

Evaluation of Two-microphone Acoustic Feedback Cancellation Using Uniform and Non-uniform Sub-bands in Hearing Aids

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Abstract—The limiting feature of hearing aids is acoustic feedback. This feedback problem has been approached by using an acoustic feedback canceller. This well known identification in a loop problem, will give a biased solution due to correlation between the desired and loudspeaker signals. Speech and music signals have long tails in the correlation. As a result, the performance of the system is considerably degraded and under certain conditions the cancellation system will be unstable. The two-microphone techniques have the potential to significantly reduce this problem. This paper introduces the applications of uniform and non-uniform sub-band techniques into the two-microphone acoustic feedback cancellation (AFC-2mics) to decorrelate input signals and individually adapt solutions in those bands. The system has been evaluated in terms of Misalignment (MisAL) and Maximum Stable Gain (MSG) using both male and female speech input signals. The simulation results show that our proposed methods provide better performance than the other existing methods.

I. INTRODUCTION

In a hearing aid, a part of the loudspeaker (receiver) signal feeds into the microphone, resulting in acoustic feedback. Due to this feedback the signal quality is considerably degraded. It also limits the achievable maximum stable gain. In the worst case, this feedback loop might cause instability (howling). A common approach to solve this problem is to employ an adaptive feedback canceller which is used to estimate the feedback path. Then the estimated feedback signal is subtracted from the microphone signal. However, this identification in a loop problem requires knowledge of some aspects of the input signal [1], in practice this is not possible resulting in biased estimation of the feedback path. In the past fifty years many solutions were introduced to decorrelate these signals such as delay insertion, frequency shifting, phase modulation, probe noise, pre-filters [2], [3], [4], [5], [6], [7], [8], [9].

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In the pre-filter approaches pre-filters are used to pre-whiten the input data of an adaptive filter, resulting in reduction in the bias of identification process [2], [7]. A dominant method is the prediction error method for acoustic feedback cancellation (AFC-PEM), where an incoming signal is assumed to be generated by passing a white noise signal through an all-pole filter. Firstly, this filter is used to estimate the incoming signal model and then the inverse of this model is employed to whiten the inputs of an adaptive filter. Consequently, the bias problem in estimation of feedback coefficients can be reduced considerably [2], [10]. Some extended works based on AFC-PEM are found in [11], [12], [13]. The AFC-PEM solution provides quite good system performance in terms of misalignment and maximum stable gain. However, in some specific scenarios the AFC-PEM method seems to depend on the accuracy of estimate of the incoming signal model, i.e., some speech signals seem to be better modelled than others [14].

Recently, transform domain (TD) processing and sub-band processing have shown to be promising techniques for acoustic feedback cancellation (AFC) in hearing aids. The DFT- or DCT-based transform domain were applied to prediction error method in [13] and to two-microphone acoustic feedback cancellation method (AFC-2mics) in [15], whereas a sub-band technique in conjunction with frequency shifting and AFC-PEM was introduced in [16]. In these methods an implementation of FIR uniform DFT/DCT-modulated filter banks [13], [15] or uniform subbands [16] (analysis filter) were used to separate the loudspeaker and microphone signals into frequency bands, then an adaptive filter was applied to each band. By designing an analysis filter with orthogonal or approximately orthogonal frequency bands, the correlation among signals in different frequency bins were lower, resulting in a considerable improvement in system performance.

The AFC-2mics method utilized two measurements of the incoming wave to whiten the cancellation error signal in order to decrease the bias in estimation of the feedback cancellation

filter. As a result, the hearing aids' performance is improved significantly. In fact, AFC-2mics approach provides a more stable solution than AFC-PEM [15], [17], [14]. By employing transform domain techniques in AFC-2mics algorithm (AFC-2mics-TD) the performance of the system can be improved considerably. Nevertheless, the performance of AFC-2mics-TD is still limited since there are overlaps between frequency bins which are considerable, also the side lobe attenuation in each filter band is not high enough relative to the main lobe. Thus, the output signals of frequency bands are not well partitioned among bands [18]. Moreover, in the AFC-2mics-TD method the output signals of filter banks are pushed into buffers of length $L_{\hat{g}}$ before they are fed into M adaptive filters, whereas the transform domain is orthogonal, not shift orthogonal.

