

Acoustic feedback control in sound reinforcement systems*

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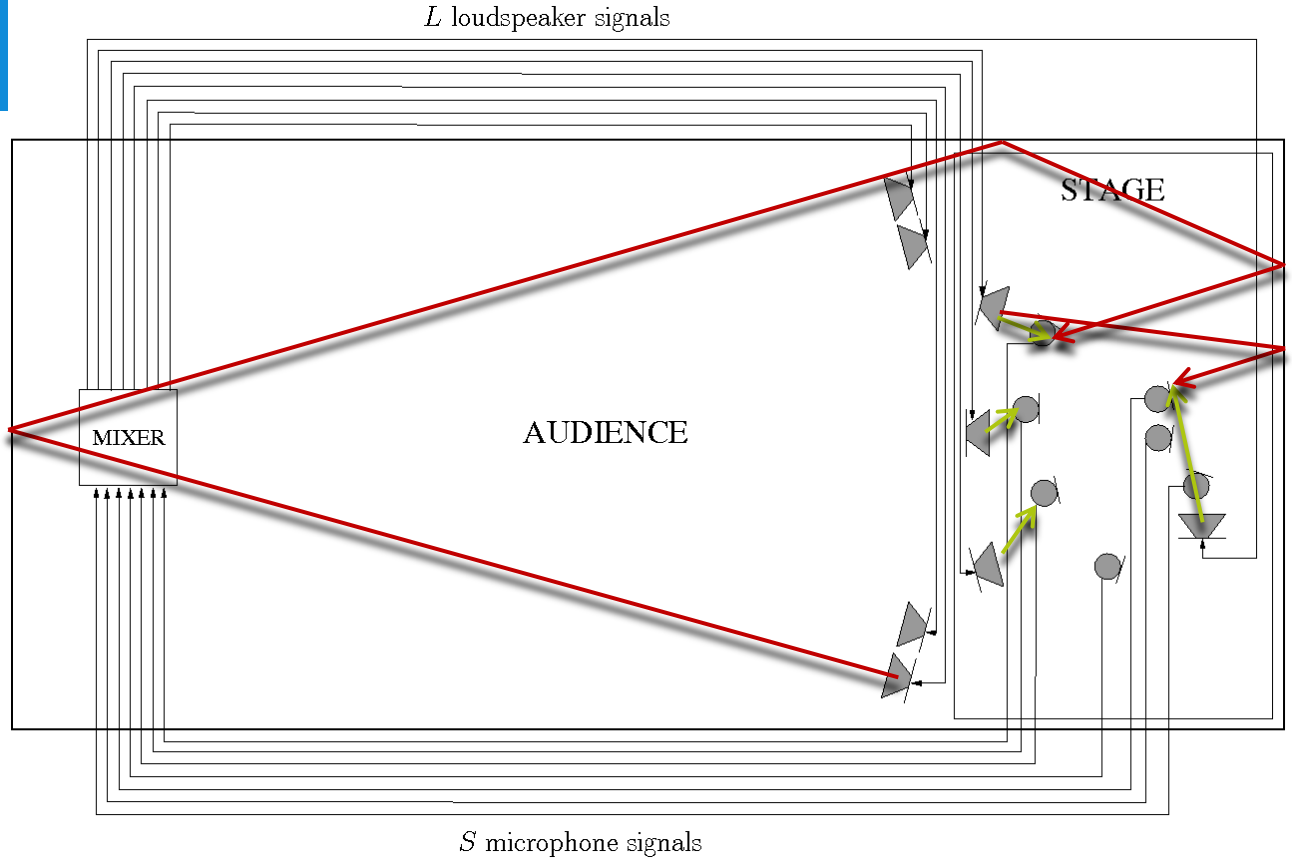
*Joint work with Marc Moonen (ESAT-SCD, Katholieke Universiteit Leuven, BE)

Outline

- Introduction
  - sound reinforcement systems
  - acoustic feedback
- Acoustic feedback control
- Notch-filter-based howling suppression (NHS)
  - introduction
  - howling detection
  - notch filter design
  - simulation results
- Adaptive feedback cancellation (AFC)
  - introduction
  - closed-loop signal decorrelation
  - adaptive filter design
  - simulation results
- Conclusion & open issues

Introduction (1)

Sound reinforcement systems (1)

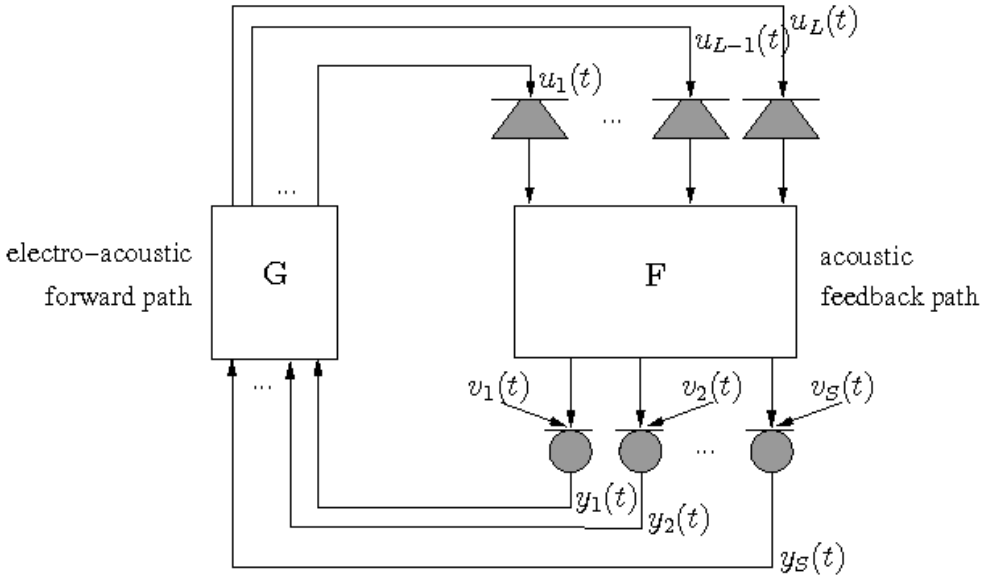


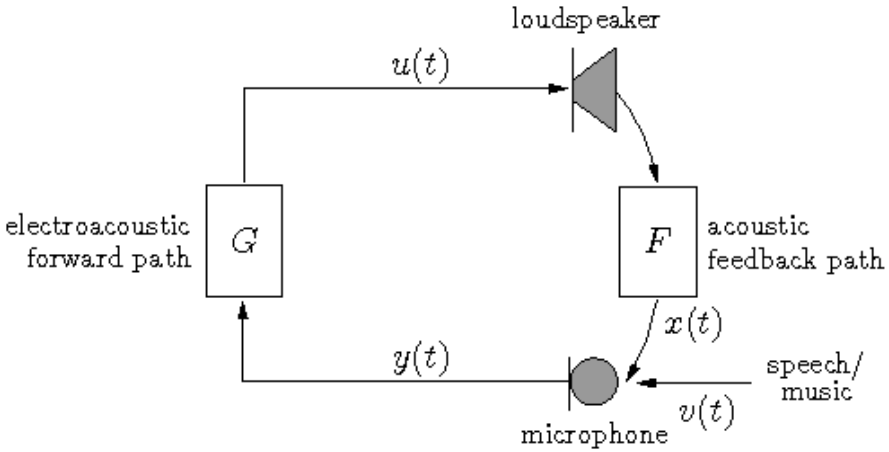
- sound sources
- microphones
- mixer & amp
- loudspeakers
- monitors
- room
- audience

Goal: to deliver sufficiently high sound level and best possible sound quality to audience

Introduction (2)

Sound reinforcement systems (2)



