Digital Audio Signal Processing

DASP

Chapter-5:
Acoustic Echo & Feedback Cancellation

Marc Moonen
Dept. E.E./ESAT-STADIUS, KU Leuven
marc.moonen@kuleuven.be
homes.esat.kuleuven.be/~moonen/

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Outline

Acoustic Echo Cancellation (AEC)

- Intro/Acoustic Channels
- Adaptive Filters for AEC
- Control Algorithm
- Stereo AEC

Acoustic Feedback Control (AFC)

- Intro/Nyquist Stability & Maximum Stable Gain
- AFC Methods
- Notch-Filter-Based Howling Suppression (NHS)
- Adaptive Feedback Cancellation (AFC)

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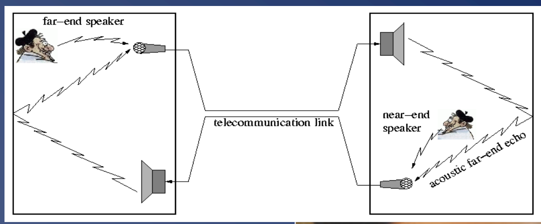
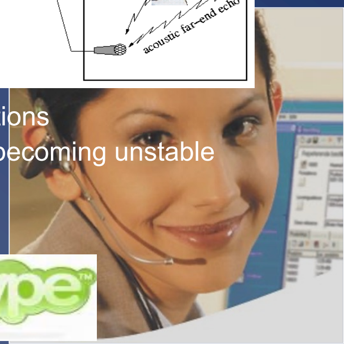
Introduction

Acoustic Echo Cancellation (AEC)



Suppress echo..
- To guarantee normal conversation conditions
- To prevent the closed-loop system from becoming unstable

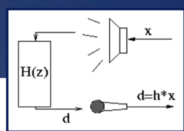
Applications
- Teleconferencing
- Hands-free telephony
- Handsets, ..

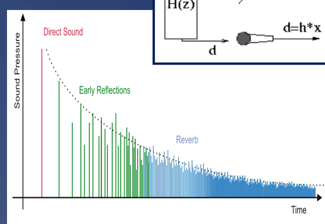


Introduction: Acoustic Channels

- Loudspeaker + acoustic path + microphone can be modeled with sufficient accuracy as a <u>linear filtering</u> operation

  PS: Loudspeaker often introduces <u>non-linear distortion</u>, not taken into account here





- Acoustic path is represented by the **<u>acoustic impulse response</u>**

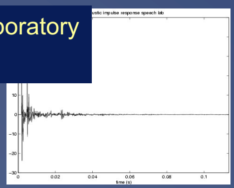
First there is a <u>dead</u> time. Then come the <u>direct path</u> impulse and some <u>early reflections</u>, which depend on the geometry of the room. Finally there is an exponentially decaying tail called <u>reverberation</u>, coming from multiple reflections on walls, objects,... Reverberation mainly depends on 'reflectivity' (rather than geometry) of the room…

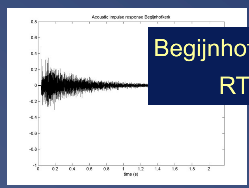
Introduction: Acoustic Channels

To characterize the 'reflectivity' of a room the reverberation time '**RT60'** is defined

- RT60 = time which the sound pressure level or intensity needs to decay to -60dB of its original value
- For a typical office room RT60 is between 100 and 400 ms, for a church RT60 can be several seconds

ESAT speech laboratory
RT60 ≈ 120 ms





Begijnhofkerk Leuven
RT60 ≈ 3730 ms

PS: Acoustic room impulse responses are highly time-varying !!!!
PS: Acoustic room impulse responses are position dependent !!!!

Acoustic Channels

Acoustic Impulse Response : FIR or IIR ?

- **If the acoustic impulse response is modeled as an..**
  - **FIR** filter → hundreds/thousands of filter taps are needed
  - **IIR** filter → filter order can be reduced, but still hundreds of filter coeffs (num. + denom.) may be needed (sigh!)
- **Furthermore...**
  - In a speech comms set-up the acoustics are highly time-varying, hence adaptive filtering techniques are called for (see DSP-CIS):
    - FIR adaptive filters : simple adaptation rules, no local minima,..
    - IIR adaptive filters : more complex adaptation, local minima, have to monitor stability, ..
- **Hence FIR models are used in practice**