This paper presents the applications of uniform and non-uniform sub-band techniques to AFC-2mics. The idea is to design a set of uniform and/or non-uniform sub-band filters in order to separate the loudspeaker, microphone and error signals of AFC-2mics into smaller frequency bands before these signals are fed into adaptive filters. As a result, the signals in the frequency bands are well decorrelated between bands and further improvements in system performance can be achieved. Simulations are implemented for a scenario where the feedback path is fixed and the speech source is moving as well as a scenario where the feedback path changes and the speech source is fixed. The simulation results show that both the proposed AFC-2mics method using uniform sub-bands (AFC-2mics-UniSB) and the proposed AFC-2mics method using non-uniform subbands which are designed based on modified octave bands (AFC-2mics-MOB) outperform the AFC-PEM. Moreover, both methods provide further improvement in convergence rate as well as steady-state MSE than AFC-2mics and AFC-2mics-TD in both cases.

This paper is organized as follows. In Section II, the review of AFC-2mics and the proposed system models of AFC-2mics-UniSB and AFC-2mics-MOB are described. Section III shows the simulation results and performance comparison among AFC-PEM, AFC-PEM-TD, AFC-2mics, AFC-2mics-TD, and proposed approaches.

II. PROPOSED SYSTEM

A. Review of AFC-2mics method

In the AFC-2mics method, two microphones and one loudspeaker are utilized for acoustic feedback cancellation. The main microphone $m_1(k)$ is placed in the ear, whereas the second microphone $m_2(k)$ is located behind the ear (BTE) such that the distance between two microphones is far enough to ensure that the second feedback path $F_2(q)$ has small effect on the system identification [14]. The microphone signals are denoted as

$$m_1(k) = x_1(k) + \hat{\mathbf{f}}_1^T(k) \mathbf{y}(k) \quad (1)$$

$$m_2(k) = x_2(k) + \hat{\mathbf{f}}_2^T(k) \mathbf{y}(k) \quad (2)$$

where $\mathbf{f}_1(k) = [f_1(k), f_1(k-1), \dots, f_1(k-L_f+1)]^T$, $\mathbf{f}_2(k) = [f_2(k), f_2(k-1), \dots, f_2(k-L_f+1)]^T$ are impulse responses (of length L_f) of the first and second feedback paths, respectively and $\mathbf{y}(k)$ is a vector containing the L_f most recent loudspeaker signals $y(k)$. The error signal $\bar{x}_1(k)$ is calculated as follows

$$\begin{aligned} \bar{x}_1(k) &= m_1(k) - \hat{\mathbf{f}}_1^T(k) \mathbf{y}(k) \\ &= x_1(k) + \left(\mathbf{f}_1^T(k) - \hat{\mathbf{f}}_1^T(k) \right) \mathbf{y}(k) \end{aligned} \quad (3)$$

where $\hat{\mathbf{f}}_1(k) = [\hat{f}_1(k), \hat{f}_1(k-1), \dots, \hat{f}_1(k-L_f+1)]^T$ is the impulse response of the first feedback canceller $\hat{F}_1(q)$. This error signal feeds back to the loudspeaker through a forward path which is denoted as $K(q) = q^{-d_k} |K|$, where d_k represents the delay and $|K|$ is the gain. In the AFC systems using single microphone, the error signal $\bar{x}_1(k)$ is utilized to control the feedback canceller. However, the bias in adaptation process might be large because the incoming signal behaves as a disturbance towards the feedback path identification. In order to reduce this bias, the second microphone is deployed. In the AFC-2mics approach the incoming signals at the first and second microphones are $x_1(k)$ and $x_2(k)$, respectively, and they have the following relationship,

