Characterization and Application of Echo Cancellation Methods

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C O N N E X I O N S

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Chapter 1

Background[1]

1.1 Introduction

Echoes are defined as the repetition of a sound waveform due to changes in the properties of the medium through which the transmitted signal propagates. In telecommunication services, where the presence of echo has a substantial effect on the perceived quality and intelligibility of speech, the notion of **echo cancellation** is of paramount importance.

The practice of echo suppression and cancellation began its development half a century ago, with commercial echo cancellation devices being developed at the AT&T Bell Labs in the 1960's. With the onset of wireless cellular networks and other modern telecommunication services, echo cancellation algorithms are still widely implemented.

Today, echo cancellation processors employ exhaustive digital signal processing algorithms to filter out obstreperous echo signals. In general, these devices sample the input speech signal, create a desired model signal of the echo trajectory in order to approximate the echo signal, and then subtract this desired signal from the echoed input signal to recover a clean, original transmission.

In our project, our group challenged the question of whether these techniques could be applied in dealing with the complex problem of cancelling a room's echo.

[1] This content is available online at <http://cnx.org:8888/content/m45382/1.2/>.

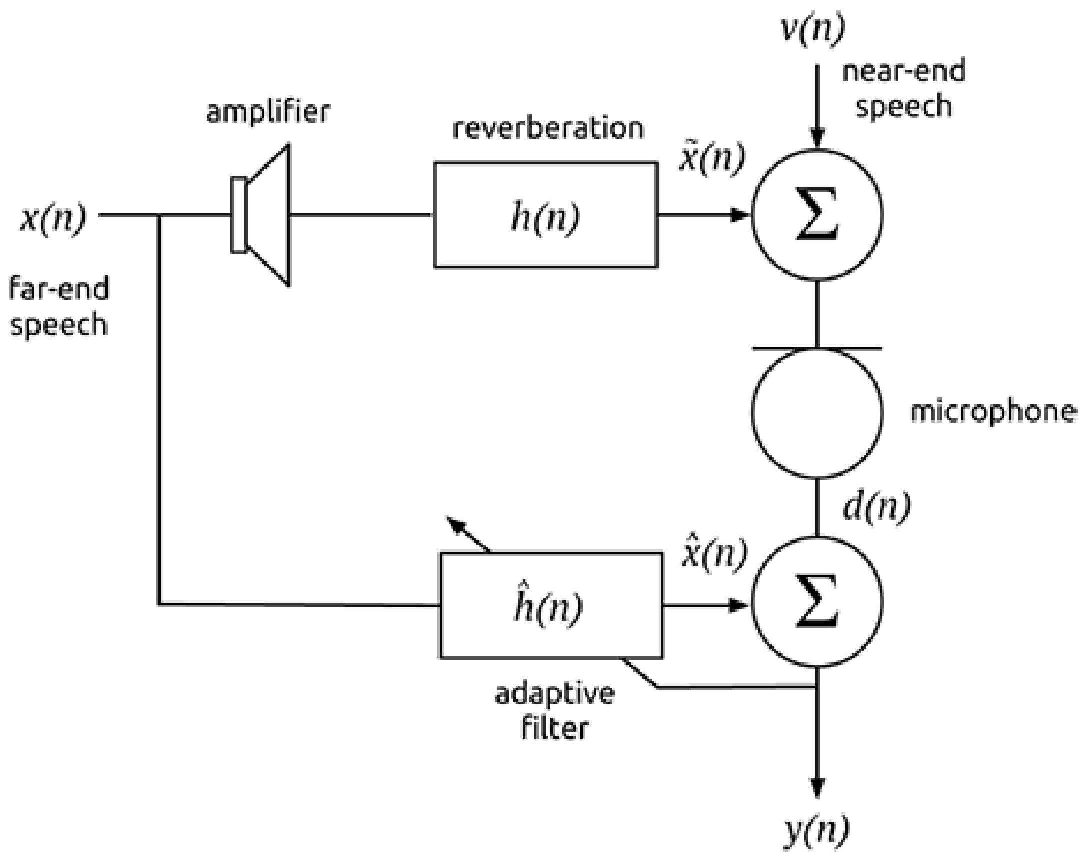

Figure 1.1: Block diagram example of typical acoustic echo cancellation.

1.2 Project Motivation

When seniors in Rice's capstone engineering design course present in the Oshman Engineering Design Kitchen (OEDK) classroom, they are recorded for future reference. The room however suffers from excess acoustic reverberation making the recordings difficult to understand. Many lecturers and students have complained about the sound quality in the classroom. However the OEDK has hired more than three contractors to diagnose the problem. Each had conflicting reports on the cause of the problems (the high ceiling, concrete floors, magnetic walls).

Example 1.1
[MEDIA OBJECT][2] Recording played in acoustically desirable room.
[MEDIA OBJECT][3] The same recording played in OEDK classroom.

