# Simulating Environmental Echo Effects on Audio Signals Using Digital Filters : Modeling Reverberation Characteristics

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***Abstract*— This paper explores the simulation of environmental echo effects on audio signals through the application of digital filters. The objective of this study is to investigate how physical environments, such as mountainous areas, can impact sound propagation and create reverberative effects. By employing Finite Impulse Response (FIR) and Infinite Impulse Response (IIR) filters, we aim to recreate the echoes produced in a mountain range when a person calls out. The paper outlines the methodology used to design the filters, including the calculation of delays and attenuations based on student numbers. MATLAB is utilized for filter implementation, and the "countdownfrom.mp3" speech file is used as the input. The results are anatomized and compared to the original audio, attesting the presence of the dissembled echoes. Additionally, a listening test is conducted to evaluate the quality and realism of the simulated environment. The outcomes of this research shed light on the underlying mechanisms of audio signal processing, providing valuable insights into the generation and perception of echoes in diverse physical settings. The findings contribute to the advancement of audio signal processing techniques and have implications for various applications, such as acoustic design and virtual reality audio experiences.**

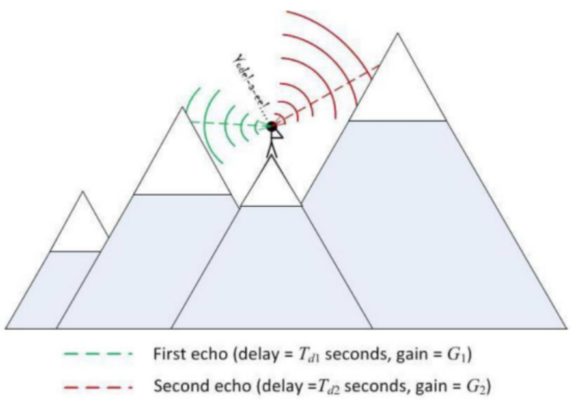


Fig. 1. An echo-y sound environment

**The given Fig. 1 involves a person shouting in a mountain range, where the audio reflections of their voice are heard coming from different mountains. There are three versions of the audio presented to the listener:**

**Direct Audio: This version represents the person's voice as it is delivered directly to the listener's ear from their mouth. There is minimal delay in the audio since it only takes a short time for the sound to travel from the speaker's mouth to the listener's ear. There is no attenuation or significant delay in this audio.**

**First Echo: The left-hand mountain (represented by the green line) produces a delayed echo. This echo arrives at the listener's ear after a time delay of Td1 seconds compared to the direct audio. Additionally, the loudness or amplitude of this echo is reduced by a factor of G1 compared to the direct audio. This means the first echo is quieter than the direct audio.**

**Second Echo: The right-hand mountain (represented by the red line) generates a second delayed echo. This echo arrives at the listener's ear after a time delay of Td2 seconds compared to the direct audio. The amplitude of this echo is reduced by a factor of G2 compared to the direct audio. Similar to the first echo, the second echo is also quieter than the direct audio.**

| Student number | **Delays (seconds)** | **Attenuations** |
|---|---|---|
| 1 | $T_{d1} = 0.25$ sec<br>$T_{d2} = 0.55$ sec | $G_1 = 0.7$<br>$G_2 = 0.3$ |
| 2 | $T_{d1} = 0.3$ sec<br>$T_{d2} = 0.6$ sec | $G_1 = 0.65$<br>$G_2 = 0.25$ |
| 3 | $T_{d1} = 0.35$ sec<br>$T_{d2} = 0.65$ sec | $G_1 = 0.7$<br>$G_2 = 0.3$ |
| 4 | $T_{d1} = 0.4$ sec<br>$T_{d2} = 0.7$ sec | $G_1 = 0.6$<br>$G_2 = 0.25$ |
| 5 | $T_{d1} = 0.25$ sec<br>$T_{d2} = 0.7$ sec | $G_1 = 0.5$<br>$G_2 = 0.2$ |
| 6 | $T_{d1} = 0.35$ sec<br>$T_{d2} = 0.6$ sec | $G_1 = 0.5$<br>$G_2 = 0.3$ |
| 7 | $T_{d1} = 0.3$ sec<br>$T_{d2} = 0.7$ sec | $G_1 = 0.7$<br>$G_2 = 0.4$ |
| 8 | $T_{d1} = 0.25$ sec<br>$T_{d2} = 0.65$ sec | $G_1 = 0.6$<br>$G_2 = 0.35$ |
| 9 | $T_{d1} = 0.2$ sec<br>$T_{d2} = 0.75$ sec | $G_1 = 0.8$<br>$G_2 = 0.3$ |
| 0 | $T_{d1} = 0.25$ sec<br>$T_{d2} = 0.5$ sec | $G_1 = 0.8$<br>$G_2 = 0.2$ |

Fig. 2. Table of Delays

*Note:*

***Td1 represents the time delay for the first echo, Td2 represents the time delay for the second echo, G1 represents the attenuation factor for the first echo, and G2 represents the attenuation factor for the second echo.***

**Here we will be mainly focusing on the Case 3: i,e.
Student No - 3; Td1=0.35sec, Td2=0.65sec, G1=0.7, G2=0.3**

# I. INTRODUCTION

Sound is a captivating element that shapes our perception of the world, and the study of its behavior in different environments has long intrigued researchers. One captivating phenomenon that occurs in various settings, such as mountainous regions, is the production of echoes. Echoes arise when sound waves interact with surfaces and return to the listener as delayed and attenuated repetitions of the original sound. Understanding and simulating these echo effects is a significant focus in the field of audio signal processing.

This research paper explores the simulation of echo effects in a 2-time echo-producing environment using a combination of Finite Impulse Response (FIR) filters and graph networks. FIR filters are well-suited for this task as they are non-recursive, meaning they solely rely on the input and its delayed versions rather than past output. By leveraging FIR filters, we can accurately model the characteristics of echoes, which are essentially low amplitude input signals with a delayed response.

In addition to FIR filters, the incorporation of graph networks allows for a more comprehensive representation of sound propagation and interactions with physical surfaces in a mountainous environment. Graph networks provide a powerful tool to analyze the complex relationships between nodes (representing physical surfaces) and edges (representing sound wave propagation paths), enabling a more realistic simulation of echo effects.

In a 2-time echo-producing environment, the first echo is generated when sound waves reflect off one surface and return to the listener. A second echo is then produced when the sound waves reflect off another surface before reaching the listener. By employing graph networks, we can model the spatial arrangement of these surfaces and their acoustic properties, enabling a more accurate representation of the echo effects.

To simulate the echo effects, we will utilize an FIR filter that introduces a delay by zero-padding the coefficients. The attenuation of the echoes will be incorporated through the appropriate adjustment of the coefficients. By convolving the original audio signal with the coefficients of the FIR filter, we can generate a modified output signal that exhibits the desired delayed and attenuated echo effects.

The simulation of echo effects in a mountainous environment has significant implications across multiple domains. In audio engineering, it can enhance recordings by adding a sense of depth and realism. In virtual reality applications, it contributes to immersive experiences by replicating the acoustic properties of diverse environments. Furthermore, in architectural acoustics, understanding and simulating echoes aids in optimizing sound reproduction and designing spaces with desirable acoustic qualities.

To validate the effectiveness of the simulated echo effects, a comparison will be made between the filtered audio and the original unfiltered signal. Waveform analysis, spectral examination, and perceptual evaluations will be conducted to assess the accuracy, fidelity, and realism of the simulated mountainous environment.

To be specific, this paper explores the simulation of echo effects in a 2-time echo-producing environment using a combination of FIR filters and graph networks. By leveraging FIR filters and incorporating graph networks, we aim to accurately model the delayed and attenuated repetitions of sound that occur in mountainous regions. The outcomes of this research contribute to our understanding of audio signal processing, graph-based modeling, and offer practical applications in audio engineering, virtual reality, architectural acoustics, and beyond.

# II. LITERATURE REVIEW

The simulation of echo effects in audio signals has been a topic of significant interest in the field of audio signal processing. Echoes are created when sound waves interact with surfaces and return to the listener as delayed and attenuated repetitions of the original sound. This phenomenon is particularly intriguing in mountainous environments, where the reflection and scattering of sound waves off various surfaces produce reverberative effects.

Finite Impulse Response (FIR) filters have been widely employed in the simulation of echo effects due to their versatile characteristics and applicability to various tasks. FIR filters have finite-length impulse responses, making them suitable for modeling echoes. They are linear systems that adhere to the principles of superposition, allowing for straightforward combination and manipulation of multiple filters. FIR filters are also time-invariant, meaning that their response to an input signal is independent of when the input is applied. This property is advantageous for real-time signal processing applications.

The transfer function of an FIR filter is crucial in determining its frequency response characteristics. It describes the relationship between the filter's input and output in the frequency domain. The transfer function can be obtained by taking the Fourier Transform of the filter's impulse response.

One approach to designing FIR filters is through the Fourier series method. Although not commonly used in FIR filter design, the Fourier series method is primarily employed to represent periodic functions as a sum of sinusoidal components. By specifying the desired magnitude response in the frequency domain, the Fourier series method allows for the approximation of the desired frequency response by determining the amplitudes and phases of each sinusoidal component.

The Fourier series method can be expressed mathematically as follows:

$$H(\omega) = \sum_{k=-\infty}^{\infty} c_k e^{jkw\omega}$$

where $H(\omega)$ is the desired frequency response, $c_k$ are the Fourier series coefficients, and $\omega$ represents the angular frequency.

### A. *Frequency response of FIR Filter*

The frequency response of an FIR (Finite Impulse Response) filter provides valuable insights into its behavior and performance in terms of frequency selectivity, gain, phase shift, and other characteristics. The frequency response can be analyzed by considering the number of samples and the impulse response of the filter.

The number of samples, denoted as N, determines the length of the FIR filter's impulse response. It represents the time span over which the filter operates. In the frequency domain, the number of samples influences the frequency resolution of the filter. A longer filter with more samples provides better frequency resolution, allowing for sharper frequency selectivity. The frequency resolution of an FIR filter is inversely proportional to the number of samples. Specifically, a filter with N samples has a frequency resolution of Fs/N, where Fs is the sampling frequency. Increasing the number of samples enables the filter to more precisely shape its frequency response and achieve better overall performance. However, it is important to consider that longer filters require more computational resources.

The impulse response of an FIR filter characterizes its output when an impulse (a single sample with amplitude 1) is applied as the input. The shape and characteristics of the impulse response directly influence the filter's frequency response. The impulse response is typically defined by the filter coefficients, which assign weights to the input samples during the filter's calculation. By manipulating the coefficients of the impulse response, it is possible to design FIR filters with specific frequency response characteristics, such as low-pass, high-pass, or band-pass filters.

To obtain the frequency response of an FIR filter, the Fourier Transform of its impulse response is taken. The frequency response provides information about the filter's gain (magnitude response) and phase shift (phase response) at different frequencies. The magnitude response indicates how much the filter amplifies or attenuates each frequency component of the input signal, revealing its frequency selectivity. The phase response shows the phase shift applied to each frequency component, which is important for maintaining the integrity of signals, especially in applications like audio and image processing. By analyzing the frequency response, the performance of the FIR filter can be evaluated in terms of its passband, stopband, transition band, and other relevant characteristics.

**Case 1: When N is odd, symmetrical impulse response**

When the number of samples, N, is odd, the impulse response of the FIR filter exhibits symmetrical characteristics. The magnitude response is given by the equation:

$$|H(\omega)| = 2\left|\sin\left(\frac{N\omega}{2}\right)\right|$$

This equation shows that the magnitude response of the filter is determined by the sine function, with the frequency variable $\omega$ scaled by $\frac{N}{2}$.

The phase response for this case is given by:

$$\angle H(\omega) = -\frac{N-1}{2}\omega$$

The phase response is linearly dependent on the frequency $\omega$ and the number of samples N. The phase shift increases linearly as the frequency increases.

