SPACE CONSTRAINED BEAMFORMER WITH A SOFT ADAPTATION BLOCKING SYSTEM AND NON-COHERENT CANCELLATION

Siow Yong Low and Sven Nordholm

{siowyong,sven}@watri.org.au
Western Australian Telecommunications Research Institute (WATRI)†,
Crawley, WA 6009, Australia

ABSTRACT

This paper introduces a new regulator to control the adaptation of the adaptive algorithm for use in a blocking system. Here, the blocking system is part of the previously proposed scheme, which cancels target signal for the estimation of noise spectrum. Contrary to the use of a hard approach to control the adaptation, a soft approach is proposed. As such, the rigidity of the adaptation is greatly reduced and the adaptation can be carried out even during non-target signal period. Results show that the proposed soft regulator achieves better target signal cancellation, which in turn leads to greater noise suppression capability.

1. INTRODUCTION

Noise is the major stumbling block for most speech communication related applications such as speech recognizers, hearing aids and hands-free communication systems. A myriad of signal processing algorithms have been proposed over the years to overcome the effects of noise [1]. Of these, microphone array based algorithms prominently stand out. This is attributed to its spatial diversity, which makes use of the spatial separation of the sources. As such, microphone arrays have the capability to spatially pass or reject sources at a specific point in space.

As outlined in [2], the generalized sidelobe canceller (GSC) succumbs to the presence of steering vector errors and consequently results in target signal cancellation. Such an undesirable effect defeats the purpose of speech acquisition by severely cancelling the target signal and limiting its noise suppression capability. In [3], the concept of space constraints is introduced to guard against the steering vector errors. Unlike a point source model, space constraints effectively compensates for the large radial vector errors in the target signal location caused by the erroneous steering vector in real life. Based on the space constraints, a robust beamformers with non-coherent processing is proposed in [4]. Similar to [2], the robust beamformer has a blocking system, which consists of adaptive filters to cancel out the target signal. However, the weight adaptation for the blocking system can only be carried out when the signal to noise ratio (SNR) is high enough to prevent noise cancellation or target signal leakage. Such a restraint requires the rigidity of controlling the adaptation by turning it on or off.

In this contribution, a "soft approach" (similar to a variable step-size approach [5]) is proposed to carry out the adaptation. As opposed to turning the adaptation on or off, the adaptation is made slower or faster. By doing so, the adaptation can be carried out continuously and at the same time reduces the leakage of the target signal. Experimental results show that the proposed soft adaptation reduces target signal leakage and consequently leads to greater noise suppression.

†WATRI is a joint institute between Curtin University of Technology and The University of Western Australia. Research (partially) supported by National ICT Australia (NICTA). NICTA is funded by the Australian Research Council (ARC).

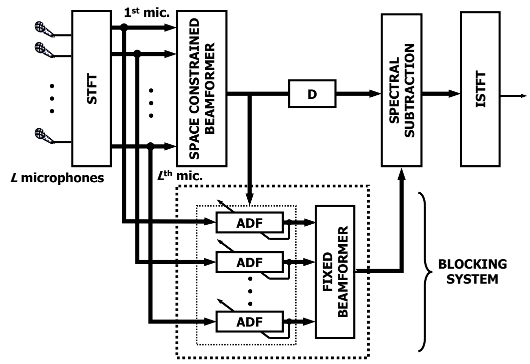

Figure 1: *The space constrained beamformer with adaptive blocking system (SCBF-ABS).*

2. SPACE CONSTRAINED BEAMFORMER WITH ADAPTIVE BLOCKING SYSTEM (SCBF-ABS)

Figure 1 shows the structure of the proposed space constrained beamformer with adaptive blocking system (SCBF-ABS), see also [4]. The abbreviation ADF denotes adaptive filters and D is the delay to shift to the center lag of the filter. Basically, the space constrained beamformer extracts the target signal by suppressing all sidelobes simultaneously in each subband. Following that, the output of the beamformer is fed into the blocking system as a reference. The blocking system consists of adaptive filters [2], which act as spatial-temporal rejection filters. The blocking system passes the interference and blocks the target signal by cancelling any components that are correlated to the reference signal. Thus the output signals at the blocking system consist mainly of the noise and the noise information can be estimated. The residue noise in the space constrained beamformer output can be further suppressed by using the estimated noise spectrum from the blocking system.

