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Feedback Active Noise Control Based Voice Enhancing Ear-Protection System

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Abstract – This paper proposes a voice enhancing ear-protection system which is based on feedback active noise control(FBANC). The proposed system selectively suppresses the background noise and preserves the talking voice by controlling the adaptive algorithm with the voice activity period detection module. The noise reduction performance of the proposed noise canceling algorithm is analytically derived for the two key performance affecting parameters, i.e., electro-acoustic coupling distance and noise bandwidth. The proposed system is also implemented with a floating-point DSP system and its performance is experimentally tested to compare with the analytically derived results. The achieved levels of noise reduction for the three different noise bandwidths cases, i.e., 10Hz, 50Hz, and 90Hz, are high to show 17.05dB, 10.54dB and 8.99dB, respectively. The feasibility of the proposed system is also shown by the peak noise reduction achieved more than 25dB while preserving the voice component in the frequency range between 200-800Hz.

Keywords: Feedback ANC, Active noise control, Voice enhancement, Ear-protection, Earmuffs

1. Introduction

Passive earmuffs are widely used in noisy environment such as industry plants, construction site and manufacturing factories. Simultaneous suppression of talking voice of other person is one of the big troubles of adopting the passive earmuffs.[1-3]

In this paper, an active-type ear-protection system is proposed, where background noise is selectively suppressed while talking voices are preserved as much by applying the feedback active noise control (ANC) algorithm. In contrast to passive noise suppress techniques such as soundproof walls or earplugs, the ANC technique is to suppress an external noise in an active way. Where, the external noise is cancelled by combining it with an anti-noise having opposite phase and same magnitude [4-6].

Many commercial ANC headsets applications are developed based on the feedforward ANC scheme. The feedforward ANC scheme utilizes the reference of the noise directly captured from the noise source, which is highly correlated with the noise. However, the feedforward ANC systems for headsets have significant stability and performance deficiencies caused by nonstationary reference inputs, measurement noise, and acoustic feedback.[7-9] The feedforward scheme can also give inconvenience to workers since the reference microphone connected with a cumbersome wire limits the ANC users' mobility.

The feedback ANC (FBANC) scheme can be more proper for headset-type ANC applications because of their convenience that a reference microphone is not required at noise source. Unlike the feedforward ANC, the adaptive feedback active noise control(FBANC) provides a more accurate noise cancelation since the microphone is placed inside the ear-cup of the headset [4-7]. Since FBANC is not provided separate reference measurement, FBANC algorithm acts as a linear predictor to generate the anti-noise sound with previous residue measurement. Therefore FBANC system can reduce only predictable and narrow band noise and the performance of FBANC system highly depends on length of internal system delay and noise bandwidth [9]. For an extreme example, white noise cannot be reduced by FBANC scheme.

The proposed voice enhancing ear-protection system is a headphones based structure where the feedback ANC generates headphones speaker sound to acoustically cancel the undesired background noise. The conventional feedback ANC system tries to tune the adaptive filter to suppress the voices together with the background noise and cancel not only the external noise but also every incomming sound including human voice which is necessary for communication with other workers. Such FBANC scheme is not suitable for the workers in noisy environments because the poor speech recognition among the ANC users causes inconvenience in their conversation.

The proposed ear-protection system is intended to selectively cancel the background noise by controlling the update of the adaptive filter in accordance with voice activity. The adaptive filter stays tuned to cancel only the background noise, being not disturbed by the occurrence of voices. In order to classify the speech/non-speech periods, a voice activity detection (VAD) module is embedded in the FBANC algorithm. During the last decade, numerous

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different strategies for detecting speech on a noisy signal have been developed and the influence of the VAD effectiveness on the performance of speech processing systems have been evaluated [10].

Design of the feedback ANC based applications requires comprehensive understanding of the effect of physical parameters including the two key effecting parameters, i.e., electro-acoustic coupling and noise bandwidth characteristics.[7-9] The typical target noises of the ANC application are narrowband noises produced by rotating machines such as engines used in transportation vehicles, turbines in auxiliary power sources, compressors in refrigerators, and vacuum pumps [11-12]. Such noises can be modeled with the 2nd order AR process model having center frequency and -3dB bandwidth. In [9], the author analyzed and derived the noise reduction performance of feedback ANC system for the 2nd order autoregressive noise model.