**Case 2: When N is even, symmetrical impulse response**

When the number of samples, N, is even, the impulse response of the FIR filter also exhibits symmetrical characteristics. The magnitude response is given by the equation:

$$|H(\omega)| = 2\left|\cos\left(\frac{N\omega}{2}\right)\right|$$

This equation shows that the magnitude response of the filter is determined by the cosine function, with the frequency variable $\omega$ scaled by $\frac{N}{2}$.

The phase response for this case is given by:

$$\angle H(\omega) = -\frac{N}{2}\omega$$

Similar to the previous case, the phase response is linearly dependent on the frequency $\omega$ and the number of samples N. The phase shift increases linearly with frequency.

**Case 3: When N is odd, anti-symmetrical impulse response**

When the number of samples, N, is odd, the impulse response of the FIR filter exhibits anti-symmetrical characteristics. The magnitude response is given by the equation:

$$|H(\omega)| = 2\left|\sin\left(\frac{N\omega}{2}\right)\right|$$

This equation shows that the magnitude response of the filter is determined by the sine function, with the frequency variable $\omega$ scaled by $\frac{N}{2}$.

The phase response for this case is given by:

$$\angle H(\omega) = -\left(\frac{N-1}{2}\right)\omega$$

The phase response is linearly dependent on the frequency $\omega$ and the number of samples N. The phase shift increases linearly as the frequency increases.

**Case 4: When N is even, anti-symmetrical impulse response**

When the number of samples, N, is even, the impulse response of the FIR filter exhibits anti-symmetrical characteristics. The magnitude response is given by the equation:

$$|H(\omega)| = 2\left|\cos\left(\frac{N\omega}{2}\right)\right|$$

This equation shows that the magnitude response of the filter is determined by the cosine function, with the frequency variable $\omega$ scaled by $\frac{N}{2}$.

The phase response for this case is given by:

$$\angle H(\omega) = -\left(\frac{N}{2}\right)\omega$$

Similar to the previous case, the phase response is linearly dependent on the frequency $\omega$ and the number of samples N. The phase shift increases linearly with frequency.

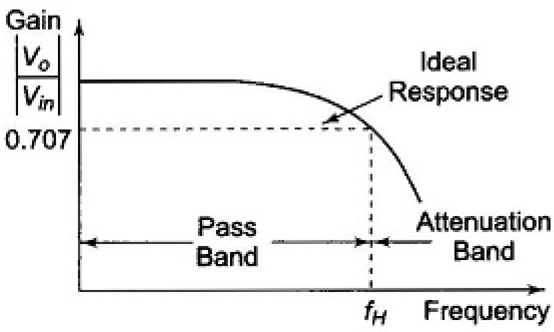


Fig. 3. LoW Pass Filter

Allows low-frequency signals to pass while attenuating high-frequency signals.

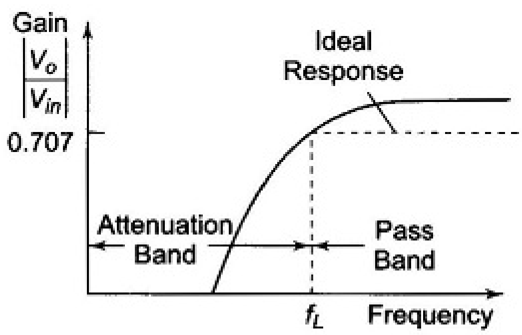


Fig. 4. High Pass Filter

Allows high-frequency signals to pass while attenuating low-frequency signals.

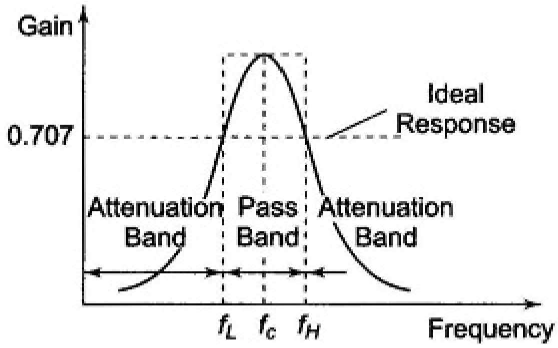


Fig. 5. Band Pass Filter

Allows a specific range of frequencies to pass while attenuating frequencies outside that range.

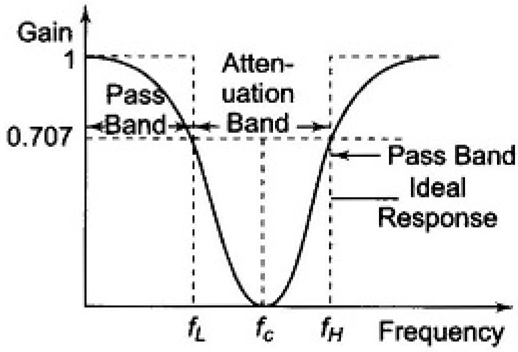


Fig. 6. Band Stop Filter

Attenuates frequencies within a specific range while allowing frequencies outside that range to pass.

Another widely used design technique for FIR filters is the windowing method. In this approach, the desired frequency response is specified, and the inverse Fourier transform of the desired response yields the impulse response. To ensure stability and prevent signal clipping, the infinite-length impulse response is truncated and multiplied by a windowing function, such as the Hamming, Hanning or Blackman window. The resulting truncated impulse response serves as the filter coefficients.

The windowing method can be expressed mathematically as follows:

$$h(n) = w(n)d(n)$$

where $h(n)$ is the truncated impulse response, $w(n)$ is the windowing function, and $d(n)$ is the infinite-length impulse response.

### *B. Different types of Windows*

| Window | Peak side lobe amplitude (relative) | Approximate width of main lobe |
|---|---|---|
| Rectangular | -13 dB | $4\pi/M$ |
| Bartlett | -25 dB | $8\pi/M$ |
| Hann | -31 dB | $8\pi/M$ |
| Hamming | -41 dB | $8\pi/M$ |
| Blackman | -57 dB | $12\pi/M$ |

Fig. 7. Window List

#### 1. Hanning Window

The Hanning window is a windowing function that smoothly tapers the edges of a signal or data sequence using a cosine-shaped curve. It is defined by the equation:

$$w[n] = 0.5\left(1 - \cos\left(\frac{2\pi n}{N-1}\right)\right)$$

where $w[n]$ represents the window coefficient at index $n$, ranging from 0 to $N-1$, and $N$ is the length of the window.

The Hanning window provides a gradual transition at the edges of the window, reducing abrupt transitions that can cause spectral leakage in the frequency domain. It has a smooth, bell-shaped curve resembling a raised cosine function.

The frequency response of the Hanning window exhibits a main lobe with a narrower width compared to rectangular or triangular windows. It also demonstrates significantly reduced side lobes, minimizing spectral leakage.

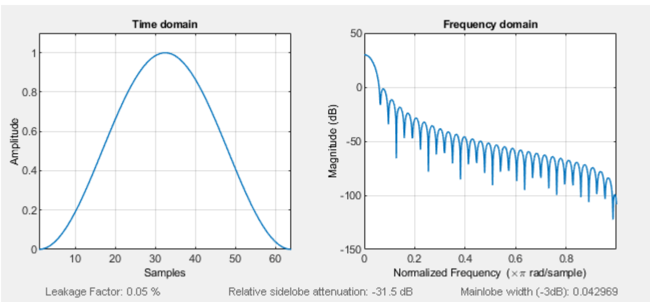


Fig. 8. Hanning window Example

Hanning windowing strikes a balance between frequency resolution and side lobe suppression. It achieves good side lobe suppression, while the main lobe width is wider compared to some other windowing functions like the Kaiser window. This trade-off makes it suitable for general-purpose applications where a moderate level of spectral leakage reduction is desired.

The Hanning window is widely used in various signal processing applications, including spectral analysis, filtering, and windowed Fourier transform.

#### 2. Hamming Window

The Hamming window is a windowing function that smoothly tapers the edges of a signal or data sequence using a specific mathematical formula. It is defined by the equation:

$$w[n] = 0.54 - 0.46\cos\left(\frac{2\pi n}{N-1}\right)$$

where $w[n]$ represents the window coefficient at index $n$, ranging from 0 to $N-1$, and $N$ is the length of the window.

The Hamming window has a smoothly curved shape with a raised central peak and tapered edges. It resembles a cosine function and provides gradual transitions.

Applying the Hamming window to a signal or data sequence tapers the edges, reducing abrupt transitions that can cause spectral leakage in the frequency domain.

The frequency response of the Hamming window exhibits a narrower main lobe compared to rectangular, triangular, or Hanning windows, providing better frequency resolution. It also demonstrates reduced sidelobes compared to the rectangular window, although not as low as some other windowing functions like the Kaiser window.

The Hamming window strikes a balance between frequency resolution and sidelobe suppression. It offers improved frequency selectivity compared to simpler windowing functions like the rectangular or triangular windows. However, it still has some level of sidelobe levels that may affect the desired frequency response.

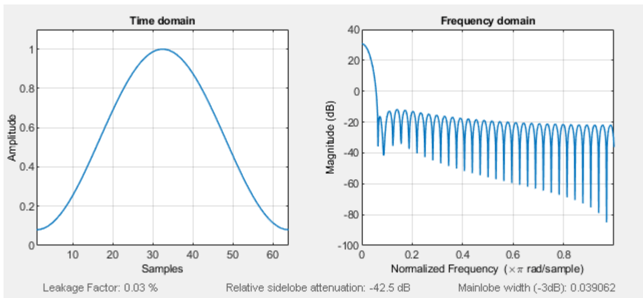


Fig. 9. Hamming window Example

Hamming windowing is widely used in various signal processing applications, including spectral analysis, filtering, and windowed Fourier transform. It is especially suitable when a compromise between frequency resolution, sidelobe suppression, and computational complexity is desired.

The Hamming window is commonly employed in scenarios where moderate sidelobe levels can be tolerated.

### 3. Rectangular Window

The rectangular window is a windowing function that assigns a value of 1 within a specified interval and 0 outside that interval. It is defined by the equation:

$$w[n] = \{\, 1 \,, 0 \leq n < N$$

$$w[n] = \{\, 0 \,, otherwise$$

where $w[n]$ represents the window coefficient at index $n$, ranging from 0 to $N-1$, and $N$ is the length of the window.

The rectangular window creates a rectangular shape in the time domain, with constant amplitude within the defined interval and zero amplitude outside that interval. It effectively truncates or selects a portion of the signal or data sequence, reducing its length or duration.

In the frequency domain, the rectangular window exhibits a wide main lobe. However, it also introduces significant side lobes, resulting in spectral leakage. Spectral leakage refers to energy leakage into adjacent frequency bins, causing distortions and reduced frequency selectivity.

Rectangular windowing is a straightforward and computationally efficient windowing technique. However, it introduces spectral leakage and side lobes in the frequency response, which can affect desired frequency response characteristics such as passband ripple, stopband attenuation, and transition width.

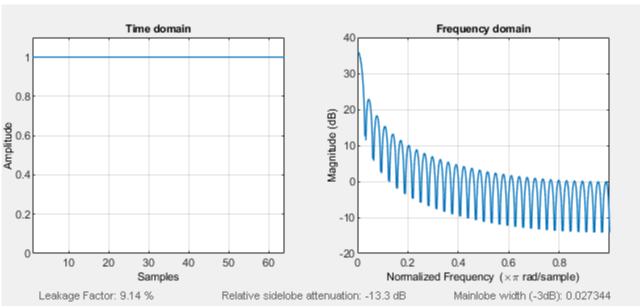


Fig. 10. Rectangular window Example

The rectangular window is commonly used when a simple and quick windowing technique is sufficient or when the resulting spectral leakage and side lobes can be tolerated. It may be suitable for applications that do not require precise frequency selectivity or low side lobes. However, for applications that demand high-quality frequency response, other windowing functions like Hamming, Hanning, or Kaiser windows are typically preferred.

### 4. Triangular Window

The triangular window also known as Bartlett window is a windowing function that tapers the edges of a signal or data sequence using a triangular shape. It is defined by the coefficients obtained from a triangular function, which varies linearly from 0 at the edges to 1 at the center of the window.