2.1. Space Constrained Beamformer

Assume that the spatial location of the target signal are spatially independent, then the vector of beamformer weights is

$$\mathbf{w}_{\mathsf{SCBF}}(\omega) = \left[\mathbf{R}_s(\omega) + \mathbf{R}_n(\omega)\right]^{-1} \mathbf{r}_{ds}(\omega), \qquad (1)$$

where $\mathbf{R}_s(\omega)$ and $\mathbf{r}_{ds}$ are the target signal spatial covariance matrix and target signal cross-covariance vector, respectively. These covariance information can be calculated analytically as in [3]. The noise covariance information, $\mathbf{R}_n(\omega)$ can be estimated from K samples of received data during the target signal "silence" periods when only the noise is active. Alternatively, the noise statistics can be estimated by using a calibration process. The beamformer output is then

$$y(\omega, k) = \mathbf{w}^H_{\mathsf{SCBF}}(\omega, k)\mathbf{x}(\omega, k), \qquad (2)$$

where the $L \times 1$ observation vector, $\mathbf{x}(\omega, k)$ is defined as $\mathbf{x}(\omega, k) = [x_1(\omega, k), x_2(\omega, k), \cdots, x_L(\omega, k)]^T$. The space constrained beamformer suppresses interference outside the specified constrained region. The output contains mainly the enhanced target signal and it serves as a reference signal for the blocking system.

2.2. Soft Adaptation Blocking System

Similar to [2] and [4], the proposed blocking system consists of a set of L adaptive filters to decorrelate the target signal from the observations. As mentioned previously, the weight adaptation for the blocking system should be carried out only when the SNR is high enough to prevent noise cancellation or target signal leakage. However, such a constraint requires the rigidity of controlling the adaptation by turning it on or off. To avoid the restraint, a soft approach is proposed instead. As opposed to turning the adaptation on or off, the adaptation is made slower or faster. By doing so, the adaptation can be carried out continuously and at the same time reduces the leakage of the target signal.

A modified LMS algorithm with a soft adaptation is used to adapt the filter coefficients in the blocking system. The blocking system vector of weights update for the l-th adaptive filter in frequency ω is

$$\mathbf{w}_{l,\mathsf{BS}}(\omega, k+1) = \mathbf{w}_{l,\mathsf{BS}}(\omega, k) + e^*_l(\omega, k)\mathbf{y}_{\mathsf{ref}}(\omega, k) g_l(\omega, k), \qquad (3)$$

where the $Q \times 1$ blocking system weights $\mathbf{w}_{l,\mathsf{BS}}(\omega, k)$ is given as

$$\mathbf{w}_{l,\mathsf{BS}}(\omega, k) = [w_{l,\mathsf{BS}}(\omega, k), w_{l,\mathsf{BS}}(\omega, k-1), \cdots, w_{l,\mathsf{BS}}(\omega, k-Q+1)]^T, \qquad (4)$$

and Q is the order of the adaptive filter. The error signal, $e_l(\omega, k)$ is defined as

$$e_l(\omega, k) = x_l(\omega, k) - \mathbf{w}^H_{l,\mathsf{BS}}(\omega, k)\mathbf{y}_{\mathsf{ref}}(\omega, k), \qquad (5)$$

and the reference signal, $\mathbf{y}_{\mathsf{ref}}(\omega, k)$ is given as

$$\mathbf{y}_{\mathsf{ref}}(\omega, k) = [y_{\mathsf{ref}}(\omega, k), y_{\mathsf{ref}}(\omega, k-1), \cdots, y_{\mathsf{ref}}(\omega, k-Q+1)]^T. \qquad (6)$$

The step-size function $g_l(\omega, k)$ is proposed as

$$g_l(\omega, k) = \exp\left\{\frac{-\mu}{\rho\left[Q\hat{\sigma}^2_{e_l}(\omega, k) + \mu\hat{\sigma}^2_{\mathbf{y}_{\mathsf{ref}}}(\omega, k) + \delta\right]}\right\}, \qquad (7)$$

where μ and ρ are the non-linear step size constant and the exponential regulator constant, respectively. The variance $\hat{\sigma}^2_{e_l}(\omega, k)$ is the estimated output power of the l-th adaptive filter, $\hat{\sigma}^2_{\mathbf{y}_{\mathsf{ref}}}(\omega, k)$ is the estimated reference output power and δ is a small constant to avoid the amplification problem. Note that the smaller the regulator value ρ is, the steeper the non-linear function decays and hence the more rigorous the adaptation is. To illustrate this, Figures 2(a) and (b) show the plot of a linear function with values between $[0, 5]$ and its corresponding transformed functions with varying values of ρ, respectively.