$$x_1(k-d_m) = G(q)x_2(k) + \zeta(k) \quad (4)$$

where $G(q)$ is a FIR filter of length L_g with impulse response $\mathbf{g} = [g_0, g_1, \dots, g_{L_g-1}]^T$, which predicts $x_1(k)$ from $x_2(k)$ and $\zeta(k)$ represents the part that is not predictable; a small delay d_m is added in the first microphone path to ensure the system is causal. The first incoming signal can be estimated by a FIR filter $\hat{G}(q)$, then it is subtracted from $\bar{x}_1(k)$. As a result, a new error signal $e(k)$ which does not include $x_1(k)$ is used to update the coefficients of both feedback canceller and the FIR filter $\hat{G}(q)$

$$e(k) = \bar{x}_1(k) - \hat{x}_1(k). \quad (5)$$

Where $\hat{x}_1(k) = \hat{G}(q)m_2(k)$ is the estimate of the first incoming signal. Thus the bias of the adaptive filter is now caused by the second feedback path which by design is weaker than the first feedback path. Therefore, the bias is reduced [15], [14].

B. Proposed AFC-2mics-UniSB and AFC-2mics-MOB

In order to improve the performance of the AFC-2mics approach we propose to use sub-band filters as depicted in Fig. 1. This proposed system model is similar to the AFC-2mics-TD [15]. However, the contribution of this paper is to apply uniform and/or non-uniform sub-band filters instead of uniform DFT/DCT-modulated filter banks in the AFC-2mics method. From [19], it is clear that if a signal $x(n)$ is divided into M sub-band signals using a set of analysis filters which fulfil the strictly complementary (SC) condition as in (6), the

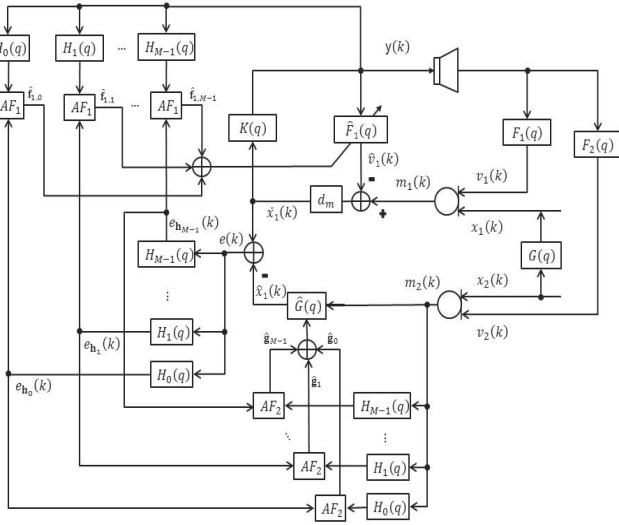


Fig. 1: Proposed AFC-2mics with sub-band techniques

original signal can be reconstructed with no distortion except for a delay by simply adding the sub-band signals

$$H_{sum}(z) = \sum_{i=1}^{M-1} H_i(z) = cz^{-n_0}, c \neq 0. \quad (6)$$

Where $H_i(z)$ is the transfer function of i^{th} frequency bin, $i = 0, 1, \dots, M-1$; M denotes the number of sub-bands and c is a constant value. In the proposed AFC-2mics-UniSB model we design a set of M uniform FIR sub-band filters (UniSB) in frequency range [250Hz, 7900Hz]. Kaiser window is used for each sub-band filter design to lower the effect of side lobes on adjacent sub-bands [20]. Besides, a set of M non-uniform sub-band filters based on modified octave bands (MOB) with edge frequencies defined as (7) is used for AFC-2mics-MOB.

$$f_{edge}^{MOB} = [250Hz, 500Hz, 1kHz, 2kHz, 3.5kHz, 5kHz, 7.9kHz]. \quad (7)$$

This set of MOB filters provides narrower bandwidth filters in low frequency bands where almost energy of speech is concentrated and wider bandwidth filters in higher frequency bands where little energy of speech is available.

