On the performance of decorrelation by prefiltering for adaptive feedback cancellation in Public Address systems

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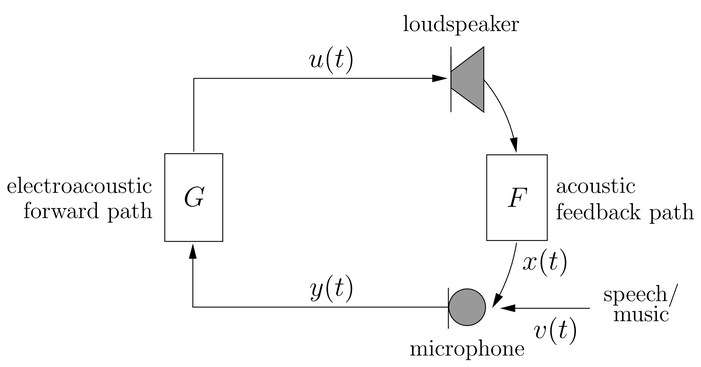
Abstract

The presence of an acoustic feedback path in Public Address (P.A.) systems limits the maximum allowable amplification if howling is to be avoided. Adaptive feedback cancellation (AFC) techniques aim at identifying the unknown feedback path, usually also by incorporating some form of decorrelation in the electroacoustic forward path. Recently, decorrelation by prefiltering was proposed for AFC in hearing aids, where the signals used for identification are filtered with the inverse source signal model. We apply this technique to closed-loop identification of room impulse responses. However, to obtain a useful estimate a much larger time window has to be used compared to the hearing aid case and the characteristics of the source signal may then vary significantly throughout the data set used. We propose a basis expansion technique to cope with this non-stationarity and we compare the proposed technique with an existing AFC method for P.A. systems.

1 Introduction

- Acoustic feedback problem: instability of closed-loop system if

$$\exists\omega: \begin{cases} |G(e^{i\omega})F(e^{i\omega})| \geq 1 \\ \angle G(e^{i\omega})F(e^{i\omega}) = n2\pi, n \in \mathbb{Z}. \end{cases} \quad (1)$$



- Approaches at feedback suppression:
  - smoothing the open-loop response $|G(e^{i\omega})F(e^{i\omega})|$ by equalization, notch filtering, modulation, ...
  - adaptive feedback cancellation (AFC): estimating the unknown feedback path transfer function $F(q,t)$ from microphone and loudspeaker signals $y(t)$ resp. $u(t)$

2 Adaptive Feedback Cancellation

- Closed-loop identification of the unknown feedback path impulse response:

$$\begin{bmatrix} u(t) & \dots & u(t-n_{\hat{F}}) \\ u(t-1) & \dots & u(t-n_{\hat{F}}-1) \\ \vdots & \ddots & \vdots \\ u(1) & \dots & u(-n_{\hat{F}}+1) \end{bmatrix} \cdot \begin{bmatrix} \hat{f}_0(t) \\ \vdots \\ \hat{f}_{n_{\hat{F}}}(t) \end{bmatrix} = \begin{bmatrix} y(t) \\ y(t-1) \\ \vdots \\ y(1) \end{bmatrix} \quad (2)$$

$$\Updownarrow$$

$$\mathbf{U}_{t\times(n_{\hat{F}}+1)}\hat{\mathbf{f}}(t)_{(n_{\hat{F}}+1)\times 1} = \mathbf{y}_{t\times 1} \quad (3)$$

yields the least squares estimate

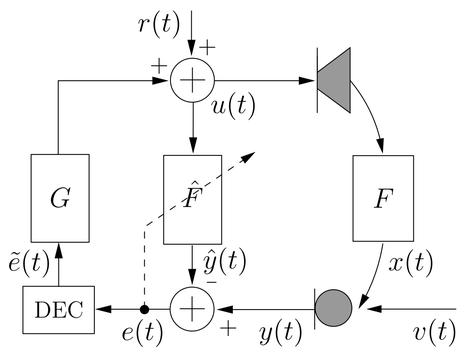
$$\hat{\mathbf{f}}_{\mathbf{LS}}(t) = (\mathbf{U}^T\mathbf{U})^{-1}\mathbf{U}^T\mathbf{y} \quad (4)$$

$$= \mathbf{f} + \underbrace{(\mathbf{U}^T\mathbf{U})^{-1}\mathbf{U}^T\mathbf{v}}_{\text{bias}}. \quad (5)$$

- For non-white source signals, the cross-correlation between $u(t)$ and $v(t)$ leads to a bias term $(\mathbf{U}^T\mathbf{U})^{-1}\mathbf{U}^T\mathbf{v}$ in the estimate, which does not disappear as $t \to \infty$.