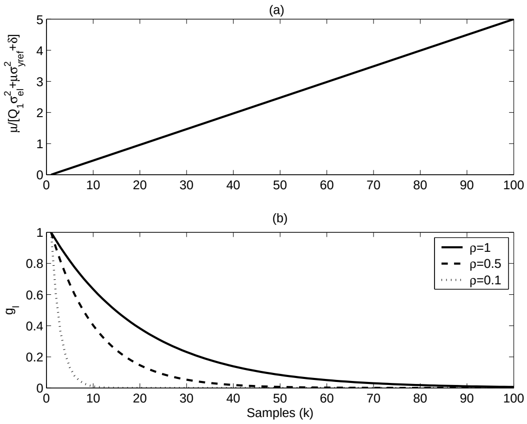

Figure 2: *(a) A linear function with values [0,5]. (b) The transformed linear function through the non-linear function by using (7) with varying ρ.*

From (7), the non-linear function exploits intervals of strong target signal by maximally adjusting its value to allow faster convergence. Likewise, if there is an absence of target signal, the non-linear function reduces its value. If the exponential function is omitted from (7), then the non-linear step size does the opposite by allowing a smaller step size when the target signal is strong and vice versa. This sort of control is particularly suitable in an adaptive noise canceller where the adaptation should be performed only when the SNR is low [6]. Thus, the exponential function translates the non-linear function into its reversal form, i.e., as $\hat{\sigma}^2_{el}(\omega, k)$ becomes larger, there will be an increase in $g_l(\omega, k)$.

The output power is updated continuously by using the square of its vector norm of length Q and smoothed as

$$\hat{\sigma}^2_{e_l}(\omega, k) = \alpha_e \hat{\sigma}^2_{e_l}(\omega, k-1) + (1-\alpha_e)\|\mathbf{e}_l(\omega, k)\|^2, \qquad (8)$$

where $\mathbf{e}_l(\omega, k) = [e_l(\omega, k), e_l(\omega, k-1), \cdots, e_l(\omega, k-Q+1)]^T$ and α_e is a smoothing factor. Desirably, the estimated reference power, $\hat{\sigma}^2_{\mathbf{y}_{\mathsf{ref}}}(\omega, k)$ should consist of the power of the target signal only. However, this quantity is unknown and it needs to be estimated from the reference signal, $\mathbf{y}_{\mathsf{ref}}(\omega, k)$ since the reference signal is by far the "cleanest" target signal information available. From (5), the term $\mathbf{w}^H_{l,\mathsf{BS}}(\omega, k)\mathbf{y}_{\mathsf{ref}}(\omega, k)$ approximates the target signal more closely since the adaptive weights are adjusted to temporally decorrelate the target signal from the input observation. Thus, the weighted reference signal is proposed to estimate the reference power as follows

$$\hat{\sigma}^2_{\mathbf{y}_{\mathsf{ref}}}(\omega, k) = \alpha_{\mathbf{y}_{\mathsf{ref}}} \hat{\sigma}^2_{\mathbf{y}_{\mathsf{ref}}}(\omega, k-1) + (1-\alpha_{\mathbf{y}_{\mathsf{ref}}})\|\mathbf{w}^H_{l,\mathsf{BS}}\mathbf{y}_{\mathsf{ref}}(\omega, k)\|^2, \qquad (9)$$

where $\alpha_{\mathbf{y}_{\mathsf{ref}}}$ is a smoothing constant. Following that, a fixed beamformer with a highpass characteristic, $\mathbf{w}_{\mathsf{FBF}}$ is then used to collapse the L outputs of the adaptive filters $\mathbf{e}_{\mathsf{BS}}(\omega, k)$ to form the noise information where $\mathbf{e}_{\mathsf{BS}}(\omega, k) = [e_1(\omega, k), e_2(\omega, k), \cdots, e_L(\omega, k)]^T$ and the output of the fixed beamformer is $y_{\mathsf{BS}}(\omega, k) = \mathbf{w}^T_{\mathsf{FBF}}\mathbf{e}_{\mathsf{BS}}(\omega, k)$.