The sub-band filter banks are applied to the loudspeaker signal $y(k)$, the second microphone signal $x_2(k)$ and the error signal $e(k)$ in the AFC-2mics approach. Then the sub-band loudspeaker and the sub-band error signals are used as inputs to the first set of adaptive filters (AF_1) in order to estimate the first feedback path $\hat{f}_1(k)$. To estimate the coefficients of the FIR filter $\hat{g}(k)$ the second sub-band microphone signal and the sub-band error signal are fed into the second set of adaptive filters (AF_2). Each adaptive filter is used per frequency band and all filters in the set have the same length. The lengths of AF_1 and AF_2 are $L_{\hat{f}_1}$ and $L_{\hat{g}}$, respectively. The length of each UniSB and MOB filter is selected as $L = 64$ in order to ensure that (6) is satisfied. Therefore, the synthesis filters in

the proposed system are simply adders.

Fig. 2a presents the frequency responses of eight uniform sub-band filters using DCT, which utilize for AFC-2mics-TD, whereas Fig. 2b and Fig. 2c illustrate the frequency responses of six uniform/non-uniform sub-band filters which apply for AFC-2mics-UniSB and AFC-2mics-MOB, respectively. It is clear that UniSB and MOB filters provide less overlap and have the first side lobe attenuations relative to the main lobe (≈ -45 dB) much higher than those of DCT filters (≈ -13 dB \dots -16 dB). Consequently, the bands of the resulting signals after UniSB and/or MOB are much better partitioned than those after DCT filters, leading to a significantly improved convergence rate. In addition, by taking advantage of the second microphone and employing UniSB and/or MOB on AFC-2mics, the bias in AFC-2mics-UniSB and AFC-2mics-MOB has been further reduced compared to AFC-2mics-TD.

Algorithm 1: AFC-2mics employing subbands for hearing aids

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for  $k = 1, 2, \dots$  do
  Analysis filters:
   $\mathbf{y}_H(k) = \mathbf{H}^T \mathbf{y}(k)$ 
   $\mathbf{e}_H(k) = \mathbf{H}^T \mathbf{e}(k)$ 
   $\mathbf{m}_H(k) = \mathbf{H}^T \mathbf{m}_2(k)$ 
  Adaptive filters using NLMS algorithm:
  for  $i = 1, 2, \dots, M$  do
     $\hat{\mathbf{f}}_{1,i}(k) = \hat{\mathbf{f}}_{1,i}(k-1) + \frac{\mu}{\mathbf{y}_{h_i}^T(k) \mathbf{y}_{h_i}(k) + \delta} \mathbf{y}_{h_i}(k) e_{h_i}(k)$ 
     $\hat{\mathbf{g}}_i(k) = \hat{\mathbf{g}}_i(k-1) + \frac{\mu}{\mathbf{m}_{h_i}^T(k) \mathbf{m}_{h_i}(k) + \delta} \mathbf{m}_{h_i}(k) e_{h_i}(k)$ 
  end for
  Synthesis filters:
   $\hat{\mathbf{f}}_1(k) = \sum_{i=1}^{M-1} \hat{\mathbf{f}}_{1,i}(k)$ 
   $\hat{\mathbf{g}}(k) = \sum_{i=1}^{M-1} \hat{\mathbf{g}}_i(k)$ 
end for
    
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Although AFC-2mics-UniSB and AFC-2mics-MOB have different designs in subband filters, the same algorithm for the acoustic feedback cancellation has been used. The complete algorithm description is given in Algorithm 1. The vectors and matrices in Algorithm 1 are defined as follows

$$\mathbf{y}(k) = [y(k), y(k-1), \dots, y(k-L+1)]^T \quad (8)$$

$$\mathbf{m}_2(k) = [m_2(k), m_2(k-1), \dots, m_2(k-L+1)]^T \quad (9)$$

$$\mathbf{e}(k) = [e(k), e(k-1), \dots, e(k-L+1)]^T \quad (10)$$

$$\mathbf{y}_H(k) = [y_{h_0}(k), y_{h_1}(k), \dots, y_{h_{M-1}}(k)]^T \quad (11)$$

