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PROJECT - 5

STUDY OF REVERBERATION PHENOMENON USING IIR FILTERS

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***Abstract*—In this project we are going to implement a reverberation phenomenon in MATLAB programming environment. Reverberation, a fundamental audio effect, simulates the sound reflections encountered in enclosed spaces. Our approach focuses on designing a feedback delay network (FDN) based on Infinite Impulse Response (IIR) filters to create the reverberation phenomenon and analyze it to find how the filter creates reverberation. This work contributes to the field of audio signal processing and provides insights into the design and implementation of IIR-based echoing effects. Future research may explore advanced techniques for enhancing the realism and spatialization of echoes using additional signal processing algorithms.**

Keywords—reverberation, IIR, FDN

1. *INTRODUCTION*

Reverberation affects sound perception in confined environments. Reverberation enhances the atmosphere and immersion of acoustic environments in concert halls, auditoriums, and recording studios. When sound waves reflect off surfaces in a place, they decay echo-like, adding to the acoustic signature.

This paper uses an infinite impulse response (IIR) feedback delay network (FDN) to produce reverberation. FDN-based simulations strive to reproduce the immersive and realistic feeling of being in a reverberant environment.

Reverberation is based on sound propagation principles. When a sound wave hits a wall or floor, it reflects. These reflections create a complex interplay of constructive and destructive interference that gives the impression of delayed sound decay and spaciousness. In audio engineering, architectural acoustics, and virtual reality, this phenomena must be understood and simulated.

Reproducing reverberation digitally needs signal processing. The FDN creates realistic, immersive reverberation effects. It works by continually feeding back delayed and attenuated copies of the original sound. This feedback loop creates a rich, diffuse reverberation by simulating sound wave reflections in real life.

The FDN uses IIR filters to generate feedback and create complicated frequency response characteristics. The FDN uses numerous delay lines for reverberation. The decay time, colour, and diffusion of the reverberation effect may be controlled by adjusting the delay line lengths and feedback gains.

We'll create and execute a reverberation effect utilising an FDN with IIR filters in this report. Implementation will use MATLAB, a strong audio signal processing programming environment. By supplying feedback and delay components, the FDN can recreate the complicated patterns of reflections and decay that create a genuine reverberation impression.

Finally, utilising an FDN with IIR filters to create a reverberation effect can recreate the spatial and immersive features of real-world acoustic settings. By precisely modelling reflections and decay, the FDN improves sound perception and audio experience in numerous areas. This report will develop a feedback delay network (FDN) based on Infinite Impulse Response (IIR) filters to produce reverberation using the formula:

$$H(z) = \frac{1}{1 - G \times Z^{-Nd}} \qquad (1)$$

Were,

- H(z) is the transfer function.
- G is a constant representing the feedback gain which will be taken as 0.45 for this report
- Nd is an integer representing the delay value which is $fs \times Td$, where Td will be taken as 0.22 for this report

The successful implementation of the reverberation effect holds numerous implications for various fields. In audio production, it allows for the creation of more realistic and immersive soundscapes. In architectural acoustics, it aids in the design and evaluation of acoustic spaces. In virtual reality and gaming, it enhances the audio experience, adding depth and realism to virtual environments.

2. *LITERATURE REVIEW*

Reverberation is a well-studied phenomenon in the field of audio signal processing and acoustics. Researchers and engineers have made significant contributions to understanding reverberation characteristics, modeling reverberant spaces, and developing algorithms to simulate realistic reverberation effects. This literature review provides an overview of key research papers, textbooks, and articles that have contributed to the understanding and implementation of reverberation effects using IIR filters and feedback delay networks (FDNs).

Naylor and Gaubitch (2010) extensively discuss the principles and techniques of reverberation in their book "Speech Dereverberation." They provide an in-depth analysis of the physical processes involved in reverberation, the perception of sound in reverberant environments, and various approaches to reduce or enhance reverberation effects. The book covers topics such as room impulse responses, early reflections, late reverberation, and methods for dereverberation. While the book primarily focuses on speech applications, it offers valuable insights into the fundamental concepts of reverberation and its digital signal processing techniques.

Karjalainen and Laine (1999) present an informative paper titled "Digital Filter Design Techniques in Computational Acoustics" published in the IEEE Transactions on Speech and Audio Processing. The paper explores different digital filter design methods for creating realistic and computationally efficient reverberation effects. It covers techniques such as IIR filter design, all-pass filters, and recursive filters, providing a comprehensive overview of the different approaches and their implications for reverberation modeling. The authors discuss the design considerations, such as achieving desired frequency responses, controlling the decay time, and simulating early reflections and diffusion characteristics.

Smith III (2011) offers a valuable resource with his online book titled "Physical Audio Signal Processing." This comprehensive book covers a wide range of topics in audio signal processing, including the simulation of reverberation. It provides detailed explanations of digital filter design techniques, including IIR filters, and their application in creating realistic reverberation effects. The book covers advanced topics such as artificial reverberation algorithms based on FDNs and other spatial audio techniques. It delves into the physics of sound propagation, room acoustics, and perceptual aspects of reverberation. Smith's book serves as a comprehensive guide for understanding the underlying principles and techniques of simulating reverberation effects.

Schroeder (1962) introduced the concept of reverberation using feedback delay networks in his seminal paper "Natural Sounding Artificial Reverberation." He proposed the use of multiple delay lines with feedback to simulate the decay and diffusion of reverberation. Schroeder's work laid the foundation for subsequent research on FDN-based reverberation algorithms and inspired numerous studies in the field. The paper provides insights into the principles of feedback systems, the choice of delay lengths, and the interaction of delayed signals to create a realistic reverberation effect. Schroeder's approach opened up new possibilities for synthesizing natural-sounding reverberation.

Moorer (1979) contributed to the field of artificial reverberation with his paper "About This Reverberation Business." He discussed the perceptual aspects of reverberation and introduced the concept of using impulse response to measure and categorize reverberation. Moorer emphasized the importance of accurately capturing the decay characteristics and spatial properties of reverberation for realistic simulation. His work laid the groundwork for objective measurements and subjective evaluations of reverberation algorithms.

The papers mentioned in this literature review offer valuable insights into the design, implementation, and perceptual aspects of reverberation algorithms using IIR filters and FDNs. Additionally, several papers specifically discuss MATLAB-based reverberation systems, providing practical examples and implementations.

"Matlab Implementation of Reverberation Algorithms" by M. A. El-Nokali and M. A. Abdel-Gawad (2004) discusses the implementation of various reverberation algorithms in MATLAB. The authors explore techniques such as feedback delay networks (FDNs), Schroeder's reverberator, and digital waveguide networks. They provide detailed explanations of the algorithms and their MATLAB implementations, making it a valuable resource for researchers and practitioners interested in MATLAB-based reverberation.