2.3. Non-Coherent Cancellation

The non-coherent approach offers better suppression capability particularly in the lower frequency range where the noise is highly coherent [7]. However, conventional non-coherent technique introduces spectral artifacts or musical tones caused by spectral discontinuity/variations due to the inaccurate estimate of the noise spectrum [1]. A new regulator is introduced to control the gain function by correctly weighing the subtraction level in each subband. As such, different levels of noise in each subband can be determined for maximum noise reduction with small target signal distortion. The gain function is calculated by using a block based method. Firstly, the noise signals, $y_{\mathsf{BS}}(\omega,k)$ from the blocking system is formed into non-overlapping P blocks of length K as

$$\mathbf{y}_{\mathsf{NOISE}}(\omega,p) = [y_{\mathsf{BS}}(\omega,Kp), y_{\mathsf{BS}}(\omega,Kp-1), \cdots, y_{\mathsf{BS}}(\omega,K(p-1)+1)], \quad (10)$$

where $p = 1, 2, \cdots, P$. Similarly, the outputs $\mathbf{y}_{\mathsf{SCBF}}(\omega,p)$ from the space constrained beamformer is formed in the manner defined in (10).

Assume that the space constrained beamformer output, $\mathbf{y}_{\mathsf{SCBF}}(\omega,p)$ consists of the desired overall output, $\mathbf{y}_{\mathsf{SCBF\text{-}ABS}}(\omega,p)$ (i.e., target signal only) and the noise information, $\mathbf{y}_{\mathsf{NOISE}}(\omega,p)$. Thus, the instantaneous overall output power can be estimated as

$$\mathbf{S}_{\mathbf{y}_{\mathsf{SCBF\text{-}ABS}}}(\omega,p) = \mathbf{S}_{\mathbf{y}_{\mathsf{SCBF}}}(\omega,p) - \mathbf{S}_{\mathbf{y}_{\mathsf{NOISE}}}(\omega,p), \quad (11)$$

where

$$\mathbf{S}_{\mathbf{y}_{\mathsf{SCBF}}}(\omega,p) = \left[|y_{\mathsf{SCBF}}(\omega,Kp)|^2, |y_{\mathsf{SCBF}}(\omega,Kp-1)|^2, \cdots, |y_{\mathsf{SCBF}}(\omega,K(p-1)+1)|^2\right], \quad (12)$$

and

$$\mathbf{S}_{\mathbf{y}_{\mathsf{NOISE}}}(\omega,p) = \left[|y_{\mathsf{BS}}(\omega,Kp)|^2, |y_{\mathsf{BS}}(\omega,Kp-1)|^2, \cdots, |y_{\mathsf{BS}}(\omega,K(p-1)+1)|^2\right]. \quad (13)$$

Equation (11) can be rewritten as

$$\mathbf{S}_{\mathbf{y}_{\mathsf{SCBF\text{-}ABS}}}(\omega,p) = \mathbf{S}_{\mathbf{y}_{\mathsf{SCBF}}}(\omega,p) \odot [1 - \mathbf{S}_{\mathbf{y}_{\mathsf{NOISE}}}(\omega,p) \oslash \mathbf{S}_{\mathbf{y}_{\mathsf{SCBF}}}(\omega,p)], \quad (14)$$