$$\mathbf{m}_H(k) = [m_{h_0}(k), m_{h_1}(k), \dots, m_{h_{M-1}}(k)] \quad (12)$$

$$\mathbf{H} = [\mathbf{h}_0, \mathbf{h}_1, \dots, \mathbf{h}_{M-1}] \quad (13)$$

$$\hat{\mathbf{f}}_{1,i}(k) = [\hat{f}_{1,i}(k), \hat{f}_{1,i}(k-1), \dots, \hat{f}_{1,i}(k-L_{\hat{f}}+1)]^T \quad (14)$$

$$\hat{\mathbf{g}}_i(k) = [\hat{g}_i(k), \hat{g}_i(k-1), \dots, \hat{g}_i(k-L_{\hat{g}}+1)]^T \quad (15)$$

where δ is a small positive value to avoid dividing by zero; $\hat{\mathbf{f}}_{1,i}(k)$, $\hat{\mathbf{g}}_i(k)$ are the estimated feedback path, the estimated FIR filter at time k in the i^{th} sub-band, $i = 0, 1, \dots, M-1$, respectively. The inputs of the i^{th} adaptive filter in the first

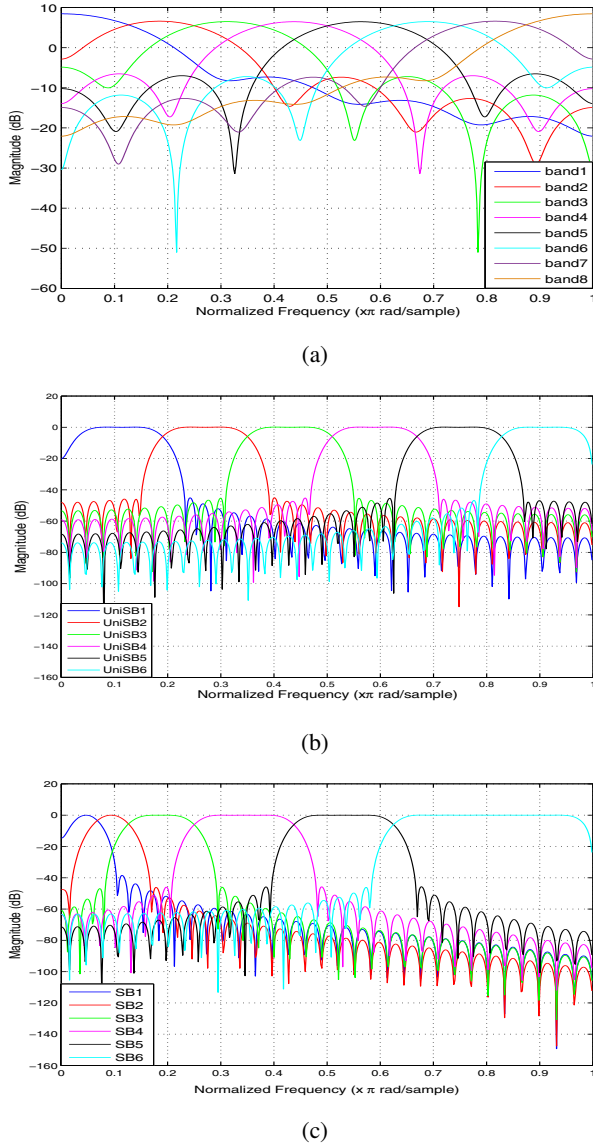


Fig. 2: Frequency responses of (a) uniform DCT-modulated filters, $M = 8$; (b) uniform sub-band filters (UniSB), $M = 6$; (c) non-uniform sub-band filters (MOB), $M = 6$.

set (AF_1) and the second set (AF_2) are denoted as follows, respectively.

$$\mathbf{y}_{\mathbf{h}_i}(k) = [y_{\mathbf{h}_i}(k), y_{\mathbf{h}_i}(k-1), \dots, y_{\mathbf{h}_i}(k-L_{\hat{f}}+1)]^T \quad (16)$$

$$\mathbf{m}_{\mathbf{h}_i}(k) = [m_{\mathbf{h}_i}(k), m_{\mathbf{h}_i}(k-1), \dots, m_{\mathbf{h}_i}(k-L_{\hat{g}}+1)]^T \quad (17)$$