Performance of the proposed ear-protection system is analytically derived to examine the effect of the two key design parameters and to provide an optimized design guide. To show the feasibility of the proposed ear-protection system, an experimental system is implemented and its performance is evaluated.

2. Feedback Active Noise Control Based Voice Enhancing Ear-protection System

The principle of the proposed ear-protection system is illustrated in the Fig. 1, where the feedback ANC algorithm produces the anti-noise sound through the headphone - speaker and cancels the undesired background noise (primary noise) coming into the ear-cup.

Selective suppression of the background noise is necessary for enhancement of voices. The voice enhancing ear-protection system's block diagram is illustrated in Fig. 2. When voices are added to the background noise, the adaptive filter tries to tune itself to suppress the voices together with the background noise. By stopping the update process of the adaptive algorithm during the voice activity

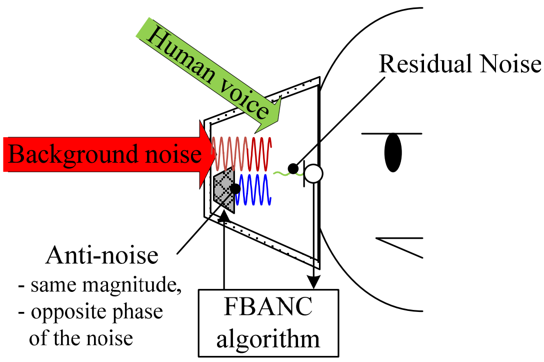


Fig. 1. Principle of the proposed voice enhancing ear-protection system

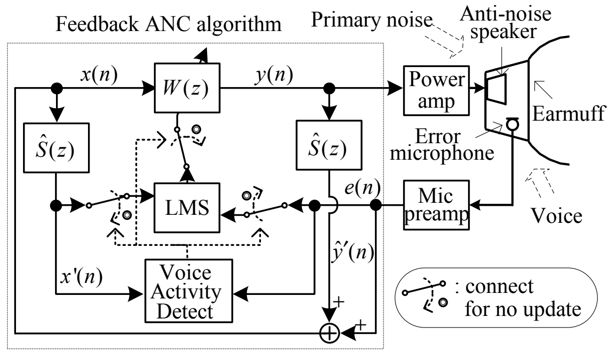


Fig. 2. The block diagram of the proposed FBANC based ear-protection system

period as detected by a separate voice detection module, the adaptive filter can stay tuned to cancel only the background noise, being not disturbed by the occurrence of voices. To avoid canceling of human voice, a voice activity detection (VAD) module detects the voice activity from the reference signal $x(n)$ then switches the adaptation mode into noise tracking mode or voice enhancing mode. When voice activity is detected, in the voice enhancing mode, the VAD module controls the FBANC algorithm to stop update the filter coefficients.

In the feedback ANC algorithm [4-5], the reference signal $x(n)$ is internally generated through the feedback path based on the error signal $e(n)$ and the filtered anti-noise signal $y'(n)$ as

$$x(n) = \hat{y}'(n) + e(n) = \hat{s}(n) * y(n) + e(n). \tag{1}$$

The residual sound $e(n)$ is sensed by the error microphone. The electro-acoustic coupling path $S(z)$ including the acoustic space, loudspeaker and microphone system, A/D and D/A converters and amplifiers is modeled as a transfer function model $\hat{S}(z)$, which is called the secondary path estimation. The secondary path and its model can be expressed as combination of pure Δ delay component and polynomial component $A(z)$ as $S(z) = z^{-\Delta}A(z)$ and $\hat{S}(z) = z^{-\Delta}\hat{A}(z)$.

The adaptive filter $W(z)$ is updated by the normalized least-mean-square(LMS) algorithm based on the error signal $e(n)$ and filtered reference signal $x'(n)$ [13]. The selective filter update algorithm is described in the following.

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \frac{\mu}{|\mathbf{x}'|^2}\mathbf{x}'e(n),\ \mu = \begin{cases} 0 & \text{for } V(n) > V_{th} \\ \mu' & \text{for } V(n) < V_{th} \end{cases} \tag{2}$$

where $\mathbf{w}(n)$ and $\mathbf{w}(n+1)$ are the adaptive filter vectors of the present and next iteration, $\mathbf{x}(n)$ is the reference signal vector, μ and μ' are the step-size and $V(n)$ and V_{th} are the current voice activity level and voice activity threshold level. The algorithm stops the update process of the adaptive algorithm during the voice activity period, $V(n) > V_{th}$.