where $|\cdot|$ represents the absolute value of each element in the vector, $\odot$ and $\oslash$ denote the element-wise multiplication and element-wise division, respectively. From (14), it is observed that a gain function can be formed to further suppress the residue noise in $\mathbf{y}_{\mathsf{SCBF}}(\omega,p)$. Nevertheless, the level of subtraction must be correctly ascertained to ensure the right amount of noise is subtracted. For convenience, the instantaneous magnitude spectrum estimate is defined as $\bar{\mathbf{S}}_{\mathbf{y}_{\mathsf{SCBF\text{-}ABS}}}(\omega,p) = \sqrt{\mathbf{S}_{\mathbf{y}_{\mathsf{SCBF\text{-}ABS}}}(\omega,p)}$, where the operator $\sqrt{\cdot}$ denotes element-wise square root. Hence, the gain function, $\mathbf{G}(\omega,p)$ for magnitude spectral subtraction in each block is proposed as

$$\mathbf{G}(\omega,p) = \left\{1 - \varphi(\omega,p)\left[\bar{\mathbf{S}}_{\mathbf{y}_{\mathsf{NOISE}}}(\omega,p) \oslash \bar{\mathbf{S}}_{\mathbf{y}_{\mathsf{SCBF}}}(\omega,p)\right]\right\} \exp\left[-j\pi\omega(1+M/2)\right], \quad (15)$$

where $\bar{\mathbf{S}}_{\mathbf{y}_{\mathsf{NOISE}}}(\omega,p) = \sqrt{\mathbf{S}_{\mathbf{y}_{\mathsf{NOISE}}}(\omega,p)}$, $\bar{\mathbf{S}}_{\mathbf{y}_{\mathsf{SCBF}}}(\omega,p) = \sqrt{\mathbf{S}_{\mathbf{y}_{\mathsf{SCBF}}}(\omega,p)}$ and M is the number of frequency bins. The parameter $\varphi(\omega,p)$ is the gain regulator and the exponential function is included to introduce a phase to the gain function for causality.

The gain regulator, $\varphi(\omega,p)$ in (15) is central to the spectral subtraction algorithm as it adjusts the desired noise reduction in each p-th block of frequency ω. Traditionally, this parameter is fixed constant throughout the blocks and the frequency range of interest. This inflexible approach limits the amount of interference reduction achievable and causes significant speech distortion. This is because the levels of the estimated noise spectrum vary randomly in each block and frequency. Naturally, with constant gain regulator, the algorithm may over- or under-subtract resulting in discontinuity in the spectrum (or time-varying tones).

To estimate $\varphi(\omega,p)$, consider the output of the spectral subtraction algorithm. The overall output of the SCBF-ABS, $\mathbf{y}_{\mathsf{SCBF\text{-}ABS}}(\omega,p)$ is given as

$$\mathbf{y}_{\mathsf{SCBF\text{-}ABS}}(\omega,p) = \mathbf{G}(\omega,p) \odot \mathbf{y}_{\mathsf{SCBF}}(\omega,p). \quad (16)$$

Substituting (15) into (16) yields

$$\mathbf{y}_{\mathsf{SCBF\text{-}ABS}}(\omega,p) = \left\{1 - \varphi(\omega,p)[\bar{\mathbf{S}}_{\mathbf{y}_{\mathsf{NOISE}}}(\omega,p) \oslash \bar{\mathbf{S}}_{\mathbf{y}_{\mathsf{SCBF}}}(\omega,p)]\right\} \odot \exp\left[-j\pi\omega(1+M/2)\right]\mathbf{y}_{\mathsf{SCBF}}(\omega,p). \quad (17)$$

For mathematical tractability, the exponential term in (17) is removed. Thus, solving for $\varphi(\omega,p)$ gives

$$\varphi(\omega,p) = \left[\bar{\mathbf{S}}_{\mathbf{y}_{\mathsf{SCBF}}}(\omega,p) \oslash \bar{\mathbf{S}}_{\mathbf{y}_{\mathsf{NOISE}}}(\omega,p)\right] [1 - \mathbf{y}_{\mathsf{SCBF\text{-}ABS}}(\omega,p) \oslash \mathbf{y}_{\mathsf{SCBF}}(\omega,p)]. \quad (18)$$