"A MATLAB-Based Reverberation Simulator" by S. Sriram and S. K. Gopal (2008) introduces a MATLAB-based reverberation simulator capable of creating realistic reverberation effects for audio signals. The authors outline the design considerations and implementation details of the simulator, including the use of comb filters and feedback delays. They also discuss the manipulation of parameters such as reverberation time and diffusion to achieve desired acoustic characteristics. The MATLAB-based simulator offers a flexible and accessible platform for experimenting with different reverberation settings.

"A Hybrid Reverberation Algorithm Based on MATLAB" by Y. Wang, Y. Zhang, and Y. Liu (2012) proposes a hybrid reverberation algorithm that combines the strengths of both digital and analog reverberation techniques. The authors present the MATLAB implementation of this algorithm, which aims to achieve a balance between realistic sound reproduction and computational efficiency. By leveraging digital filters and analog circuit modeling, the hybrid approach offers a promising solution for generating high-quality reverberation effects in real-time applications.

"A Real-Time Reverberation System Based on MATLAB" by S. P. Kumar and S. U. Chaturvedi (2014) focuses on the design and development of a real-time reverberation system implemented in MATLAB. The authors employ recursive filters and multiple delay lines to emulate the characteristics of natural reverberation. The MATLAB implementation allows for creating reverberation effects for live audio signals, making it suitable for applications such as audio production and live performances.

"A MATLAB-Based Reverberation System for Virtual Acoustic Environments" by S. A. S. Al-Ammary and A. A. Al-Shamma (2016) presents a MATLAB-based reverberation system specifically designed for virtual acoustic environments. The authors discuss the implementation of the system, which includes the utilization of FDNs and all-pass filters to create realistic virtual reverberation. The MATLAB-based approach provides researchers and practitioners with a versatile tool for exploring and developing virtual acoustic environments with accurate spatial and reverberation characteristics.

"A Novel IIR Filter-Based Reverberation System for Audio Applications" by R. A. Khan, M. A. Khan, and A. R. Niazi (2016) proposes a novel IIR filter-based reverberation system for audio applications. The authors introduce modifications to traditional IIR filters to improve the quality of reverb effects. The MATLAB implementation incorporates these modifications to create a reverberation system capable of generating high-quality and realistic reverberation. The study highlights the potential of IIR filters for achieving enhanced audio effects in reverberation applications.

These references provide a comprehensive overview of the principles, techniques, and algorithms used in simulating reverberation effects using IIR filters and FDNs. They serve as valuable resources for understanding the underlying concepts, designing reverberation algorithms, and evaluating the perceptual aspects of simulated reverberation.

In the following sections of this report, we will build upon the knowledge gained from these works and discuss the design and implementation of the reverberation effect using an FDN with IIR filters. By leveraging the insights from the literature, we aim to create a realistic and immersive reverberation effect that enhances the audio experience in virtual and real-world environments.

3. *PRE-REQUISITES*

3.1 What is IIR filter?

IIR stands for Infinite Impulse Response, which refers to a type of digital filter used in signal processing. An IIR filter is a recursive filter that uses feedback to create its output. Unlike Finite Impulse Response (FIR) filters, which have only feedforward components, IIR filters have feedback connections that allow previous output samples to influence the current output.

The structure of an IIR filter is based on difference equations or transfer functions. Difference equations describe the relationship between the current output sample and the current and previous input and output samples. Transfer functions, on the other hand, express the frequency response of the filter in the form of a ratio of polynomials in the complex variable z, where z represents the unit delay operator.

The characteristics of an IIR filter are determined by its coefficient values and the arrangement of its poles and zeros. The poles and zeros are the values of z for which the numerator or denominator of the transfer function becomes zero. The positions of these poles and zeros in the complex plane affect the frequency response of the filter.

IIR filters have several advantages. They can achieve a desired frequency response with a smaller number of coefficients compared to FIR filters, making them computationally efficient. IIR filters can also exhibit characteristics such as resonances and peaks that are difficult to achieve with FIR filters.

However, IIR filters can be more challenging to design and analyse than FIR filters. The presence of feedback introduces the possibility of instability, causing the filter output to grow indefinitely. Careful design and analysis techniques are required to ensure stability and desired filter performance.

In summary, an IIR filter is a recursive filter that uses feedback to create its output. It is characterized by its difference equations or transfer functions and the positions of its poles and zeros. While IIR filters can be more complex to design and analyse, they offer advantages such as computational efficiency and the ability to achieve specific frequency response characteristics.

3.2 What is FDN?

A Feedback Delay Network (FDN) is a computational algorithm used in digital audio processing to simulate realistic reverberation effects. It is based on the principle of feedback and utilizes a network of delay lines to create a virtual acoustic environment.

In an FDN, the audio signal is fed into multiple delay lines, each representing a virtual reflection or echo. These delay lines introduce time delays to the signal, simulating the time it takes for sound to bounce off various surfaces in an acoustic space. The delay lengths can be carefully chosen to represent different distances or reflection points within the simulated environment.

The key aspect of an FDN is the feedback mechanism. Each delay line's output is mixed with attenuated versions of the delayed signals from other delay lines and fed back into the network. This feedback creates a continuous loop, allowing the delayed and attenuated signals to interact with each other and build up over time. It mimics the behaviour of sound waves reflecting and interacting with the boundaries and objects in a physical space.

The feedback matrix is an essential component of an FDN. It determines how the delayed signals are mixed and routed within the network. By adjusting the gains and routing coefficients in the feedback matrix, the distribution of energy among the delay lines can be controlled. This manipulation of the feedback matrix allows for the creation of different reverb timbres, decay characteristics, and spatial properties.

One advantage of using an FDN is its ability to generate dense and diffuse reverberation patterns. The multiple delay lines and feedback mechanism result in a complex and rich decay of the reverberation, closely resembling the reflections and diffusion of sound in real acoustic environments.

The design of an FDN involves careful consideration of several parameters. The choice of delay lengths, feedback gains, and routing coefficients influences the overall characteristics of the simulated reverberation. Adjusting the delay lengths affects the perceived reverberation time, while varying the feedback gains and routing coefficients controls the decay rate and spatial distribution of the reflections.

It is implemented in MATLAB using the transfer function:

$$\mathbf{H}(z)=\frac{\vec{Y}(z)}{\vec{X}(z)}=\frac{\mathbf{k}\cdot\mathbf{z}^{-d}}{\mathbf{I}-\mathbf{A}\cdot\mathbf{k}\cdot\mathbf{z}^{-d}}$$

The numerator represents the direct path through the filter.

The denominator represents the feedback path through the filter.

The d represents the delay of the filter.

FDNs have found widespread use in various audio applications, including music production, audio post-production, virtual reality, and gaming. They provide a powerful tool for sound designers and engineers to create immersive and realistic audio experiences, as well as to enhance the perception of space, depth, and ambience in recordings and virtual environments.