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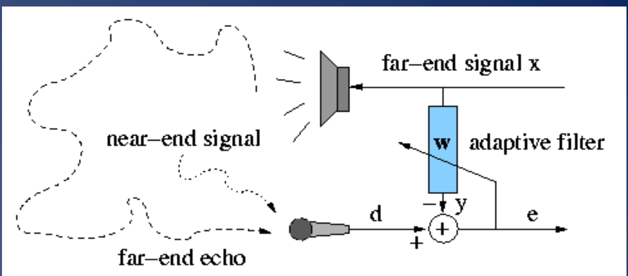
Adaptive filters for AEC

Basic set-up

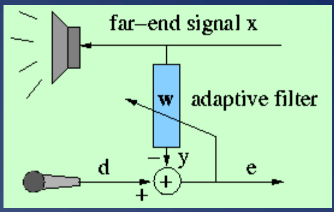


- Adaptive filter produces a model for acoustic room impulse response + an estimate of the echo contribution in microphone signal, which is then subtracted from the microphone signal
- Thanks to adaptivity
  - time-varying acoustics can be tracked
  - performance superior to performance of `conventional’ techniques (e.g. voice controlled switching, loss control, etc.)

Adaptive filters for AEC: NLMS

- **NLMS update equations**

$$y[k] = \mathbf{x}_k^T \mathbf{w}_k$$
$$e[k] = d[k] - y[k]$$
$$\mathbf{w}_{k+1} = \mathbf{w}_k + \frac{\mu}{\mathbf{x}_k^T \mathbf{x}_k + \delta} \mathbf{x}_k e[k]$$



in which

$$\mathbf{x}_k = \begin{bmatrix} x[k] \\ \vdots \\ x[k-(N-1)] \end{bmatrix}, \qquad \mathbf{w}_k = \begin{bmatrix} w[0] \\ \vdots \\ w[N-1] \end{bmatrix}$$

N is # filter taps, k is the discrete-time index
μ is the adaptation stepsize and δ is a regularization parameter

Adaptive filters for AEC: NLMS

- **Pros and cons of NLMS**

+ cheap algorithm : O(N)
+ small input/output delay (= 1 sample)

– for colored far-end signals (such as speech) convergence of the NLMS algorithm is slow (cfr λ_{max} versus λ_{min}, etc…., see DSP-CIS)
– large N then means even slower convergence

¤ NLMS is thus often used for the cancellation of short echo paths

Adaptive filters for AEC

- As some input/output delay is acceptable in AEC (e.g. ITU says 16ms), algorithms can be derived that are <u>even cheaper</u> than NLMS, by exchanging implementation cost for extra processing delay, sometimes even with <u>improved performance</u> :

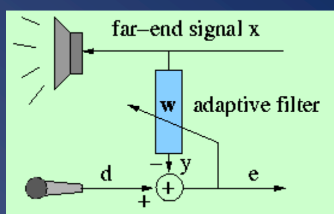
  - **Frequency-domain adaptive filtering (<u>FDAF</u>)**
  - **Partitioned Block FDAF (<u>PB-FDAF</u>)**

  \+ cost reduction
  \+ optimal (stepsize) tuning for each subband/frequency bin separately results in improved performance

Adaptive filters for AEC: <u>B</u>lock-<u>LMS</u>

- To derive the *frequency-domain* adaptive filter the **<u>BLMS</u>** algorithm is considered first

$$\mathbf{e}_n = \mathbf{d}_n - \mathbf{X}_n^T \mathbf{w}_n$$
$$\mathbf{w}_{n+1} = \mathbf{w}_n + \mu \mathbf{X}_n \mathbf{e}_n$$



in which

$$\mathbf{X}_n = \begin{bmatrix} x[nL+1] & \cdots & x[(n+1)L] \\ \vdots & \ddots & \vdots \\ x[nL+1-(N-1)] & \cdots & x[(n+1)L-(N-1)] \end{bmatrix}, \quad \mathbf{d}_n = \begin{bmatrix} d[nL+1] \\ \vdots \\ d[(n+1)L] \end{bmatrix}, \quad \mathbf{w}_n = \begin{bmatrix} w[0] \\ \vdots \\ w[N-1] \end{bmatrix}$$

N is # filter taps, L is block length, *n* is block time index
BLMS = gradient averaging over block of *L* samples (i.o. 1 sample in LMS)