In addition, the error signal $\mathbf{e}_{\mathbf{h}_i}(k)$ which is used to control the update of the adaptive filters AF_1 and AF_2 is defined as

$$\mathbf{e}_{\mathbf{h}_i}(k) = [e_{\mathbf{h}_0}(k), e_{\mathbf{h}_1}(k), \dots, e_{\mathbf{h}_{M-1}}(k)]^T \quad (18)$$

The sub-band filters with length of L have impulse responses $\mathbf{h}_i(k)$ as described in (19).

$$\mathbf{h}_i = [h_{i,0}, h_{i,1}, \dots, h_{i,L-1}]^T \quad (19)$$

III. SIMULATION RESULTS

The simulations are implemented for real speech signal which is constructed by concatenating a total of 80s of real male and female speech patterns extracted from Noizeus database [21]. The feedback paths are measured for both cases of normal and closest feedback paths. In the former case the hearing aid fits into the ear, and in the latter case a flat object is placed close to the ear ($< 1\text{cm}$). Moreover, the incoming signals are recorded at two microphones. The performance of all system models is evaluated based on two terms of Misalignment (MisAL) and Maximum Stable Gain (MSG) which are defined in frequency domain [10] as follows

$$\text{MisAL} = 20 \log_{10} \frac{\int_0^\pi |F(\omega) - e^{-j\omega d_{fb}} \hat{F}(\omega)|^2 d\omega}{\int_0^\pi |F(\omega)|^2 d\omega} \quad (20)$$

$$\text{MSG} = -20 \log_{10} \left(\max_{\omega} |F(\omega) - e^{-j\omega d_{fb}} \hat{F}(\omega)| \right) \quad (21)$$

where d_{fb} represents a delay in the feedback canceller's path.

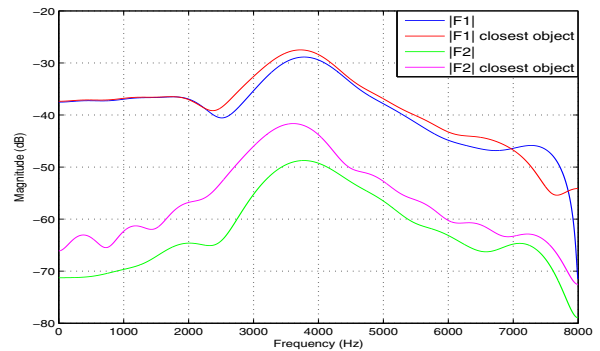
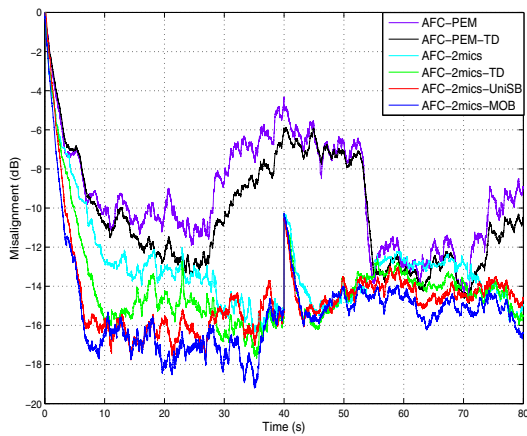


