ASSESSING THE ACOUSTIC FEEDBACK CONTROL PERFORMANCE OF ADAPTIVE FEEDBACK CANCELLATION IN SOUND REINFORCEMENT SYSTEMS

Toon van Waterschoot and Marc Moonen

Dept. E.E./ESAT, SCD-SISTA, Katholieke Universiteit Leuven
Kasteelpark Arenberg 10, B-3001 Leuven, Belgium
phone: +32 16 321709, fax: +32 16 321970
email: {tvanwate,moonen}@esat.kuleuven.be
web: http://homes.esat.kuleuven.be/˜tvanwate

ABSTRACT

Adaptive feedback cancellation (AFC) is considered to be a promising solution to the acoustic feedback problem in sound reinforcement systems. A fundamental problem in AFC is related to the closed-loop nature of a sound reinforcement system, which results in a considerable signal correlation between the far-end and near-end signal. To avoid a biased and slowly converging feedback path estimation, the AFC approach is usually realized by combining an adaptive filter with a decorrelation method. In the AFC literature, decorrelation methods have only been evaluated w.r.t. the resulting adaptive filter misadjustment, and moreover, few results are available concerning the proper choice of the decorrelation parameters. In this paper, results of a comparative evaluation of existing decorrelation methods are reported, in terms of two measures that actually determine the acoustic feedback control performance, namely the maximum stable gain (MSG) increase and the sound quality. It appears that the choice of the decorrelation method and its parameters has a profound influence on these performance measures. Moreover, when decorrelation is applied in the closed signal loop, a trade-off between the resulting MSG increase and sound quality is unavoidable.

1. INTRODUCTION

The acoustic feedback problem is a long-standing problem in sound reinforcement systems. When a sound signal is captured by a microphone, and subsequently amplified and played back through a loudspeaker, the loudspeaker sound is often fed back to the microphone either through a direct acoustic coupling or indirectly as a consequence of reverberation. The existence of such an acoustic feedback path results in a closed signal loop, which limits the performance of a sound reinforcement system in two ways. First of all, there is an upper limit to the amount of amplification that can be applied if the system is required to remain stable, which is referred to as the maximum stable gain (MSG). Second, the sound quality is affected by occasional howling when the MSG is exceeded, or, even when the system is operating below the MSG, by ringing and excessive reverberation.

This research work was carried out at the ESAT laboratory of Katholieke Universiteit Leuven, in the frame of K.U.Leuven Research Council: CoE EF/05/006 Optimization in Engineering (OPTEC) and the Belgian Programme on Interuniversity Attraction Poles initiated by the Belgian Federal Science Policy Office IUAP P6/04 (DYSCO, 'Dynamical systems, control and optimization', 2007-2011), and the Concerted Research Action GOA-AMBioRICS, and was supported by the Institute for the Promotion of Innovation through Science and Technology in Flanders (IWT-Vlaanderen). The scientific responsibility is assumed by its authors.

State-of-the-art methods for acoustic feedback control can be categorized into four classes [1]: phase modulation methods, gain reduction methods, spatial filtering methods, and room modeling methods. Adaptive feedback cancellation (AFC) is a room modeling method, in which the acoustic feedback path is modeled using an adaptive finite impulse response (FIR) filter. In this way, the feedback signal can be predicted and subtracted from the microphone signal. While gain reduction methods have long time been the most widespread solution to acoustic feedback in sound reinforcement systems, the AFC approach has recently gained a lot of attention due to its successful application in hearing aids (see [2] and references therein). A fundamental problem encountered in AFC is the signal correlation between the far-end and near-end signal, which leads to biased and high-variance acoustic feedback path estimates when standard least-squares(LS)-based adaptive filtering algorithms are used. For this reason, the AFC approach usually entails a decorrelation method to reduce the far-end to near-end signal correlation. Decorrelation can be performed either in the closed signal loop, by injecting a noise signal [3]-[5], including a nonlinear [5] or time-varying [5]-[7] signal operation, or inserting a processing delay, or in the adaptive filtering circuit, by having the adaptive filter preceded by a processing delay [8],[9] or a pair of decorrelating prefilters [10]-[13].