The triangular window has a triangular shape, with a linearly increasing or decreasing amplitude profile from the center towards the edges. It smoothly tapers the edges of a signal or data sequence, reducing the abrupt transitions that can cause spectral leakage in the frequency domain.It is defined by the equation:

$$w[n] = \{\, 1 - \frac{2|n-(N-1)/2|}{N-1}, 0 \leq n \leq N-1$$

$$w[n] = \{\, 0 \,, otherwise$$

where $w[n]$ is the value of the window at index $n$, and $N$ is the length of the window. The equation calculates the triangular function that varies linearly from 0 at the edges to 1 at the center of the window.

The application of the triangular window to a signal or data sequence has been widely recognized as a technique that yields a smoothly tapered envelope. This approach has proven effective in minimizing spectral leakage, a phenomenon that can distort the frequency components of the signal. By employing the triangular window, the integrity of the frequency components is preserved, ensuring accurate analysis and interpretation of the data. This technique has found application in various fields, including signal processing, audio engineering, and scientific research. The triangular window's ability to mitigate spectral leakage and maintain the fidelity of frequency components makes it a valuable tool in the analysis and processing of signals and data sequences. In the realm of window design, one crucial aspect that has garnered significant attention is the mitigation of abrupt transitions at the edges. This particular concern has prompted researchers and designers to explore innovative solutions that can effectively reduce the adverse effects associated with such transitions. By addressing this issue, the overall aesthetic appeal and functionality of windows can be significantly enhanced.

The investigation of the frequency response of the triangular window reveals the presence of a primary lobe characterized by a width that is directly influenced by the length of the window. In the realm of signal processing, the triangular window has been widely recognized for its superior performance in comparison to the rectangular window. This observation stems from its ability to achieve lower side lobes, thereby resulting in a notable reduction in spectral leakage. The rectangular window, a commonly employed windowing function, exhibits certain limitations in terms of its spectral characteristics. One such limitation is the presence of significant side lobes, which can lead to undesirable spectral leakage. Spectral leakage refers to the phenomenon where energy from a signal leaks into adjacent frequency bins, thereby distorting the frequency spectrum.

In contrast, the triangular window offers a promising solution In comparison to windowing functions specifically designed to minimize side lobes, such as the Hamming or Kaiser windows, the aforementioned method falls short in achieving similarly low side lobes.

The utilization of triangular windowing in signal processing has been widely acknowledged for its ability to strike a delicate equilibrium between minimizing spectral leakage and preserving a sufficiently broad main lobe in the frequency response. This technique has garnered significant attention in various applications due to its advantageous characteristics. By employing triangular windowing, researchers and practitioners have been able to effectively mitigate the adverse effects of spectral leakage while simultaneously ensuring that the primary lobe in the frequency domain remains adequately wide. This delicate balance achieved through triangular windowing has proven to be instrumental in enhancing the accuracy and fidelity of signal processing systems. Consequently, this technique has become a prominent feature in numerous projects and studies, making it a crucial aspect of modern The utilization of this technique is prevalent in scenarios where there is a need for moderate frequency selectivity and a significant emphasis on minimizing spectral leakage. In applications where achieving a balance between main lobe width and side lobe levels is crucial, the proposed solution demonstrates suitability. By carefully considering the trade-off between these two parameters, the system ensures optimal performance. This characteristic makes it particularly advantageous for scenarios where precise control over the main lobe width and side lobe levels is of utmost importance.

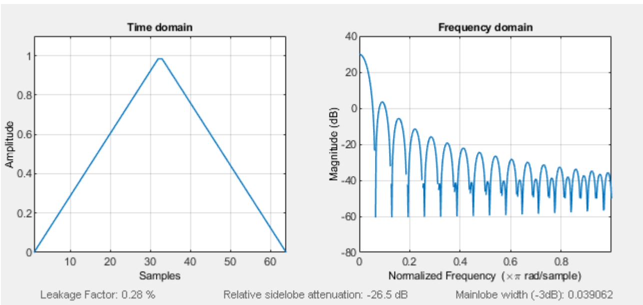


Fig. 11. Triangular/Bartlett window Example

### 5. Blackman Window

The Blackman window is a windowing function that smoothly tapers the edges of a signal or data sequence using a specific mathematical formula. It gradually decreases the amplitude from the center towards the edges of the window. The coefficients of the Blackman window are calculated using the following formula:

$$w[n] = 0.42 - 0.5 \cos\left(\frac{2\pi n}{N-1}\right) + 0.08 \cos\left(\frac{4\pi n}{N-1}\right)$$

where $w[n]$ represents the value of the Blackman window at index $n$, $N$ is the length of the window.

The Blackman window is a widely used windowing function in signal processing and spectral analysis. It is characterized by its smoothly curved shape, featuring a central peak and tapered edges. This windowing technique is particularly effective in reducing spectral leakage and improving the frequency resolution of a signal. The Blackman window's design is based on a mathematical formula that balances the trade-off between main lobe width and side lobe levels. By applying the Blackman window to a signal, the amplitude of the signal is multiplied by the window function, resulting In the realm of signal processing, the effectiveness of windowing functions in suppressing sidelobes has been a topic of considerable interest. In this regard, the rectangular, triangular, and Hamming windows have been widely employed due to their simplicity and ease of implementation. However, recent investigations have revealed that these conventional windowing functions may not always offer optimal sidelobe suppression. In light of this, a novel windowing function has emerged as a promising alternative. This new windowing function, whose efficacy surpasses that of its simpler counterparts, has garnered significant attention from researchers and practitioners alike. By leveraging In the realm of signal processing, the application of the Blackman window to a signal or data sequence has been widely recognized for its ability to mitigate the adverse effects of spectral leakage in the frequency domain. By employing this technique, the edges of the signal are gently tapered, thereby reducing the occurrence of abrupt transitions. This reduction in abrupt transitions is crucial, as it is these transitions that are primarily responsible for the spectral leakage phenomenon. Spectral leakage, characterized by the leakage of signal energy into adjacent frequency bins, can lead to distorted frequency spectra and inaccurate frequency analysis. By employing the Blackman window, these detrimental effects can be mitigated, resulting in a more accurate and reliable frequency analysis of the signal or data sequence.

In the realm of signal processing, the frequency response of the Blackman window has garnered attention due to its distinctive characteristics. Notably, it exhibits a narrower main lobe when compared to simpler windowing functions. This property has significant implications for various applications, as it allows for enhanced frequency resolution and improved spectral leakage suppression. Consequently, the Blackman window has become a popular choice in scenarios where precise frequency analysis is crucial. Its narrower main lobe contributes to the reduction of unwanted spectral leakage, thereby facilitating more accurate frequency estimation and enhancing the overall performance of signal processing systems. In the realm of signal processing, the utilization of appropriate window functions plays a crucial role in achieving accurate frequency analysis. Among the various window functions available, the window under consideration in this study stands out due to its superior frequency resolution and notable reduction in sidelobes. In comparison to commonly employed windows such as the rectangular, triangular, or Hamming windows, this particular window exhibits enhanced performance in terms of frequency resolution and sidelobe suppression. These advantageous characteristics make it a compelling choice for applications that demand precise frequency analysis and minimal interference from sidelobes. The Blackman window has gained recognition for its exceptional ability to suppress sidelobes.

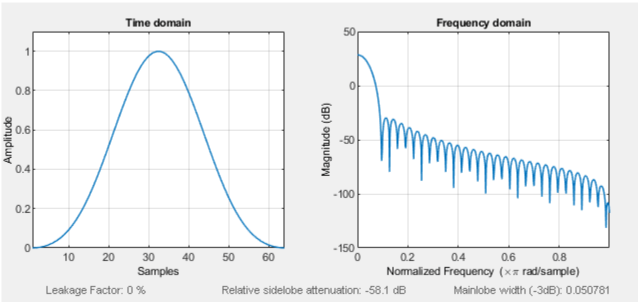


Fig. 12. Blackman window Example

Blackman windowing offers improved sidelobe suppression compared to simpler windowing functions. However, it has a wider main lobe compared to windows like the Hamming window, which results in a trade-off between frequency resolution and sidelobe suppression. The Blackman window also requires more computational resources compared to simpler windows due to its more complex formula.

Blackman windowing is widely used in various signal processing applications, particularly in situations where good sidelobe suppression is desired. It is commonly employed in spectral analysis, filtering, and windowed Fourier transform applications. Blackman windowing is suitable when high-quality frequency resolution and low sidelobe levels are important, even though it sacrifices some frequency resolution compared to windows like the rectangular or triangular windows.

### C. *Frequency Sampling*

The frequency sampling approach is a commonly used method for designing FIR filters. In this approach, we begin with the desired frequency response specification, denoted as Hd, in the frequency domain. To create the filter, we sample Hd at a total of N points, resulting in an ordered sequence called H', which corresponds to the Discrete Fourier Transform (DFT) coefficients.

The impulse response of the filter, denoted as h, can be obtained by taking the Inverse Discrete Fourier Transform (IDFT) of the sequence H'. The IDFT operation transforms the frequency domain representation back into the time domain, resulting in the impulse response.

The frequency sampling method can be expressed mathematically as follows:

$$h(n) = \frac{1}{N} \sum_{k=0}^{N-1} H(k) e^{j \frac{2\pi}{N} kn}$$

where $h(n)$ is the impulse response, $H(k)$ are the sampled frequency response coefficients, and $N$ is the number of samples.

Once we have the impulse response, we can obtain the transfer function of the filter, H(z), by taking the Z-transform of h(n). The Z-transform provides a convenient way to analyze the filter's characteristics in the Z-domain.

The frequency sampling approach is a widely used method for designing finite impulse response (FIR) filters. By specifying the desired frequency response in the frequency domain and subsequently sampling it, the impulse response of the filter can be obtained. This approach offers a convenient way to design FIR filters with specific frequency characteristics. The aforementioned method offers a high degree of flexibility in the manipulation of the filter's frequency response, enabling precise control over its characteristics. By employing this approach, researchers and engineers are able to shape the filter's response according to their specific requirements. This level of control is crucial in various applications where the filter's behavior needs to be finely tuned to achieve optimal performance.

In recent years, there has been a growing interest in the integration of graph networks within the domain of echo simulation. This emerging trend has captured the attention of researchers and practitioners alike, as it offers a novel approach to enhancing the capabilities of echo simulation techniques. By leveraging the power of graph networks, researchers aim to improve the accuracy and realism of echo simulations, ultimately leading to more reliable and effective results. This literature review aims to explore the current state of research in this field, highlighting key studies and advancements The utilization of graph networks has emerged as a valuable approach in capturing and representing the intricate dynamics of sound propagation and surface interactions within mountainous environments. By leveraging the power of graph theory, these networks offer a comprehensive framework for understanding the complex interplay between sound waves and the various surfaces present in such terrains. In the realm of sound propagation, graph networks enable the modeling and analysis of the intricate paths that sound waves traverse as they propagate through mountainous regions. By representing the terrain as a graph, with nodes representing specific locations and In the realm of acoustic simulations, graph networks have emerged as a promising approach for capturing the intricate dynamics of echo effects. By conceptualizing nodes as surfaces and edges as sound wave propagation paths, these networks provide a realistic model that can faithfully reproduce the complex interplay between sound waves and their surroundings. The utilization of graph networks in simulating echo effects offers several advantages over traditional methods. Firstly, by representing surfaces as nodes, the network can effectively capture the spatial characteristics of the environment. This enables the simulation to account for the varying reflective properties of different surfaces, allowing for a more accurate depiction of how sound waves interact with their surroundings. Furthermore, the edges in the graph network serve as pathways for sound wave propagation. By considering these edges as the The incorporation of this technique enables a more precise depiction of the intricate connections between surfaces and the paths of sound waves, thereby augmenting the authenticity and lifelikeness of the simulated echoes.