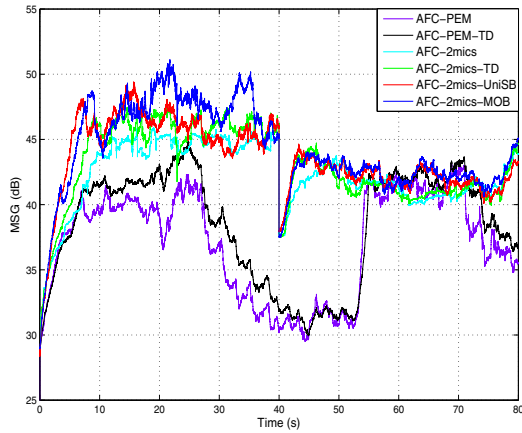
Fig. 3: Feedback characteristics

A classic Normalized Least Mean Square (NLMS) algorithm is employed to all adaptive filters. The following parameters are chosen for all system models: $|K| = 30\text{dB}$, $d_k = 32$ samples, $d_{fb} = 16$ samples, $L_{\hat{f}} = 22$, sampling frequency $f_s = 16\text{kHz}$ and the length of true feedback path $L_f = 38$. In the AFC-PEM and AFC-PEM-TD methods the Levinson-Durbin algorithm with subsequent frame of 160 samples is utilized in order to predict the coefficients of a 21-order AR model of the incoming signal [2]. In AFC-2mics and its modifications $L_{\hat{g}} = 10$, and $d_m = 1$ are selected.

Two scenarios with a hearing aid mock-up located in the right ear are evaluated for all system models. In the first scenario, the feedback path changes from normal to the closest feedback paths at the 40th second, the speech source position is in front of the human face (0 degree). In the second scenario, the feedback paths keep unchanged as normal feedback paths for whole 80 seconds, but the speech source position moves clockwise (CW) from 0 degree (in the first 40 seconds) to 180 degree (in the second 40 seconds). The step sizes μ used in



(a) Misalignment

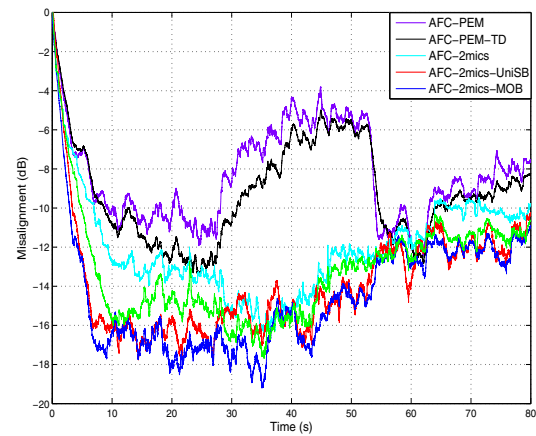


(b) MSG

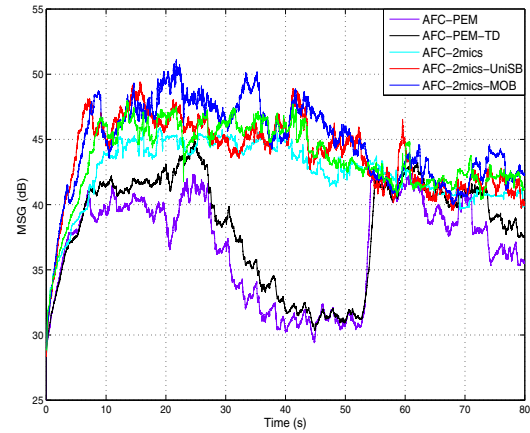
Fig. 4: Performance of AFC-2mics, AFC-2mics-TD, AFC-2mics-UniSB, AFC-2mics-MOB and AFC-PEM, AFC-PEM-TD when the feedback path changes at 40th second, speech source position is 0dgs

different approaches are selected such that all performances have the same initial convergence rates. In addition, the same step size is used for both estimations of the first feedback path $\hat{F}_1(q)$ and FIR filter $\hat{G}(q)$ in each system model. For instance, in our simulations $\mu = 0.0005$, $\mu = 0.00005$ and $\mu = 0.001$ are chosen for AFC-PEM, AFC-PEM-TD, AFC-2mics, respectively, whereas AFC-2mics-TD, AFC-2mics-UniSB and AFC-2mics-MOB use the same step sizes $\mu = 0.0001$.

