Quality Enhancement of Voice Communication & Audio experience

Application Assignment

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Outline

Although we have sufficient resources and already existing technologies in the field of audio and voice enhancement, there are a lot of opportunities for implementing advanced algorithms using Digital Signal Processing for audio and voice enhancements.

Sound Effect Normalization
- SEN Basics and implementation

Acoustic Echo Cancellation
- Background

Mono to Stereo Conversion
- Definitions
- Conversion approach & Results

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Sound Effect Normalization

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Sound Effect Normalization(SEN)

Sound Effect Normalization(SEN) can be used to improve speech intelligibility in movies through de-emphasis of background sounds and music. Most of the times it is applicable for television shows and sports where we have a lot of background noise due to people shouting and music playing.
Digital Audio Effects:- Audio effects can be classified by the way do their processing. We studied to of them:

- **Basic Filtering :** Lowpass, Highpass filter etc, Equaliser
- **Time Varying Filters :** Wah-wah

Basic Digital Audio Filtering Effects: Equalisers

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  But for many audio applications this a little too restrictive.

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- Therefore **Equalisers** are used.
  Equalisers, by contrast, enhance/diminish certain frequency bands whilst leaving others unchanged.
  They are built using a series of **Shelving** and **Peak** filters.
  Usually second-order filters are employed.

Shelving Filter

Boost or cut the low or high frequency bands with a cut-off frequency, Fc and gain G.

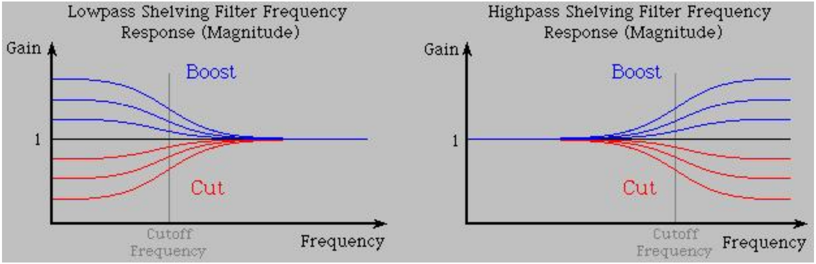

Figure: Shelving Filter Frequency Response

Shelving Filter (Cont.)

A first-order shelving filter may be described by the transfer function:

$$H(z) = 1 + H_0/2 * (1 + A(z)) \qquad .....(ForLowPass)$$
$$H(z) = 1 + H_0/2 * (1 - A(z)) \qquad .....(ForHighPass)$$

Shelving Filter (Cont.)

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where A(z) is a first-order all pass filter passes all frequencies but modifies phase:
For Boost:

$$A(z) = \frac{z^{-1} + a_B}{1 + a_B z^{-1}}$$

Similarly, for Cut we use a_C

Shelving Filter (Cont.)

The gain, G, in dB can be adjusted accordingly:

$$H_0 = V_0 - 1$$

$$\textit{where} \qquad V_0 = 10^{(G/20)}$$

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and the cut-off frequency for boost or cut are given by:

$$a_B = \frac{tan(2\pi fc/fs) - 1}{tan(2\pi fc/fs) + 1}$$

$$a_C = \frac{tan(2\pi fc/fs) - V_0}{tan(2\pi fc/fs) + V_0}$$

Peak Filter

Boost or cut mid-frequency bands with a cut-off frequency, Fc, a bandwidth, fb and gain G.

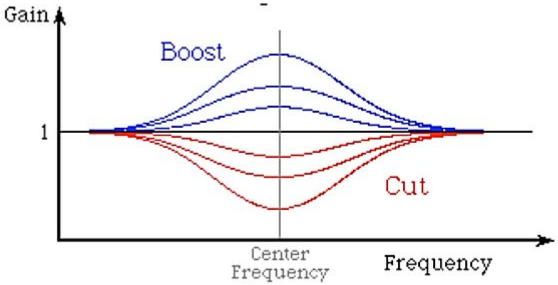

Figure: Peak Filter Frequency Response(Magnitude)

Peak Filter (Contd.)

A second-order peak filter may be described by the transfer function:

$$H(z) = 1 + H_0/2 * (1 + A(z))$$

Peak Filter (Contd.)