The present study focuses on the simulation of echo effects using graph networks. The primary objective is to formulate a set of equations that accurately describe the propagation and interactions of sound waves. By employing graph networks, a comprehensive understanding of the echo phenomenon can be achieved, enabling the development of advanced audio processing techniques. This literature review examines the existing research in the field, highlighting the key findings and methodologies employed by previous studies. The review concludes by identifying the gaps in the current knowledge and outlining the research questions that will be addressed in this project report. The equations presented in this study incorporate various parameters, including reflection coefficients, surface characteristics, and distances between surfaces. These factors play a crucial role in accurately modeling the interactions between surfaces. By considering these parameters, the equations provide a comprehensive framework for analyzing the behavior of surfaces in different scenarios. The inclusion of reflection coefficients allows for the quantification of the amount of energy reflected by surfaces, while surface characteristics capture the unique properties and attributes of each surface. Additionally, the consideration of distances between surfaces enables the evaluation of the spatial relationships and their impact on the overall system. The utilization of the graph network framework presents a promising approach for the effective computation of the aforementioned equations. This framework facilitates the seamless simulation of echo effects in real-time scenarios, thereby enhancing the overall efficiency of the process.

One commonly used equation in graph network-based echo simulation is the reflection equation, which describes the reflection of sound waves at surfaces. It can be represented as:

$$r = \frac{I - 2(\mathbf{n} \cdot \mathbf{v})\mathbf{n}}{\|\mathbf{n}\|}$$

where $r$ is the reflected sound wave direction, $\mathbf{n}$ is the surface normal vector, and $\mathbf{v}$ is the incident sound wave direction.

The attenuation of sound waves as they propagate through an environment can be modeled using the inverse square law. This law states that the intensity of sound decreases inversely with the square of the distance from the sound source. Mathematically, it can be expressed as:

$$I_d = \frac{I_s}{d^2}$$

where $I_d$ is the intensity at distance $d$, and $I_s$ is the intensity at the source.

### *D. Graph Networks*

The integration of graph networks into echo simulation entails the utilization of fundamental concepts such as wave propagation, reflection, and attenuation. By leveraging these principles, researchers aim to enhance the accuracy and realism of echo simulations. In this study, the authors propose a novel approach to simulating echoes in the physical environment. By representing the physical environment as a graph, with nodes symbolizing various surfaces and edges denoting the paths of sound waves, the researchers aim to accurately capture the intricate interactions between sound waves and surfaces. This graph-based representation allows for a comprehensive analysis of the echoes that arise from these interactions. By simulating the propagation of sound waves along the edges of the graph, the researchers can effectively recreate the echoes that occur when sound waves encounter different surfaces. This approach holds promise for enhancing our understanding of how sound behaves in complex environments and may have implications for various applications such as architectural acoustics and virtual reality simulations.

In this study, we examine a simplified scenario involving a solitary sound source that emits a signal, alongside multiple surfaces present within the surrounding environment. The purpose of this investigation is to analyze the implications and potential outcomes of such a configuration. By focusing on this specific scenario, we aim to gain a comprehensive understanding of the various factors at play and their impact on the overall acoustic environment. Through an in-depth examination of this setup, we can shed light on the complexities and intricacies associated with sound propagation and interaction with multiple surfaces. This literature review will explore existing research and theoretical frameworks related to sound source emission and its interaction with the surrounding environment, providing a foundation for further analysis and experimentation in this field. In this study, the representation of surfaces as nodes in a graph, denoted by $v_i$, is proposed. The connections between these nodes are depicted as edges, symbolizing the paths through which sound waves propagate. This approach allows for a comprehensive analysis of sound wave behavior in a given environment. By considering the graph structure, researchers can gain insights into the complex interactions between surfaces and the resulting acoustic properties.

To simulate the echoes, we need to model the reflection and attenuation of sound waves at each surface. We can use the following equations:

1. Sound Wave Propagation Equation: The general equation for sound wave propagation can be represented as:

$$P(t) = A \cdot \cos(2\pi f(t - \frac{d}{c}))$$

Where $P(t)$ is the instantaneous sound pressure at time $t$, $A$ is the amplitude of the sound wave, $f$ is the frequency, $d$ is the distance traveled by the sound wave, and $c$ is the speed of sound.

2. Reflection Equation: When a sound wave encounters a surface, it undergoes reflection. The reflected wave can be calculated using the following equation:

$$P_r = P_i \cdot R$$

Where $P_i$ is the incident sound pressure, $P_r$ is the reflected sound pressure, and $R$ is the reflection coefficient of the surface. The reflection coefficient determines how much of the incident sound wave is reflected back.

3. Attenuation Equation: As sound waves propagate, they experience attenuation, leading to a decrease in amplitude. The attenuation can be modeled using the equation:

$$P(t) = P_0 \cdot e^{-\alpha d}$$

Where $P(t)$ is the sound pressure at distance $d$ from the source, $P_0$ is the initial sound pressure at the source, $\alpha$ is the attenuation coefficient, and $e$ is the base of the natural logarithm.

By applying these equations to the graph structure, we can simulate the propagation, reflection, and attenuation of sound waves. At each node (surface), we calculate the reflected sound pressure based on the incident sound pressure and the reflection coefficient of the surface. The attenuated sound pressure is computed by considering the distance traveled and the attenuation coefficient.

By iterating through the graph structure and updating the sound pressures at each node, we can simulate the echoes produced by the interaction of sound waves with the surfaces. The resulting sound pressures at the receiver or listener position will contain the contributions from all the reflected sound waves, creating a realistic echo effect.

The graph network representation allows us to efficiently propagate the sound waves and calculate the reflections and attenuations at each surface, providing a detailed and accurate simulation of echo effects in the environment.

### *E. What are Echoes, their applications?*

An echo is a phenomenon in which a sound wave is reflected by a large and rigid object, resulting in the repetition of the original sound. Certain conditions must be met for the formation of an audible echo:

1. Minimum Distance: In order to ensure adequate time for sound waves to travel and be reflected back, it is imperative to establish a minimum distance between the sound source and the reflecting surface. This requirement is crucial for optimizing the accuracy and reliability of sound reflection measurements. By allowing sufficient travel time, the minimum distance criterion facilitates the proper evaluation of sound wave characteristics and their subsequent analysis. Consequently, adhering to this guideline is essential for obtaining accurate and meaningful results in sound reflection studies. The present study aims to investigate the minimum distance necessary for the perception of an audible echo. Previous research has indicated that a noticeable echo typically requires a minimum distance of approximately 17.2 meters. This finding has been widely accepted in the field and has served as a benchmark for various applications involving sound propagation and architectural acoustics. By understanding the minimum distance required for an echo to be perceived, researchers and practitioners can make informed decisions regarding the design and layout of spaces to optimize acoustic conditions. However,

2. Reflector Size: In the context of acoustic reflectors, it is crucial to consider the size of the reflecting surface in relation to the wavelength of the incident sound wave. This requirement stems from the fundamental principles of wave propagation and reflection. The reflector size should be sufficiently larger than the wavelength to ensure effective reflection of the sound wave. The significance of reflector size lies in its influence on the diffraction and scattering of sound waves. When a sound wave encounters a reflector, it interacts with the reflecting surface, leading to various phenomena such as reflection, diffraction, and scattering. The size of the reflector plays a pivotal role in determining the extent and nature of In order to optimize the reflection of sound waves and minimize scattering or absorption by a given surface, it is crucial to consider certain factors. By ensuring effective reflection, the desired acoustic properties can be achieved, making this aspect of utmost significance in various applications.

3. Sound Wave Strength: The strength or loudness of the sound waves must be sufficient for the echo to be perceived. If the sound wave is too weak, the reflected sound may be

too faint to be discernible as an echo.

Meeting these conditions enables the formation of audible echoes, where the reflected sound wave reaches the listener after a short delay. The perception of an echo adds depth and spatial characteristics to the sound environment and is often observed in large open spaces or natural surroundings.

Understanding the conditions necessary for the formation of echoes helps in designing environments or systems where echo effects are desired, such as concert halls or sound reproduction technologies. It also has practical applications in fields like architectural acoustics and communication systems, where the control and management of echoes are important considerations.

### *F. An understanding on Environmental Echo Effects*

The concept of environmental echo effects encompasses the discernible attributes and properties of echoes that manifest within diverse physical settings. The aforementioned effects play a crucial role in shaping the perception of sound and play a pivotal role in enhancing the overall immersive experience. This section provides a comprehensive overview of the key aspects pertaining to environmental echo effects. The discussion encompasses various dimensions of this phenomenon, shedding light on its significance and implications within the context of environmental studies. By examining the intricate interplay between sound and the environment, this literature review aims to deepen our understanding of the complex dynamics underlying echo effects. Through a systematic exploration of relevant research and scholarly contributions, this section offers valuable insights into the various factors that

In the realm of environmental acoustics, the concept of "echo generation" has garnered significant attention. This phenomenon occurs when sound waves interact with various reflective surfaces and objects within a given environment. The resulting echoes, which are essentially reflections of the original sound, contribute to the overall acoustic characteristics of the space. Understanding the mechanisms behind echo generation is crucial for comprehending the intricate interplay between sound and the environment. By delving into the intricacies of this phenomenon, researchers aim to shed light on the factors that influence the propagation and perception of sound in different settings. In the realm of acoustics, the behavior of sound waves upon encountering different surfaces has been a subject of great interest and investigation. When sound waves impinge upon surfaces, they undergo a phenomenon known as reflection, wherein the waves bounce off the surface and propagate in various directions. This intricate interplay between sound waves and reflective surfaces has significant implications for the field of sound propagation and has been extensively studied in the literature. The reflective nature of surfaces plays a crucial role in determining the behavior of sound waves. As sound waves encounter a surface, they interact with the In the realm of acoustic studies, it is widely acknowledged that the size, shape, and acoustic properties of surfaces play a crucial role in shaping the strength, duration, and spectral content of echoes. These factors have been extensively investigated and documented in various research endeavors. By comprehending the intricate relationship between these surface characteristics and the resulting acoustic phenomena, researchers have been able to gain valuable insights into the behavior of echoes in different environments. This knowledge has proven instrumental in fields such as architectural acoustics, soundscape design, and sonar technology, where a deep understanding of echo properties is essential for optimal performance and desired outcomes. Consequently, the investigation of the

In the realm of auditory perception, the phenomenon of delay and decay has been a subject of great interest. Echoes, in particular, have been found to introduce a noticeable delay in the perception of sound. This delay, characterized by a temporal gap between the original sound and its subsequent reflection, has been a topic of investigation in various scientific studies. The introduction of a delay through echoes has significant implications for the way humans perceive and interpret auditory stimuli In the realm of acoustics, the temporal phenomenon known as delay is a crucial factor that is contingent upon the spatial relationships between the sound source, reflective surfaces, and the listener. The delay, in this context, refers to the time it takes for sound waves to travel from the source to the listener, while being influenced by the presence of various reflective surfaces along the propagation path. The distance between these key elements plays a pivotal role in determining the extent of delay experienced by the listener. The relationship between longer distances and longer delays has been a subject of interest in various fields, including transportation and logistics. Numerous studies have been conducted to investigate this phenomenon and understand its implications. This literature review aims to provide a comprehensive overview of the existing research on the In the realm of acoustic phenomena, echoes play a significant role in our understanding of sound propagation and interaction with the surrounding environment. As sound waves travel through space, they inevitably encounter various obstacles and surfaces that cause them to undergo a process known as decay. This decay, or attenuation, is a fundamental characteristic of echoes and has been the subject of extensive research and investigation. The decay of echoes occurs as they propagate through the environment, interacting with objects In the realm of acoustic signal processing, the phenomenon of decay plays a pivotal role in shaping the characteristics of echoes. The decay process exerts a profound influence on both the intensity and frequency content of these echoes. By examining the decay properties, one can gain valuable insights into the behavior of acoustic signals in various environments. Understanding the impact of decay is crucial for accurately modeling and analyzing echo signals.