The primary noise $d(n)$ can be modeled as $d(n) = n(n) + v(n)$ where $n(n)$ is the background noise and $v(n)$ is the voice component. By the selective filter update algorithm (2), the adaptive filter $W(z)$ is tuned only by the statistical property of the background noise $n(n)$. In [14], the solution of the optimum adaptive filter $W_{opt}(z)$ was derived as

$$W_{opt}(z) = \frac{[z^{\Delta}G(z)]_{+} / G(z)}{A(z)} \tag{3}$$

where $G(z)$ is the spectral factorization of noise spectral density $\mathbb{N}(z) = E[|N(z)|^2] = G(z)G(z^{-1})$. $N(z)$ is the z-transform of noise $n(n)$. The operation $[\ ]_{+}$ indicates that the causal part is taken from the inverse transform of the quantity in the square bracket. The numerator of (3), $[z^{\Delta}G(z)]_{+}$ indicates the spectral component of noise $n(n)$ except for the amount of the innovations that occur during the latest Δ-step interval. [9]. By taking z-transform of (1) and putting (3), $E(z)$, which is the z-transform of error signal $e(n)$, can be derived as

$$\begin{aligned} E(z) &= [1 - z^{-\Delta}[z^{\Delta}G(z)]_{+} / G(z)]D(z) \\ &= [1 - z^{-\Delta}[z^{\Delta}G(z)]_{+} / G(z)][N(z) + V(z)] \end{aligned} \tag{4}$$

where $V(z)$ is z-transform of voice $v(n)$. The mean-squared value of (4) can be derived as

$$\begin{aligned} E[|E(z)|^2] &= |G(z) - z^{-\Delta}[z^{\Delta}G(z)]_{+}|^2 \\ &\quad + |G(z) - z^{-\Delta}[z^{\Delta}G(z)]_{+}|^2 \left|\frac{V(z)}{G(z)}\right|^2 \end{aligned} \tag{5}$$

In (5), the first term of right side is the minimum mean-squared prediction error for background noise $n(n)$ which is for the innovations that occur during the latest Δ-step interval. The second term's $\left|\frac{V(z)}{G(z)}\right|^2$ shows the voice enhancement effect of the proposed algorithm. If the dominant frequency components of voice and noise occupy in different frequency ranges, the voice-dominant frequency range is emphasized, but the noise-dominant range is de-emphasized. The second term's left part $|G(z) - z^{-\Delta}[z^{\Delta}G(z)]_{+}|^2$, which is the minimum mean-squared prediction error of background noise, tends to become white noise when the secondary path delay Δ is very short. On the other hand, it is tends to be close to noise spectral $\mathbb{N}(z)$ when the delay is long enough to cover the coherence-time of correlation of noise [9]. The voice enhancement can be effective when the dominant frequency range of noise doesn't overlap with the voice-dominant range, and the delay Δ is shorter than the coherence-time of noise correlation.

In order to control the adaptive ANC algorithm to selectively track characteristics of undesired noise, a voice activity detection (VAD) module is embedded in the FBANC algorithm. The voice activity period is detected when the energy level exceeds the threshold level. The most formant component of human voice exists between 400 and 800Hz range and some consonant and sibilance components exist in 4000Hz to 7000Hz range. Typically, since the sound of human voice lasts for more than 150ms, when the voice component is detected, the proposed VAD algorithm turns the voice activity level $V(n)$ to 1 (one) immediately and when the voice component is disappeared, the voice activity level $V(n)$ is slowly decreased to 0 (zero) for 150ms. Therefore the filter update is not disturbed by the occurrence of voices even when the voice component is not strong enough to be detected at tale of voice formant sound. The value of voice activity threshold level V_{th} can set between 1 and 0. In the proposed algorithm, the threshold level V_{th} is set to 0.1.

The contribution of the proposed algorithm is that noise is canceled out in acoustic-domain by active noise control technique while voice component is enhanced by controlling the update process of the adaptive algorithm during the voice activity period. On the other hand, most other former noise reduction algorithms for voice enhancement such as spectral subtraction algorithm are intended to be applied in signal-domain. Which means they can extract voice signal by filtering out noise component from measured speech signal but cannot provide actual noise canceling in acoustic domain. Therefore, the proposed system can be applied to ear-protection system for workers in noisy environments.