2.1 Decorrelation in the forward path

- In existing adaptive feedback cancellation algorithms some decorrelating signal operation is typically included in the forward path, e.g.
  - a persistently exciting **probe signal** $r(t)$ is injected at the loudspeaker,
  - or an appropriate **delay** d is added to the forward path: $\tilde{e}(t) = e(t-d)$,
  - or a **nonlinearity** is added to the forward path, e.g.
    - a frequency shift: $\tilde{e}(t) = e(t)\cos(2\pi\Delta_f t) - \hat{e}(t)\sin(2\pi\Delta_f t)$
    - a phase modulation: $\tilde{e}(t) = e(t)e^{jk\sin\omega_m t}$
    - a half wave rectifier: $\tilde{e}(t) = e(t) + \alpha(e(t) + |e(t)|)$
- Making an optimal trade-off between decorrelation and audible signal distortion is often difficult.



2.2 Decorrelation in the identification algorithm

- From a system identification point of view, source signal $v(t) = H(q,t)w(t)$ acts as a noise signal to the identification. Prefiltering the loudspeaker and microphone signals with a time-varying filter $L(q,t)$:

$$\mathbf{L}_{t\times(t+n_L)}\mathbf{U}_{(t+n_L)\times(n_{\hat{F}}+1)}\hat{\mathbf{f}}(t) = \mathbf{L}_{t\times(t+n_L)}\mathbf{y}_{(t+n_L)\times 1} \quad (6)$$

with $\mathbf{L}$ a prefiltering matrix whose structure is defined by

$$\mathbf{L}_{t\times(t+n_L)} = \begin{bmatrix} l_0(t) & l_1(t) & \dots & l_{n_L}(t) & 0 & \dots & 0 \\ 0 & l_0(t-1) & \dots & l_{n_L-1}(t-1) & l_{n_L}(t-1) & \dots & 0 \\ \vdots & \vdots & \ddots & \vdots & \vdots & \ddots & \vdots \\ 0 & 0 & \dots & 0 & 0 & \dots & l_{n_L}(1) \end{bmatrix}$$

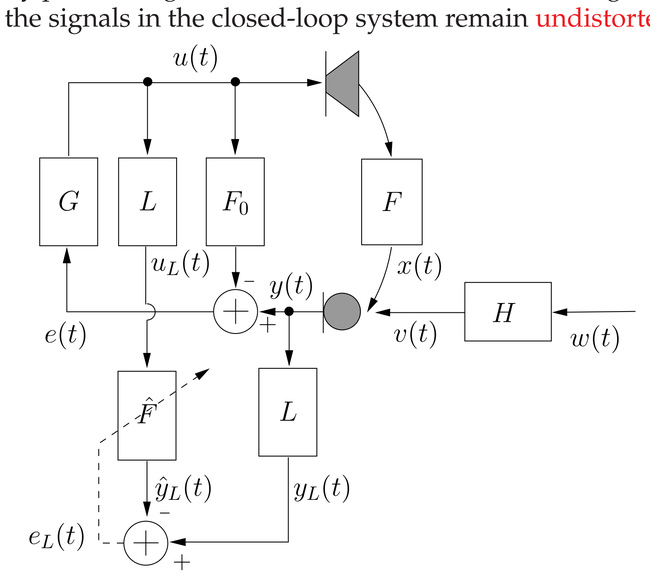
yields the least squares estimate

$$\hat{\mathbf{f}}_{\mathbf{LS},L}(t) = (\mathbf{U}^T\mathbf{L}^T\mathbf{L}\mathbf{U})^{-1}\mathbf{U}^T\mathbf{L}^T\mathbf{L}\mathbf{y} \quad (7)$$

$$= \mathbf{f} + \underbrace{(\mathbf{U}^T\mathbf{L}^T\mathbf{L}\mathbf{U})^{-1}\mathbf{U}^T\mathbf{L}^T\mathbf{L}\mathbf{v}}_{\text{bias}} \quad (8)$$

which is unbiased if $L(q,t)$ is a consistent estimate of the inverse source signal model $H^{-1}(q,t)$, $\forall t$.

- By performing decorrelation in the identification algorithm, the signals in the closed-loop system remain undistorted.



3 Time-Varying AR Modelling

- Audio signals are often modelled as autoregressive (AR) processes, with possibly time-varying coefficients:

$$H(q,t) = \frac{1}{1 + a_1(t-1)q^{-1} + \dots + a_p(t-p)q^{-p}} \quad (9)$$

- If the non-stationarity is ignored, a fixed (high-pass) prefilter $L(q)$ can be used to achieve some decorrelation. The bias term $(\mathbf{U}^T\mathbf{L}^T\mathbf{L}\mathbf{U})^{-1}\mathbf{U}^T\mathbf{L}^T\mathbf{L}\mathbf{v}$ will not disappear completely though.
- Concurrent estimation of $F(q)$ and $H^{-1}(q,t)$ from the closed-loop data $\{u(k), y(k)\}$ generally leads to the minimization of a non-linear cost function.
- Alternatively $H^{-1}(q,t)$ and $-H^{-1}(q,t)F(q)$ can be estimated in a two-channel adaptive filtering scheme. It is then assumed that $H^{-1}(q,t)$ does not vary significantly over the time window used. This is reasonable for hearing aid applications but not when dealing with room acoustics.
- By expanding the TVAR coefficients onto a set of basis functions:

$$a_i(t-i) = \sum_{l=0}^{m} a_{il} f_l(t-i) \quad i = 1, \dots, p, \quad (10)$$

a time-invariant parameter set

$$\theta_{H^{-1}} \triangleq \begin{bmatrix} a_{10} & \dots & a_{1m} & \dots & a_{p0} & \dots & a_{pm} \end{bmatrix}^T \quad (11)$$

can be used to describe the dynamics of the source signal over a longer time window. The problem is then reduced to concurrent estimation of $F(q)$ and $\theta_{H^{-1}}$ from the basis expanded data set $\{U(k), Y(k)\}$ with