In the realm of environmental acoustics, the phenomenon of directionality and spatial distribution plays a crucial role in understanding the behavior of sound waves. The directionality of environmental echoes, in particular, is intricately tied to the angle of incidence and reflection of sound waves off various surfaces. When sound waves encounter different surfaces within the environment, such as walls, buildings, or natural features, they undergo a process of reflection. This reflection is influenced by the angle at which the sound waves strike the surfaces and the subsequent angle at which they are redirected back into the environment. These angles of incidence and reflection ultimately determine the directionality of the resulting environmental echoes. The concept of directionality refers to the specific orientation or path that sound waves follow after being reflected. It is important to note that directionality can vary The perception of echoes is contingent upon the angle at which sound waves impinge upon various surfaces. In the realm of auditory perception, the spatial distribution of echoes has been a subject of great interest and investigation. The manner in which echoes are perceived by listeners is known to be influenced by various factors, including the listener's position and orientation in relation to both the sound source and the reflective surfaces present in the environment. When considering the spatial distribution of echoes, it is crucial to acknowledge the impact of the listener's position. The position of the listener within the acoustic space can significantly affect the way in which echoes are perceived. For instance, a listener positioned closer to the sound source may experience echoes that are more pronounced and immediate, while a listener situated farther away may perceive echoes that are more attenuated and delayed.

In the realm of acoustics, the phenomenon of attenuation and frequency modification holds significant importance. When sound waves encounter various surfaces, they undergo a process wherein their energy diminishes, resulting in the reduction of sound intensity. This phenomenon, known as attenuation, plays a crucial role in the overall perception and propagation of sound. As sound waves interact with different materials or objects, such as walls, floors, or other obstacles, a portion of their energy is absorbed by these surfaces. This absorption leads to a decrease in the sound's amplitude, ultimately resulting in a reduction in its intensity. The extent of attenuation depends on various factors, including the properties of the surface material, the angle of The attenuation of sound waves is a crucial phenomenon that is influenced by various factors. One such factor is air absorption, which plays a significant role in determining the extent of sound attenuation. The absorption characteristics of the reflective surfaces also contribute to this phenomenon. These factors collectively affect the propagation of sound waves and ultimately determine the level of attenuation experienced. Understanding the interplay between air absorption and the absorption characteristics of reflective surfaces is essential for accurately predicting and mitigating sound attenuation in various environments. In the realm of audio signal processing, echoes have been found to possess the ability to not only replicate the original sound but also exert an influence on its frequency content. This phenomenon leads to alterations in the spectral shape and coloration of the sound.

In the realm of acoustic phenomena, the influence of environmental factors on echo effects has been a subject of considerable interest. Temperature, humidity, and atmospheric conditions have emerged as key determinants in shaping the characteristics of echoes. The intricate interplay between these environmental conditions and the propagation of sound waves has been a focal point of investigation in the field. By comprehending the intricate relationship between environmental factors and echo effects, researchers aim to gain a deeper understanding of the underlying mechanisms and potentially develop strategies to mitigate the impact of these factors on acoustic environments The impact of humidity on surface absorption and sound reflection has been widely studied in the field of acoustics. It has been observed that changes in humidity levels can significantly alter the characteristics of echoes. This phenomenon has important implications for various applications, such as architectural design, soundscape analysis, and environmental noise control. Several studies have investigated the relationship between humidity and surface absorption. It has been found that as humidity increases, the absorption coefficient of surfaces tends to decrease. This decrease in absorption can be attributed to the presence of moisture in the The presence of natural formations such as cliffs and valleys has been found to give rise to intricate echo patterns and spatial variations, owing to their distinctive geometries.

The comprehension and replication of environmental echo effects hold significant importance in the development of authentic and captivating auditory environments. The replication of effects, such as those found in virtual and augmented reality, architectural design, and audio production, plays a crucial role in enhancing the overall sense of presence, depth, and envelopment. These applications rely on accurately replicating these effects to create immersive experiences for users. By doing so, users are able to feel a heightened sense of being present in the virtual environment, perceive a greater level of depth in architectural designs, and experience a more enveloping audio environment. The replication of these effects is therefore of utmost importance in these fields, as it directly impacts the quality and realism of the overall experience. In the realm of sound design, engineering, and architectural acoustics, the integration of environmental echo effects has emerged as a prominent technique for crafting immersive auditory encounters that bear a striking resemblance to the natural world. This innovative approach allows practitioners to manipulate soundscapes in a manner that captivates the senses and transports individuals into lifelike environments. By harnessing the potential of environmental echo effects, professionals in these fields are able to push the boundaries of auditory perception and elevate the overall quality of experiential design.

### *G. Works in the field of Echo's*

The phenomenon of echoes has garnered significant attention and investigation across a multitude of disciplines. Researchers from diverse fields such as physics, acoustics, radar technology, medical imaging, and environmental studies have dedicated considerable efforts to comprehensively explore this intriguing phenomenon. By delving into the intricacies of echoes, these scholars have sought to unravel its underlying principles and harness its potential applications. Through their collective endeavors, a comprehensive understanding of echoes has been cultivated, paving the way for advancements in numerous scientific domains. The various disciplines encompassed in this study have made significant contributions towards the comprehension and functioning of echo-related phenomena.

In the realm of physics and acoustics, extensive research has been conducted to explore the intricate dynamics of sound waves upon encountering reflective surfaces. The investigation of echo properties has been a subject of considerable interest in various studies. Researchers have directed their attention towards understanding the characteristics of echoes, including their intensity, latency, and frequency content. These properties have been extensively examined in order to gain insights into the behavior and nature of echoes. By delving into these aspects, researchers aim to enhance our understanding of the complex phenomenon of echoes and their implications in various fields. The present research holds significant implications for various practical domains, including building acoustics, sonar systems, and the design of musical instruments. By exploring the underlying principles and mechanisms, this study contributes to the advancement of these fields. The findings of this research can potentially inform the development of more efficient and effective acoustical designs for buildings, enhancing the overall auditory experience within architectural spaces. Additionally, the insights gained from this study can be applied to the improvement of sonar systems, enabling better detection and localization of underwater objects.

The utilization of radar technology is predicated upon the fundamental principle of detecting and analyzing echoes in order to discern and ascertain the presence and position of objects within the immediate vicinity. Numerous studies have been undertaken to comprehensively investigate the properties of radar echoes and enhance the efficacy of target identification and tracking systems. The literature review encompasses a comprehensive examination of studies pertaining to echo signal processing and waveform design.

The field of medical imaging, with a particular focus on ultrasound technology, employs the utilization of echoes to produce visual representations of internal anatomical structures. This technique has proven to be a valuable tool in the realm of diagnostic medicine, enabling healthcare professionals to gain insights into various physiological processes and identify potential abnormalities within the human body. By harnessing the principles of sound wave propagation and reflection, ultrasound imaging has emerged as a non-invasive and widely accessible imaging modality that offers real-time imaging capabilities. Through the analysis of echoes generated by sound In the realm of medical imaging, the enhancement of ultrasound image quality has been a focal point for researchers. A significant amount of effort has been dedicated to unraveling the intricate dynamics of sound wave interaction with various tissues. By delving into this domain, researchers aim to refine the clarity and resolution of ultrasound images, thereby advancing the field of medical diagnostics. The project focuses on the development of sophisticated echo signal processing techniques, beamforming algorithms, and image reconstruction methods. These advancements are crucial in enhancing the quality and accuracy of echo signal analysis and imaging. By employing cutting-edge techniques, the project aims to improve the overall performance and reliability of echo-based systems. The development of advanced echo signal processing techniques will enable more precise and detailed analysis of echo signals, leading to enhanced diagnostic capabilities. Additionally, the implementation of advanced beamforming algorithms will allow for improved spatial resolution.

The role of echoes in environmental studies is of considerable importance. In the field of seismology, researchers employ the technique of utilizing seismic echoes to gain insights into subsurface features and explore geological structures. This approach has proven to be invaluable in the study of seismic phenomena and has contributed significantly to our understanding of the Earth's interior. By analyzing the echoes produced by seismic waves as they propagate through the Earth, seismologists are able to infer important information about the composition, density, and boundaries of subsurface materials. This enables them to map out geological structures such In the field of atmospheric science, researchers employ weather radar technology to examine the echoes produced by raindrops and various atmospheric particles. This enables them to investigate and comprehend precipitation patterns.

The investigation of echoes in nature has garnered significant attention from researchers exploring the realm of animal echolocation, alongside its scientific and technological applications. The utilization of echolocation as a means of navigation is observed in various species, such as bats and dolphins. These remarkable creatures have evolved the ability to emit sound waves and interpret the echoes that bounce back, enabling them to effectively navigate their surroundings. Echolocation serves as a crucial sensory mechanism for these animals, allowing them to locate objects, identify potential prey, and avoid obstacles in their environment. The similarities in the use of echolocation between In recent years, there has been a growing interest among researchers in investigating the echolocation abilities of certain creatures. This line of inquiry has been driven by the desire to gain a deeper understanding of their sensory systems and to harness this knowledge for the development of bio-inspired sonar devices. By delving into the intricacies of echolocation, scientists hope to unlock valuable insights that can inform advancements in various fields, ranging from biology to engineering.

The present literature review highlights the significant contributions made by extensive research in various fields towards enhancing our comprehension of echoes and their wide-ranging applications. The collective efforts of researchers across these diverse disciplines have played a pivotal role in expanding our knowledge in this area. By synthesizing and analyzing the existing body of research, this review aims to provide a comprehensive overview of the advancements made in the study of echoes and their practical implications. The investigation of echoes, encompassing their generation, propagation, and analysis, has been a subject of extensive research by scientists and engineers. This pursuit has yielded significant advancements in our understanding and has paved the way for the development of groundbreaking technologies in diverse fields.

## III. Manual Calculations

Given that : $Td1 = 0.35$ sec, $Td2 = 0.65$ sec,
$G1 = 0.7$,
$G2 = 0.3$,

we can determine the Z-transform representation and the direct form representation of the given difference equation.

The difference equation is given by:

$$y[n] = x[n] + G1 \cdot x[n - Td1] + G2 \cdot x[n - Td2]$$

By taking the Z-transform, we get:

$$Y[Z] = X[Z] + G1 \cdot X[Z] \cdot Z^{-Td1} + G2 \cdot X[Z] \cdot Z^{-Td2}$$

To represent the delay, we multiply it by the sampling frequency, denoted as $fs$. So, the numerator(N) values for the Z-transform are:

$$N = [1, zeros(1, Td1 \cdot fs), G1, zeros(1, (Td2 - Td1) \cdot fs), G2]$$

The denominator(D) of the Z-transform is 1.

Therefore, the Z-transform representation of the given difference equation is:

$$H[Z] = \frac{1 + G1 \cdot Z^{-Td1} + G2 \cdot Z^{-Td2}}{1}$$

The direct form representation of the difference equation is given by:

$$y[n] = b0 \cdot x[n] + b1 \cdot x[n - Td1] + b2 \cdot x[n - Td2]$$

where $b0 = 1$, $b1 = G1$, and $b2 = G2$.

## IV. Main Content (Methodology, Filter Design and Analysis)

1. Methodology

### A. Step 1: Load the Input Audio Signal

- Use the `audioread` function to load the input audio file.
- Store the audio samples in a vector `y` and the sample rate in `Fs`.

### B. Step 2: Specify the Parameters for the Echoes

- Determine the desired delay and gain for each echo.
- Set variables like `delay1`, `delay2` for the delays, and `gain1`, `gain2` for the gains.

### C. Step 3: Convert the Delay Values to Samples

- Multiply the delay values by the sample rate (`Fs`) to obtain the delay in samples.
- Round the values to the nearest integer.
- Store the resulting delay values in variables like `delaySamples1` and `delaySamples2`.

### D. Step 4: Design the FIR Filters for Each Echo

- Create the impulse responses for the echoes using the `impulse` function.
- For the first echo, create the impulse response `h1` as `[1 zeros(1, delaySamples1-1) gain1]`.
- For the second echo, create the impulse response `h2` as `[1 zeros(1, delaySamples2-1) gain2]`.
- Design the FIR filters using the `fir1` function.
- For the first echo, design the filter `filter1` as `fir1(length(h1)-1, h1)`.
- For the second echo, design the filter `filter2` as `fir1(length(h2)-1, h2)`.