3. Performance Analysis of the Ear-protection System

In this section, the noise reduction performance of the proposed noise canceling algorithm is analytically derived in relation to the two key parameters, i.e., noise bandwidth and the secondary path delay.

The combined effect of length of the secondary path delay and noise bandwidth, which is inversely proportional to the coherence time of noise correlation, is the most important factor of noise reduction performance of FBANC. When assumed pure Δ-step delay model of secondary path and its errorless estimation, the FBANC model having secondary path delay Δ can be simplified as an equivalent Δ-step ahead linear predictor model[4-5]. The noise reduction effect of FBANC is guaranteed only when the secondary path delay is sufficiently shorter than the coherence-time of noise correlation.[9] Also, the length of the secondary path delay is determined by size of spatial coverage range of noise cancelling so called zone of quiet (ZOQ) [7,8].

To exclusively investigate the effect of the two parameters, the secondary path and the noise are modeled with a pure delay and the AR processes, respectively. The first order AR model, AR(1), is defined as [13]

$$d(n) = l \cdot d(n-1) + i(n) \quad \text{for } |l| < 1 \tag{6}$$

where $i(n)$ is a white Gaussian noise, the AR coefficient l is the value of the pole on the real axis in z-plane. The minimum distance from the pole location of the AR model to the unit circle in z-plane, $1-|l|$, directly indicates the -3 dB bandwidth of the AR process under their narrowband assumption of $1-|l| \ll 1$.

Under the assumption of the pure delay secondary path model, $S(z) = z^{-\Delta}$, the feedback ANC can reduced to the Δ -step linear predictor [4-5]. The noise reduction performance of FBANC can be obtained from the feedback ANC's equivalent Δ -step ahead linear predictor model, reflecting the noise bandwidth with the AR model parameter.

Since the equivalent model is a typical feedforward linear predictor, the theoretical maximum noise reduction (NR) bound can be readily derived in the sense of Wiener as

$$\begin{aligned} NR_{\max} &\triangleq \frac{E\left[|d(n)|^2\right]}{E\left[|e_{\min}(n)|^2\right]} = \frac{E\left[|d(n)|^2\right]}{E\left[|d(n) - \mathbf{w}_o{}^H \mathbf{x}(n)|^2\right]} \\ &= \frac{\sigma_d^2}{\sigma_d^2 - \mathbf{r}(-\Delta)^H \mathbf{R}^{-1} \mathbf{r}(-\Delta)}, \end{aligned} \tag{7}$$

Where $\mathbf{w}_o = \mathbf{R}^{-1}\mathbf{r}(-\Delta)$ is the optimum filter coefficient vector, $e_{\min}(n)$ is the minimum prediction error in the optimum condition, $\mathbf{x}(n) = \left[d(n-\Delta)\ d(n-\Delta-1) \cdots d(n-\Delta-M+1)\right]^T$ is the reference signal vector, and E[•] denotes the expectation. The mean of the primary noise $d(n)$ is assumed to be zero, and the variance is σ_d^2, the correlation matrix $\mathbf{R} = E\left[\mathbf{x}(n)\mathbf{x}^H(n)\right]$, and $\mathbf{r}(-\Delta) = E\left[\mathbf{x}(n)d^*(n)\right]$, of which elements are the autocorrelation of primary noise corresponding to the delay Δ. The derived result (7) shows that the maximum noise reduction bound depends on $\mathbf{r}(-\Delta)$, i.e., autocorrelation of the primary noise with the delay Δ.

A closed form equation of maximum available noise reduction is derived as the ratio between the average power of the primary noise $d(n)$ and that of the residual noise $e(n)$ as

$$NR_{AR(1).\max} = \frac{\sigma_d^2}{\sigma_d^2 - \mathbf{r}(-\Delta)^H \mathbf{R}^{-1} \mathbf{r}(-\Delta)} = \frac{\sigma_d^2}{\sigma_d^2 - \sigma_d^2 l^{2\Delta}} = \frac{1}{1-l^{2\Delta}} \tag{8}$$

where Δ : delay of the secondary path,
l : AR coefficient,
$1-|l|$: -3 dB noise bandwidth for $1-|l| \ll 1$, and
$r(-m) = \sigma_d^2 l^m$: autocorrelation of (6) for $m \geq 0$.