Adaptive filters for AEC: Block-LMS

- Both the BLMS convolution and correlation operation are computationally demanding. They can be implemented more efficiently in the *frequency domain* using fast convolution techniques, i.e. overlap-save/overlap-add :

$$\mathbf{e}_n = \mathbf{d}_n - \boxed{\mathbf{X}_n^T \mathbf{w}_n}$$

$$\mathbf{w}_{n+1} = \mathbf{w}_n + \mu \boxed{\mathbf{X}_n \mathbf{e}_n}$$

convolution

$$\begin{bmatrix} \mathbf{0}_{M-L} & \mathbf{0} \\ \mathbf{0} & \boxed{\mathbf{I}_L} \end{bmatrix} \mathbf{F}^{-1} \underline{\mathbf{X}}^{(n)} \underline{\mathbf{w}}^{(n)}$$

overlap-save

correlation

$$\begin{bmatrix} \boxed{\mathbf{I}_N} & \mathbf{0} \\ \mathbf{0} & \mathbf{0}_{M-N} \end{bmatrix} \mathbf{F}^{-1} \underline{\mathbf{X}}^{(n)*} \mathbf{F} \begin{bmatrix} \mathbf{0} \\ \mathbf{e}_n \end{bmatrix}$$

with

$$\underline{\mathbf{X}}^{(n)} = diag\left\{ \mathbf{F} \begin{bmatrix} x[(n+1)L - M + 1] \\ \vdots \\ x[(n+1)L] \end{bmatrix} \right\}, \quad \underline{\mathbf{w}}^{(n)} = \mathbf{F} \begin{bmatrix} \mathbf{w}_n \\ \mathbf{0} \end{bmatrix}, \quad \mathbf{F}(m,n) = e^{-j\frac{2\pi\, mn}{M}}$$

M-point DFT-matrix

Adaptive filters for AEC: FDAF

Overlap-save FDAF

$$\underline{\mathbf{X}}^{(n)} = diag\left\{ \mathbf{F} \begin{bmatrix} x[(n+1)L - M + 1] \\ \vdots \\ x[(n+1)L] \end{bmatrix} \right\}$$

$$\mathbf{y}^{(n)} = \begin{bmatrix} \mathbf{0}_{M-L} & \mathbf{0} \\ \mathbf{0} & \mathbf{I}_L \end{bmatrix} \mathbf{F}^{-1} \underline{\mathbf{X}}^{(n)} \underline{\mathbf{w}}^{(n)}$$

$$\mathbf{d}^{(n)} = \begin{bmatrix} \mathbf{0}_{M-L} \\ \mathbf{d}_n \end{bmatrix}, \quad \mathbf{d}_n = \begin{bmatrix} d[nL+1] \\ \vdots \\ d[(n+1)L] \end{bmatrix}$$

$$\mathbf{e}^{(n)} = \mathbf{d}^{(n)} - \mathbf{y}^{(n)}$$

$$\underline{\mathbf{w}}^{(n+1)} = \underline{\mathbf{w}}^{(n)} + \mu \mathbf{F} \begin{bmatrix} \mathbf{I}_N & \mathbf{0} \\ \mathbf{0} & \mathbf{0}_{M-N} \end{bmatrix} \mathbf{F}^{-1} \underline{\mathbf{X}}^{(n)*} \mathbf{F} \mathbf{e}^{(n)}$$

Will only work if

$$\boxed{M \geq N + L - 1}$$

(*M* is DFT-size)

Adaptive filters for AEC: FDAF

¤ Typical parameter setting for the FDAF :

$$N = L, \quad M = 2L, \quad M = 2^p, \quad p \in \mathbb{N}$$

¤ FDAF is functionally equivalent to BLMS (!)

\+ FDAF is significantly cheaper than (B)LMS (cfr FFT/IFFT i.o. DFT/IDFT) for a typical parameter setting

If N=1024 : $\dfrac{\text{cost LMS}}{\text{cost FDAF}} \approx 20$

\- Input/output delay is equal to *2L-1=2N-1*, which may be unacceptably large for realistic parameter settings : e.g. if N=1024 and fs=8000Hz → delay is 256 ms !

Adaptive filters for AEC: PB-FDAF

- **Overlap-save PB-FDAF :** N-tap filter split into (N/P) filter sections, P-taps each, then apply overlap-save to each section (`P takes the place of N’).