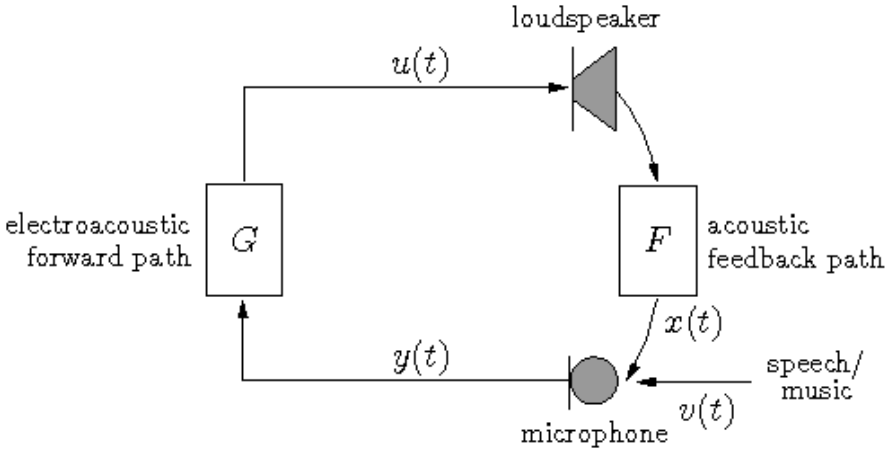


We will mostly restrict ourselves to the **single-loudspeaker-single-microphone** case

Introduction (3)

Sound reinforcement systems (3)

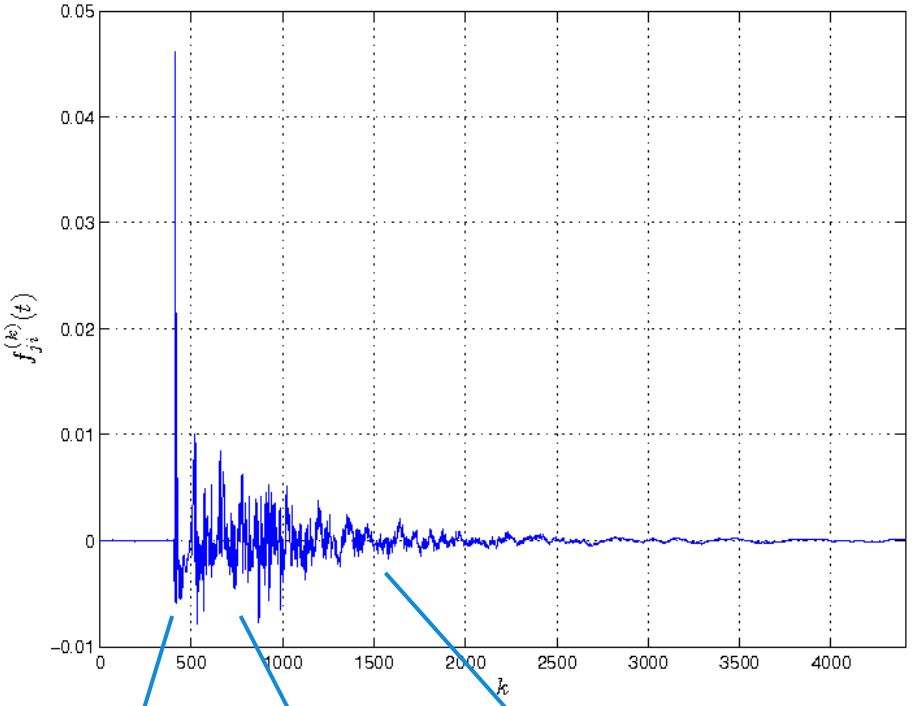
- Assumptions (for now):
  - loudspeaker has linear & flat response
  - microphone has linear & flat response
  - forward path (amp) has linear & flat response
  - acoustic feedback path has linear response
- But: **acoustic feedback path has non-flat response**



Introduction (4)

Sound reinforcement systems (4)

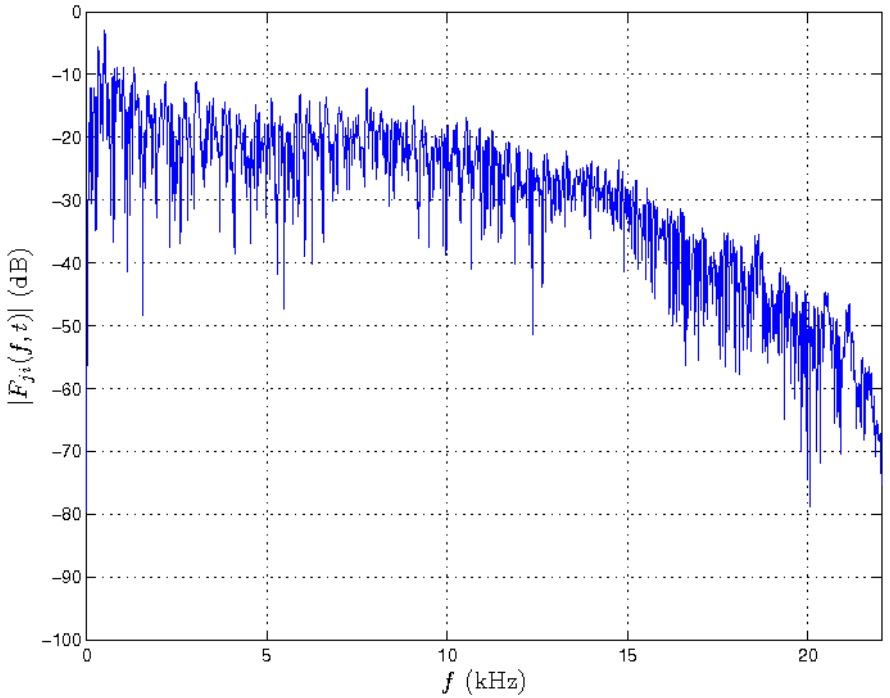
- Acoustic feedback path response: example room (3 x 3 x 4 m)



direct coupling

early reflections

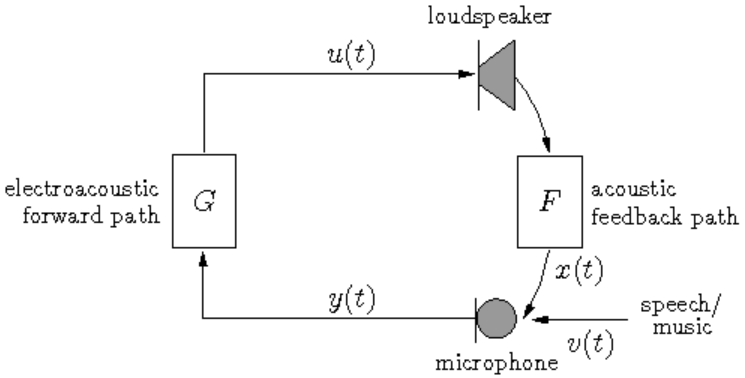
diffuse sound field



peaks/dips = anti-nodes/nodes of standing waves
peaks ~10 dB above average, and separated by ~10 Hz

Introduction (5)

Acoustic feedback problem (1)



- Nyquist stability criterion:
  - if there exists a radial frequency ω for which

$$\begin{cases} |G(\omega,t)F(\omega,t)| & \geq & 1 \\ \angle G(\omega,t)F(\omega,t) & = & n2\pi, \quad n \in \mathbb{Z} \end{cases}$$

    then the closed-loop system is **unstable**
  - if the unstable system is excited at the critical frequency ω, then an oscillation at this frequency will occur **= howling**
- Maximum stable gain (MSG):
  - maximum forward path gain before instability