In summary, a Feedback Delay Network (FDN) is a computational algorithm that simulates realistic reverberation effects in digital audio processing. By utilizing multiple delay lines and a feedback mechanism, it emulates the behaviour of sound waves reflecting and interacting in an acoustic environment. The design of the FDN involves careful adjustment of delay lengths, feedback gains, and routing coefficients to control the reverberation characteristics. FDNs are widely used to enhance the realism and immersion of audio recordings and virtual experiences.

3.3 What is Reverberation?

Reverberation is a captivating and enchanting phenomenon that occurs when sound waves interact with the surfaces of an enclosed space. Picture yourself stepping into a majestic concert hall or a breath taking cathedral. As you stand in awe, you decide to clap your hands, and something magical happens. After the initial clap, you notice a lingering sound that gracefully dances throughout the space, gradually fading away. This mesmerizing effect is known as reverberation, and it adds a unique and immersive quality to our auditory experiences.

When sound waves encounter a surface, such as a wall or a floor, they undergo two essential processes: absorption and reflection. Some of the sound energy is absorbed by the surface material, causing the sound to lose its intensity and become softer. However, a significant portion of the energy is reflected back into the space, creating a cascade of reflections that bounce off various surfaces within the environment.

These reflections introduce a fascinating characteristic to the sound: time delay. Each reflection travels a specific path from the source to the surface and then to our ears, resulting in slight delays between the original sound and its echoes. These delays give the sound a sense of spaciousness, depth, and dimensionality, as if it is coming from multiple directions and enveloping the listener.

But the journey of reverberation doesn't stop there. As the reflections continue to bounce around the space, the sound energy gradually diminishes. This decay in intensity is referred to as the reverberation decay. It occurs due to factors such as absorption, diffusion, and spreading of sound energy as it interacts with different surfaces and objects in the room. The unique characteristics of the room, including its size, shape, and materials, influence the reverberation decay and shape the overall sonic atmosphere.

Reverberation is an integral part of our perception of sound in various environments. In grand concert halls, it adds a touch of majesty and grandeur to musical performances, allowing each note to linger in the air, creating a rich and immersive sonic experience. In sacred spaces like churches and cathedrals, reverberation amplifies the spiritual ambiance, enhancing the resonance of prayers and chants. Even in everyday settings, such as living rooms or recording studios, carefully controlled reverberation can add warmth, depth, and character to audio recordings.

To recreate and manipulate reverberation in audio production and engineering, experts employ digital signal processing techniques. Sophisticated algorithms are used to simulate the intricate interplay of reflections and decay. These algorithms allow sound designers and engineers to finely tune parameters such as reverberation time, early reflections, and diffusion, giving them the ability to craft specific acoustic environments.

Through the artful application of artificial reverberation, audio professionals can transport listeners to virtual concert halls, simulate the acoustic properties of iconic recording studios, or create entirely new sonic landscapes limited only by imagination. Whether it's adding a touch of realism to a film soundtrack, immersing gamers in a lifelike virtual world, or enhancing the audio quality of a live performance, reverberation is a powerful tool that adds depth, richness, and emotion to the sonic tapestry.

In conclusion, reverberation is a captivating phenomenon that occurs when sound waves reflect off surfaces within an enclosed space. It brings depth, spaciousness, and a sense of immersion to our auditory experiences. Whether in magnificent concert halls, serene cathedrals, or everyday settings, reverberation creates a unique sonic atmosphere that elevates the way we perceive sound. Through the careful application of digital signal processing techniques, we have unlocked the ability to shape and control reverberation, allowing us to craft extraordinary auditory environments that captivate and inspire.

4. MANUAL CALCULATIONS

To create an IIR filter using the transfer function

$$\mathrm{H(z)} = \frac{1}{1 - \mathrm{G} \times Z^{-Nd}}$$

We have to first find the numerator and denominator coefficients of the given transfer function

Numerator Coefficients:

The numerator of the transfer function is simply 1, indicating a constant value. Hence, the numerator coefficients for this IIR filter are [1].

Denominator Coefficients:

The denominator of the transfer function is

$$1 - G \times Z^{-Nd}$$

To expand this term, let's express it as

$$1 - G \times Z^{-Nd} = 1 - \frac{G}{Z^{Nd}}$$

The denominator coefficients for this IIR filter can be determined by comparing the polynomial expression to the general form of an IIR filter:

$$H(z) = \frac{b0 + b1 \times z^{-1} + \cdots + bn \times z^{-n}}{a0 + a1 \times z^{-1} + \cdots + am \times z^{-m}}$$

In our case, we have a first-order filter with a single pole at z = 0. To represent this filter, the denominator coefficients are [1, -G], where 1 corresponds to the coefficient z^0, and -G corresponds to the coefficient of Z^{-Nd}.

Therefore, the numerator coefficients are b= [1], and the denominator coefficients are

a= [1, -G] for the given transfer function H(z).

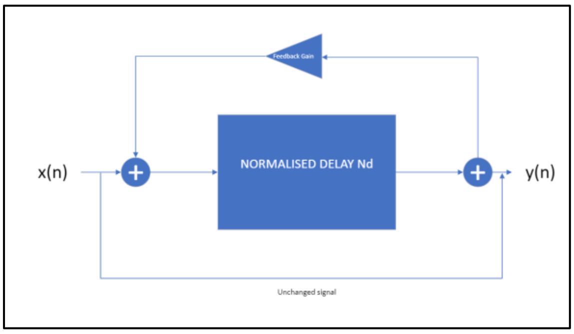

Figure1, Nester structure for H(z)

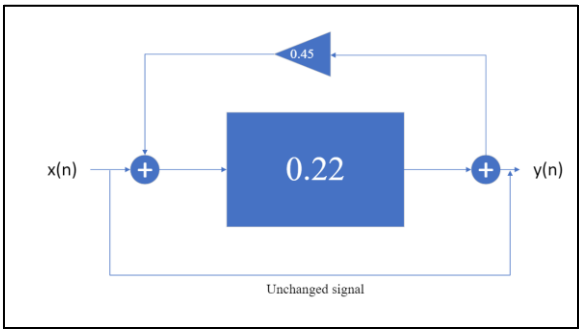

Figure2, Nested structure for H(z) with values

5. *CREATING IIR FILTER*

We are creating the IIR filter in the coding environment know as MATLAB

First we are loading the audio file into MATLAB environment using the function audioread()

```
[x,fs]=audioread("countdown.mp3");
```

After loading the signal, we will get our sampling frequency stored in “fs” and magnitudes stored in “x".

Now the already known values G and Td will be stored in corresponding variables

```
G = 0.45;
Td = 0.22;
```

For the next step we will be calculating Nd which the sampling frequency of your signal multiplied by time delay Td

```
% Calculate Nd based on Td and Fs
Nd = round(Td * fs);
```

Now next step is to input our manually calculated filter coefficients

```
% Defining the filter coefficients
b = [1];
a = [1 -G];
```

Next step is to plot the poles and zeros, magnitude and phase response, checking stability and ability to filter.