The purpose of this paper is to present an evaluation of the acoustic feedback control performance that can be achieved with the AFC approach and different decorrelation methods. In the literature, the AFC performance is typically quantified in terms of the adaptive filter misadjustment [10]-[13]. While the misadjustment is an effective measure to quantify the impulse response mismatch between the true and estimated acoustic feedback path, it hardly provides information on the acoustic feedback control performance in terms of MSG increase and sound quality. Moreover, a comparative evaluation of different decorrelation methods has not yet been reported. Finally, each decorrelation method is governed by one or more parameters, yet the influence of different parameter values has not been studied in detail. These three issues are addressed in the evaluation presented in this paper, i.e., to quantify the AFC performance in terms of the MSG increase and sound quality, to compare different decorrelation methods, and to study the influence of the decorrelation parameter values. To this end, the AFC concept is briefly explained in Section 2 and the existing decorrelation methods are reviewed in Section 3. Then, in Section 4, results of a comparative AFC evaluation with different decorrelation methods and decorrelation parameter values are presented.

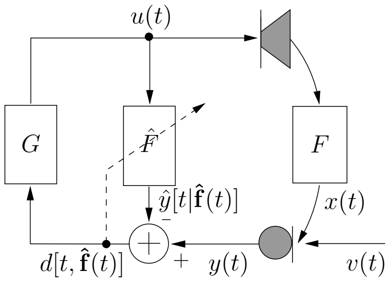

Figure 1: Adaptive feedback cancellation (AFC) concept.

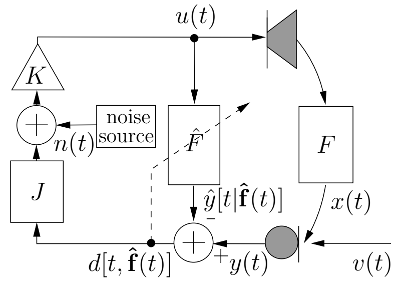

Figure 2: AFC with decorrelation by noise injection.

Finally, Section 5 concludes the paper.

2. ADAPTIVE FEEDBACK CANCELLATION

In a sound reinforcement system, the microphone signal $y(t)$ consists of a near-end signal component $v(t)$ and a feedback signal component $x(t)$. The AFC approach is aimed at predicting the feedback signal component and then subtracting this prediction from the microphone signal. The predicted feedback signal, denoted as $\hat{y}[t,\hat{\mathbf{f}}(t)]$, is obtained by filtering the far-end signal $u(t)$ with a model $\hat{F}(q,t)$ of the acoustic feedback path, see Fig. 1, where q denotes the time shift operator, i.e., $q^{-k}u(t) = u(t-k)$. This model is calculated using an adaptive filter, that is designed to identify the feedback path impulse response $\mathbf{f}(t)$ and track its changes. The feedback path and adaptive filter impulse responses are defined at time t as

$$\mathbf{f}(t) = \begin{bmatrix} f_0(t) & f_1(t) & \dots & f_{n_F}(t) \end{bmatrix} \quad (1)$$
$$\hat{\mathbf{f}}(t) = \begin{bmatrix} \hat{f}_0(t) & \hat{f}_1(t) & \dots & \hat{f}_{n_{\hat{F}}}(t) \end{bmatrix} \quad (2)$$

respectively. We will further assume that $n_{\hat{F}} = n_F$.