This shows that the noise reduction of 1st-order AR noise is determined by two parameters, Δ and l. As shown in (8), the Noise Reduction performance increases as the secondary-path delay Δ decreases and the bandwidth $1-|l|$ decreases.

The noise from rotating machines such as a vehicle engine, a generator, or a ventilator can be modeled with the 2nd order autoregressive model. The physical parameters for characteristics of rotating machine noise, i.e., center frequency, bandwidth, and the Q-factor, can be explicitly reflected by the AR coefficients. The 2nd order AR process model having center frequency ω_0 and -3dB bandwidth $1-|l|$ can be expressed as

$$d(n) = 2l\cos(\omega_0) \cdot d(n-1) - l^2 \cdot d(n-2) + i(n) \tag{9}$$

The noise reduction performance of the 2nd order autoregressive noise model is analyzed and derived in [9] as

$$NR_{AR(2).\max} = \frac{1}{1-l^{2\Delta}\left[1+\dfrac{\sin(2\omega_o)\sin(2\omega_o\Delta)+2\sin^2(\omega_o\Delta)(\cos(2\omega_o)-l^2)}{2\sin^2(\omega_o)\cdot(1+l^2)/(1-l^2)}\right]}. \tag{10}$$

This equation gives the upper bound of noise reduction in terms of the secondary-path delay and the bandwidth and the center frequency of a rotating machine-type noise. The noise reduction performance is also increased by decreasing the secondary-path delay and the bandwidth as the noise reduction for the first-order AR noise does.

4. Measurements and Results

The performance of the proposed system is also experimentally tested to compare with the analytically derived results. For the experimental test, a headphones structured earmuffs is implemented and the picture of overall experimental setting is shown in Fig. 3.

The headphones are installed on a dummy head and though the dummy head, the sensing tip of the measurement microphone is faced with the speaker inside the headphones. Primary noise is generated by a loud speaker located in front of the dummy head. The ANC algorithm is implemented on floating-point DSP system.

Primary noises are generated using the 2nd order AR models having 200Hz center frequency and the three different -3dB bandwidths, 10Hz, 50Hz and 90Hz, respectively. Such test primary noises model the typical

ANC target noises such as airplane engine noise, rotating machine noises in plants, whose dominant components are at relatively low frequency band. [1,5-7,16]

The ear-protection system captured the noise and sampled it with sampling frequency 96 kHz and the sampled data is decimated to 16 kHz for ANC operation. The same noise sound is applied two times to the implemented system, once without running ANC and once with running ANC. The microphone sensed residue signals are recoded using an independent computer. The noise reduction performance is measured by obtaining the ratio of the average powers of the recoded residue of the two cases with Eq. (11)

$$NR = \frac{E[|d(n)|^2]}{E[|e(n)|^2]} \tag{11}$$

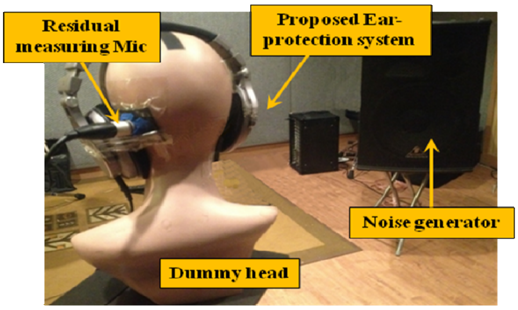
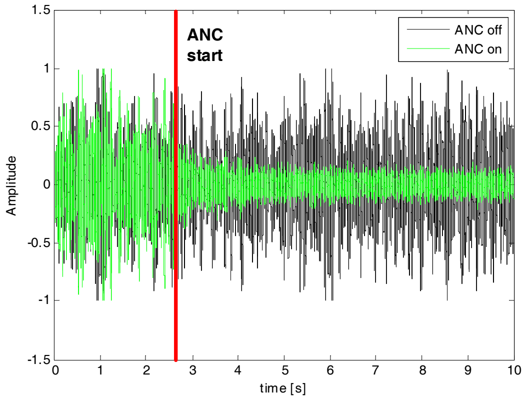
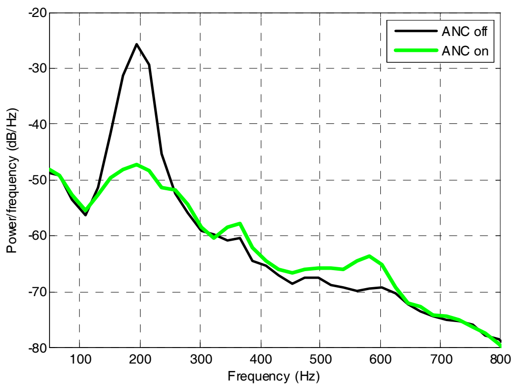