From (18), it is observed that during source silence period, the overall output, $\mathbf{y}_{\mathsf{SCBF\text{-}ABS}}(\omega,p)$ should be approximately zero. Thus, the gain regulator can be approximated as

$$\varphi(\omega,p_{\mathsf{ss}}) = \left[\bar{\mathbf{S}}_{\mathbf{y}_{\mathsf{SCBF}}}(\omega,p_{\mathsf{ss}}) \oslash \bar{\mathbf{S}}_{\mathbf{y}_{\mathsf{NOISE}}}(\omega,p_{\mathsf{ss}})\right], \quad (19)$$

where p_{ss} denotes source silence period block. During speech active period, however, the gain regulator is not updated and it is set to retain its previous value. This is possible because noise is generally long term stationary or slowly time varying relative to speech. Also, during the presence of target signal (speech) period, most of the musical tones are masked by the strong speech component itself. An exponential averaging is used to update the gain regulator as follows

$$\bar{\varphi}(\omega,p_{\mathsf{ss}}) = \alpha_\varphi \bar{\varphi}(\omega,p_{\mathsf{ss}}-1) + (1-\alpha_\varphi)\varphi(\omega,p_{\mathsf{ss}}), \qquad p_{\mathsf{ss}} \geq 2, \quad (20)$$

where α_φ is a smoothing constant. By the same token, the gain function is also smoothed as

$$\bar{\mathbf{G}}(\omega,p) = \alpha_G \bar{\mathbf{G}}(\omega,p-1) + (1-\alpha_G)\mathbf{G}(\omega,p), \qquad p \geq 2, \quad (21)$$

where the smoothing factor, α_G controls the length of the exponential memory.

3. EVALUATIONS

The performance evaluation of the proposed SCBF-ABS was made in a real room of dimensions $3.5 \times 3.1 \times 2.3$ m^3 using a six-element omnidirectional linear array. The signals were sampled at 8 kHz with an inter-element distance of 0.04 m and the reverberation time of the room was 250 ms. The target signal was positioned 0.6 m at an angle of 90° from the centre of the array. Two loudspeakers emitting machinery noise were placed facing the array. The signal to noise ratio was calculated to be in the vicinity -7 dB.

SNR	GSC	SCBF	SCBF-ABS 1 (Conventional)	SCBF-ABS 2 (Proposed)
-5	-29.0	-23.7	-23.6	-23.5
0	-29.0	-23.9	-23.8	-23.8
5	-29.1	-24.2	-23.9	-24.0
10	-29.2	-24.3	-24.1	-24.2

Table 1: *Distortion measures for different SNRs.*

Two performance indices are introduced to quantify objectively the distortion in the source signal and the noise suppression. The normalized distortion, $\mathcal{D}$ is defined in decibels (dB) as

$$\mathcal{D} = 10\log_{10}\left[\frac{1}{\mathcal{M}}\sum_{m=0}^{\mathcal{M}-1}|C\hat{P}_{out,s}(\omega_m) - \hat{P}_{in,s}(\omega_m)|\right] \quad (22)$$

where $\omega_m = 2\pi m/\mathcal{M}$, is the discretized normalized frequency, $\mathcal{M}$ is the number of FFT points and the constant C normalizes the suppression measure to the structure's source signal gain. The normalized noise/jammer(s) suppression, $\mathcal{S}$ in dB is

$$\mathcal{S} = 10\log_{10}\left[\frac{\sum_{m=0}^{\mathcal{M}-1}\hat{P}_{in,n}(\omega_m)}{\sum_{m=0}^{\mathcal{M}-1}\hat{P}_{out,n}(\omega_m)}\right] - 10\log_{10}(C), \quad (23)$$

where $\hat{P}_{in,n}(\omega_m)$ and $\hat{P}_{out,n}(\omega_n)$ are the spectral power estimates of the reference sensor observation and the output, respectively when the noise/jammer(s) is active alone.

Figure 3 presents the noise suppressions with different SNR for the GSC, the SCBF, the SCBF-ABS (conventional), and the proposed SCBF-ABS (soft approach). The plot shows that the SCBF-ABS yields the highest noise suppression capability compared to all the other approaches. The incorporation of the "soft energy detector" minimizes the target signal leakage, which prevents target signal cancellation and consequently leads to higher noise suppression. Figures 4(a) and 4(b) plot the original spectral components of the target signal and the magnitude of the non-linear function given in (7) for frequency 1700 Hz, respectively. Clearly, the step-size function increases its value during the presence of target signal and correspondingly decreases its value when there is no target signal. By doing so, the target signal cancellation is kept to the minimum and the noise is passed, which in turn leads to better noise suppression capability.