The closed-loop frequency response of the system shown in Fig. 1 is given by

$$\frac{U(\omega,t)}{V(\omega,t)} = \frac{G(\omega,t)}{1 - G(\omega,t)[F(\omega,t) - \hat{F}(\omega,t)]} \quad (3)$$

where $U(\omega,t)$ and $V(\omega,t)$ denote the short-term far-end and near-end signal spectrum, and $G(\omega,t)$, $F(\omega,t)$, and $\hat{F}(\omega,t)$ denote the short-term electro-acoustic forward path, acoustic feedback path, and adaptive filter frequency response, respectively. From the Nyquist stability criterion [1], the following expression for the MSG can be derived,

$$\mathrm{MSG}(t)\ [\mathrm{dB}] = -20\log_{10}\left[\max_{\omega\in\mathscr{P}} |J(\omega,t)[F(\omega,t) - \hat{F}(\omega,t)]|\right]. \quad (4)$$

where $\mathscr{P}$ denotes the set of frequencies at which the loop phase is a multiple of 2π, and $J(\omega,t)$ denotes the forward path processing before the amplifier, i.e., $G(\omega,t) = J(\omega,t)K(t)$ with $K(t)$ the amplifier gain. From (4), it immediately follows that the better the fit between the estimated and actual feedback path frequency response, particularly at critical frequencies of the closed-loop system, the larger the achievable MSG increase.

While the concept of AFC is relatively simple and similar to the well-known acoustic echo cancellation (AEC) approach, its realization is not straightforward. In the identification of the acoustic feedback path model $\hat{F}(q,t)$, a fundamental problem appears which is due to the closed-loop nature of the system. The LS estimate of the acoustic feedback path impulse response $\mathbf{f}(t)$ can be shown to be biased due to the correlation between the far-end and near-end signal [1]. The resulting effect is that the adaptive filter does not only predict and cancel the feedback component in the microphone signal, but also (part of) the near-end signal component. As a consequence, the feedback-compensated signal $d[t,\hat{\mathbf{f}}(t)]$ is a distorted estimate of the near-end signal $v(t)$. Moreover, since the AFC has to operate in a continuous double-talk situation, the adaptive filter convergence may be extremely slow. For this reason, the AFC is typically combined with a decorrelation method.

3. DECORRELATION METHODS

3.1 Decorrelation in the closed signal loop

Decorrelation of the far-end and near-end signal can be achieved by inserting a decorrelating signal operation in the closed signal loop. Four such decorrelation methods have been proposed: noise injection, time-varying processing, nonlinear processing, and forward path delay.

3.1.1 Noise injection [3]-[5]

A white noise signal $n(t)$ is added to the feedback-compensated signal after the forward path processing (but before the forward path amplification), see Fig. 2, i.e.,

$$u(t) = K(t)\Big[J(q,t)d[t,\hat{\mathbf{f}}(t)] + n(t)\Big]. \quad (5)$$

The effect of the noise injection is that the far-end to near-end signal correlation is decreased, hence the bias will be reduced but not completely eliminated. With the aim of reducing the influence of the noise injection on sound quality, the noise spectrum can be shaped such as to render the noise less perceptible, e.g., by A-weighting [3] or psychoacoustic noise shaping [4]. Unfortunately, noise shaping decreases the decorrelation effect, making the noise injection less effective in removing the bias.

3.1.2 Time-varying processing [5]-[7]

Any linear time-varying filter (LTV) $H(q,t)$ can be used as a decorrelation device in the forward path, see Fig. 3, i.e.,

$$u(t) = G(q,t)\Big[H(q,t)d[t,\hat{\mathbf{f}}(t)]\Big]. \quad (6)$$

Frequency shifting (FS) is the most widely used LTV decorrelation method [6],[7]. An FS filter has an LTV frequency response $H(\omega,t) = e^{j\omega_m t}$, with ω_m the radial frequency shift,

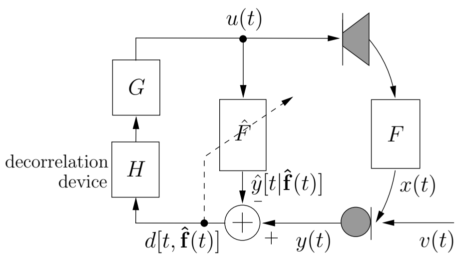

Figure 3: AFC with decorrelation by a time-varying/nonlinear/delay operation in the forward path.

and can be realized by operating on the analytical representation of the feedback-compensated signal $d[t,\hat{\mathbf{f}}(t)]$ [1]. While the perceptible signal distortion introduced by the FS operation appears to be acceptable for speech signals [7], the FS decorrelation technique was found to be perceptually inadequate for audio applications [4].