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$$H(z) = 1 + H_0/2 * (1 + A(z))$$

where A(z) is a second-order all pass filter
For Boost:

$$A(z) = \frac{-a_B + (d - da_B)z^{-1} + z^{-2}}{1 + (d - da_B)z^{-1} + a_B z^{-2}}$$

Similarly for Cut, we use a_C

Peak Filter (Contd.)

A second-order peak filter may be described by the transfer function:

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For Boost:

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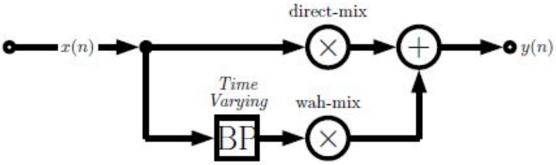
Similarly for Cut, we use a_C
The center/cut-off frequency, d, is given by:
d = -cos($2\pi fc/fs$)

Time-varying Filters

Wah-wah : A bandpass filter with a time varying centre (resonant) frequency and a small bandwidth. Filtered signal mixed with direct signal.
The signal flow for a wah-wah is as follows:

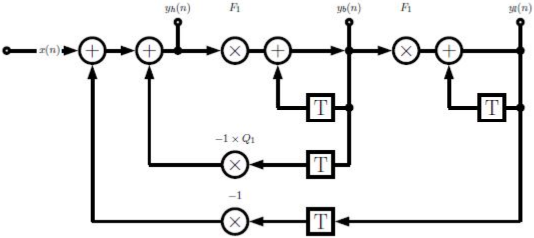


where BP is a time varying frequency bandpass filter.
It alters the tone and frequencies of the audio signal to create a distinctive sound, mimicking the human voice saying the onomatopoeic name "wah-wah". Hence the name.

Time Varying Filter Implementation: State Variable Filter

In time varying filters, independent control over the cut-off frequency and damping factor of a filter is needed.
A **State Variable Filter** is implemented for the same.
One further advantage is that lowpass, bandpass and highpass filter output are obtained simultaneously.
A state variable filter is a type of active filter. It consists of one or more integrators, connected in some feedback configuration.
A state variable filter realizes the state-space model directly.

The State Variable Filter



where:

$$\begin{aligned} \text{x(n)} &= \text{input signal} \\ \text{yl(n)} &= \text{lowpass signal} \\ \text{yb(n)} &= \text{bandpass signal} \\ \text{yh(n)} &= \text{highpass signal} \end{aligned}$$

and tuning coefficients F1 and Q1 related to the cut-off frequency, fc, and damping, d:

$$\text{F1} = 2\sin(2\pi fc/fs), and Q1 = 2d$$

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Background & Motivation

- Echo is a phenomenon where a delayed and distorted version of an original sound or electrical signal is reflected back to the source. Since the advent of telephony echoes have been a problem in communication networks
- Electrical echo due to impedance mismatch
- Acoustic echo is due to the coupling between the loudspeaker and microphone
- Did you know?
  - The range of human hearing is generally considered to be 20 Hz to 20 kHz, but it is far more sensitive to sounds between 1 kHz and 4 kHz. For example, listeners can detect sounds as low as 0 dB SPL at 3 kHz, but require 40 dB SPL at 100 hertz (an amplitude increase of 100)

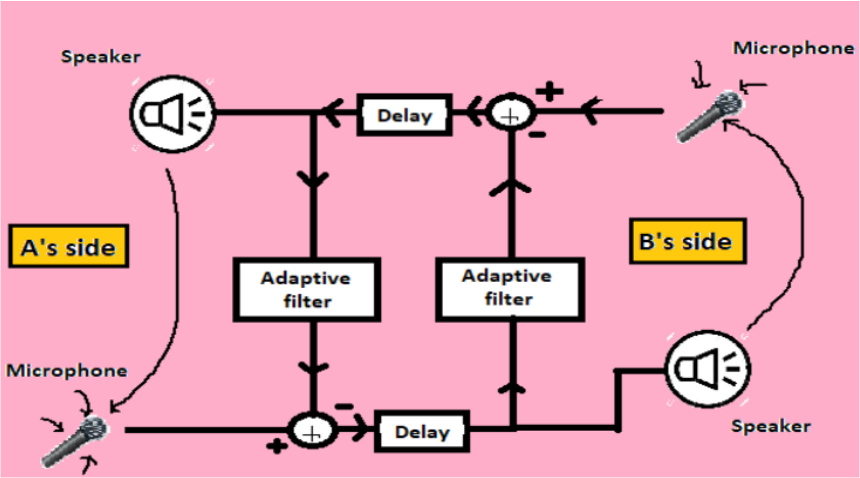

Figure: Basic block diagram of AEC Implementation

AEC Generation block

- A measured microphone signal d(n) contains two signals: The near-end speech signal r(n) The far-end echoed speech signal d(n)
- The goal is to remove the far-end echoed speech signal from the microphone signal.