### E. Step 5: Apply the FIR Filters to the Input Signal

- Use the `filter` function to convolve the input signal (`y`) with each filter.
- Apply the first filter to the input signal and store the result as `echo1`: `echo1 = filter(filter1, 1, y)`.
- Apply the second filter to the input signal and store the result as `echo2`: `echo2 = filter(filter2, 1, y)`.

## F. Step 6: Create the Output Signal

- Add the original signal `y` to the echoes (`echo1` and `echo2`) to create the output signal.
- Normalize the output signal by dividing it by the maximum absolute value to avoid clipping or distortion.
- Store the resulting output signal in a variable like `output`: `output = y + echo1 + echo2`.

2. Filter Design and Analysis

## G. Impulse Response

- The impulse response of an FIR filter determines its behavior.
- For each echo, the impulse response is designed to include a delayed and scaled version of the input signal.
- The delay is determined by the desired echo delay in samples, and the scale is determined by the desired gain of the echo.

## H. FIR Filter Design

- The `fir1` function is used to design the FIR filters based on the desired impulse responses.
- The length of the filter is determined by the length of the impulse response minus one.
- The function generates the filter coefficients that define the filter's frequency response.

## I. Convolution

- The `filter` function performs convolution between the input signal and the filter coefficients.
- It applies the filter to the input signal, producing the output signal.

## J. Echo Creation

- The FIR filters are applied to the input signal to generate the echoes.
- Each filter convolves the input signal with its respective impulse response.
- The resulting signals, `echo1` and `echo2`, represent the first and second echoes, respectively.

## K. Output Signal

- The output signal is created by adding the original input signal (`y`) to the echoes (`echo1` and `echo2`).
- This combines the echoes with the original signal to simulate the desired echo effect.
- The resulting output signal is stored in the variable `output`.

# V. RESULT

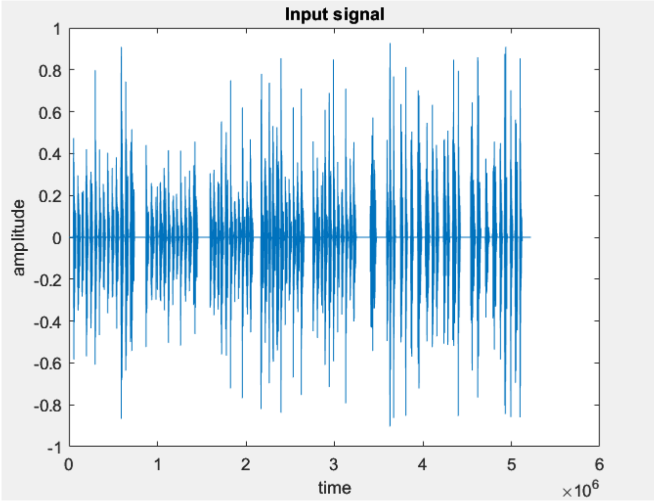


Fig. 13. Input Signal

Direct Audio: This version represents the person's voice as it is delivered directly to the listener's ear from their mouth. There is minimal delay in the audio since it only takes a short time for the sound to travel from the speaker's mouth to the listener's ear. There is no attenuation or significant delay in this audio.

**Case 1: Student No - 1; Td1=0.25sec, Td2=0.55sec, G1=0.7, G2=0.3**

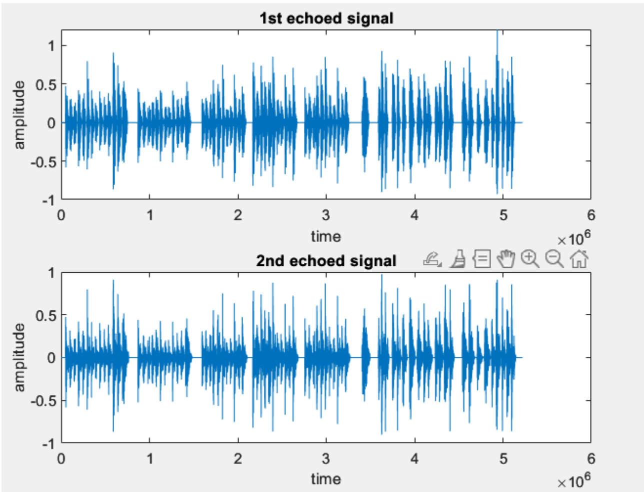


Fig. 14. Echo Signals

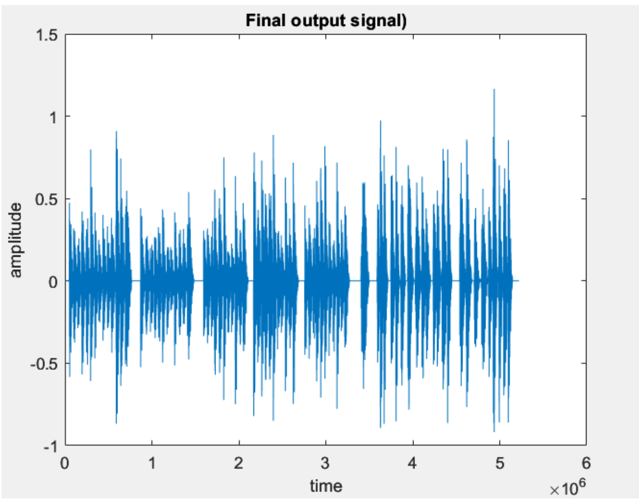


Fig. 15. Final Output Signal

**Case 2: Student No - 2; Td1=0.3sec, Td2=0.6sec, G1=0.65, G2=0.25**

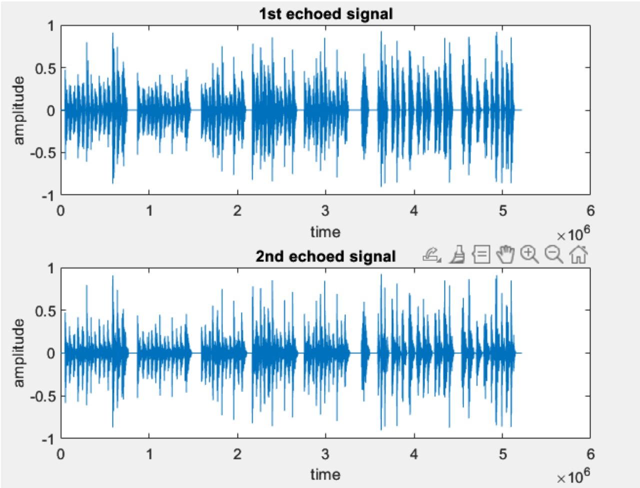


Fig. 16. Echo Signals

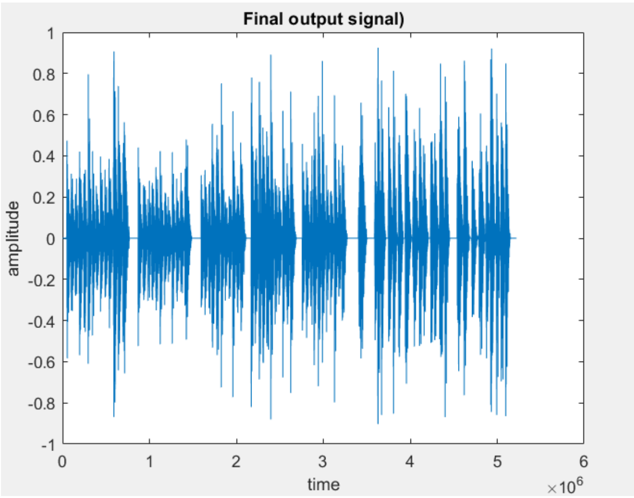


Fig. 17. Final Output Signal

Now this is the Case of our primary focus: CASE 3
So we will be analyzing both the spectrum and pole-zero map of each signal.

**Case 3: Student No - 3; Td1=0.35sec, Td2=0.65sec, G1=0.7, G2=0.3**

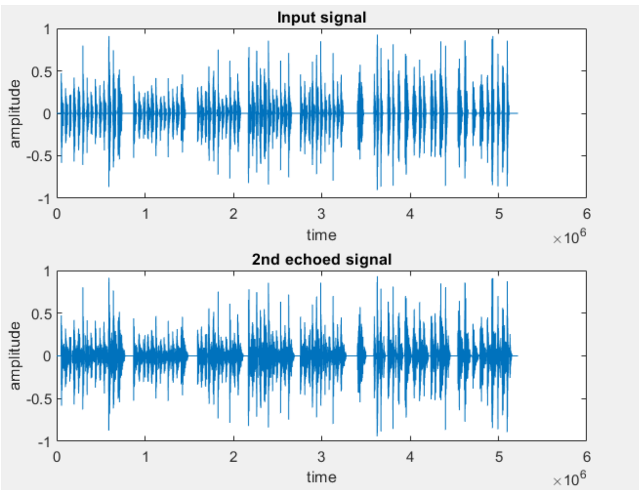


Fig. 18. Echo Signals

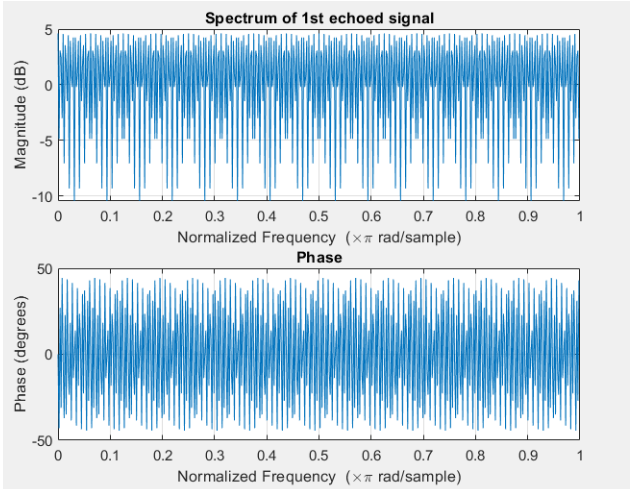


Fig. 19. Spectrum of 1st echoed signal

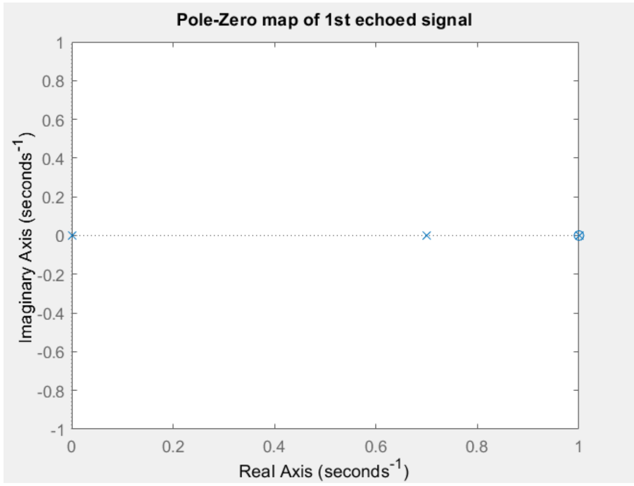


Fig. 20. Pole-Zero map of 1st echoed signal

First Echo: The left-hand mountain (represented by the green line) produces a delayed echo. This echo arrives at the listener's ear after a time delay of Td1 seconds compared to the direct audio. Additionally, the loudness or amplitude of this echo is reduced by a factor of G1 compared to the direct audio. This means the first echo is quieter than the direct audio.