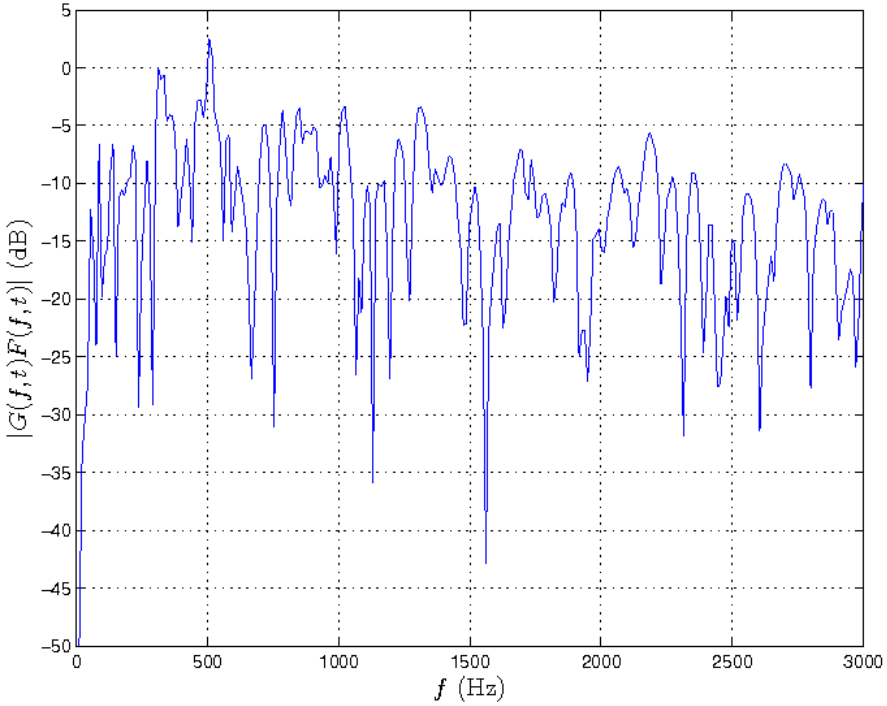
$$\begin{aligned} \mathrm{MSG}(t) &= -20\log_{10}\left[\max_{\omega\in\mathcal{P}}|F(\omega,t)|\right] && \text{(if } G \text{ has flat response)} \\ &\approx -10\log_{10}\left[\log_{10}(BT_{60}/22)\right] - 3.8 && \text{[Schroeder, 1964]} \end{aligned}$$

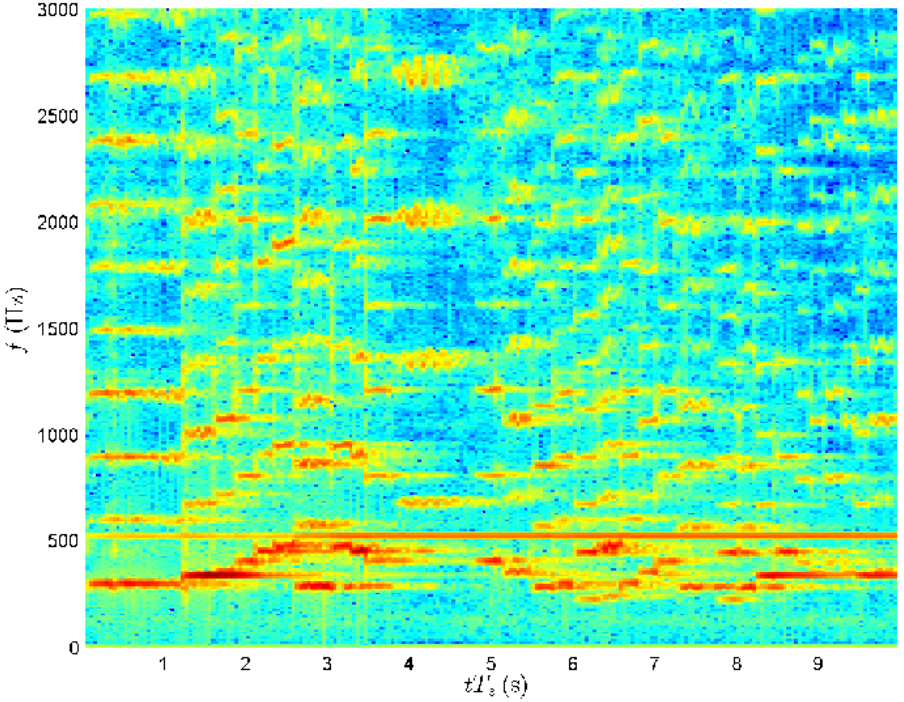
  - a 2-3 dB gain margin is desirable to avoid **ringing**

Introduction (6)

Acoustic feedback problem (2)

- Example of closed-loop system instability:







Acoustic feedback control

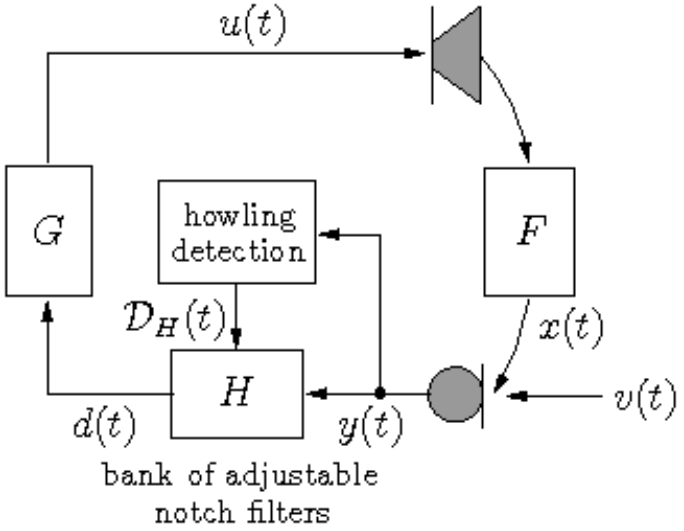
Classification of existing methods

- phase modulation (PM) methods
  - smoothing of "loop gain" (= closed-loop magnitude response)
  - phase/frequency/delay modulation, frequency shifting
  - well suited for reverberation enhancement systems (low gain)
- spatial filtering methods
  - (adaptive) microphone beamforming for reducing direct coupling
- gain reduction methods
  - (frequency-dependent) gain reduction after howling detection
  - most popular method for sound reinforcement applications
- room modeling methods
  - adaptive inverse filtering (AIF): adaptive equalization of acoustic feedback path response
  - adaptive feedback cancellation (AFC): adaptive prediction and subtraction of feedback (≠howling) component in microphone signal

Notch-filter-based howling suppression (1)

Introduction

- gain reduction methods:
  - automation of the actions a human operator would undertake
- classification of gain reduction methods:
  - automatic gain control (full-band gain reduction)
  - automatic equalization (1/3 octave bandstop filters)
  - NHS: notch-filter-based howling suppression (1/10-1/60 octave filters)
- NHS subproblems:
  - howling detection
  - notch filter design



Notch-filter-based howling suppression (2)

Howling detection (1)

- howling detection procedure:
  - divide microphone signal in overlapping frames

$$\mathbf{y}(t) = [y(t+P-M) \ \ldots \ y(t+P-1)]^T$$

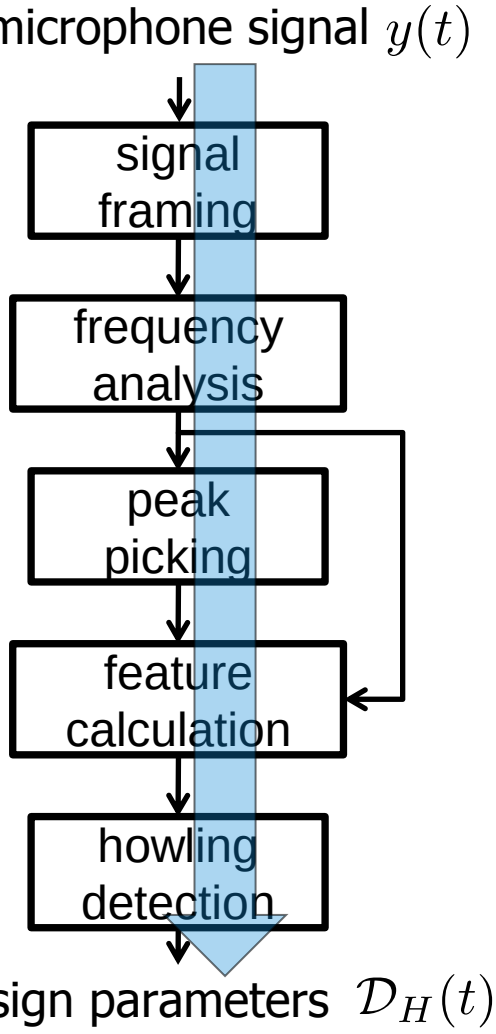
  - estimate the microphone signal spectrum (DFT)

$$\mathbf{Y}(t) = [Y(\omega_0, t) \ \ldots \ Y(\omega_{M-1}, t)]^T$$

  - select a number of candidate howling components

$$\mathcal{D}_{\breve{\omega}}(t) = \{\breve{\omega}_i\}_{i=1}^{N}$$

  - calculate a set of discriminating signal features
  - decide on presence/absence of howling



