residual error signal and desired audio signal as expressed as

eD(k)=dD(k)−uD(k)

= dD(k)−[yu (k)+au (k)], (16)

where dD(k) is the low-rate unwanted noise component in eD(k) ; yu (k) is the low-rate anti-noise signal y(n) after passing through secondary-path S(z) , decimator and interpolator including the required digital lowpass filter; au (k) is the low rate undesired audio signal component in e (k)

As shown in equation (16), the low-rate error signal eD(k)

contains the undesired audio signal au (k) that becomes interference to the ANC adaptation process. The adaptive FIR filter S^(z) is proposed to cancel the audio interference au D(k).

The audio canceller filter S^(z) uses the low-rate audio signal aD(k) as the reference signal to generate the estimate of au (Dk) as expressed as

D a^u (k) = ∑Lc–1 ŝS (k)aD(k–l) = ^sT(k)aD(k), (17)

where ŝ S(k) is the l th coefficient of the adaptive FIR filter

S^(z) of length Lc at time k,

s^(k) =[ŝ O(k) ŝ 1(k) ··· ŝ L–1(k)]T ; and

aD(k)= [a(k) a(k−1) ··· a(k−Lc+1)]T.

The adaptation process of S^(z) uses LMS algorithm as expressed as

s^(k+1)=s^(k)+µ1aD(k)eu (k), D (18)

where µ1 is the step size for this adaptation. Assuming that the audio signal a(n) is of persistent excitation and uncorrelated with the primary noise d(n), then the LMS algorithm can converge to the optimum filter s^o if and only if

0< µ < 2 , (19)

3OOLcPa(n)

where Pa(n) is the power of aD(n).

The audio canceller filter S^(z) adaptively estimates and removes the audio components au (k) in eD(k), producing a true error signal e D k as expressed as

eu (k) = eD(k)+a^u (k), (20)

Substituting (16) to (20), we obtain

eu (k) = dD(k)−[yu (k)+au (k)]+a^u (k) (21)

D D D D

Then, after convergence is achieved such that a^u (k) ≅ au (k), we can obtain

eu (k) = dD(k)−yu (k) (22)

As the residual error of primary noise dD(k) cancelled by anti-noise yu (k), which is the true error signal required for the FXLMS algorithm in the ANC part.

Both decimators' transfer function are identical and referred as Dc (z). Since both transfer function Dc (z)H^(z) and

H(s)Dc(z) are excited by ta^hu e(ks)ame usi(gn)al a(t),^ and generate

the same output such that ≅ a k , then S(z) is clearly

the estimate of S(z). During ANC online operation, S(z) is highly possible to change, and the audio canceller filter S^(z) which is also the estimate S(z) can adaptively track the change. Therefore, the proposed multirate audio-integrated feedback ANC system using the multirate digital-mixing technique has the capability of online secondary-path modeling.

In the multirate audio-integrated ANC system, the anti- noise signal passes not only the secondary-path S(z), but also passes interpolator and decimator including the two identical lowpass filter. Therefore, the fixed FIR filter M^ (z) as the estimate of the interpolator and decimator including the two lowpass filter model is required in the FXLMS algorithm. We refer this as multirate transfer function as M(z).

The fixed FIR filter M^ (z) is offline estimated as shown in (Figure 3) Both M(z) and M^(z) are excited by the white noise signal v(k). The error signal eN(k) can be obtained as

eN(k) = vu(k) + v ^ u(k), (20) where vu(k) and v^ u(k) are the output of M(z) and M ^ (z) , respectively. The adaptation process of M^ (z) uses LMS algorithm as expressed as

m ^ (k+1) = m ^ (k) + µ2 v(k)eN(k), (21)

where m ^ (k) = [N ^ (k) N ^ (k) ··· N^ (k)]T ; N ^ (k) is the N

l th coefficient of the adaptive FIR filter M^(z) of length LN at time k, and v(k)=[v(k) v(k−1) ··· v(k−LN+1)]T , µ2 is the step size for this adaptation. After convergence is achieved, the coefficients of M ^ (z) is fixed and copied to the online multirate audio-integrated feedback ANC system.

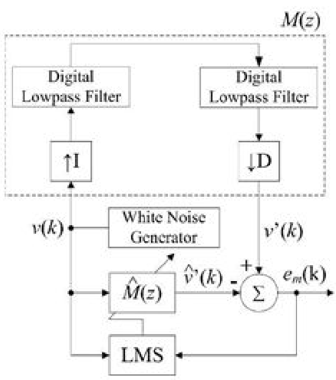

Figure 3: Multirate transfer function estimation.

The coefficients of ANC filter W(z) with length L are updated using FXLMS algorithm, as expressed as

w(k+1)=w(k)+µ3x′(k)eu (k),D (22)

where µ3 is the step size for this adaptation; x′(k)=[xu(k) xu(k−1) ··· x′(k−L+1)]T is the input signal x(k) filtered by series of S ^ (z) and M ^ (z).