Fig. 4 presents the misalignment and MSG of all system models mentioned above for the first scenario. It is clear that the AFC-PEM and AFC-PEM-TD are sensitive to the input signal, especially when the input signal is female speech (26 s-52 s and 68 s-80 s). The reason is that female speech seems to be more difficult to pre-whiten than male speech signal, resulting in larger bias in identification process of the feedback path. However, all AFC methods using two microphones work well with varying input signals. From the simulations it can be



(a) Misalignment



(b) MSG

Fig. 5: Performance of AFC-2mics, AFC-2mics-TD, AFC-2mics-UniSB, AFC-2mics-MOB and AFC-PEM, AFC-PEM-TD when the feedback path is fixed and speech source moves from 0dgs to 180dgs

seen that they outperform the AFC-PEM and AFC-PEM-TD. By using transform domain processing with uniform DCT-modulated filter banks the AFC-2mics-TD obtains much better performance than AFC-2mics. The proposed AFC-2mics-UniSB and AFC-2mics-MOB not only outperform the AFC-2mics, but also provides further performance improvement compared to the AFC-2mics-TD for both MisAL and MSG. This is due to the increased separation of the different frequency bands. In fact, the two proposed methods obtain much higher convergence rate than any above mentioned method. Furthermore, AFC-2mics-UniSB provides up to 2 dB gain improvement in steady-state MSE compared to AFC-2mics-TD, although its performance has a small degradation when the incoming signal is a female speech signal. Besides, AFC-2mics-MOB achieves further improvement in MisAL (up to 4 dB for the case of normal feedback path and up to 2 dB for the case of closest feedback path) and MSG (up to 4 dB

TABLE I: Normalized average power estimates of loudspeaker signals and error signals after filter banks

$\bar{p}_{y_i,TD}$	1, 0.5015, 0.1672, 0.0912, 0.0456, 0.0304, 0.0182, 0.0122
$\bar{p}_{y_i,UniSB}$	1, 0.1333, 0.0667, 0.0222, 0.0222, 0
$\bar{p}_{y_i,MOB}$	1, 0.4643, 0.6429, 0.1786, 0.0714, 0.0714
$\bar{p}_{e_i,TD}$	1, 0.8627, 0.7074, 0.8223, 0.5555, 0.3744, 0.2624, 0.1675
$\bar{p}_{e_i,UniSB}$	1, 0.3568, 0.6511, 0.2530, 0.2028, 0.0837
$\bar{p}_{e_i,MOB}$	0.7162, 0.6065, 1, 0.9472, 0.9638, 0.8602

gain for both cases) compared to AFC-2mics-TD and it also works well with both male and female speech signals. The reason is that the AFC-2mics-MOB system is designed such that more narrow subbands located in low frequency range and few large subbands located in high frequency range. Therefore, the input signal is separated well in low frequency range where most energy of speech signal is concentrated, resulting in less correlation among those subbands.

The normalized average power estimates from maximum values among sub-bands of loudspeaker and error signals after filter banks for three latter methods AFC2mics-TD, AFC2mics-uniSB and AFC2mics-MOB are presented in Table. I. It can be seen that in the AFC2mics-MOB method both loudspeaker signal $y(n)$ and the error signal $e(n)$ have more equalized average powers per bands than those in AFC2mics-TD and AFC2mics-uniSB methods. Therefore, the update equations in Algorithm 1 have more equalized gradients over bands in case of AFC2mics-MOB method, resulting in the better convergence rate.

Fig. 5 shows the performance of all considered system models for the second scenario. It can be seen that when the speech source moves CW to position of 180° the performance of all above system models degrades significantly. However, the proposed approaches still outperform AFC-PEM and AFC-PEM-TD and have better performance (up to 2 dB gain) than AFC-2mics and AFC-2mics-TD methods.

IV. CONCLUSION

The paper has introduced two new approaches for acoustic feedback cancellation in hearing aids. These approaches employ sub-band filters either uniform or non-uniform to have better band separation compared to transform domain methods. Consequently, a further decorrelation of the speech signals in different frequency bins is obtained. Furthermore, by employing a non-uniform filter bank the adaptive filters in each band will experience better equalized signal inputs. Thus, the errors will have similar weighting, providing a smaller misalignment and higher maximum stable gain. Simulation results have shown that the proposed methods have much better performance than previously suggested methods.

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