4. CONCLUSION

This paper has presented a new non-linear function to control the adaptation in adaptive blocking system of the SCBF-ABS. The proposed soft regulator proves to be efficient in minimizing the leakage of the target signal as the adaptation is made slower/faster according to the target signal level. Experimental results demonstrate that the proposed method achieves better noise suppression compared to the conventional hard regulator.

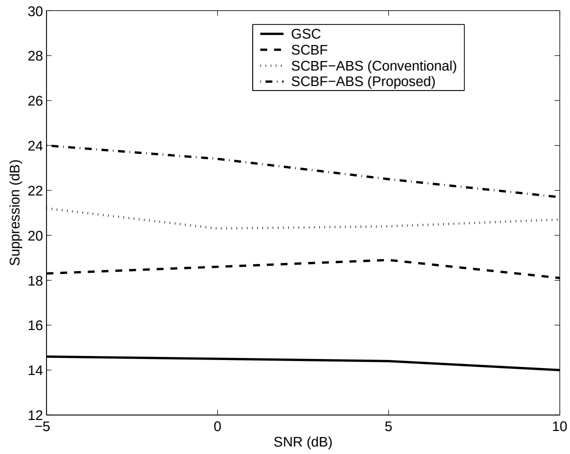

Figure 3: *Noise suppression measures for different SNRs.*

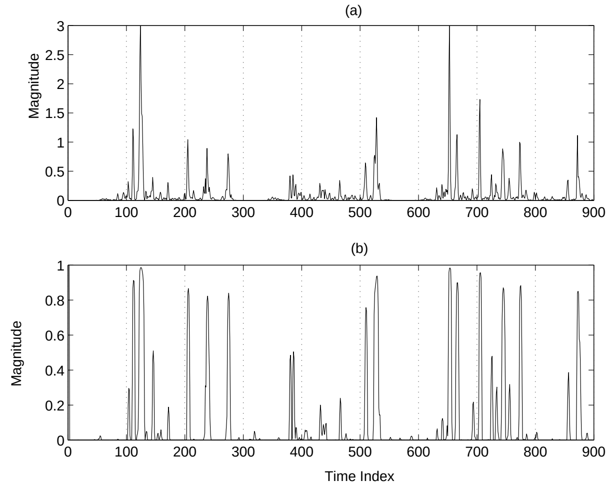

Figure 4: *The magnitudes of the (a) original target signal and (b) the step size function by using (7) at frequency 1700 Hz for the SCBF-ABS.*

5. REFERENCES

[1] J. Benesty, S. Makino, and J. Chen, *Speech Enhancement*, Signals and Communication Technology. Springer-Verlag, Berlin, March 2005.

[2] O. Hoshuyama, A. Sugiyama, and A. Hirano, "A robust adaptive beamformer for microphone arrays with a blocking matrix using constrained adaptive filters," *IEEE Trans. on Signal Process.*, vol. 47, no. 10, pp. 2677–2684, June 1999.

[3] N. Grbić and S. Nordholm, "Soft constrained subband beamforming for handsfree speech enhancement," *IEEE Int. Conf. on Acoust., Speech and Signal Process.*, vol. 1, pp. 885–888, 2002.

[4] S. Y. Low, N. Grbić, and S. Nordholm, "Robust microphone array using subband adaptive beamformer and spectral subtraction," *IEEE Int. Conf. on Communication Systems*, vol. 2, pp. 1020–1024, June 2002.

[5] R. H. Kwong and E. W. Johnston, "A variable step size LMS algorithm," *IEEE Trans. on Signal Process.*, vol. 40, no. 7, pp. 1633–1642, July 1992.

[6] J. E. Greenberg, "Modified LMS algorithms for speech processing with an adaptive noise canceller," *IEEE Trans. on Speech and Audio Process.*, vol. 6, no. 4, pp. 338–350, July 1998.

[7] J. Meyer and K. U. Simmer, "Multichannel speech enhancement in a car environment using wiener filtering and spectral subtraction," *IEEE Int. Conf. on Acoust., Speech, and Signal Process.*, vol. 2, pp. 1167–1170, April 1997.