3.1.3 *Nonlinear processing [5]*

In the context of stereo AEC, the correlation between the stereo channels has been reduced by applying nonlinear decorrelating operations to the far-end signals. These nonlinear operations can also be used to reduce the far-end to near-end signal correlation in AFC. In particular, half-wave rectification has been applied to AFC decorrelation [5], i.e.,

$$u(t) = G(q,t)\left[d[t,\hat{\mathbf{f}}(t)] + \alpha\left(\frac{d[t,\hat{\mathbf{f}}(t)] + |d[t,\hat{\mathbf{f}}(t)]|}{2}\right)\right] \quad (7)$$

The parameter α can be tuned to trade off decorrelation and perceptible signal distortion.

3.1.4 *Forward path delay*

In hearing aid AFC applications [2], inserting a processing delay of d_1 samples in the electro-acoustic forward path has been proposed to decorrelate the far-end and near-end signal,

$$u(t) = G(q,t)d[t-d_1,\hat{\mathbf{f}}(t-d_1)]. \quad (8)$$

This approach is particularly useful for near-end signals that have an autocorrelation function that decays rapidly, e.g., voiceless speech signals, provided that the delay value d_1 is chosen accordingly.

3.2 Decorrelation in the adaptive filtering circuit

Decorrelation can also be applied in the adaptive filtering circuit, by inserting an adaptive filter delay or using decorrelating prefilters.

3.2.1 *Adaptive filter delay [8],[9]*

Due to the time needed for the loudspeaker sound to propagate through a direct coupling to the microphone, the acoustic feedback path impulse response typically exhibits an initial delay, the value of which is proportional to the loudspeaker-microphone distance. If this initial delay (or a lower bound for it) is known a priori and corresponds to d_2T_s s with T_s the sampling interval, then the first d_2 coefficients in the acoustic feedback path model can be forced to zero,

$$\hat{F}(q,t) = \hat{f}_{d_2}(t)q^{-d_2} + \hat{f}_{d_2+1}(t)q^{-(d_2+1)} + \ldots + \hat{f}_{n_{\hat{F}}}q^{-n_{\hat{F}}}.$$

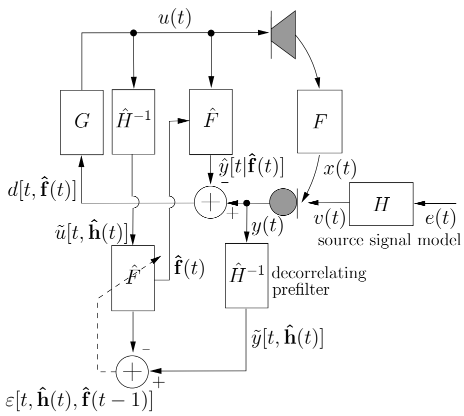

Figure 4: AFC with decorrelating prefilters in the adaptive filtering circuit.

If the far-end and near-end signal cross-correlation function is small for time lags larger than d_2 samples, then the remaining bias can be considered negligible.