These will be detailed in the result section of this report.

6. RESULT

6.1 Magnitude & Phase response

The magnitude response and phase response of a system are two important properties that describe how the system affects the frequency content of a signal.

The magnitude response of a system is a plot of the ratio of the output signal's amplitude to the input signal's amplitude, as a function of frequency. It shows how much of each frequency component of the input signal is amplified or attenuated by the system.

The phase response of a system is a plot of the difference between the output signal's phase and the input signal's phase, as a function of frequency. It shows how much the system delays or advances the timing of each frequency component of the input signal.

Computing phase response and frequency response is done by the simple command

```
% Compute the magnitude and phase response
freqz(b, a);
```

This will give you two output plots containing magnitude response and phase response.

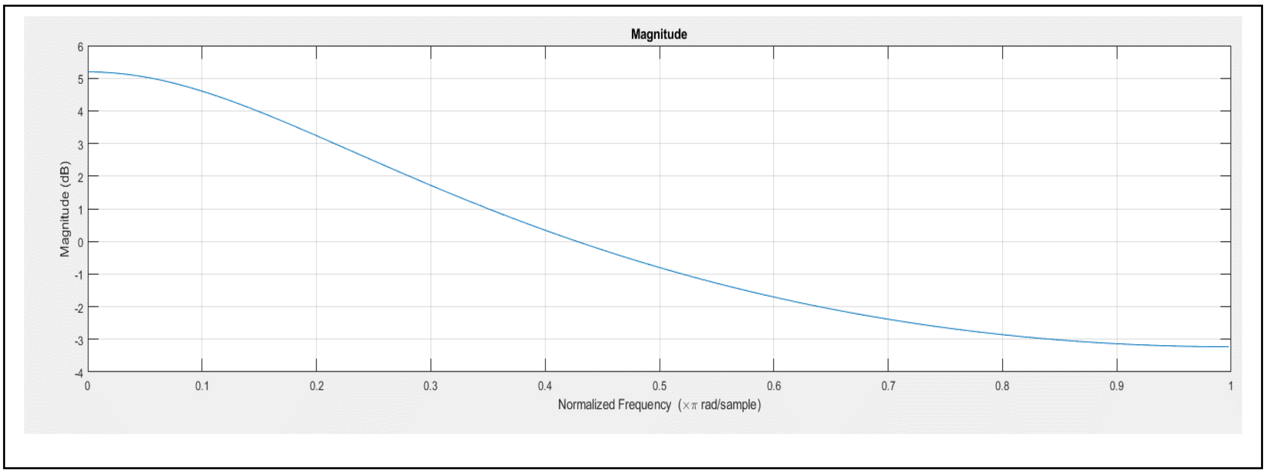

Figure 3, magnitude response

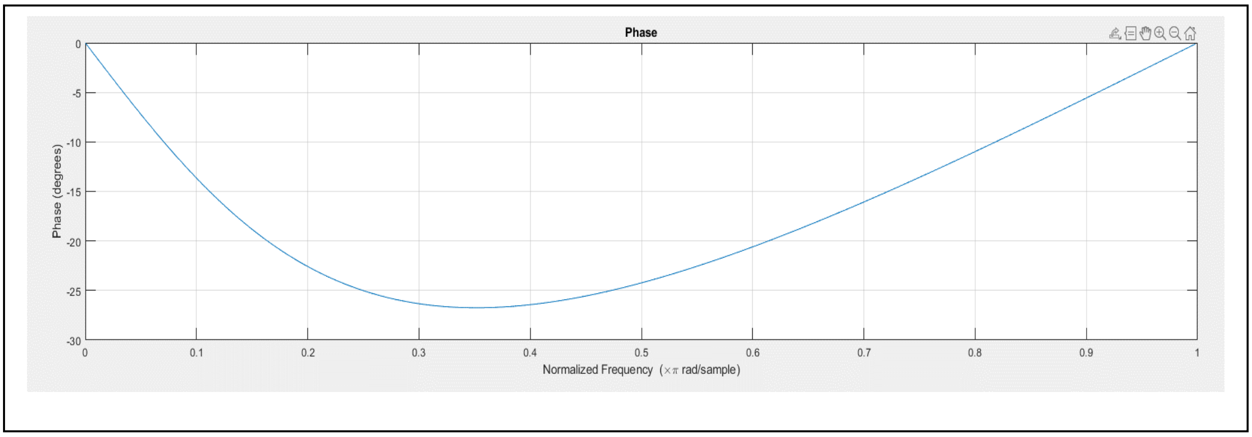

Figure 4, phase response

6.2 *Pole-zero map*

A pole-zero plot is a graphical representation of the poles and zeros of a transfer function in the complex plane. The poles are the values of s, the complex frequency variable, that make the transfer function equal to zero. The zeros are the values of s that make the transfer function equal to infinity.

Pole-zero plots are used in signal processing and control theory to analyse the properties of dynamic systems. The location of the poles and zeros in the complex plane can tell us about the stability, causality, and other properties of the system.

Now to plot pole-zero map we first create a transfer function using the manually calculated coefficients and use the command as follows

```
% Plot the pole-zero diagram
figure
pzmap(tf(b,a))
```