Because the problem areas are not uniform, and physically altering the room would be expensive, we approached the problem from a digital signal processing perspective. Implementing an echo cancellation filter

[2]This media object is an audio file. Please view or download it at <OriginalFemale.wav>

[3]This media object is an audio file. Please view or download it at <EchoedFemale.wav>

to the microphone's signal should improve the quality of the presentation's audio when being played back by students or other future listeners and allow for seniors to receive better feedback on their presentation skills.

1.3 Main Goals

- Research and implement methods of echo cancellation, especially those used in telecommunications, to be used for rectifying the classroom's acoustic problem.
- Make a recommendation for approaching echo cancellation in OEDK classroom.

Chapter 2

Reverberation in Enclosed Spaces[1]

2.1 Reverberation

When sound is made in an enclosed space, echoes of the signal reflect of the room's surfaces and form a **reverberated signal**, a superposition of the signal generated and delayed versions of itself with amplitude decay. These reverberations make listening unpleasant and can sometimes make a presentation unintelligible. The OEDK classroom suffers greatly from reverberations which are distracting when listening to recordings of presentations.

Two Room Impulse Responses

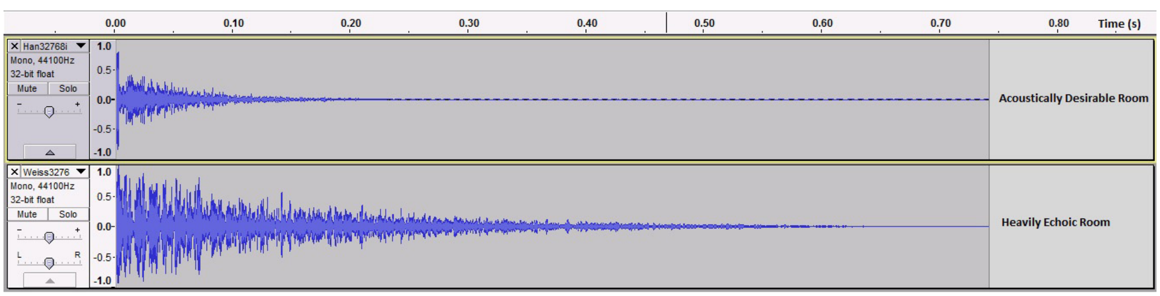

Figure 2.1: Echoes of the impulse can especially be seen in the echoic room, where the signal takes longer to decay.

2.2 Reverberation Time

The **reverberation time** of a room is the time it takes for sound to decay by 60 decibels. The reverberation time of a room can be approximated by the **Sabine Equation**:

$$RT_{60} = \frac{4\ln 10^6}{c}\frac{V}{Sa} \approx 0.1611\ \mathrm{m}^{-1}\frac{V}{Sa} \tag{2.1}$$

[1]This content is available online at <http://cnx.org:8888/content/m45387/1.1/>.

where c is a constant relating to the speed of sound in the room, V is the volume of the room, S is the surface area, and a is the average of the Sabine Absorption coefficients of all the objects in the room.

The Sabine absorption coefficient is a ratio of how much sound is absorbed or reflected off the surface of an object. This ratio changes based on the frequency of the sound. Rooms tend have more reverb at lower frequencies.

Empirically, the reverberation time of a room can be measured by recording the response of the room to a unit impulse or pink/white noise and seeing how long it takes for the sound to decay by 60 decibels. The measured reverberation times of the OEDK classroom, an acoustically desirable room, and a heavily echoic room are displayed below.

	Reverberation Time (s)	**Description**
Acoustically Desirable Room	.23	Carpeted room in Hanszen College filled with soft furniture.
OEDK Classroom	.57	Large classroom with concrete floor, magnetic walls, and a high ceiling.
Heavily Echoic Room	.62	Tiled classroom in Weiss College with glass walls and single table.

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Measured Reverberation Times in 3 Rooms

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Table 2.1

Example 2.1
[MEDIA OBJECT][2] Impulse in Acoustically Desirable Room.
[MEDIA OBJECT][3] The same impulse played in Heavily Echoic Room. The difference in reverberation time is audible.

Reverberation time is a good way to diagnose how badly a room echoes a generated signal. The actual echo cancellation process involves digitally filtering out the decaying copies of the generated speech signals that cause the room's poor reverberation times.

[2]This media object is an audio file. Please view or download it at <LowReverbImpulse.wav>
[3]This media object is an audio file. Please view or download it at <HighReverbImpulse.wav>

Chapter 3

The Deconvolution Method[1]

3.1 Deconvolution

The output of a linear time-invariant (LTI) system is the convolution of the input signal with the impulse response of the system. If the classroom is modeled as an LTI system, then the output echoed signal, $y(t)$, is the convolution of the input signal, $x(t)$, with the room's impulse response, $h(t)$.