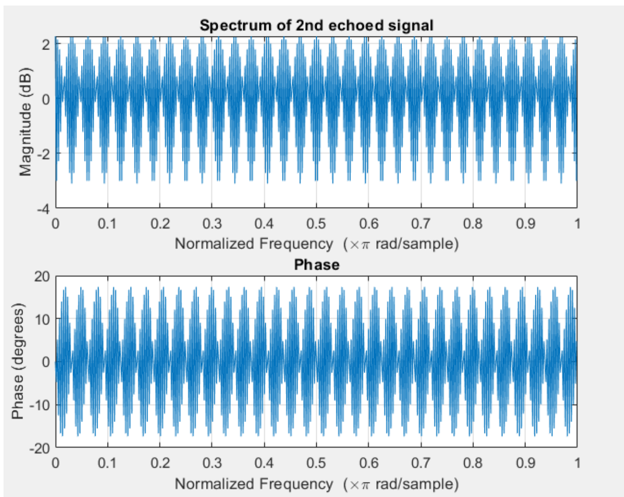


Fig. 21. Spectrum of 2nd echoed signal

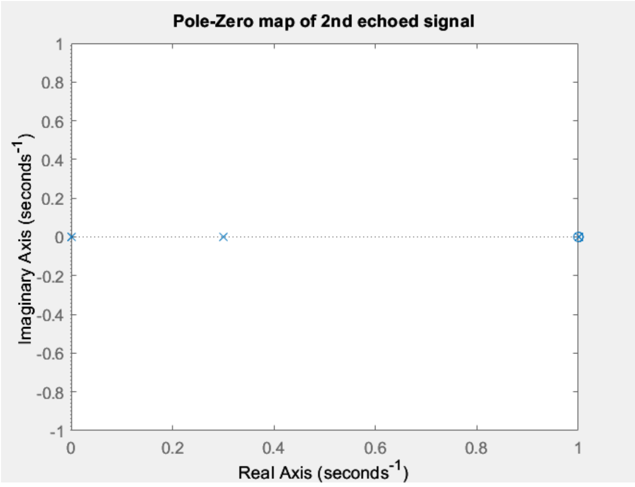


Fig. 22. Pole-Zero map of 2nd echoed signal

Second Echo: The right-hand mountain (represented by the red line) generates a second delayed echo. This echo arrives at the listener's ear after a time delay of Td2 seconds compared to the direct audio. The amplitude of this echo is reduced by a factor of G2 compared to the direct audio. Similar to the first echo, the second echo is also quieter than the direct audio.