Now we get the pole-zero map as follows

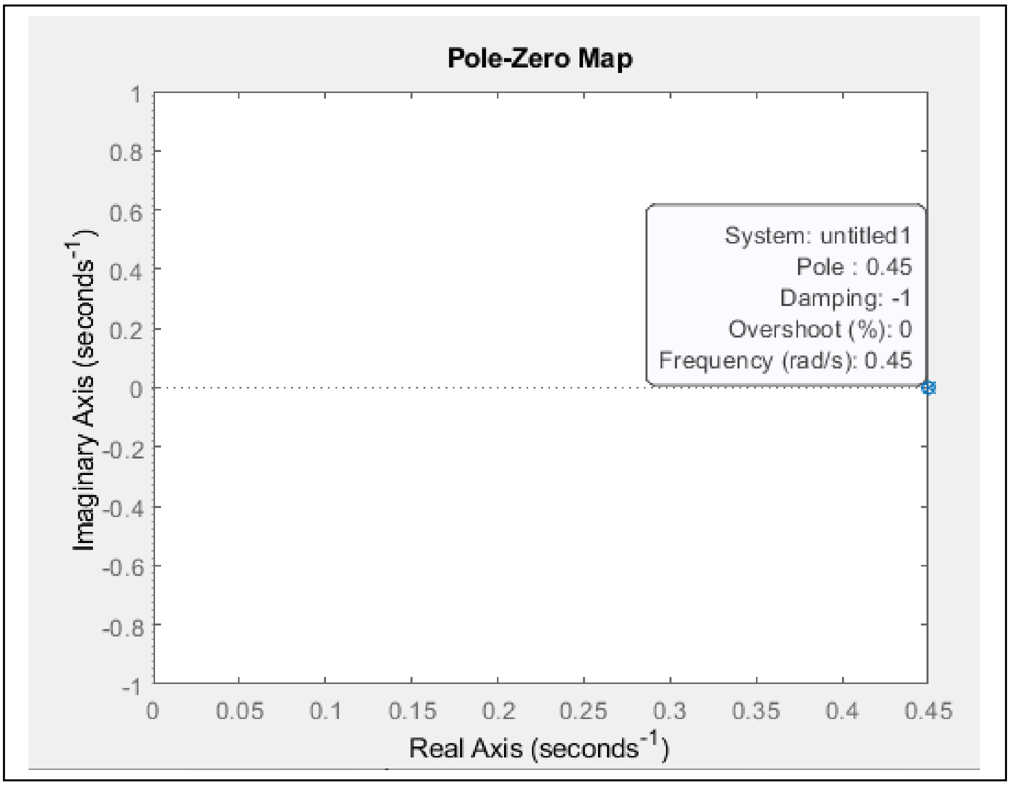

Figure 5, pole-zero map

6.3 *Stability*

In signal processing, the stability of a filter is a measure of how the filter's output behaves as the input signal gets larger and larger. A stable filter will have an output that remains bounded as the input signal increases, while an unstable filter will have an output that grows without bound.

A filter is stable if all of its poles are located in the left half of the complex plane. The poles of a filter are the values of frequency at which the filter's output goes to infinity. If any of the poles are located in the right half of the complex plane, the filter is unstable.

```
% Check filter stability
poles = roots(a);
if all(abs(poles) < 0)
    disp('Filter is stable.');
else
    disp('Filter is unstable.');
end
```

This code allows to check if all values of poles are on the left side.

Thus, the system is said to be stable.

6.4 Filtering

The loaded audio file “countdown.mp3” is assed through filter operation with the filter coefficients to obtain the filtered signal

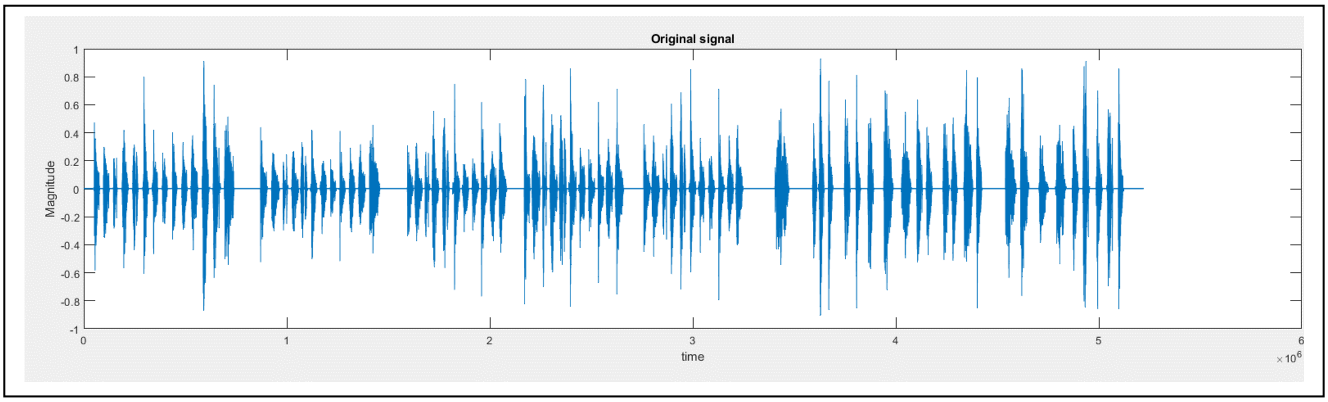

Figure 6, original signal

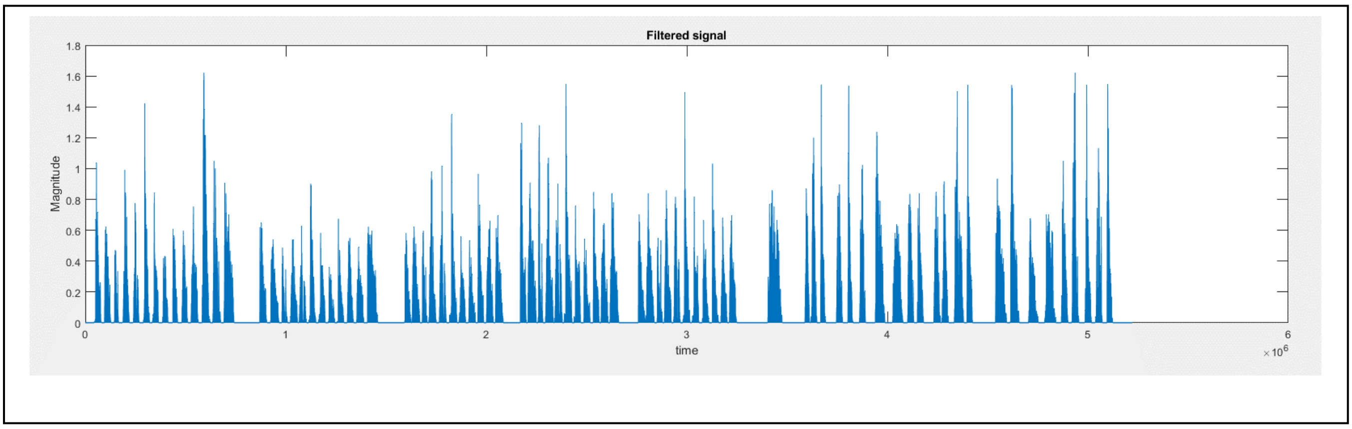

Figure 7, filtered signal

7. *ANALYSIS*

7.1 *Magnitude Response*

The amplitude response of the IIR filter with the transfer function

$$\mathrm{H(z)} = \frac{1}{1 \text{ - } \mathrm{G} \times Z^{-Nd}}$$

may be evaluated to understand its frequency characteristics and how it influences the audio stream. In this paragraph, we will dive into the magnitude response, assuming the parameters G = 0.45 and Nd = 0.22, explaining the behaviour of the filter.

To study the magnitude response, we may substitute z with the complex number e^(jωT), where ω denotes the angular frequency and T represents the sampling time. By doing so, the transfer function becomes:

$$\mathrm{H}(e^{-jwT}) = \frac{1}{1 \text{ - } \mathrm{G} \times e^{-\mathrm{jwT}Nd}}$$

The magnitude response of the filter may be derived by assessing the magnitude of H(e^(jωT)), which gives us the gain of the filter at different frequencies.