3.2.2 *Decorrelating prefilters [10]-[13]*

From a system identification point of view, the bias in the LS estimate of the acoustic feedback path model can be eliminated by using an appropriate near-end signal model in the identification. Assuming a (time-varying) parametric near-end signal model $H(q,t)$,

$$v(t) = H(q,t)e(t) \quad (9)$$

the unbiased identification approach consists in prefiltering the far-end and microphone signals with an estimate $\hat{H}^{-1}(q,t)$ of the inverse near-end signal model before feeding these signals to the adaptive filtering algorithm. This approach is depicted in Fig. 4, where the prefiltered far-end and microphone signals are calculated as

$$\tilde{y}[t,\hat{\mathbf{h}}(t)] = \hat{H}^{-1}(q,t)y(t) \quad (10)$$
$$\tilde{u}[t,\hat{\mathbf{h}}(t)] = \hat{H}^{-1}(q,t)u(t) \quad (11)$$

and $\hat{\mathbf{h}}(t)$ contains the parameters of $\hat{H}^{-1}(q,t)$.

The concurrent estimation of the near-end signal model and the acoustic feedback path model can be performed using a prediction-error-method(PEM)-based AFC algorithm as proposed in [12]-[13].

4. EVALUATION

The evaluation is based on AFC simulations with speech and audio signals, at a sampling frequency $f_s = 16$ kHz and $f_s = 44.1$ kHz, respectively. After the initial AFC convergence, the amplifier gain $K(t)$ is raised to 7 dB above the MSG without AFC. The instantaneous MSG and sound quality is then measured during a time interval of 15 s (30 s in the audio simulation), in the middle of which a feedback path change is simulated corresponding to a 1 m microphone displacement. The acoustic feedback path impulse responses used in the simulation, were measured in a room with a reverberation time of 125 ms and truncated to $n_F = 2000$ ($n_F = 4410$ in the audio simulation). An NLMS adaptive algorithm is used with a step size parameter 0.02 for speech and 0.005 for audio. The performance measures used are the mean MSG increase (ΔMSG), defined as the difference of the instantaneous MSG averaged over time with the MSG without AFC, and the mean frequency-weighted log-spectral signal distortion (SD), defined in [1],[14] as an AFC sound quality measure. These measures are plotted for the different decorrelation methods as a function of the corresponding decorrelation parameter in Figs. 5 and 6.

For *noise injection*, the decorrelation parameter $\mathrm{SNR} = 10\log_{10}[\sum_t v^2(t)/\sum_t n^2(t)]$ takes on the values $\{-2.5, 0, 2.5, 5, 7.5, 10\}$ dB. Noise injection delivers the largest MSG increase of all decorrelation methods, but the worst sound quality. For audio, SNR = 0 dB appears to yield the best trade-off between MSG increase and sound quality, while a trade-off SNR value is more difficult to find for speech. Decorrelation by *time-varying processing* is achieved by applying an FS operation with $f_m = \{1, 3, 5, 10, 15, 20\}$ Hz. The sound quality increases monotonically with increasing f_m, while the MSG does not vary too much for different f_m values. A reasonable MSG increase and sound quality are obtained for frequency shift values $f_m \leq 10$ Hz. When including *nonlinear processing* by half-wave rectification, the decorrelation is governed by the parameter α (see (7)), with values $\{0.001, 0.005, 0.01, 0.05, 0.1, 0.5\}$. The corresponding acoustic feedback control performance appears to be extremely poor. Decorrelation by inserting a *forward path delay* d_1 or an *adaptive filter delay* d_2 has been evaluated with $d_{1,2} = \{0.3125, 0.625, 1.25, 2.5, 5, 10\}$ ms for speech and $d_{1,2} = \{0.7256, 1.4512, 2.9025, 5.805, 11.61, 23.22\}$ ms for audio. Both methods perform reasonably well for speech, but poorly for audio. The optimal delay value for speech is in the range 1–5 ms. *Decorrelating prefilters* consisting of a cascade of a pitch prediction model and an all-pole model [12],[13] were evaluated for different all-pole model orders $n_C = \{5, 10, 15, 20, 25, 30\}$. The resulting MSG increase is relatively high, and the sound quality is the best among all decorrelation methods. The acoustic feedback control performance appears to be quasi independent of n_C.