Will only work if $\boxed{M \geq P + L - 1}$

$$\underline{\mathbf{X}}_p^{(n)} = diag\left\{\mathbf{F}\begin{bmatrix} x[(n+1)L - pP - M + 1] \\ \vdots \\ x[(n+1)L - pP] \end{bmatrix}\right\}, \quad p = 0 : \frac{N}{P} - 1$$

$$\mathbf{y}^{(n)} = \begin{bmatrix} \mathbf{0}_{M-L} & \mathbf{0} \\ \mathbf{0} & \mathbf{I}_L \end{bmatrix} \mathbf{F}^{-1} \sum_{p=0}^{\frac{N}{P}-1} \underline{\mathbf{X}}_p^{(n)} \underline{\mathbf{w}}_p^{(n)}$$

$$\mathbf{d}^{(n)} = \begin{bmatrix} \mathbf{0} \\ \mathbf{d}_n \end{bmatrix}, \quad \mathbf{d}_n = \begin{bmatrix} d[nL+1] \\ \vdots \\ d[(n+1)L] \end{bmatrix}$$

$$\mathbf{e}^{(n)} = \mathbf{d}^{(n)} - \mathbf{y}^{(n)}$$

$$\underline{\mathbf{w}}_p^{(n+1)} = \underline{\mathbf{w}}_p^{(n)} + \mu \mathbf{F} \begin{bmatrix} \mathbf{I}_P & \mathbf{0} \\ \mathbf{0} & \mathbf{0}_{M-P} \end{bmatrix} \mathbf{F}^{-1} \underline{\mathbf{X}}_p^{(n)*} \mathbf{F} \mathbf{e}^{(n)}, \quad p = 0 : \frac{N}{P} - 1$$

Adaptive filters for AEC: PB-FDAF

¤ Typical parameter setting :

$$P = L, \quad M = 2L, \quad M = 2^q, \quad q \in \mathbb{N}$$

- ¤ PB-FDAF is intermediate between LMS and FDAF (N/P=1)
- ¤ PB-FDAF is functionally equivalent to BLMS
- + PB-FDAF is cheaper than (B)LMS : $\frac{\text{cost LMS}}{\text{cost PBFDAF}} \approx 6$
  If N=1024, P=L=128, M=256 ➔
- + Input/output delay is $2L-1$ which can be chosen small, in the example above the delay is 32 ms, if fs=8000Hz
- + Instead of a simple stepsize μ, 'subband' dependent stepsizes μ_i can be applied to increase convergence speed
- ¤ used in commercial AECs

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Control Algorithm

- Adaptation speed (μ) in LMS-type algorithms should be adjusted…
  - to the far-end signal power, in order to avoid instability of the adaptive filter (see DSP-CIS)
    $\rightarrow$ stepsize normalization (e.g. NLMS)
  - to the amount of near-end activity, in order to prevent the filter to move away from the optimal solution (see DSP-CIS on 'excess MSE')
    $\rightarrow$ double-talk detection

Double talk refers to the situation where both the far-end and the near-end speaker are active.

Control Algorithm

3 modes of operation:

1. Near-end activity (single or double talk) (*Ed* large)
   $\rightarrow$ $\boxed{e=d-y, \quad \mu=\text{small or } 0}$ $\rightarrow$ FILT
2. No near-end activity, only far-end activity (*Ex* large, *Ed* small)
   $\rightarrow$ $\boxed{e=d-y, \quad \mu=\mu_{\max}}$ $\rightarrow$ FILT+ADAPT
3. No near-end activity, no far-end activity (*Ex small, Ed* small)
   $\rightarrow$ $\boxed{e=d, \quad \mu=0}$ $\rightarrow$ NOP

- *Ex* is short-time energy of the far-end signal (loudspeaker)
- *Ed* is short-time energy of the desired signal (microphone)

$$Ex=\sum_{i=0}^{L} x[k-i]^2$$

$$Ed=\sum_{i=0}^{L} d[k-i]^2$$

Control Algorithm

Double-talk Detection (DTD)

- Problem: detection of (near-end) speech during (far-end) speech
- Desired properties: Limited number of false alarms, small delay, low complexity, etc...
- Different approaches exist, which are based on energy, correlation, spectral contents, etc…
- Example: Energy-based DTD
  - Method-1
    If $\boxed{Ed > \tau\, Ex}$ → double talk (τ is a chosen threshold)
  - Method-2
    If $\boxed{\rho > 1}$ → double talk (with $\rho = \frac{Ex\,Ee}{Ex^2 + Ey^2}$)

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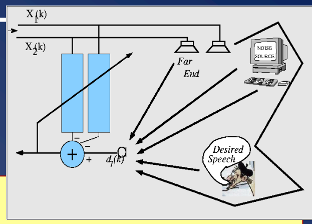
Stereo-AEC

Stereo-AEC Conditioning Problem:



S-AEC input vectors are

$$\underbrace{\mathbf{x}_k}_{2N\times 1} = \left[\mathbf{x}_{k,1}^T \quad \mathbf{x}_{k,2}^T\right]^T$$
$$= \left[x_1[k] \quad x_1[k-1] \quad \ldots \quad x_1[k-N+1] \quad | \, x_2[k] \quad \ldots \quad x_2[k-N+1]\right]^T$$

Mono : autocorrelation of x-signal (e.g. speech) has an impact on convergence (see DSP-CIS)
Stereo : also cross-correlation between signals x1 and x2 plays a role now…

➔ Large(r) eigenvalue spread ($\lambda_{max} >> \lambda_{min}$, i.e. large(r) condition number) of correlation matrix -> large(r) impact on convergence !