Notch-filter-based howling suppression (3)

Howling detection (2)

- spectral signal features for howling detection:
  1. Peak-to-Threshold Power Ratio (PTPR)
     = howling should only be suppressed when it is sufficiently loud
  2. Peak-to-Average Power Ratio (PAPR)
     = howling eventually has large power compared to speech/audio
  3. Peak-to-Harmonic Power Ratio (PHPR)
     = howling does not exhibit a harmonic structure (≠ in case of clipping!)
  4. Peak-to-Neighboring Power Ratio (PNPR)
     = howling is a non-damped sinusoid, having approx. zero bandwidth
- temporal signal features for howling detection
  1. Interframe Peak Magnitude Persistence (IPMP)
     = howling components typically persist longer than speech/audio
  2. Interframe Magnitude Slope Deviation (IMSD)
     = howling exhibits an exponential amplitude buildup over time

Notch-filter-based howling suppression (4)

Howling detection (3)

- howling detection as a binary hypothesis test:

  $\mathcal{H}_0$: howling does not occur (Null hypothesis)

  $\mathcal{H}_1$: howling does occur (Alternative hypothesis)

- detection performance:
  - probability of detection ~ **reliability**

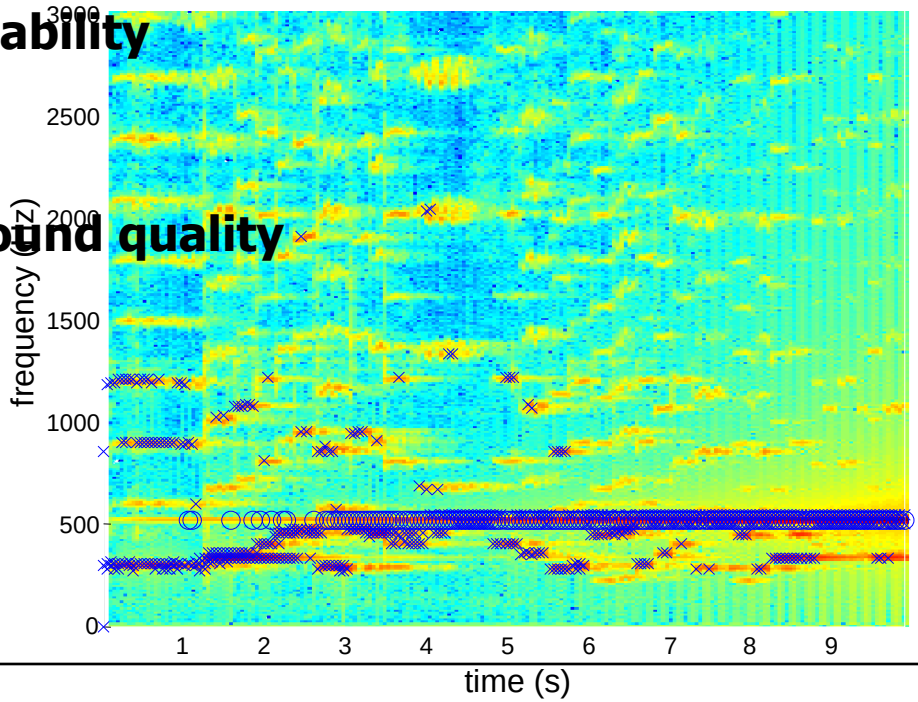
$$P_D = P(\mathcal{H}_1;\mathcal{H}_1) = \frac{N_{TP}}{N_P}$$

  - probability of false alarm ~ **sound quality**

$$P_{FA} = P(\mathcal{H}_1;\mathcal{H}_0) = \frac{N_{FP}}{N_N}$$

- example of detection data set:

o = positive realizations (N_P = 166)
x = negative realizations (N_N = 482)



Notch-filter-based howling suppression (5)

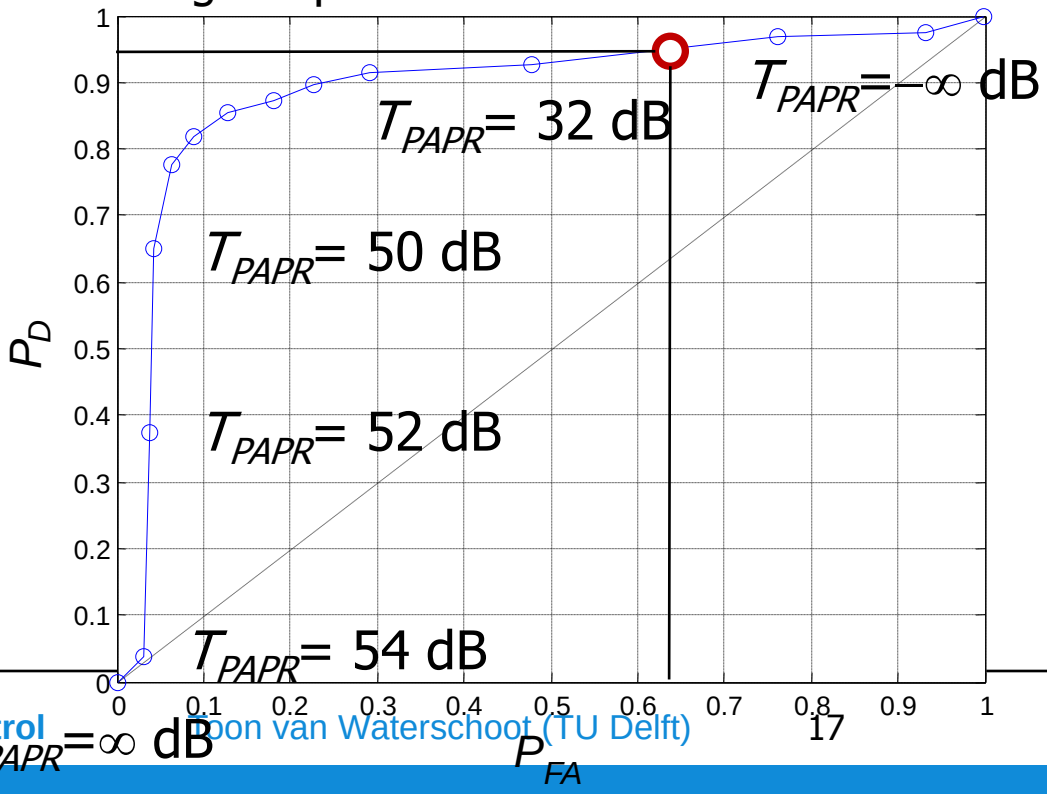
Howling detection (4)

- example of **single-feature** howling detection criterion:

$$\mathrm{PAPR}(\breve{\omega}_i, t) \geq T_{\mathrm{PAPR}}[\mathrm{dB}] \Rightarrow \mathrm{Reject}\ \mathcal{H}_0$$

- evaluation measures:
  - ROC curve: P_D vs. P_{FA} for entire range of possible threshold values
  - P_{FA} for fixed P_D = 95 %

criterion	P_{FA}
PTPR	70 %
PAPR	63 %
PHPR	37 %
PNPR	33 %
IPMP	54 %
IMSD	40 %



Notch-filter-based howling suppression (6)

Howling detection (5)