IV. Experiment results and analysis

Real-time experiments were conducted to evaluate the performance proposed multirate audio-integrated feedback ANC system implemented on audio headrest system [10] shown in Fig. 4. The experimental setup was built which consisting of two T1-1828S rectangular loudspeakers with dimensions of 13.5 x 5.5 x 2.2 cm to serve as the secondary loudspeakers. Two Shure MX183 microphones were used as the error microphones. The two error microphones and secondary loudspeakers are summed as one in analog domain, thus single channel multirate audio-integrated feedback ANC system like Figure 2 can be applied. The anti-aliasing and reconstruction filter were built from 8th-order switched- capacitor filter with cut-off frequency set at 750 Hz. A floating- point digital signal processor, TMS320C6713, was used to conduct the real-time experiments. The noise was played by a primary loudspeaker located at 1 m from the back of the head. The ANC headrest system was located in a laboratory without special acoustic treatment. A KEMAR was put as a human model to simulate a real acoustical situation.

Figure 4: Experimental Setup.

The unwanted noise is multiple sinusoids with frequencies 200, 300, 400, 500, and 600 Hz. In order to clearly evaluate the ANC performance, all experiment results display only the frequency range up to 1000 Hz. The audio signal comes from analog source which is converted to digital signal with sampling frequency 32 kHz, while the ANC system worked at sampling frequency of 2 kHz. Therefore, the sampling frequency conversion factors for decimator D and interpolator I are 16.

First, we used the multiple sinusoids with frequencies 250, 350, 450, and 550 as the desired audio signal to show the effectiveness of the audio interference cancelation algorithm. Figure 5 shows the noise reduction with the audio interference cancellation, achieved at the error microphones. The dashed-red line is error signal when ANC is off, and the solid-blue line is when the ANC is on. the This figure shows that the proposed multi-rate processing audio-integrated ANC system only successfully reduce the unwanted noise at 200, 300, 400, 500, and 600 Hz, and the desired audio remains unreduced.

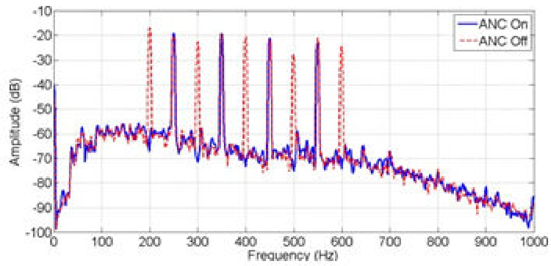

Figure 5: Magnitude spectra showing noise reduction at error microphones.

In second experiment, an example of real piece of music, which is continuously changing, was used as the practical audio signal. The spectrogram in (Figure 6) shows the residual noise signal recorded from the error microphones. The darker red color lines at 200, 300, 400, 500, 600 Hz represent the unwanted noise. In the beginning, the system worked without playing music. When the ANC was on, the color of lines at 200, 300, 400, 500, 600 Hz is faded, which means that the unwanted noise was effectively reduced. At about 10.5 second, the music was played. As can be seen, there are more darker red color which represents the audio component. When the ANC was on, the color of lines at 200, 300, 400, 500, 600 Hz is also faded but the color of audio component is not. It means that the unwanted noise was effectively reduced while the desired audio was maintained.

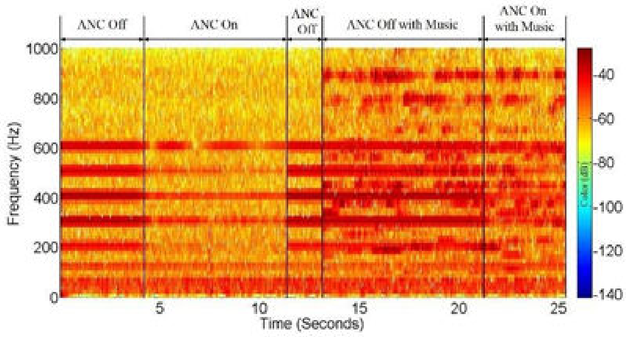

Figure 5: Spectrogram of error signal measured at error microphones.

Conclusion

This paper emphasizes that the proposed multirate audio- integrated feedback ANC system can effectively work with low sampling frequency for the ANC function and high sampling frequency for the audio function. The multirate transfer function should be included in the FXLMS algorithm.

The dielectric properties for Fe/TiO2 and Ni-Fe/TiO2 composite as a function of temperature and frequency were studied. The values of dielectric constant and dielectric loss decrease with increasing frequency. The values of dielectric permittivity for Fe/TiO2 exhibit relatively lower dielectric constant than Ni-Fe/TiO2. A relaxation peak has been recognized and shifted to higher frequency with increasing temperature. The ac conductivity increased with frequency which could be related to the hopping conduction process. The values of activation energy Ea for Fe/TiO2 are higher than Ni-Fe/TiO2 and decreased with increasing frequency. The high dielectric properties and low activation energy may reflect the importance of Ni-Fe/TiO_2 in dielectric applications.

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