Stereo-AEC

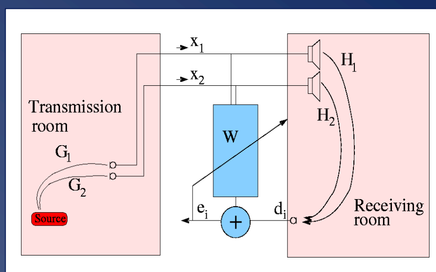
Stereo-AEC Conditioning Problem:

Consider transmission room impulse responses G1,G2 (length Q)

Assume $N=Q$ then :

explain!

$$\left[\mathbf{x}_{k,1}^T \quad \mathbf{x}_{k,2}^T\right]\begin{bmatrix}\mathbf{G}_2 \\ -\mathbf{G}_1\end{bmatrix} = 0$$



Hence filter input correlation matrix will be rank-deficient
(with `null-space', hence $\lambda_{min}=0$, hence worst possible (∞) eigenvalue spread)
Hence LMS converges to a filter vector with spurious contribution
$\alpha.[G_2^T \ -G_1^T]^T$ (corresponding to non-converging 'mode' defined by λ_{min}) , which depends on transmission room, i.e. G1 & G2, which themselves are time-varying!

Stereo-AEC

In practice : $N < Q$

Hence $$\begin{bmatrix} \mathbf{x}_{k,1}^T & \mathbf{x}_{k,2}^T \end{bmatrix} \begin{bmatrix} \mathbf{G}_2^{truncated} \\ -\mathbf{G}_1^{truncated} \end{bmatrix} = \delta \cong 0$$

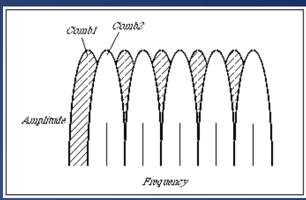
So that correlation matrix will be (only) ill-conditioned (instead of rank-deficient)

…which however is still bad news

Stereo-AEC

Stereo-AEC Fixes:

- Reduce correlation between the loudspeaker signals by…
  - Complementary comb filters
  - White noise insertion
  - Colored (masked) noise insertion (p27)
  - Non-linear processing (p28)



Comb-1 for x1, comb-2 for x2

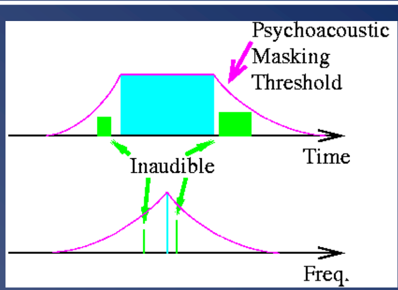
Disadvantages :
  - Signal distortion
  - Stereo perception may be affected

- In addition : use algorithms that are less sensitive to the condition number than NLMS, e.g. RLS, ...

Stereo-AEC

Stereo-AEC Fixes:

Colored noise insertion



Remove all signal content below the masking threshold
Fill with noise (both channels independently)

➡ Correlation between input channels decreases

- Poor performance for speech
- Good performance for <u>music</u>
- Computationally intensive

Stereo-AEC

Stereo-AEC Fixes:

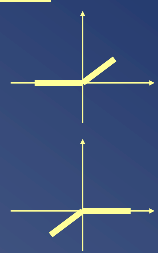
Non-linear processsing

$$x_i^{'}(k) = x_i(k) + \alpha f_i(x_i(k)) \quad i = 1..2$$

$f_i(\bullet)$ is often a half wave rectifier

$$f_1(\bullet) = \frac{\bullet + |\bullet|}{2}$$

$$f_2(\bullet) = \frac{\bullet - |\bullet|}{2}$$



$\alpha = 0.5$ is necessary for good performance, but audible

Good results for <u>speech</u>, audible artifacts in music

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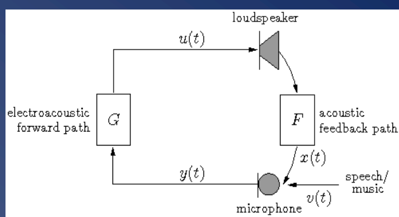
(*) Reference: T. van Waterschoot & M. Moonen, "Fifty years of acoustic feedback control: state of the art and future challenges," *Proc. IEEE*, vol. 99, no. 2, 2011, pp. 288-327.