- improved detection with **multiple-feature** howling detection criteria:
  - logical conjunction of two or more single-feature criteria
  - design guideline: combine features with high P_D, regardless of P_{FA}
- examples of multiple-feature criteria:
  - PHPR & IPMP [Lewis *et al.* (Sabine Inc.), 1993]
  - FEP = PNPR & IMSD [Osmanovic *et al.*, 2007]
  - PHPR & PNPR, PHPR & IMSD, PNPR & IMSD, PHPR & PNPR & IMSD [van Waterschoot & Moonen, 2008]

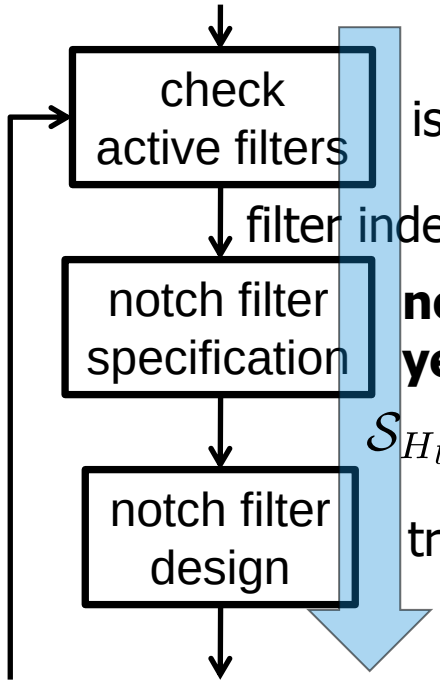
single-feature criterion	P_{FA}	multiple-feature criterion	P_{FA}
PTPR	70 %	PHPR & IPMP	65 %
PAPR	63 %	FEP	24 %
PHPR	37 %	PHPR & PNPR	14 %
PNPR	33 %	PHPR & IMSD	25 %
IPMP	54 %	PNPR & IMSD	5 %
IMSD	40 %	PHPR & PNPR & IMSD	3 %

Notch-filter-based howling suppression (7)

Notch filter design

- notch filter design procedure:

set of notch filter design parameters $\mathcal{D}_H(t)$



is a notch filter already active around howling frequency?

no? new filter: center frequency = howling frequency
yes? active filter: decrease notch gain

$\mathcal{S}_{H_l}(t) = \{\omega_{c,l}, B_l, G_{c,l}\}$

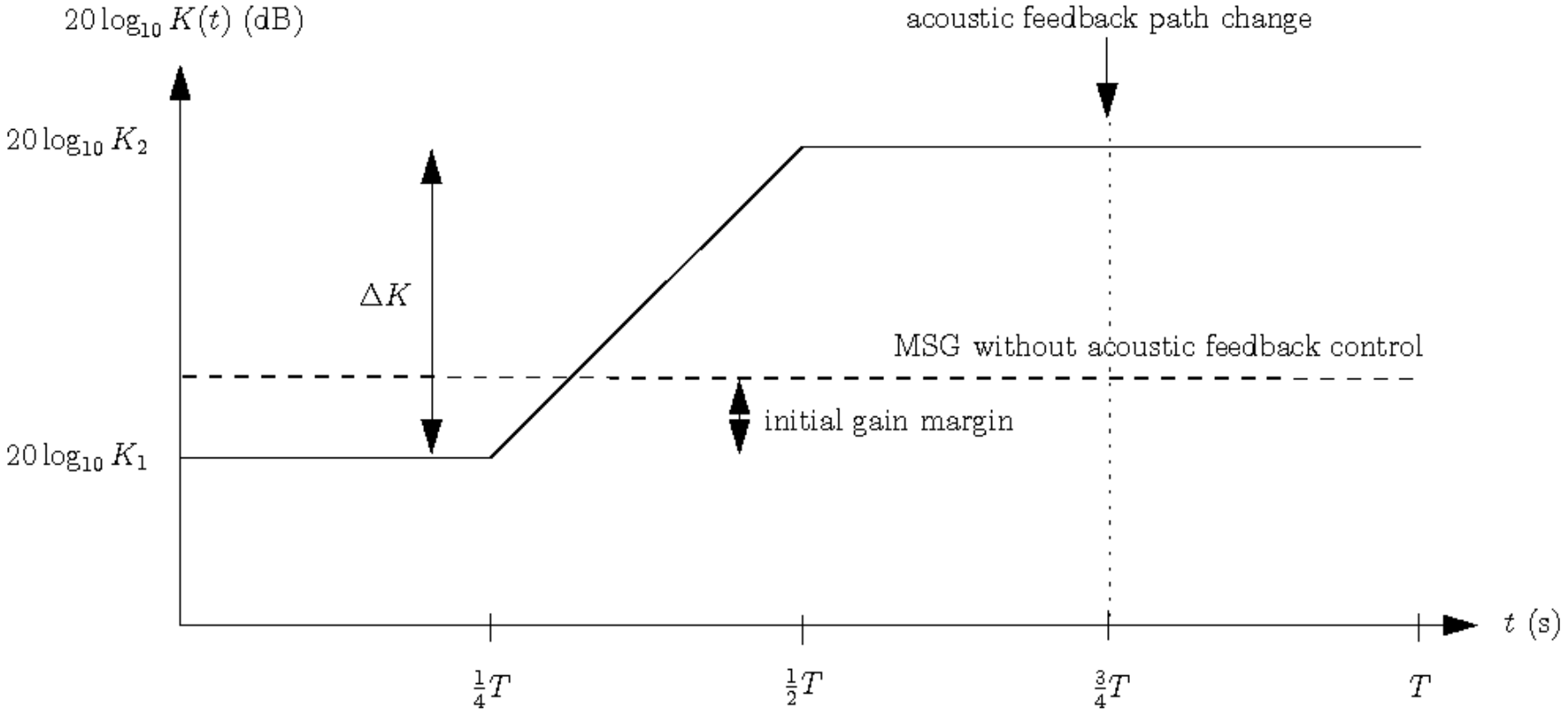
translate filter specifications into filter coefficients

bank of notch filters transfer function $H(q,t) = \prod_{l=1}^{n_H/2} H_l(q,t)$

Notch-filter-based howling suppression (8)

Simulation results (1)

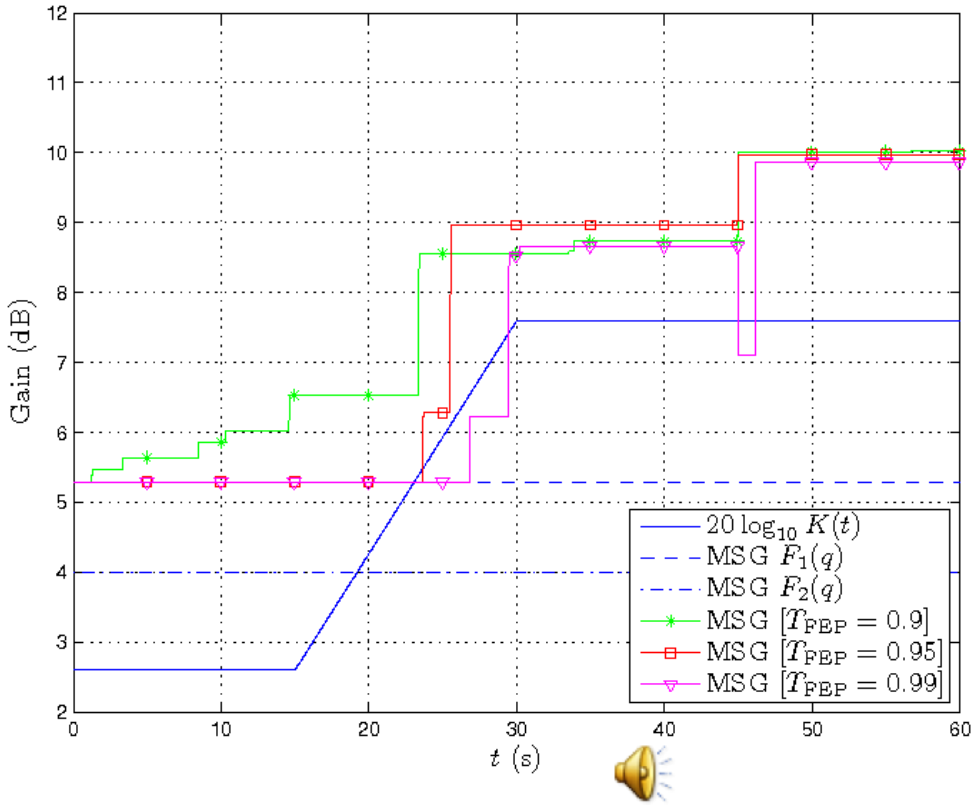
- simulation layout:

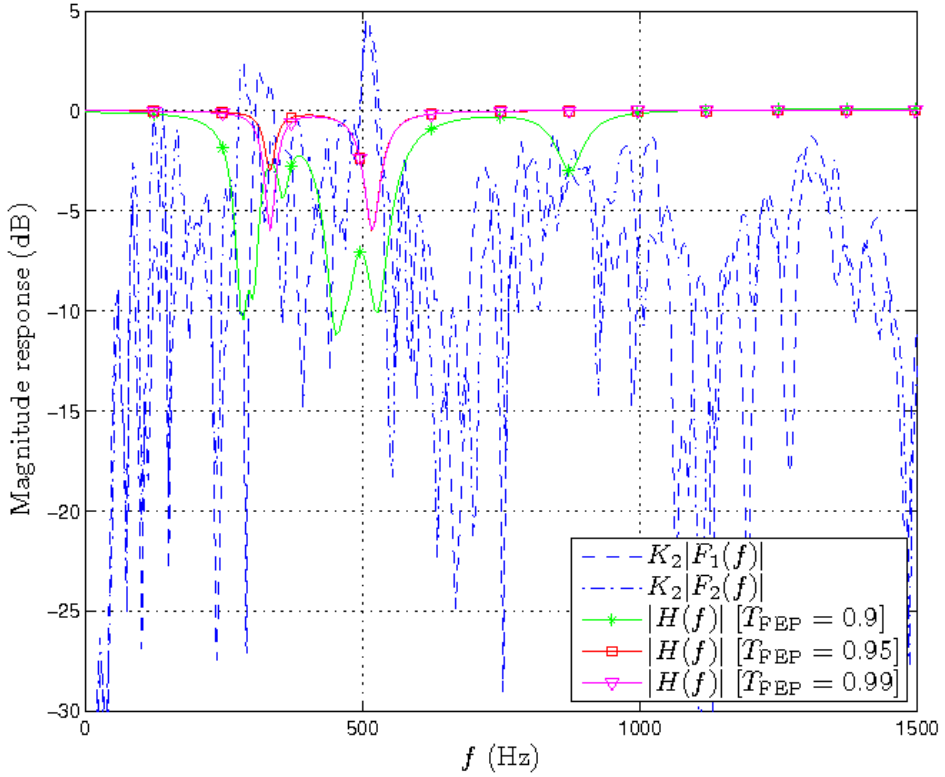


Notch-filter-based howling suppression (9)

Simulation results (2)

- simulation results for three different threshold values:

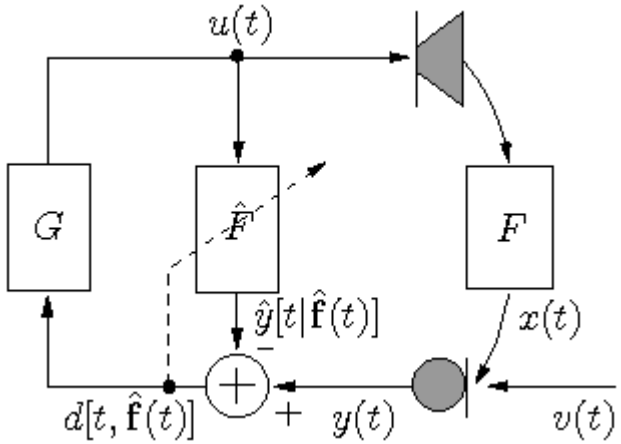




Adaptive feedback cancellation (1)

Introduction (1)

- AFC concept:
  - predict and subtract entire feedback signal component (≠howling component!) in microphone signal
  - requires adaptive estimation of acoustic feedback path model
  - similar to acoustic echo cancellation, but much more difficult due to closed signal loop



Adaptive feedback cancellation (2)

Introduction (2)

- AFC basics:
  - consider a finite impulse response (FIR) acoustic feedback path

$$\mathbf{f}(t) = \begin{bmatrix} f^{(0)}(t) & f^{(1)}(t) & \dots & f^{(n_F)}(t) \end{bmatrix}^T$$

and similarly a FIR acoustic feedback path **model**

$$\hat{\mathbf{f}}(t) = \begin{bmatrix} \hat{f}^{(0)}(t) & \hat{f}^{(1)}(t) & \dots & \hat{f}^{(n_{\hat{F}})}(t) \end{bmatrix}^T \quad (n_F = n_{\hat{F}})$$

  - least squares (LS) **estimation** of acoustic feedback path model gives

$$\begin{aligned} \hat{\mathbf{f}}(t) &= (\mathbf{U}^T\mathbf{U})^{-1}\mathbf{U}^T\mathbf{y} \\ \mathbf{y} &= \begin{bmatrix} y(t) & y(t-1) & \dots & y(1) \end{bmatrix}^T \\ \mathbf{U} &= \begin{bmatrix} \mathbf{u}(t) & \mathbf{u}(t-1) & \dots & \mathbf{u}(1) \end{bmatrix}^T \\ \mathbf{u}(t) &= \begin{bmatrix} u(t) & u(t-1) & \dots & u(t-n_{\hat{F}}) \end{bmatrix}^T \end{aligned}$$

Adaptive feedback cancellation (3)

Closed-loop signal decorrelation (1)

- AFC correlation problem:
  - LS estimation **bias** vector

$$\text{bias}\{\hat{\mathbf{f}}(t)\} = E\{(\mathbf{U}^T\mathbf{U})^{-1}\mathbf{U}^T\mathbf{v}\} \neq \mathbf{0}$$

  - non-zero bias results in (partial) source signal cancellation
  - LS estimation **covariance** matrix

$$\text{cov}\{\hat{\mathbf{f}}(t)\} = E\{(\hat{\mathbf{f}}(t) - \mathbf{f}(t))(\hat{\mathbf{f}}(t) - \mathbf{f}(t))^T\} = \left[E\{\mathbf{U}^T{\mathbf{R}_\mathbf{v}}^{-1}\mathbf{U}\}\right]^{-1}$$

    with source signal covariance matrix

$$\mathbf{R}_\mathbf{v} = E\{\mathbf{v}\mathbf{v}^T\}$$

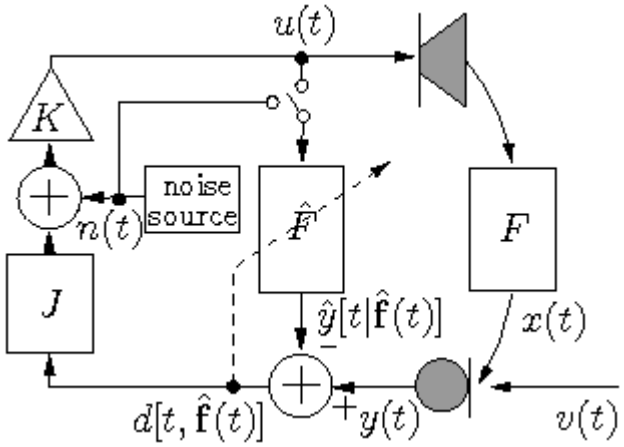
  - large covariance results in slow adaptive filter convergence

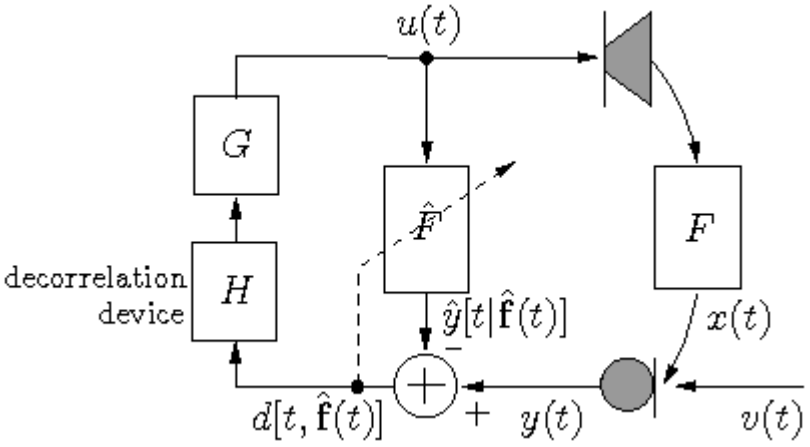
- **decorrelation** of loudspeaker and source signal is crucial issue!