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

e(n) in turns adjust the coefficients

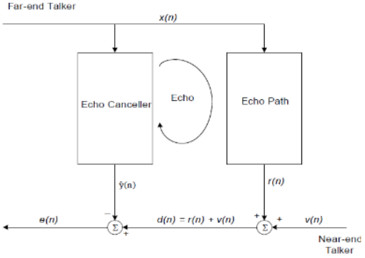

Figure: AEC Generation

Adaptive filter

- The closed loop digital adaptive filter uses feedback in the form of an error signal to refine its transfer function
- Least mean squares (LMS) algorithms are a class of adaptive filter used to mimic a desired filter by finding the filter coefficients that relate to producing the least mean square of the error signal
- [Here] the adaptive filter tends to minimize

$$[e(n)]^2$$

and accordingly adjust

$$h^*[n]$$

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What is Mono & Stereo sound?

- Stereo uses two (or more) independent audio channels to reproduce sound creating an illusion of multi-directional audible perspective
- Mono-aural sound is a single channel sound intended to reproduce sound perceived as coming from single position
- The practical difference is that mono has only one spatial dimension; something can be either close to (loud) og far away (quiet) from the listener. Stereo has two dimensions: close/far and left/right, giving a sense of space while listening to a stereo track

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Mono to Stereo Conversion approach

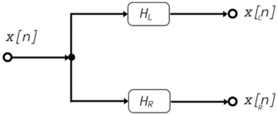

Figure: Basic blockscheme of the pseudo-stereo system

- Down-mix compatibility

$$
\begin{aligned}
x_M[n] &= x_L[n] + x_R[n] \\
&= x[n] * h_L[n] + x[n] * h_R[n] \\
&= x[n-D] \\
\Rightarrow \quad X_M(e^{j\Omega}) &= X_{(}e^{j\Omega})H_L(e^{j\Omega}) + X_{(}e^{j\Omega})H_R(e^{j\Omega}) \\
&= X(e^{j\Omega})e^{-j\Omega D}
\end{aligned}
$$

$$
\Rightarrow \quad H_L(e^{j\Omega}) + H_R(e^{j\Omega}) = e^{-j\Omega D}
$$

Continued...

- For a neutral sounding stereo output without coloration artifacts the sum of the magnitude frequency responses must be constant

$$\left|H_L(e^{j\Omega})\right| + \left|H_R(e^{j\Omega})\right| = 1$$

- For obtaining real-valued impulse responses both pseudo-stereo filters must feature conjugate symmetry

$$H_{L/R}(e^{j\Omega}) = H^*_{L/R}(e^{-j\Omega})$$

- Left channel decorrelation filter

$$H_L[k] = (1/2 + (1/\pi)\arctan(\omega^2 R[k]))e^{-j(2\pi kD/N)}$$

where R[k] is DFT of standard Gaussian Noise.

Time domain implementation

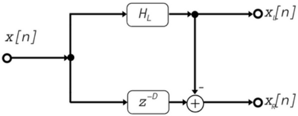

Figure: Blockscheme of the time-domain realization

- To get the left channel FIR filter impulse response, we have

$$h_L[n] = \mathcal{F}^{-1}(H_L[k])$$

- For the right channel,

$$x_R[n] = x[n-D] - x_L[n]$$

where

$$D = (N+1)/2$$

is the group delay of the FIR filter of length N.

Frequency domain implementation

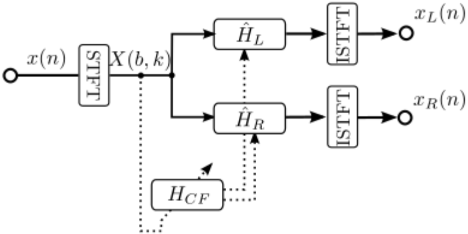

Figure: Blockscheme of the frequency-domain realization

- The time-frequency representations of the output channels are computed with a dot-wise product and a following over-lapping inverse STFT synthesis of the left and right output channel.
$$X_{L/R}(b,k) = X(b,k).H_{L/R}[k]$$

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