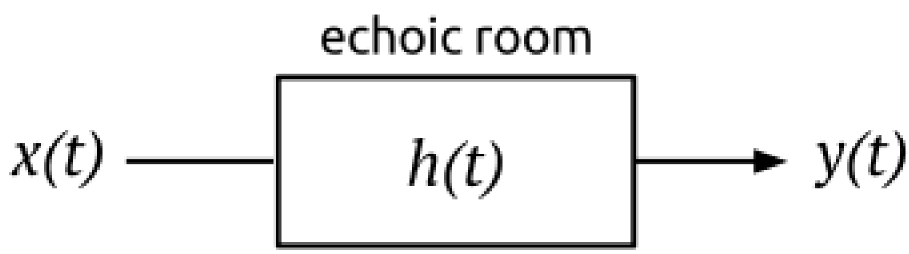

Figure 3.1: Block diagram of the echoic room as an LTI system with impulse response $h(t)$ where $y(t)$ is the echoed version of the input $x(t)$.

The process of **deconvolution** involves designing an inverse filter $\hat{\mathrm{h}}(\mathrm{t})$ that is convolved with the echoed output signal to retrieve the original signal $x(t)$. This can be done in either the time domain or frequency domain:

- **Time domain:** Use the `deconv()` method in Matlab on the echoed signal and the impulse response of the classroom in order to extract the de-echoed signal.
- **Frequency domain:** Take the Fast Fourier Transform (FFT) of both the impulse response of the room and the echoed signal. Point-wise divide the echoed signal by the transfer function, then take the inverse FFT of the result to extract the de-echoed signal.

[1]This content is available online at <http://cnx.org:8888/content/m45389/1.1/>.

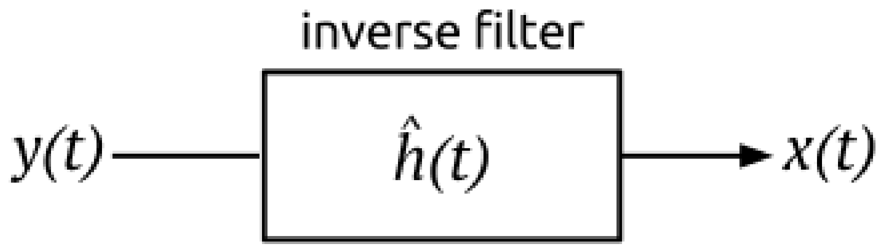

Figure 3.2: Inputting the echoed signal into the inverse filter yields the de-echoed signal.

However in practice, the system adds noise to input signal, meaning that if the signal-to-noise ratio is too low, the inverse filter will yield a noisy signal that poorly approximates the input.

3.2 Finding the Impulse Response

A system is characterized at all frequencies by taking its **impulse response**. This is done by exciting a system with a **Dirac delta function**. The Dirac Delta Function is defined as:

$$\delta(x) = \{ \begin{array}{cc} +\infty, & x = 0 \\ 0, & x \neq 0 \end{array} \} \tag{3.1}$$

$$\int_{-\infty}^{\infty} \delta(x)\, dx = 1 \tag{3.2}$$

NOTE: The Dirac Delta Function is a distribution that is infinitely tall and infinitely narrow at 0, and the area under the Dirac Delta Function is defined to be 1.

However, it is practically impossible to excite a room with the ideal Dirac Delta function. Consequently, we used three methods to approximate the room's impulse response:

- **Balloon Pop:** filling a latex balloon with air and bursting it.
- **Pseudo Dirac:** using the `dirac()` function in Matlab to generate a vector of zeros with a single one in the center, then playing it using the `sound()` function.
- **Sine-sweep Method:**
  a. Generate a logarithmically increasing sine signal over a desired frequency range (20 Hz to 20 KHz for this application)
  b. Create an inverse chirp filter that time reverses the chirp and shifts it to become a causal signal (so that it exists in positive time). Then, divide the magnitude of the spectrum of the inverse filter by the square of the magnitude of the spectrum of the chirp signal.

  NOTE: The time shift inverts the phase of the chirp leading to linear phase after the convolution and the second set of operations neutralizes the squaring of the magnitude of the spectrum caused by the convolution.

3. Convolve the chirp response of the room with the inverse chirp filter to get the impulse response of the room.

This logarithmically increasing sine signal can be characterized in the time domain by the following equation:

$$x(t) = \sin\left[\frac{T\omega_1}{\ln\left(\frac{\omega_2}{\omega_1}\right)}^{\left(e^{\frac{t}{T}\ln\left(\frac{\omega_2}{\omega_1}\right)}-1\right)}\right] \tag{3.3}$$

where ω_1 is the initial radian frequency and ω_2 is the final radian frequency of the sweep of duration T.

Example 3.1: Sine Sweep
[Media Object][2]

[2]This media object is an audio file. Please view or download it at <oedksweep.wav>