Adaptive feedback cancellation (4)

Closed-loop signal decorrelation (2)

- Decorrelation in the closed signal loop:
  - noise injection
  - time-varying processing
  - nonlinear processing
  - forward path delay
- Inherent **trade-off** between decorrelation and sound quality





Adaptive feedback cancellation (5)

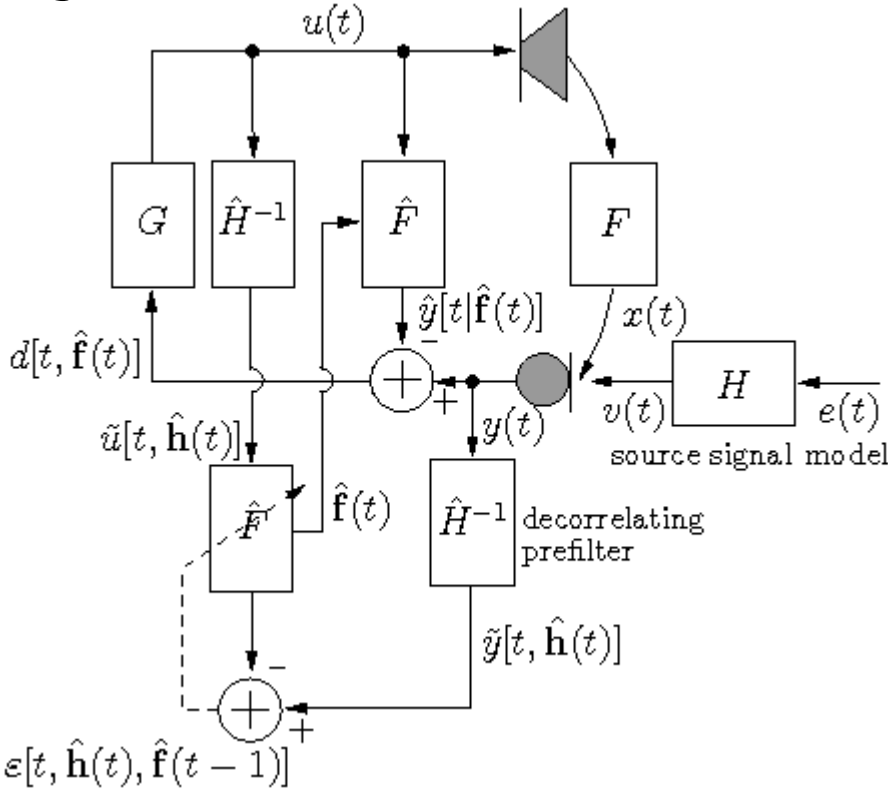
Closed-loop signal decorrelation (3)

- Decorrelation in the adaptive filtering circuit:
  - adaptive filter delay
  - decorrelating prefilters

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

based on source signal model

$$v(t) = H(q,t)e(t) = \underbrace{H_2(q,t)}_{\text{noise}}\underbrace{H_1(q,t)}_{\text{tonal}}$$



- Sound quality not compromised
- Additional information required:
  - acoustic feedback path delay
  - source signal model

Adaptive feedback cancellation (6)

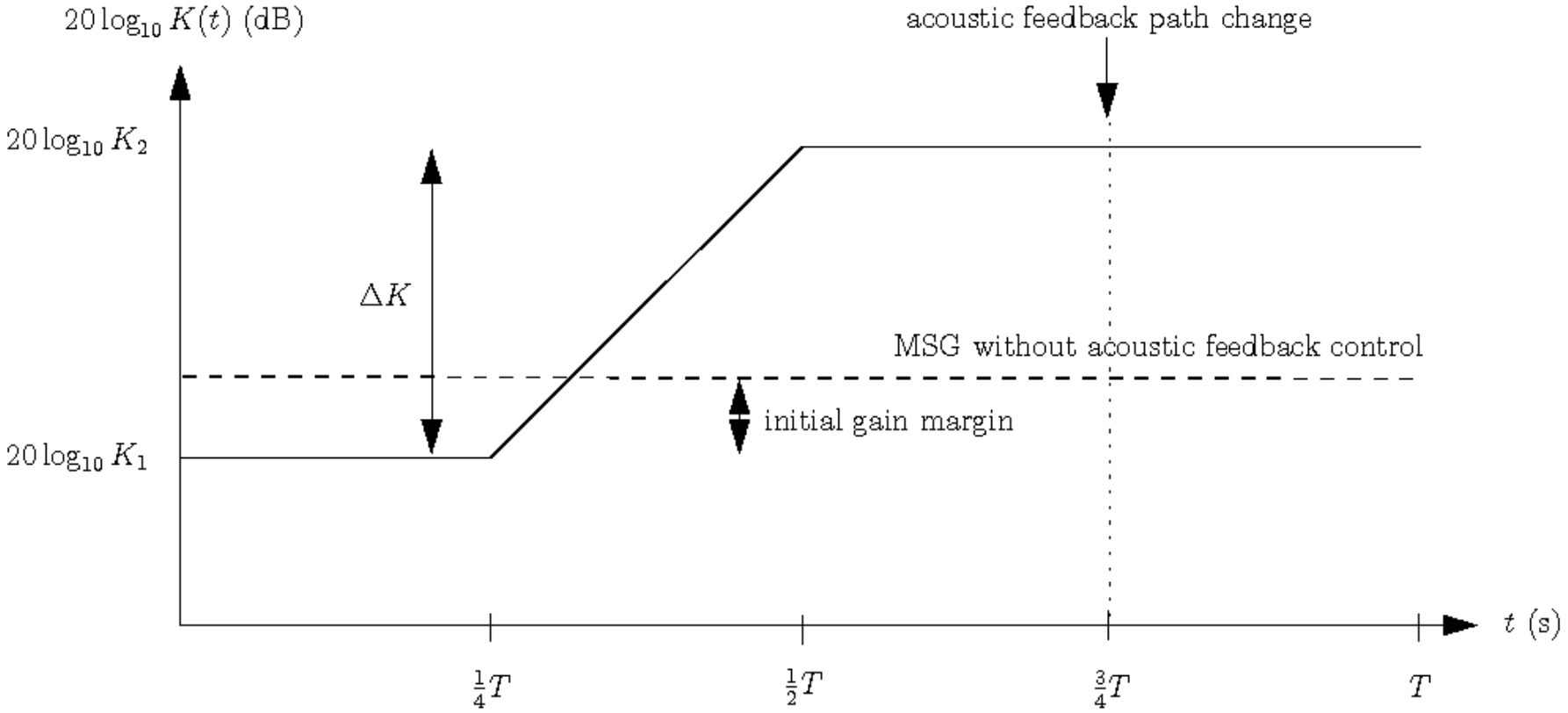
Adaptive filter design

- LS-based adaptive filtering algorithms:
  - recursive least squares (RLS)
  - affine projection algorithm (APA)
  - (normalized) least mean squares ((N)LMS)
  - frequency-domain NLMS
  - partitioned-block frequency domain NLMS
  - …
- prediction-error-method(PEM)-based adaptive filtering algorithms:
  - joint estimation of acoustic feedback path and source signal model
  - requires forward path delay and exploits source signal nonstationarity
  - available in all flavours (RLS, APA, NLMS, frequency domain, …)
  - 25-50 % computational overhead compared to LS-based algorithms