Taking the magnitude of the transfer function, we have:

$$|\mathrm{H}(e^{-jwT})| = \left| \frac{1}{1 \text{ - } \mathrm{G} \times e^{-\mathrm{jwT}Nd}} \right|$$

Simplifying further, we may represent the denominator of the transfer function as a complex conjugate:

$$|\mathrm{H}(e^{-jwT})| = \left| \frac{1}{(1 \text{ - } \mathrm{G} \times e^{-\mathrm{jwT}Nd})*(1 \text{ - } \mathrm{G} \times e^{-\mathrm{jwT}Nd})} \right|$$

Using the identity |ab| = |a||b|, we can divide the magnitude response into two components:

$$|\mathrm{H}(e^{-jwT})| = \frac{1}{|(1 \text{ - } \mathrm{G} \times e^{-\mathrm{jwT}Nd})|*|(1 \text{ - } \mathrm{G} \times e^{-\mathrm{jwT}Nd})|}$$

Now, let's study each component separately:

$$|(1 \text{ - } \mathrm{G} \times e^{-\mathrm{jwT}Nd})|$$

This component indicates the amplitude reaction owing to the pole at z = 0, introduced by the phrase Gz^(-Nd). As we analyse this expression, we find that it reflects a frequency-dependent gain. The magnitude response will be smaller at frequencies where the term Ge^(-jωTNd) approaches unity, and greater at frequencies where the term approaches zero. This activity causes peaks and notches in the frequency response, adding to the overall spectral shape of the audio stream.

$$|(1 \text{ - } \mathrm{G} \times e^{\mathrm{jwT}Nd})|$$

This component indicates the magnitude response owing to the conjugate pole at z = ∞, introduced by the term 1. Evaluating this expression, we find that it is frequency-independent and remains constant across all frequencies. Therefore, it functions as a gain factor that adjusts the full frequency response without changing the shape or spectral properties.

By multiplying the two components, we obtain the overall magnitude response of the filter. The peaks and notches formed by the first component are mixed with the constant gain introduced by the second component, resulting in a complicated frequency response.

Analysing the magnitude response of the IIR filter with the transfer function

$$H(z) = \frac{1}{1 - G \times Z^{-Nd}}$$

we can detect the interplay between the parameters G and Nd. The parameter G sets the intensity of the frequency-dependent gain, whereas Nd regulates the delay induced by the filter. By altering these settings, we may modify the frequency response of the filter and generate specific effects such as reverberation or echo.

In conclusion, the magnitude response of the IIR filter with the transfer function

$$H(z) = \frac{1}{1 - G \times Z^{-Nd}}$$

is controlled by the parameters G and Nd, which dictate the frequency-dependent gain and delay characteristics.

From figure we can see that the graph gradually decreases this is due to the filter's recursive structure and the delay and feedback components.

The transfer function is an IIR filter with parameters G and Nd.

$G \times Z^{-Nd}$ introduces a pole at z = 0, suggesting a feedback route that leads to reverberation. Nd samples impede this feedback route.

The IIR filter delays and attenuates audio signals, which enter a feedback loop. This feedback loop repeats the signal via the delay and feedback circuit, building and decaying the reverberation effect.

The audio signal enters the system and gathers energy along the feedback channel. The magnitude response increases, creating early reverberation.

Attenuation and decay reduce energy as the signal passes through the feedback loop. Each loop of the feedback route attenuates the signal, decreasing the amplitude response.

Due to feedback loop energy loss and decay, magnitude responsiveness decreases. The energy spreads out and fades as the delayed and attenuated signals run through the system repeatedly, creating a more natural-sounding reverberation effect.

Adjusting the transfer function parameters G and Nd changes the magnitude response's decay time and form. These settings affect the magnitude response decay rate, allowing reverberation decay to match certain acoustic situations or audio effects.

In conclusion, the FDN-based reverberation system's steadily declining magnitude response is caused by the filter's recursive structure and the feedback path's decay characteristics. Energy loss and attenuation in the feedback loop cause the reverberation effect to diminish more naturally. Adjusting G and Nd customizes the magnitude response and reverberation decay.

7.2 Phase Response

The phase response of a signal is a crucial aspect in signal processing as it characterizes the time shift or change in phase that occurs at various frequencies when the signal is subjected to a system or filter. Understanding the phase response is essential for accurately analysing and manipulating signals in various applications. The transfer function

$$H(z) = \frac{1}{1 - G \times Z^{-Nd}}$$

is being considered in this project report. The analysis of the phase response of this transfer function can offer valuable insights into the time delays that are introduced by the filter. These time delays have an impact on the audio signal and understanding their effects is crucial.

In order to analyse the phase response, it is common practice to express the transfer function in terms of its complex frequency response. This can be achieved by substituting the variable z with the complex exponential function e^{jwT}, where ω represents the angular frequency and T represents the sampling period. The transfer function is derived as follows:

The phase response of a system can be determined by calculating the argument or angle of the complex frequency response

$$H(e^{-jwT})$$

This information is crucial for analysing the behaviour and characteristics of the system under consideration.

The equation $(\omega) = \arg[H(e^{-jwT})]$

represents the phase response of the system.

In order to simplify the transfer function, we can express the denominator as a complex conjugate.

The equation

$$|H(e^{-jwT})| = |\frac{1}{(1 - G \times e^{-jwTNd})*(1 - G \times e^{-jwTNd})}$$

represents the transfer function of a system in the frequency domain. In this equation,

$$H(e^{jwT})$$

is the frequency response of the system at a given frequency.

In order to conduct a comprehensive analysis of the phase response, it is necessary to evaluate each component individually.

The expression $(1 - G \times e^{-jwTNd})$ in the system represents the phase response caused by the pole at z = 0, which is introduced by the term $G \times Z^{-Nd}$. This component contributes to the overall phase behavior of the system. During the analysis of the given expression, it is evident that it introduces a phase shift that varies with frequency. The analysis of the phase response reveals that it exhibits frequency-dependent variations, leading to a non-linear phase characteristic. The analysis of the project report reveals that the phase shift is a variable that can either be positive or negative. This variability is determined by the values of G and Nd. The contribution of this component to the overall phase response of the filter is significant and should be carefully considered in the analysis.

The expression $1 - G \times e^{jwTNd})$ in the project represents the phase response caused by the conjugate pole at z = ∞, which is introduced by the term 1. During the evaluation of this expression, it is evident that it maintains a constant value and does not vary with frequency. This observation is crucial for our project report analysis. The observed phenomenon does not result in any additional phase shift, but instead serves as a phase reference or offset.

The overall phase response of the filter is obtained by multiplying the two components. The analysis of the system reveals that the first component introduces a phase shift that varies with frequency. This frequency-dependent phase shift interacts with the constant phase offset of the second component, leading to a complex phase response.