Introduction

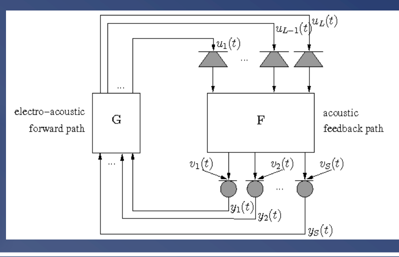
Acoustic Feedback Control (AFC)

Single channel AFC =
- One loudspeaker
- One microphone



Multi-channel AFC = ...

(not treated here)



Applications
- Hearing aids
- Sound reinforcement

AFC Basics

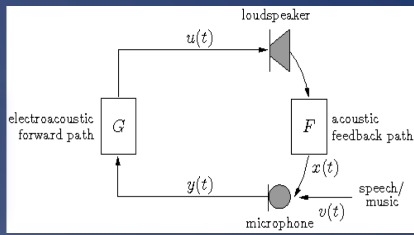
- **"Desired" system transfer function:**

$$\frac{U(\omega,t)}{V(\omega,t)} = G(\omega,t)$$

- **Closed-loop system transfer function:**

$$\frac{U(\omega,t)}{V(\omega,t)} = \frac{G(\omega,t)}{1 - G(\omega,t)F(\omega,t)}$$

  – Spectral coloration
  – Acoustic echoes
  – Risk of instability



- **Loop response:**
  – Loop gain $|G(\omega,t)F(\omega,t)|$
  – Loop phase $\angle G(\omega,t)F(\omega,t)$

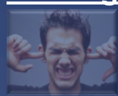
AFC Basics

- **Nyquist stability criterion:**
  – If there exists a radial frequency ω for which

$$\begin{cases} |G(\omega,t)F(\omega,t)| \geq 1 \\ \angle G(\omega,t)F(\omega,t) = n2\pi, \;\; n = \ldots, -1, 0, 1, \ldots \end{cases}$$

  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

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

if *G* has flat response

[Schroeder, 1964]
B=bandwidth

  – Desirable gain margin 2-3 dB (= MSG – actual forward path gain)

AFC Methods

1. **Phase modulation (PM) methods** (not addressed here)
   – Apply frequency/phase modulations in forward path
2. **Spatial filtering methods**
   – Microphone beamforming to reduce direct coupling (Lecture 2)
3. **Gain reduction methods**
   – (Frequency-dependent) gain reduction after howling detection
   – Example: **Notch-filter-based howling suppression** (NHS)
4. **Room modeling methods**
   – Adaptive inverse filtering (AIF): adaptive equalization of acoustic feedback path response (not addressed here)
   – **Adaptive feedback cancellation** (AFC): adaptive prediction and subtraction of feedback component in microphone signal

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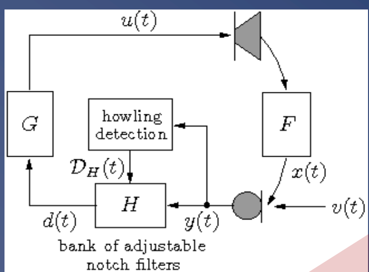
Notch-Filter-Based Howling Suppression (NHS)

