ROBUST AND LOW-COST CASCADED NON-LINEAR ACOUSTIC ECHO CANCELLATION

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May 22-27, 2011
Outline

1 Introduction
   - Echo Problem
   - Acoustic Echo Cancellation

2 Acoustic Path Model
   - Modelling of Acoustic Path Component
   - Cascaded and Parallel System

3 Non-linear Acoustic Echo Cancellation
   - Minimum Mean Square Error Estimator (MMSE)
   - Least Mean Square (LMS) Adaptive Tracking

4 Tests Results and Comments
   - Variable Acoustic Path
   - Variable Delay of the Acoustic Path

5 Conclusions
Acoustic Echo Approach

Echo problem
- echo $\rightarrow$ loudspeaker signal picked up by the microphone
- echo and network delay $\rightarrow$ disturb communication

Approach

Acoustic Echo Cancellation (AEC)

Figure: AEC approach
Linear Acoustic Echo Cancellation

characteristics

- Linear environment:
  \[ y(n) = h(n) \ast x(n) \]

- Non-linear environments:
  \[ y(n) = h(n) \ast x(n) \]

**Figure**: Non-linear loudspeaker

**Figure**: Loudspeaker non-linearity effect, \[ y_P(n) = x(n) + \beta x^3(n) \]

**Figure**: Loudspeaker non-linearity plot.
Baseline Non-linear Acoustic Echo Cancellation

**characteristics**
- non-linearity in acoustic path
- requires more computation capacity
- slow convergence
- depends on the model accuracy

**Figure:** Non-linear AEC system
Non-linearity Source and Model

### non-linear part

**loudspeaker**
- non-linear filter
- small impulse responses
- slow variability

### linear part

**acoustic channel**
- linear filter
- longer impulse response
- high variability

**microphone**
- linear filter
- small impulse response
- low variability

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**Figure**: Proposed model
Objectives

**proposed approach**

- pre-processor $\Rightarrow$ loudspeaker model [1, 2, 3]
- loudspeaker model $\Rightarrow$ non-linear power filter [4, 5]
- power filter model from loudspeaker measurements [5]
- advantage: robustness during acoustic channel changes

**Figure: Parallel non-linear AEC approach**

**Figure: Cascaded non-linear AEC approach**
Parallel or cascaded approach?

- \( \tilde{h}_p(n) \Rightarrow \hat{h}_p(n) \ast \hat{h}(n) \)
- Parallel approach reach low minimum error
- Parallel approach re-estimate all taps during acoustic channel changes

**Figure:** Parallel and cascaded approach
Filter Estimation
Signal Expression

- echo signal:
  \[ y_p(n) = h_p(n)x_p(n) \]
  \[ y(n) = h(n) \sum_{p=1}^{P} y_p(n) \]

- estimated echo signal:
  \[ \hat{y}_p(n) = \hat{h}_p(n)x_p(n) \]
  \[ \hat{y}(n) = \hat{h}(n) \sum_{p=1}^{P} \hat{y}_p(n) \]

- error signal:
  \[ e(n) = y(n) - \hat{y}(n) \]

**Figure:** Cascaded non-linear AEC system
Filters Estimation
MMSE Estimator of Filters [6]

**AEC Linear Filter**

\[
\hat{h} = r_{y,y_P} R_{y_P}^{-1}
\]

where:
- \( r_{y,y_P} \) : cross-correlation
- \( R_{y_P} \) : auto-correlation

**Comments**
- depends on the loudspeaker signal
- \( R_{y_P} \Rightarrow \) determines convergence rate

**Figure:** Proposed model
Filters Estimation
MMSE Estimator of Filters [6]

**Pre-processor Sub-filters**

\[
\hat{h}_{p_k} = (r_{y,\tilde{y}_{p_k}} - r_{y_{p\neq p_k},\tilde{y}_{p_k}}) R_{y_{p_k}}^{-1}
\]  \(1\)

\(\tilde{y}_k(n) = x_{p_k}^p(n) * h(n)\)

\(r_{y,\tilde{y}_{p_k}}, r_{y_{p\neq p_k},\tilde{y}_{p_k}}\) : cross-correlation

\(R_{y_{p_k}}\) : auto-correlation

**Comments**

- \(r_{y_{p\neq p_k},\tilde{y}_{p_k}}\) \(\Rightarrow\) reduces performance
- correlation of input powers
- required of orthogonalization [4]

**Figure**: Proposed model
Filters Estimation
Adaptive Estimation \[1, 6\]

**AEC Linear Filter**

\[
\hat{h}(n+1) = \hat{h}(n) + \mu \hat{Y}_P(n)e(n)
\]

**Pre-processor Sub-filters**

\[
\hat{h}_p(n+1) = \hat{h}_p(n) + \mu_p \hat{X}_p(n)\hat{h}^T(n)e(n)
\]

**Comments**
- depends on the pre-processor output
- initialization: \( \exists \ p \ \text{s.t.} \ h_p(0) \neq 0 \)

**Comments**
- depends on the estimate of the linear filter
- stability (lower step-size)
Test Set-up

- pre-processor sub-filters
  \( P = 5 \)
- \( h_{p=1,2,3,4,5}(n) \): 100 taps
- \( h(n) \): 200 / 300 taps
- SNR (echo signal to noise ratio) 30 dB / 40 dB
Echo Path Changes (EPC) (10s)

common parameters
- initial: better performance
- $1^{st}$ EPC: pre-processor less accurate
- $2^{nd}$ EPC: pre-processor more accurate, better ERLE for $N_5$ than $N_3$
- proposed model re-converge faster and become more robust to path changes

Figure: Comparison of ERLE, 200 taps, SNR=40 dB
Echo Path Changes (EPC) (10s)

1st acoustic channel change

2nd acoustic channel change

- NLMS
- Proposed: $N_p = 3$
- Proposed: $N_p = 5$
- Power filter

13/17
Echo Path Delay Changes (EPC:10s, delay:2.5ms)

**Best tests case**
- initial: Similar performance
- 1\textsuperscript{st} EPC: pre-processor not accurate
- 2\textsuperscript{nd} EPC: pre-processor more accurate
- EPC improve the pre-processor accuracy

**Figure**: Comparison of ERLE, delay change, 300 taps, SNR=30 dB
Echo Path Delay Changes (EPC:10s, delay:2.5ms)

- **1st** test case: similar performance
- **2nd** test case: pre-processor more accurate

EPC improve the pre-processor accuracy

![Comparison of ERLE, delay change, 300 taps, SNR=30 dB](image)
Conclusions

- cascaded approach to N-AEC
- robustness against path changes
- robustness in delay changes

Perspectives

- reduce complexity
- pre-processor control with adaptive step-size
THANKS FOR YOUR ATTENTION
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