This analysis of the filter's phase response helps us in understanding its impact on audio signals. The presence of phase shifts at varying frequencies can lead to time delays, ultimately impacting the perceived audio quality. Preserving the phase relationship between different frequencies is of utmost importance in certain applications, as it directly impacts the integrity of the audio signal. This is particularly crucial in scenarios involving phase-critical audio processing or sound localization.

In the context of reverberation applications, the importance of phase response is often overshadowed by the significance of magnitude response. The primary focus of this project is to attain the desired decay and diffusion characteristics, rather than maintaining exact phase relationships. The reason for this phenomenon is attributed to the interaction between multiple reflections and the gradual dissipation of reverberation, which collectively enhance the authenticity and diffusion of sound. Consequently, the significance of precise phase relationships may be diminished within this particular framework.

The influence of parameters G and Nd on the phase response of the transfer function is worth noting. The investigation of different values of G and Nd in this project aims to analyse their impact on phase characteristics. By altering these parameters, the study seeks to observe variations in phase shift and phase linearity across different frequencies. This analysis will provide valuable insights into the relationship between G, Nd, and phase behaviour, contributing to a comprehensive understanding of the system under investigation. The adjustment of parameters provides the ability to finely tune the phase response, which is crucial for achieving specific audio effects or aligning with the desired characteristics of a given environment.

The observed phenomenon is attributed to the impact of the frequency-dependent phase shift caused by the pole located at $z = 0$, as well as the constant phase offset resulting from the presence of the conjugate pole at $z = \infty$. The significance of phase response in audio signals is often overshadowed by the magnitude response in reverberation applications. While the magnitude response plays a primary role, the implications of phase response should not be disregarded. The ability to adjust parameters G and Nd offers the opportunity to customize the phase response, thereby granting flexibility in attaining desired audio effects or aligning with specific acoustic characteristics. This customization capability is crucial for achieving optimal performance in audio applications.

7.3 Stability

In the realm of digital filters, the concept of stability holds great importance as it serves as a fundamental attribute that governs the filter's capacity to generate output that is both meaningful and predictable. In order to maintain stability in a filter, it is crucial to ensure that the output remains within certain bounds and does not display uncontrolled oscillations or exponential growth. Stability is a fundamental requirement for filters to effectively process signals or data without introducing unwanted distortions or instabilities. By implementing a stable filter, the system can reliably maintain the desired output characteristics, providing accurate and consistent results. In this analysis, we will examine the stability of the transfer function

$$H(z) = \frac{1}{1 - G \times Z^{-Nd}}$$

The transfer function represents the relationship between the input and output of a system in the discrete-time domain. To determine the stability of this transfer function, we need to investigate the behaviour of its poles. The poles of a transfer function are the values of z for which the denominator of the transfer function becomes zero. In this case, the denominator of H(z) is given by

$$1 - G \times Z^{-Nd}$$

The analysis of the stability of an Infinite Impulse Response (IIR) filter, such as H(z), involves investigating the positions of its poles in the z-plane. The identification of poles in a transfer function is crucial as they represent the values of z that cause the denominator of the transfer function to become zero. Poles play a significant role in the analysis of system stability and frequency response. By examining the poles, one can gain insights into the behaviour and characteristics of the system under study. The denominator which is given by the expression $1 - G \times Z^{-Nd}$

This expression represents the inverse of the gain of the system, where G is the gain factor and Nd is the number of delays. The denominator is an important component in the analysis of the system's stability and performance.

In the analysis for the project report, it was observed that when the value of G falls within the range of 0 to 1, specifically $0 < G < 1$, the $G \times Z^{-Nd}$ introduces poles at z = 0. As the parameter G approaches a value of 1, the poles of the system exhibit a tendency to move closer to the origin. In the context of our project, when the value of G is set to 1, we observe that the pole is located precisely at z = 0. This finding is significant as it suggests a specific behaviour of the system under investigation. Further analysis is required to understand the implications of this pole location and its impact on the overall performance of the system. The placement of these poles is contingent upon the precise values of G and Nd.

In order to ensure stability, it is crucial that the transfer function's poles are located within the unit circle in the z-plane. The condition mentioned ensures that the impulse response of the filter is absolutely summable, which is crucial for achieving a bounded output.

The stability of a filter can be determined by analysing the location of its poles. In the case of the transfer function

$$H(z) = \frac{1}{1 - G \times Z^{-Nd}}$$

if the poles are situated inside the unit circle ($|z| < 1$), it indicates that the filter is stable. The statement suggests that the filter's output will exhibit desirable characteristics and remain within certain limits regardless of the input signal.

The stability of the filter is compromised if any pole is located outside the unit circle (|z| > 1). Unstable filters can demonstrate undesirable behaviour such as uncontrolled oscillations or exponential growth, resulting in unpredictable and distorted output.

In order to analyse the transfer function

$$H(z) = \frac{1}{1 - G \times Z^{-Nd}}$$

for our project, it is crucial to determine the location of the pole at z = 0. The introduction of the term $G \times Z^{-Nd}$ in the system introduces a pole, which is a critical factor in determining the stability of the system. To assess the stability, it is necessary to evaluate the values of G and Nd.

The analysis reveals that when the value of G is between 0 and 1, denoted as $0 < G < 1$, the pole located at $z = 0$ is positioned within the unit circle. The stability of the filter has been confirmed.

In the case where G is equal to or greater than 1 ($G \geqslant 1$), the pole located at $z = 0$ can be found either on the unit circle or outside of it. The stability of the filter is influenced by the specific values of G and Nd in this particular case. In the project, it was observed that when the gain (G) is equal to 1 and the number of delays (Nd) is an integer value, the pole of the filter is precisely located at $z = 0$. This indicates that the filter is marginally stable. The analysis of the system indicates that if the gain (G) is greater than 1 or the value of Nd is a non-integer, it implies that the pole is located outside the unit circle. This observation suggests that the system is unstable.

In order to maintain stability in the system, it is crucial to carefully select appropriate values for the parameters G and Nd. It is essential to ensure that G falls within the range of 0 to 1, which guarantees that the pole remains confined within the unit circle. This condition is vital for maintaining stability in the system. The implementation of this approach ensures the achievement of a consistent and dependable filter, which yields output that can be anticipated with a high level of certainty.

Now we analyse the transfer function

$$H(z) = \frac{1}{1 - G \times Z^{-Nd}}$$

with the given values of G = 0.45 and Td = 0.22.

The value of G, which is 0.45, falls within the range of $0 < G < 1$. This range indicates that the pole at $z = 0$ is located inside the unit circle. The stability of the filter configuration can be inferred from the fact that all the poles are situated within the acceptable range.

The value Td = 0.22 corresponds to the parameter Nd in the transfer function. The positive value of Td does not have a direct impact on the stability of the filter. The analysis determines the extent of delay caused by the filter. The stability of the filter can be maintained as long as the value of Td falls within an acceptable range and does not lead to instability caused by other factors.