Fig. 3. Experimental setup of the FBANC based ear-protection system



(a) Comparison of the primary and the residual noise at convergence



(b) Comparison of spectra of the primary and the residual noise

Fig. 4. An example of the experimentally measured result of the FBANC based ear-protection system

An example of the experimentally measured result and its spectrum are shown in Fig. 4.

To verify the usage of pure delay model in the proposed analysis, simulations are performed for both cases of pure delay secondary path model and practical secondary path model having delay and spectral distortion. The practical spectral distortion component is measured from the implemented ear-protection system and the measured frequency response is shown in Fig. 5.

The simulation and experimental results are also summarized with analytical results in Table 1. The analytic results are obtained using the Eq. (10) and in the analytic results, the excessive error of the LMS algorithm was reflected for the comparison with the simulation results [13]. The secondary path delay is set to 0.25 msec.

It is verified that the analytic results agree with the simulation results and the experimental results showing less than 2.0 dB difference. Where, the analytic results include the excessive error of the LMS algorithm by adding the excessive error to the derived Eq. (10). The usefulness of the proposed system is also shown by the system's relatively high noise reduction performance.

Voice signals are added to the background noise signal. The update of the adaptive filter is selectively enabled to

Table 1. Noise Reduction Performance of the Proposed Ear-protection System

	3-dB Bandwidth of AR noise		
	10 Hz	50 Hz	90 Hz
Analytic results	16.96 [dB]	11.40 [dB]	8.264 [dB]
Simulation w/pure delay model	16.00 [dB]	10.78 [dB]	8.253 [dB]
Simulation w/measured secondary path model	17.09 [dB]	10.99 [dB]	8.91 [dB]
Experimental results	17.05 [dB]	10.54 [dB]	8.99 [dB]

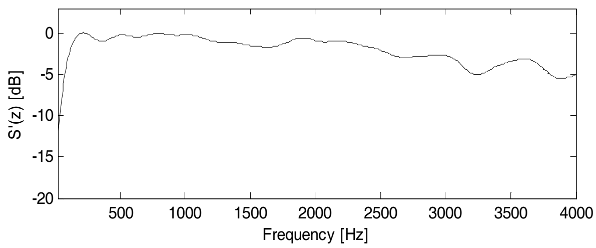
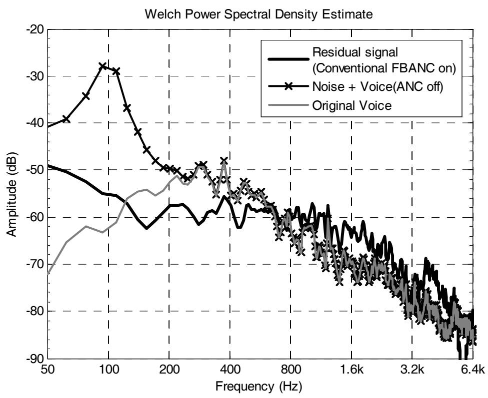


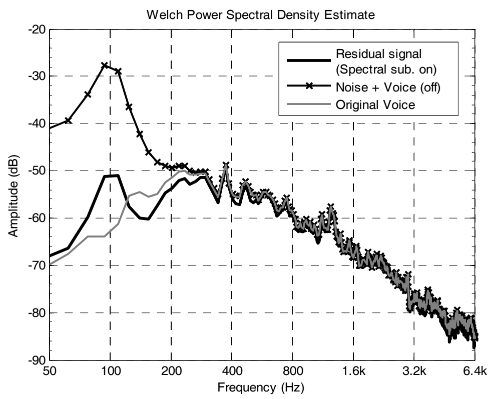
Fig. 5. The frequency response measured from the implemented voice enhancement ear-protection system, which is applied in simulations

avoid voice talking periods. The experimentally measured results are shown in Fig. 6 and Table 2.