- Gain reduction after howling detection



- NHS subproblems:
  – Howling detection
  – Notch filter design

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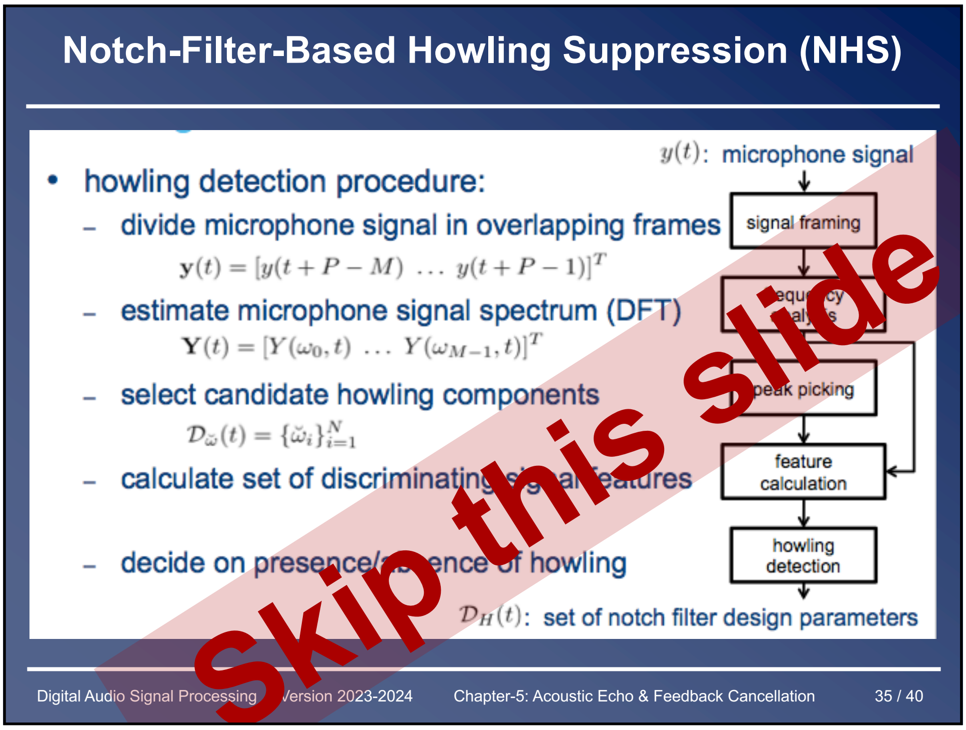
Notch-Filter-Based Howling Suppression (NHS)
howling detection procedure:
divide microphone signal in overlapping frames
$\mathbf{y}(t) = [y(t+P-M) \; \dots \; y(t+P-1)]^T$
estimate microphone signal spectrum (DFT)
$\mathbf{Y}(t) = [Y(\omega_0, t) \; \dots \; Y(\omega_{M-1}, t)]^T$
select candidate howling components
$\mathcal{D}_{\breve{\omega}}(t) = \{\breve{\omega}_i\}_{i=1}^N$
calculate set of discriminating signal features
decide on presence/absence of howling
$y(t)$: microphone signal
signal framing
peak picking
feature calculation
howling detection
$\mathcal{D}_H(t)$: set of notch filter design parameters
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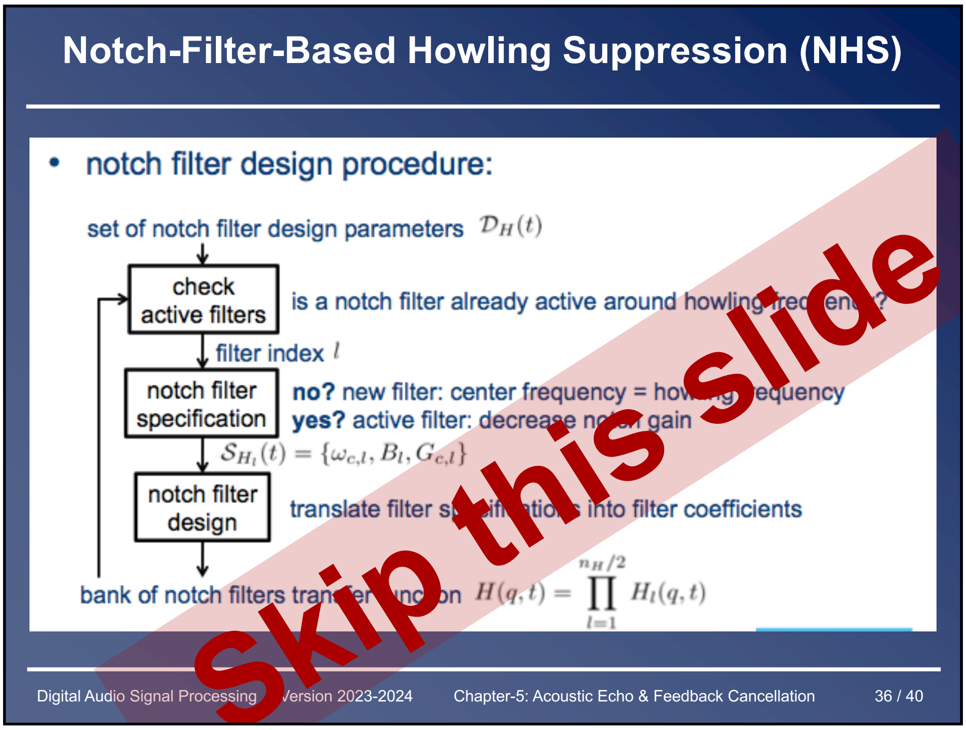
Notch-Filter-Based Howling Suppression (NHS)
notch filter design procedure:
set of notch filter design parameters $\mathcal{D}_H(t)$
check active filters
is a notch filter already active around howling frequency?
filter index l
notch filter specification
no? new filter: center frequency = howling frequency
yes? active filter: decrease notch gain
$\mathcal{S}_{H_l}(t) = \{\omega_{c,l}, B_l, G_{c,l}\}$
notch filter design
translate filter specifications into filter coefficients
bank of notch filters transfer function $H(q,t) = \prod_{l=1}^{n_H/2} H_l(q,t)$
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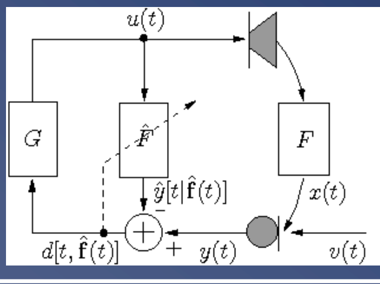
Adaptive Feedback Cancellation (AFC)