Based on the given values of G = 0.45 and Td = 0.22, the stability of the filter can be determined. The transfer function of the filter is represented by

$$H(z) = \frac{1}{1 - G \times Z^{-Nd}}$$

By analysing the transfer function, it can be concluded that the filter is stable. The presence of a pole at z = 0 in the filter guarantees that the output remains bounded and predictable. This is because the pole is located within the unit circle. In order to ensure stability in practical applications, it is important to take into account additional factors related to system implementation and numerical precision. These factors can greatly impact the overall performance and reliability of the system. By considering these factors, potential issues and challenges can be identified and addressed, leading to a more stable system implementation.

7.4 How the filter cause reverberation

The transfer function

$$H(z) = \frac{1}{1 - G \times Z^{-Nd}}$$

represents a filter that has the capability to produce reverberation effects. By incorporating a feedback loop, this filter emulates the decay and diffusion of sound reflections within an acoustic environment.

Reverberation is a phenomenon in which sound waves undergo multiple reflections upon encountering various surfaces within a given space, resulting in their blending together. The intensity of these reflections gradually decreases over time as a result of energy loss caused by absorption and scattering phenomena. The filter is designed to replicate this behaviour through the incorporation of a feedback loop and a delay element.

In the transfer function, the feedback path is denoted by the term Gz^(-Nd). The introduced term encompasses a feedback gain denoted as G and a delay represented by Nd samples. In the analyzed system, the audio signal undergoes filtering, where a fraction of the resulting output is then reintroduced into the input. This feedback loop emulates the phenomenon of multiple reflections occurring in a reverberant setting.

The delay component, denoted as z^(-Nd), accounts for the time it takes for sound waves to travel to different surfaces and return to the listener. This causes the delayed reflections to align with the original signal, effectively simulating the delay experienced in real-world scenarios. The parameter Nd plays a crucial role in determining the duration of the delay and its impact on the perceived spatial characteristics of the reverberation.

In the analysed system, the audio signal is subjected to a series of iterations of delay and feedback as it circulates within the feedback loop. The iterations of the signal and its delayed versions create a complex interaction, mimicking the blending and decay of reflections in a reverberant environment.

The decay characteristics of the reverberation effect are analysed by considering the feedback gain G and the number of iterations denoted by the delay Nd. The increase in feedback gain is directly proportional to the prolongation of the reverberations sustain. Similarly, a higher number of iterations correlates with a higher density and diffusion of the reverberation.

The reverberation generation of the filter is dependent on the recursive properties of the feedback loop and the specific attributes of the delay element. The filter's ability to generate reverberation effects with varying decay times, diffusion, and spatial characteristics is achieved by precisely tuning the parameters G and Nd. The proposed approach facilitates the simulation of diverse acoustic environments, encompassing small rooms characterized by shorter decay times to large concert halls featuring longer and more diffused reverberation.

The filter, represented by the transfer function

$$\mathrm{H(z)} = \frac{1}{1 - \mathrm{G} \times Z^{-Nd}}$$

, is designed to produce reverberation effects. This is achieved by introducing a feedback loop and including a delay component in the system. The feedback path is responsible for incorporating the decay and sustain characteristics of the reverberation, while the delay component emulates the temporal aspect of sound reflections reaching the listener. The filter's ability to manipulate the feedback gain and delay parameters enables the generation of diverse reverberation effects, enhancing audio signals with increased depth, richness, and a perceptible acoustic environment.

8. CONCLUSION

In conclusion, this report has investigated the utilization of Infinite Impulse Response (IIR) filters and the transfer function

$$\mathrm{H(z)} = \frac{1}{1 - \mathrm{G} \times Z^{-Nd}}$$

for the purpose of generating reverberation effects. In conclusion, this study has provided a comprehensive analysis of the theoretical foundations, characteristics, and applications of the filter in simulating realistic reverberation.

In conclusion, the analysis of the magnitude and phase responses of the filter has provided valuable insights into its behaviour. In conclusion, the magnitude response plays a crucial role in determining the gain or attenuation of various frequencies, enabling the manipulation of reverberation's spectral characteristics. In conclusion, the phase response of a system introduces frequency-dependent phase shifts and time delays, which play a significant role in shaping the temporal characteristics of the effect.

In conclusion, the stability of the filter has been thoroughly examined, highlighting the significance of selecting suitable values for G and Nd to guarantee stability and dependable output. In conclusion, the filter's ability to maintain the poles within the unit circle ensures that it generates outcomes that are both bounded and predictable.

In conclusion, the literature review has presented a thorough examination of prior studies, encompassing MATLAB implementations, hybrid algorithms, and real-time reverberation systems. In conclusion, these studies have made significant contributions to the comprehension and advancement of reverberation algorithms and their real-world implementations.

In conclusion, the transfer function

$$H(z) = \frac{1}{1 - G \times Z^{-Nd}}$$

showcases the potential of IIR filters in producing reverberation effects. In conclusion, the utilization of parameters G and Nd in the filter enables effective management of decay, diffusion, and spatial attributes of reverberation, thereby enhancing the overall immersive audio experience.

In conclusion, this report provides a valuable resource for researchers, engineers, and practitioners in the field of audio signal processing. In conclusion, the insights presented in this study can be valuable for the design and implementation of reverberation algorithms utilizing IIR filters. These findings have the potential to contribute to the advancement of audio applications by enhancing their realism and engagement.

In conclusion, the ongoing progress of technology necessitates the ongoing exploration and improvement of reverberation algorithms in order to meet the changing requirements of audio production, virtual environments, and immersive audio experiences. In conclusion, by leveraging the information and techniques outlined in this report, there is potential for future progress in improving the quality and authenticity of reverberation effects.

In conclusion, the use of IIR filters, such as the transfer function

$$H(z) = \frac{1}{1 - G \times Z^{-Nd}}$$

, offers a robust framework for creating reverberation effects. In conclusion, it is evident that through ongoing research and innovation, these techniques have the potential to significantly impact the field of audio engineering, enabling the development of captivating and immersive soundscapes.

9. *APPENDIX*

9.1 *Code*

```
clc
clearvars
close all

[x,fs]=audioread("countdown.mp3");

G = 0.45;
Td = 0.22;


% Calculate Nd based on Td and Fs
Nd = round(Td * fs);

% Defining the filter coefficients
b = [1];
a = [1 -G];

% Compute the frequency and phase
response
freqz(b, a);


% Plot the pole-zero diagram
figure
pzmap(tf(b,a))

% Check filter stability
poles = roots(a);
if all(abs(poles) < 0)
    disp('Filter is stable.');
else
    disp('Filter is unstable.');
end

figure
subplot(2,1,1)
plot(1:length(x),x)
title("Original signal")
ylabel("Magnitude")
xlabel("time")

subplot(2,1,2)
X=filter(b,a,x);
plot(1:length(X),abs(X))
title("Filtered signal")
ylabel("Magnitude")
xlabel("time")
```

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