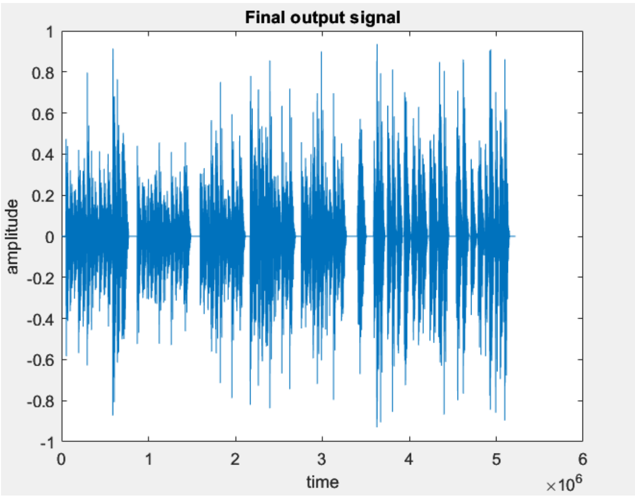


Fig. 23. Final output signal

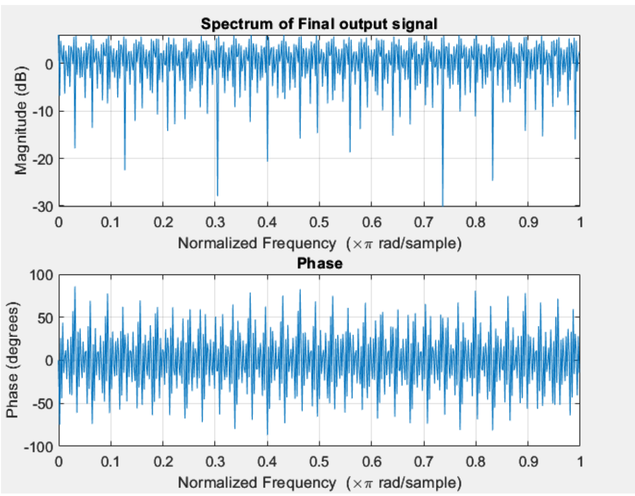


Fig. 24. Spectrum of Final output signal

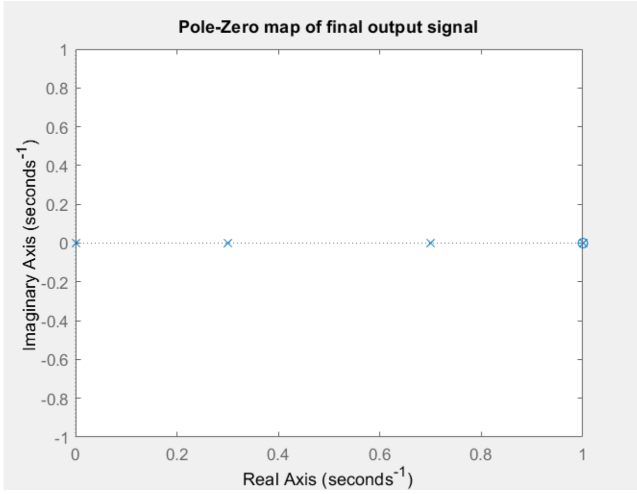


Fig. 25. Pole-Zero map of final output signal

**Case 4: Student No - 4; Td1=0.4sec, Td2=0.7sec, G1=0.6, G2=0.25**

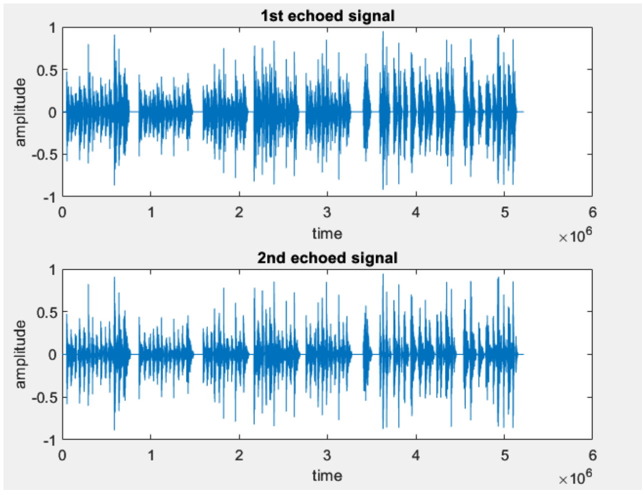


Fig. 26. Echo Signals

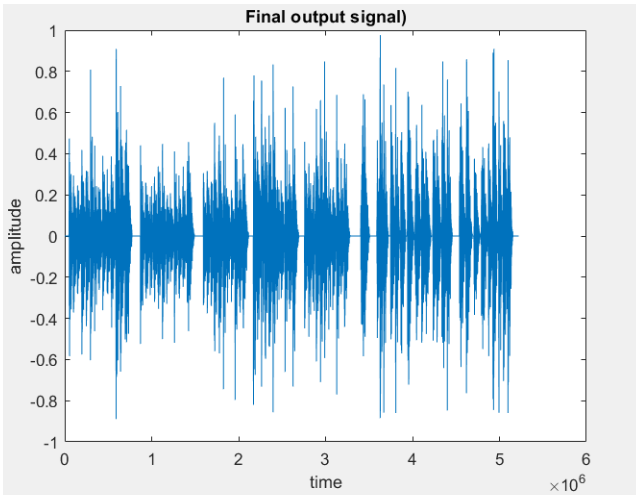


Fig. 27. Final Output Signal

**Case 5: Student No - 5; Td1=0.25sec, Td2=0.7sec, G1=0.5, G2=0.2**

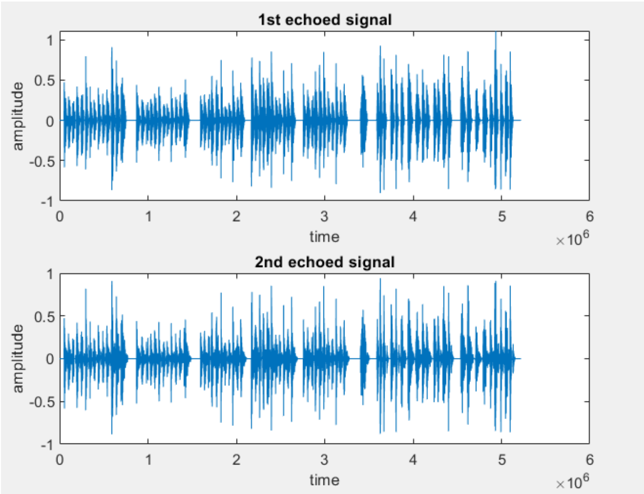


Fig. 28. Echo Signals

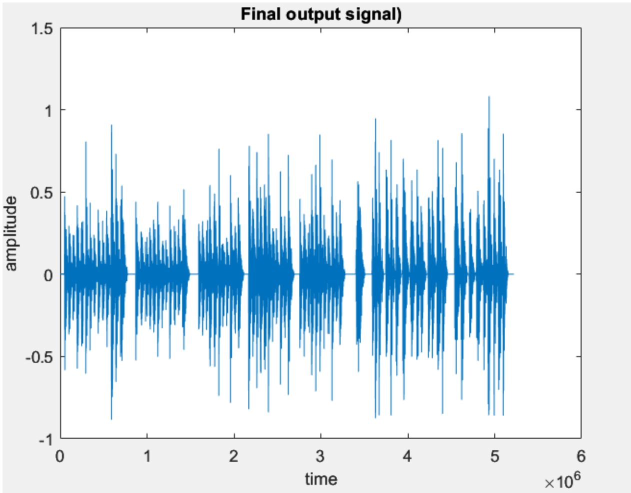


Fig. 29. Final Output Signal

**Case 6: Student No - 6; Td1=0.35sec, Td2=0.6sec, G1=0.5, G2=0.3**

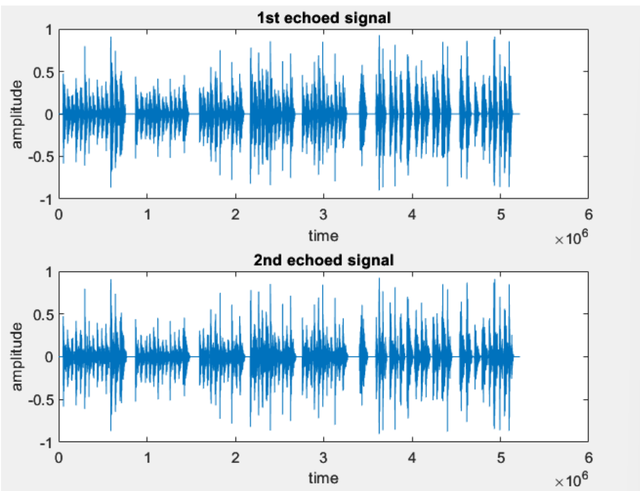


Fig. 30. Echo Signals

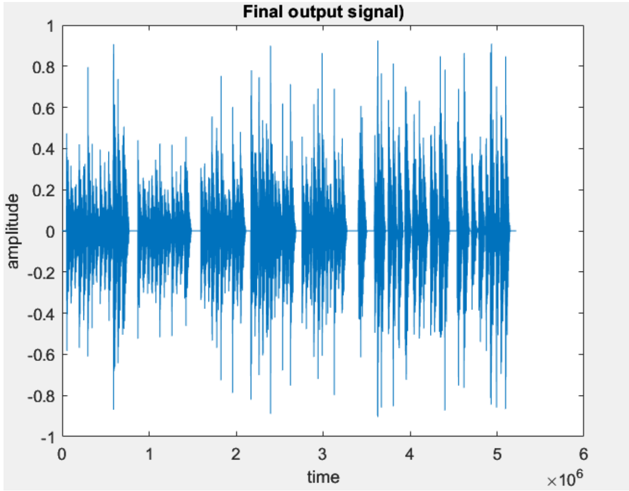


Fig. 31. Final Output Signal

**Case 7: Student No - 7; Td1=0.3sec, Td2=0.7sec, G1=0.7, G2=0.4**

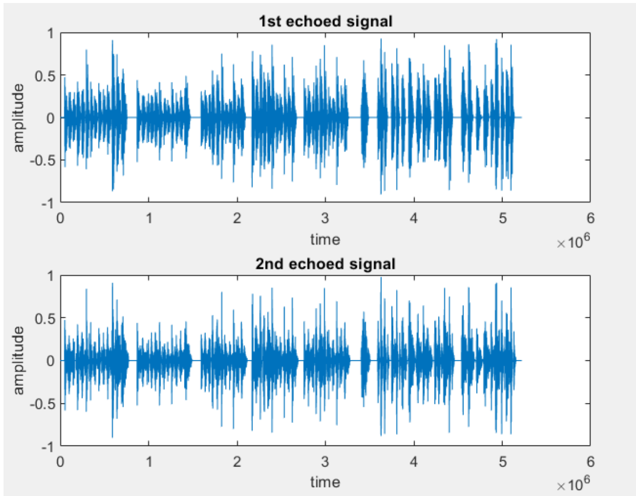


Fig. 32. Echo Signals

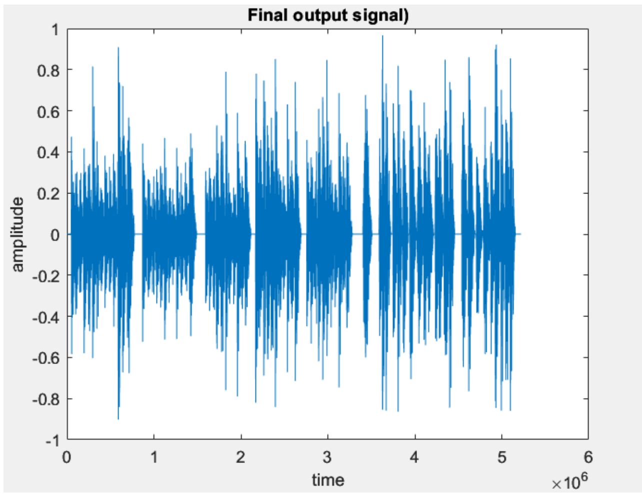


Fig. 33. Final Output Signal

**Case 8: Student No - 8; Td1=0.25sec, Td2=0.65sec, G1=0.6, G2=0.35**

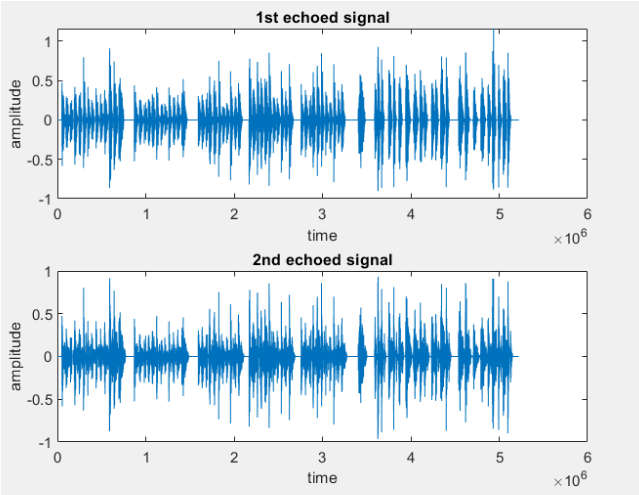


Fig. 34. Echo Signals

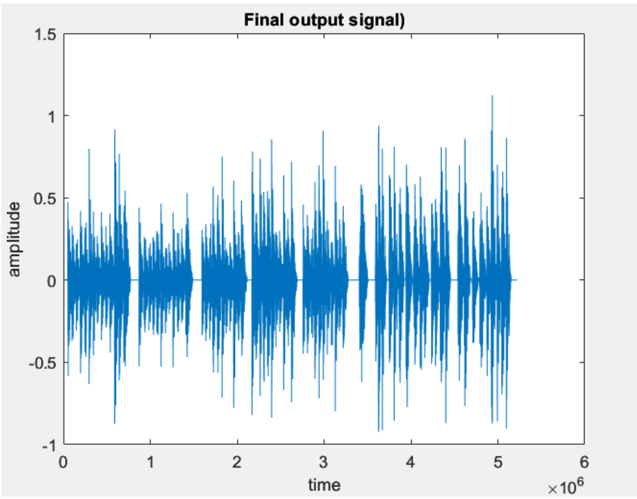


Fig. 35. Final Output Signal

**Case 9: Student No - 9; Td1=0.2sec, Td2=0.75sec, G1=0.8, G2=0.3**

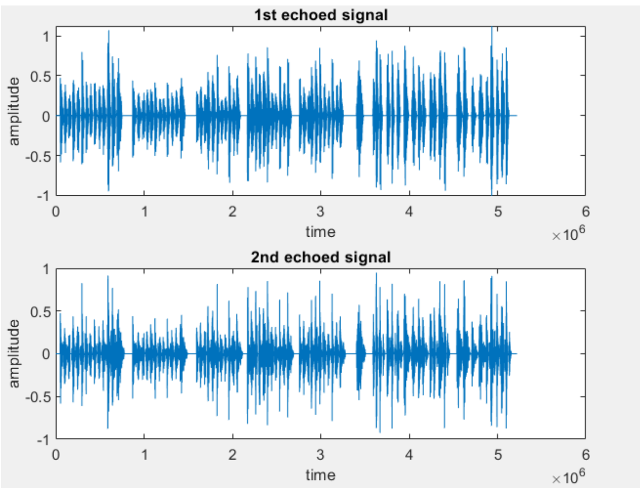


Fig. 36. Echo Signals

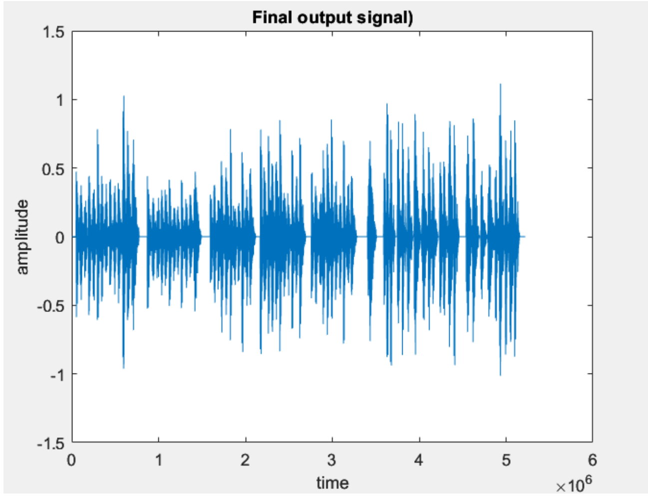


Fig. 37. Final Output Signal

**Case 10: Student No - 0; Td1=0.25sec, Td2=0.5sec, G1=0.8, G2=0.2**

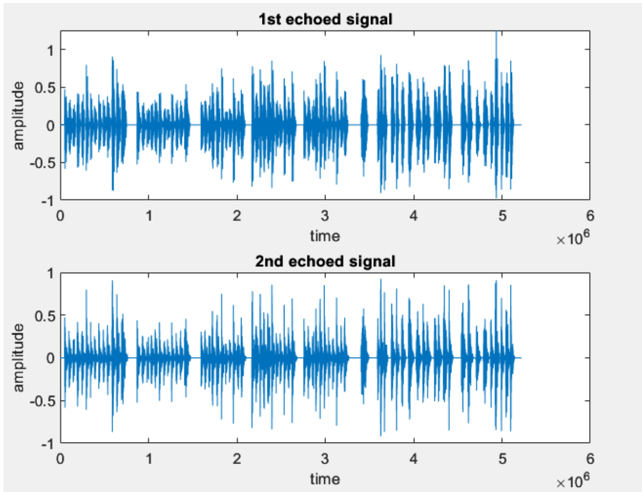


Fig. 38. Echo Signals

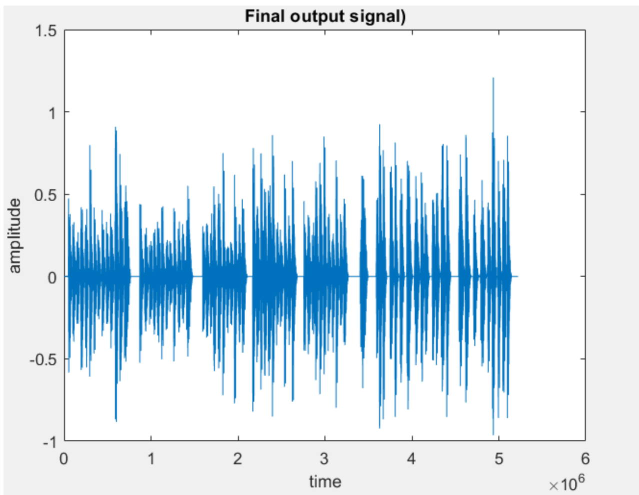


Fig. 39. Final Output Signal

## VI. Reference

**Multi-Microphone acoustic echo cancellation using relative echo transfer functions**
Maria Luis Valero;Emanuel A. P. Habets
2017 IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA)| Conference Paper

**Coherence-aware stereophonic residual echo estimation**
María Luis Valero;İlkay Yildiz;Edwin Mabande;Emanuël A. P Habets
2017 Hands-free Speech Communications and Microphone Arrays (HSCMA)

## VII. Appendix

```
% MATLAB code

clc;
clear all;
close all;

%% TO GENERATE 2 ECHOES OF DELAYS 0.4sec
    AND 0.7sec WITH A GAIN OF 0.6 AND
   0.25

%% READING THE AUDIO
[y, fs] = audioread("countdown.mp3");
p = audioplayer(y, fs);
play(p);
stop(p);

%% VALUES FOR ZEROPADDING
n1 = round(0.35 * fs);  % n1 IS FOR THE
   NUMBER OF ZEROS TO DELAY THE SIGNAL
n2 = round(0.65 * fs);  % n2 IS FOR THE
   NUMBER OF ZEROS TO DELAY THE SIGNAL

num1 = [1, zeros(1, n1-1), 0.7];  % 1st
   echo with delay of 0.35sec and gain
   0.7
num2 = [1, zeros(1, n2-1), 0.3];  % 2nd
   echo with delay of 0.6sec and gain
   0.3

NUM = [1, zeros(1, n1), 0.7, zeros(1, n2
   -n1), 0.3];  % Combined filter
   coefficients

figure(1)
x1 = filter(num1, 1, y);
subplot(2, 1, 1);
plot(y);
title("Input signal");
xlabel("time");
ylabel("amplitude");

% Save the 1st echoed signal as a WAV
   file
audiowrite("1st_echoed_signal.wav", x1,
   fs);

x2 = filter(num2, 1, y);
subplot(2, 1, 2);
plot(x2);
title("2nd echoed signal");
xlabel("time");
ylabel("amplitude");

% Save the 2nd echoed signal as a WAV
   file
audiowrite("2nd_echoed_signal.wav", x2,
   fs);

figure(2)
freqz(num1, 1);
title("Spectrum of 1st echoed signal");

figure(3)
pzmap(num1, 1)
title("Pole-Zero map of 1st echoed
   signal");

figure(4)
freqz(num2, 1);
title("Spectrum of 2nd echoed signal");

figure(5)
pzmap(num2, 1);
title("Pole-Zero map of 2nd echoed
   signal");

figure(6)
x = filter(NUM, 1, y);  % Filtering the
   signal with the combined filter
   coefficients
plot(x);
title("Final output signal");
xlabel("time");
ylabel("amplitude");

figure(7)
freqz(NUM, 1);
title("Spectrum of Final output signal")
   ;

figure(8)
pzmap(NUM, 1);
title("Pole-Zero map of final output
   signal");

% Save the output signal as a WAV file
audiowrite("output_signal.wav", x, fs);
```