- Predict and subtract entire feedback signal (i.o. only howling component) in microphone signal
- Requires adaptive estimation of acoustic feedback path model
- Similar to AEC, but more difficult due to closed signal loop

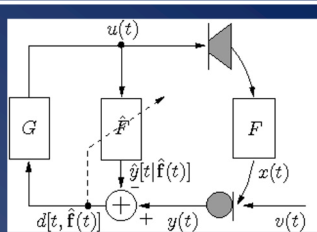
  (-> correlation of loudspeaker and source signal



Adaptive Feedback Cancellation (AFC)

- **AFC correlation problem**
  - Optimal filter (Wiener filter) will have a non-zero bias

$$
\begin{aligned}
\hat{\mathbf{f}}_{WF} &= \hat{\mathbf{R}}_u^{-1}\hat{\mathbf{r}}_{uy} \\
&= \hat{\mathbf{R}}_u^{-1}\hat{\mathbf{r}}_{ux} + \hat{\mathbf{R}}_u^{-1}\hat{\mathbf{r}}_{uv} \\
&= \mathbf{f} + \underbrace{\hat{\mathbf{R}}_u^{-1}\hat{\mathbf{r}}_{uv}}_{\text{bias}\neq 0}
\end{aligned}
$$



  - Non-zero bias results in (partial) source signal (v) cancellation

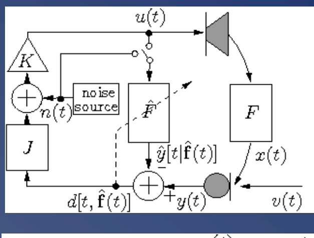
- Need **decorrelation** of loudspeaker and source signal

Adaptive Feedback Cancellation (AFC)

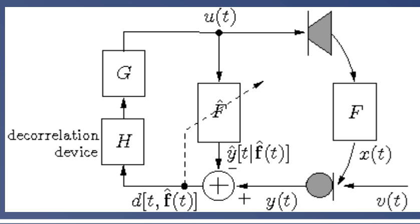
Two methods…

1. Decorrelation in the signal loop

- Noise injection ➔➔➔➔
- Time-varying processing
- Nonlinear processing
- Forward path delay



Inherent **trade-off** between decorrelation and sound quality



Adaptive Feedback Cancellation (AFC)

2. Decorrelation in the adaptive filtering circuit

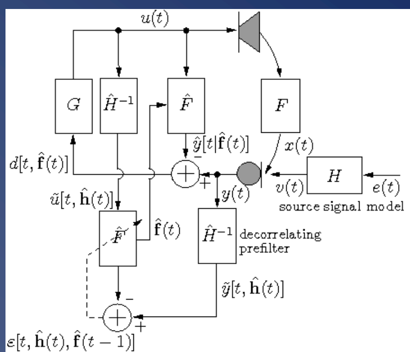
- Decorrelating prefilters to remove bias in adaptive filter

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

based on source signal model

$$v(t) = H(q,t)e(t)$$

leads to unbiased estimate !



- Sound quality not compromised
- Prediction-error-method (PEM)

(details omitted)

- joint estimation of acoustic feedback path and source signal model
- 25-50% computational overhead compared to standard ad.filtering algorithms