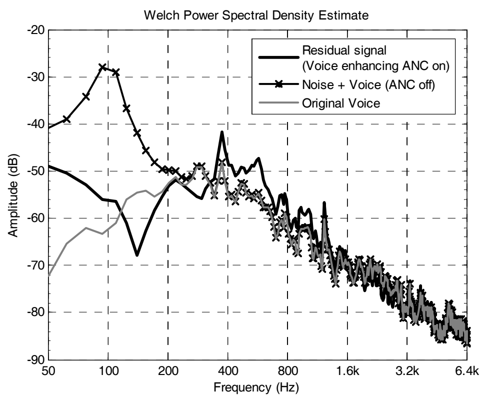
In Fig. 6. the power spectra of the voice added noise are plotted with the power spectrums obtained from the applications of two different ANC operations and a former



(a) Voice degradation showed in range of 400 to 800 Hz when applied with the conventional FBANC system



(b) Noise reduction showed when applied former noise reduction algorithm spectral subtraction.



(c) Voice enhancement showed in range of 400 to 800 Hz when applied with the proposed FBANC system

Fig. 6. Comparison of noise reductions and voice enhancement effect for the conventional FBANC (a), a former noise reduction algorithm(spectral subtraction) (b) and the proposed voice enhancing algorithm (c)

Table 2. Voice enhancement effect and noise reduction performance of the proposed ear-protection system

	Noise reduction (95Hz - 105Hz)	Voice enhancement (200Hz-800Hz)
Conventional FBANC	25.76 [dB]	-2.21 [dB] (Degraded)
Spectral subtraction	22.58 [dB]	-2.23 [dB] (Degraded)
Proposed voice enhancing ANC	27.53 [dB]	2.62 [dB] (Enhanced)

noise reduction algorithm, i.e., the conventional ANC with continuous update, the proposed system with the selective update and spectral subtraction algorithm[15]. For effective demonstration of the voice enhancement effect, the center frequency of AR noise model is allocated to 100Hz and its -3dB bandwidth is set to 5Hz.

In Table 2, the noise reduction performance and the voice enhance effect are measured by obtaining the ratio of the area of the power spectra distributed in the range of 95 to 105Hz and 200 to 800 Hz, respectively. Within each selected frequency range, the AR noise and the voice signal contain 70% of their energy. The performances in the limited frequency ranges are obtained with the following Eq. (12).

$$NR_{[f_1,f_2]} = \frac{\int_{f_1}^{f_2} E[|D(f)|^2]df}{\int_{f_1}^{f_2} E[|E(f)|^2]df} \tag{12}$$

where $D(f), E(f)$: Fourier transform of the primary noise and the residual signal,
f_1, f_2 : The lower and upper bounds of the target noise reduction frequency range.

It is shown that the peak noise reduction achieved by the application of the ANC is more than 25dB for both FBANC cases and 22dB for Spectral subtraction algorithm. The result for the conventional FBANC shows that the both the voice and the noise are suppressed by the ANC in Fig. 6(a). In this case, the spectrum area of voice is decreased by 2.21 dB. It is also verified that the proposed selective update of ANC filter significantly enhances voices by 2.62 dB as shown in Table 2. In Fig. 6(c), it is also shown to be inferred from the noticeable enhancement of power spectrum in the frequency range between 200-800 Hz where most of the speech energy is distributed.

5. Conclusion

This paper presents a FBANC based ear-protection system which selectively reduces the background noise while preserving the clarity of voice. The experimental tests showed the peak noise reduction achieved more than 15dB while the voice component is enhanced by 5.02 dB in the frequency range between 400-800Hz. The noise reduction performance of the proposed noise canceling algorithm is both analytically derived and experimentally tested and their agreements are verified. The feasibility of the proposed system is shown by the experimental test, where the achieved levels of noise reduction for the three different noise bandwidths cases, i.e., 10Hz, 50Hz, and 90Hz, are high to show 17.05dB, 10.54dB and 8.99dB, respectively.

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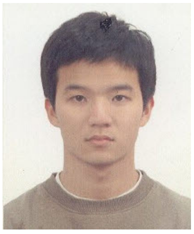

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