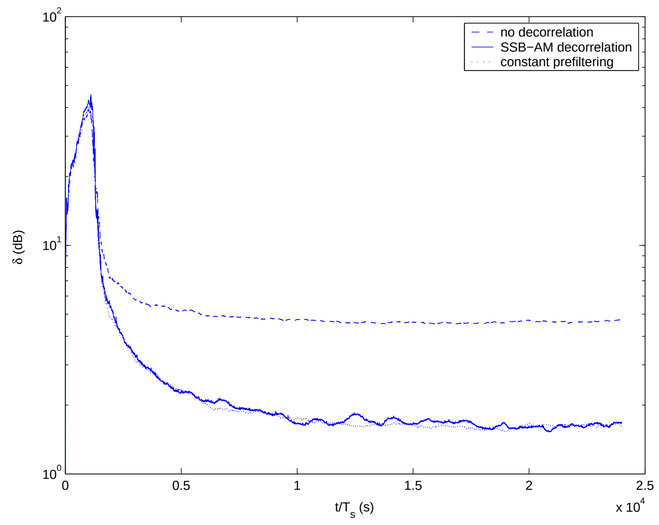
$$U(k) \triangleq \begin{bmatrix} f_0(k)u(k) \\ \vdots \\ f_m(k)u(k) \end{bmatrix}, \; Y(k) \triangleq \begin{bmatrix} f_0(k)y(k) \\ \vdots \\ f_m(k)y(k) \end{bmatrix}. \quad (12)$$

4 Simulation Results

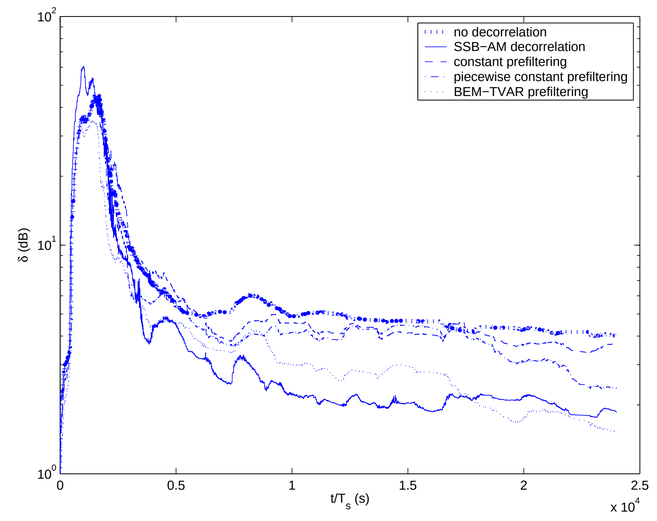
- MATLAB simulations with $f_s = 8kHz$, $F(q)$ a fixed room impulse response of length $n_F + 1 = 1000$, $G(q)$ the cascade of a unit time delay and a constant, linear amplification factor.
- Only closed-loop identification was performed, without substracting the estimated feedback signal from the microphone signal.
- An exponentially windowed RLS algorithm with forgetting factor $\lambda = 0.9997$ yields a time window of roughly $3(n_F + 1) = 3000$ samples.
- The following AFC variants have been compared:
  - no decorrelation
  - decorrelation by including an SSB-AM frequency shifter in the forward path (reference algorithm)
  - decorrelation by constant prefiltering ($L(q)$ fixed)
  - decorrelation by piecewise constant prefiltering ($L(q)$ fixed in non-overlapping frames of 3000 samples)
  - decorrelation by BEM-TVAR prefiltering ($L(q,t)$ calculated from $\theta_{H^{-1}}$ with $\theta_{H^{-1}}$ fixed in non-overlapping frames of 3000 samples)
- In the algorithms applying prefiltering, source signal $v(t)$ was assumed to be known and AR model order $p = 12$. In practice only an estimate of $v(t)$ is available through the error signal $e(t)$.
- Performance measure: normalized bias

$$\delta(t) = 20\log_{10}\frac{\|\hat{\mathbf{f}}_{\mathbf{LS}}(t) - \mathbf{f}\|}{\|\mathbf{f}\|} \quad (13)$$

- Convergence curves for a speech-shaped stationary p-th order AR source signal:



- Convergence curves for a true speech source signal:



AFC	Acoustic Feedback Cancellation
BEM	Basis Expansion Method
P.A.	Public Address
RLS	Recursive Least Squares
SSB-AM	Single Sideband Amplitude Modulation
(TV)AR	(Time-Varying) Autoregressive