Adaptive feedback cancellation (7)

Simulation results (1)

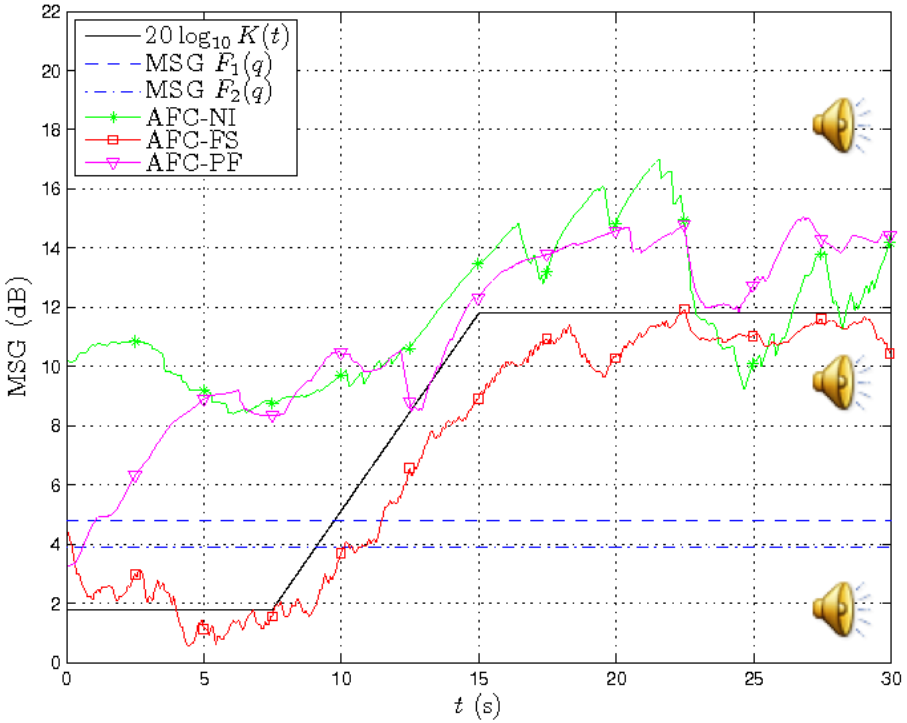
- simulation layout (revisited):



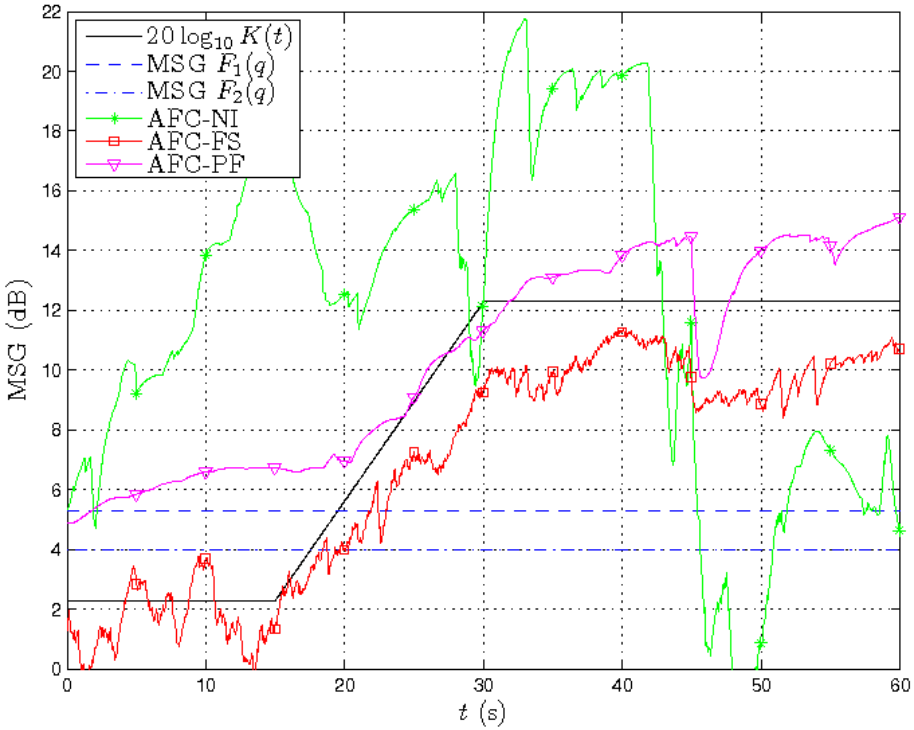
Adaptive feedback cancellation (8)

Simulation results (2)

- simulation results for three different decorrelation methods:



speech



music

Conclusion (1)

Acoustic feedback control methods

- phase modulation methods:
  - suited for low-gain applications such as reverberation enhancement
- spatial filtering methods:
  - removal of direct coupling if multiple microphones are available
- gain reduction methods: notch-filter-based howling suppression
  - very popular for sound reinforcement applications
  - accurate howling detection is crucial for sound quality and reliability
  - reasonable MSG increase (up to 5 dB) can be attained
- room modeling methods: adaptive feedback cancellation
  - upcoming method as computational resources become cheaper
  - decorrelation in adaptive filtering circuit required for high sound quality
  - MSG increase up to 20 dB is generally achieved

Conclusion (2)

Open issues

- multi-channel systems:
  - acoustic feedback problem not uniquely defined in multi-channel case
  - most methods were developed for single-channel case only
  - computational complexity may explode
- adaptive feedback cancellation:
  - computational complexity and adaptive filter convergence speed remain problematic due to very high filter orders (~1000 coefficients)
  - adaptive filter behavior in case of undermodeling not well understood
  - FIR model is inefficient for modeling acoustic resonances
- hybrid methods:
  - how to combine different methods such that desirable features are retained while undesirable properties are avoided?
  - interplay between different methods not well understood
  - and again: computational complexity…

Additional literature

- review paper:
  - T. van Waterschoot and M. Moonen, "Fifty years of acoustic feedback control: state of the art and future challenges," *Proc. IEEE*, vol. 99, no. 2, pp. 288-327 , Feb. 2011.
- phase modulation:
  - J. L. Nielsen and U. P. Svensson, "Performance of some linear time-varying systems in control of acoustic feedback," *J. Acoust. Soc. Amer.*, vol. 106, no. 1, pp. 240–254, Jul. 1999.
- spatial filtering:
  - G. Rombouts, A. Spriet, and M. Moonen, "Generalized sidelobe canceller based combined acoustic feedback- and noise cancellation," *Signal Process.*, vol. 88, no. 3, pp. 571–581, Mar. 2008.
- notch-filter-based howling suppression:
  - T. van Waterschoot and M. Moonen, "Comparative evaluation of howling detection criteria in notch-filter-based howling suppression," *J. Audio Eng. Soc.*, vol. 58, no. 11, pp. 923-940, Nov. 2010.
  - T. van Waterschoot and M. Moonen, "A pole-zero placement technique for designing second-order IIR parametric equalizer filters," *IEEE Trans. Audio Speech Lang. Process.*, vol. 15, no. 8, pp. 2561–2565, Nov. 2007.
- adaptive feedback cancellation:
  - G. Rombouts, T. van Waterschoot, K. Struyve, and M. Moonen, "Acoustic feedback suppression for long acoustic paths using a nonstationary source model," *IEEE Trans. Signal Process.*, vol. 54, no. 9, pp. 3426–3434, Sept. 2006.
  - T. van Waterschoot and M. Moonen, "Adaptive feedback cancellation for audio applications," *Signal Process.*, vol. 89, no. 11, pp. 2185–2201, Nov. 2009.
  - G. Rombouts, T. van Waterschoot, and M. Moonen, "Robust and efficient implementation of the PEM-AFROW algorithm for acoustic feedback cancellation," *J. Audio Eng. Soc.*, vol. 55, no. 11, pp. 955–966, Nov. 2007.

Questions?