5. CONCLUSION

The acoustic feedback control performance of the AFC approach has been evaluated for different decorrelation methods and parameters in a sound reinforcement scenario. The achievable MSG increase and sound quality have been compared in simulations with speech and audio signals. Decorrelation by noise injection results in the highest MSG increase at the cost of sound quality. Including a frequency shift of $f_m \leq 10$ Hz yields a reasonable acoustic feedback control performance, while a nonlinear operation such as half-wave rectification appears unsuited for AFC. Including a forward path or adaptive filter delay of 1–5 ms is appropriate for speech but not for audio. Finally, the use of decorrelating prefilters should be preferred from a sound quality point of view, and moreover results in a relatively high MSG increase.

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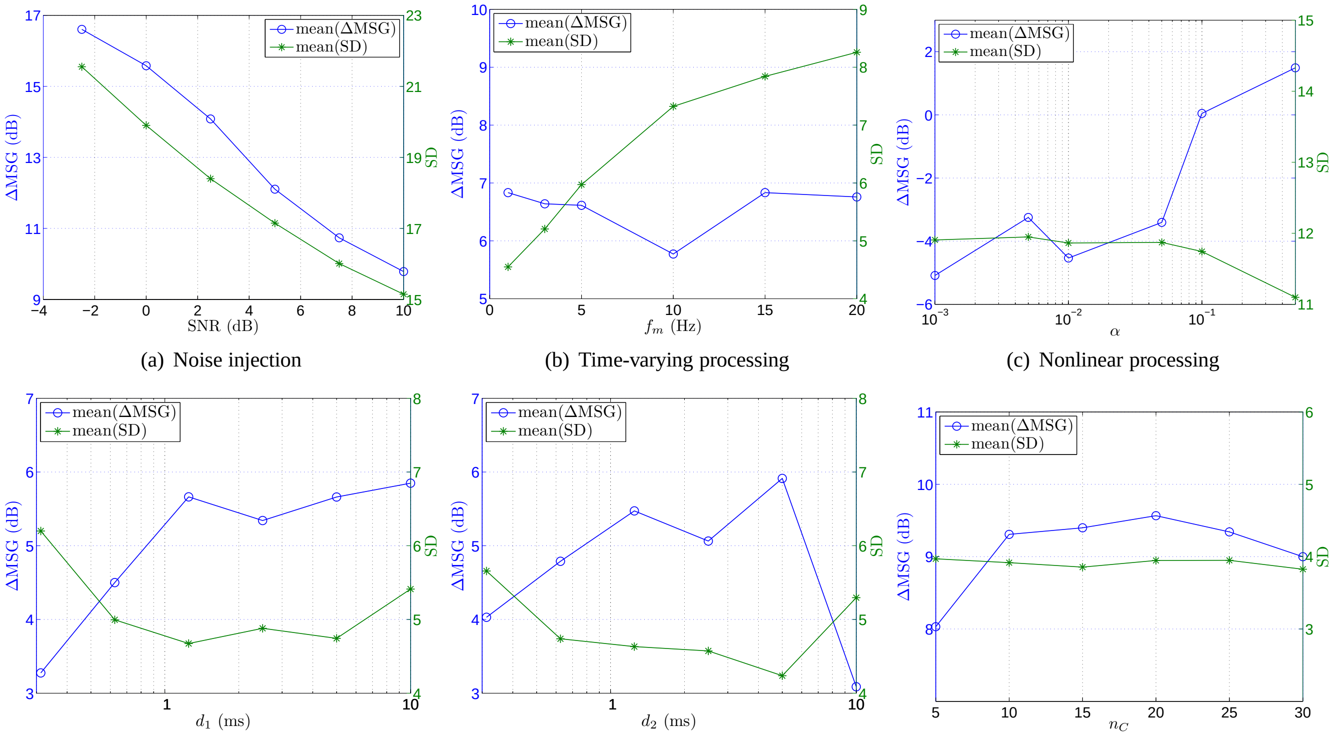
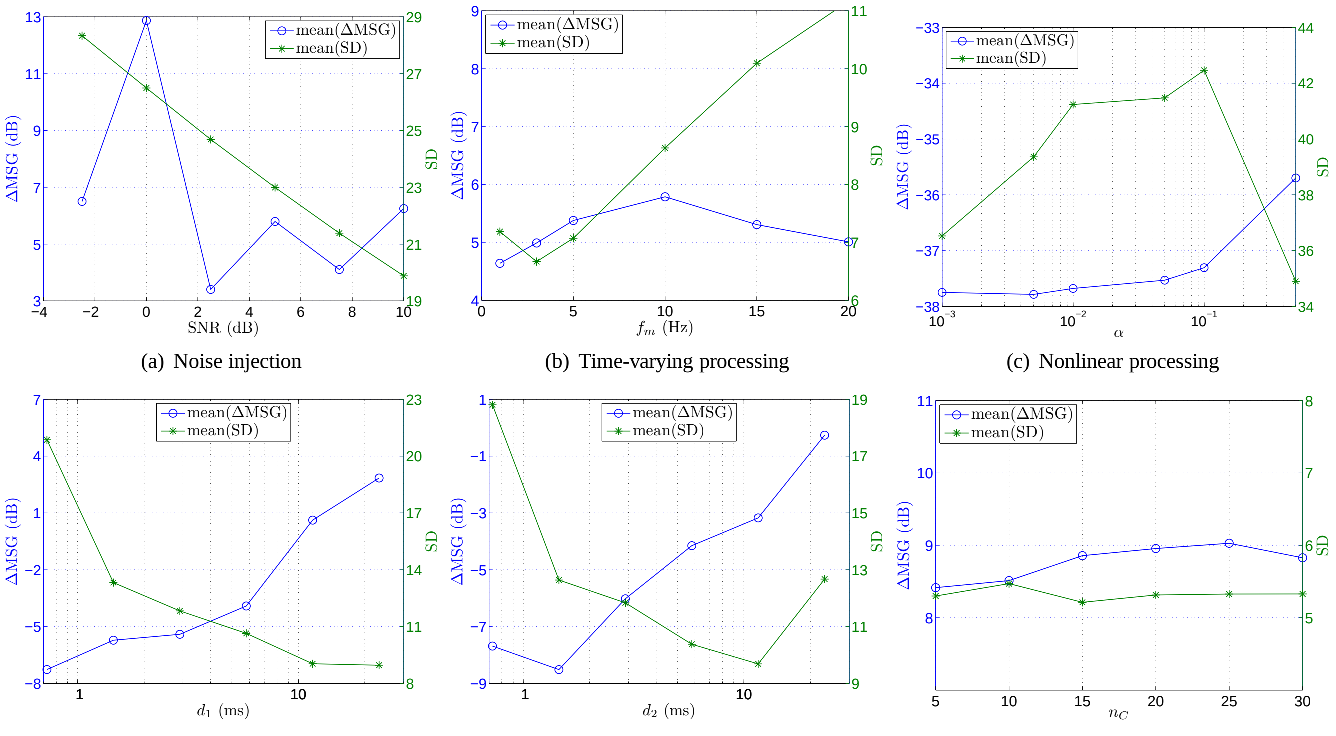


(a) Noise injection (b) Time-varying processing (c) Nonlinear processing

(d) Forward path delay (e) Adaptive filter path delay (f) Decorrelating prefilters

Figure 5: Mean MSG increase and SD values vs. decorrelation parameter values: speech simulations.



(a) Noise injection (b) Time-varying processing (c) Nonlinear processing

(d) Forward path delay (e) Adaptive filter path delay (f) Decorrelating prefilters

Figure 6: Mean MSG increase and SD values vs. decorrelation parameter values: audio simulations.