Spectrogram Study of Echo and Reverberation– A Novel Approach to Reduce Echo and Reverberation in Room Acoustics

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Abstract. This paper presents new approach to enhance the quality of sound signal perceived by the listener in a concert hall considering Spectrogram and Single Spectrum Plot. We simulate a natural concert hall using Digital Echo and Schroeder Reverberator models. We analyze the results using Spectrogram and Single Spectrum Plots and finally propose a model for better analysis of room acoustics.

Keywords: Acoustics, Echo, Reverberation, SIMULINK®, Single Spectrum Plot and Spectrogram.

1 Introduction

An Echo is a reflection of sound, arriving at the listener some time after the direct sound. This is followed by a group of closely packed reflections with decaying amplitude referred to as Reverberation. The display of the magnitude of the Short-Time Fourier Transform (STFT) is usually referred to as the Spectrogram. However, since the STFT is a function of two variables, the display of its magnitude would normally require three dimensions. Often, it is plotted in two dimensions, with the magnitude represented by the darkness of the plot. In the STFT magnitude display, the vertical axis represents the frequency variable (ω) and the horizontal axis represents the time index (n) [5]. In this paper, we perform digital implementation of Echo and Schroeder Reverberator models using SIMULINK® to obtain Spectrograms of input and output signals. For each Spectrogram, we determine Single Spectrum Plot and finally compare the Single Spectrum Plot obtained for output signal against input Single Spectrum Plot to obtain the Error Plot. From the Error Plot analysis, we propose a model to analyze Echo and Reverberation to improve quality of sound signal perceived.

2 Literature Review and Audio Effects Used in Sound Production

Echo can be implemented digitally using FIR or IIR filter. It can be Single or Multiple Echo Filters [5]. There are several models proposed for Reverberation which include

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the algorithms proposed by Schroeder, Moorer, Gardner, Dattoro [2] and Jot [3]. Schroeder Reverberator forms the basis for all other Reverberator models. In this paper, we concentrate mainly on FIR Single Echo Filter, Schroeder's Comb Reverberator and Allpass Reverberator. Echoes are simply generated by delay units. The direct sound and a single Echo appearing after R (delay parameter) sampling periods later can be generated by FIR filter [5] whereas the gain parameter α with $|\alpha| < 1$ represents the signal loss caused by propagation and reflection. Schroeder's Comb Reverberator has a delay cell with an output redirected to the input. In second Reverberator model, Schroeder replaced Comb Filter with an Allpass Filter to develop more realistic Reverberation. According to Schroeder, approximately 1000 Echoes per second are necessary to create a Reverberation that sounds natural and free of flutter [1].

3 Digital Implementation and Analysis Using SIMULINK®

We implement Digital Echo and Reverberator models using SIMULINK®. The Echo – Simulink model is represented in Fig. 1. Fig. 2 represents Comb Filter Reverberator and Fig. 3. represents Allpass Reverberator. Fig. 4 represents the proposed Error Detection Model.

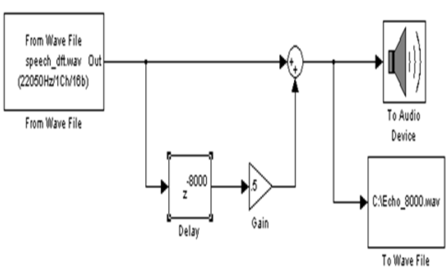


Fig. 1. Echo Model

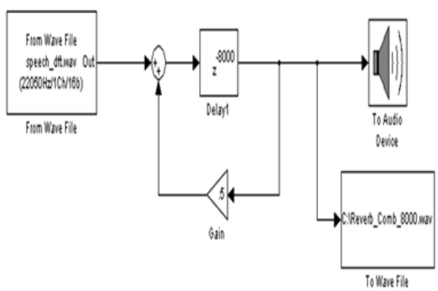


Fig. 2. Comb Filter Reverberator

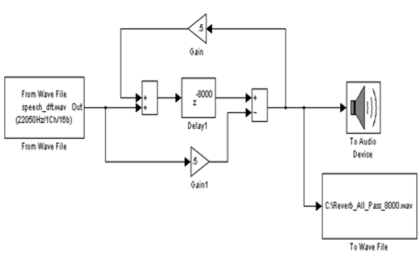


Fig. 3. Allpass Reverberator

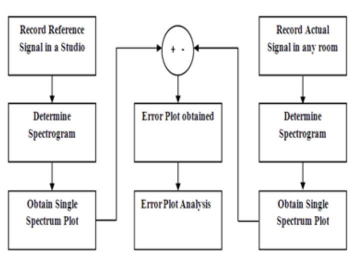
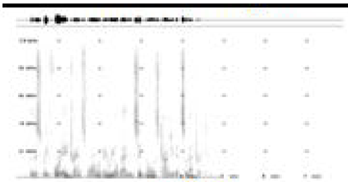


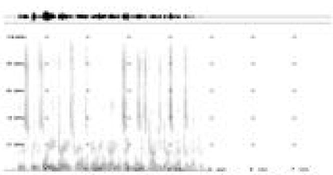
Fig. 4. Error Detection Model

We use PCM 16 bit signed, 352 kbps, 22050 Hz speech signal as input signal in all three existing models and for each model we run the simulation for 8 seconds and record their corresponding output samples. We determine Spectrogram for input and output samples to obtain Single Spectrum plot. Table 1 represents the Spectrograms obtained for the input and output of Echo, Comb Reverberator and Allpass Reverberator respectively. For each Single Spectrum Plot, power difference between input and output signal is determined using Equation 1.

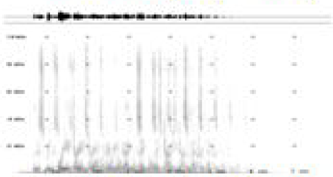
Table 1. Spectrograms



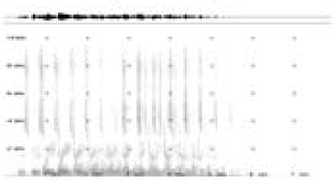
a. Input



b. Echo Output (R=8000)



c. Comb Reverberator Output (R= 8000)



d. Allpass Reverberator Output (R=8000)

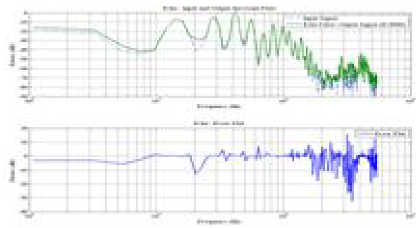


Fig. 5. Specturm Plot and Error Plot of Echo Filter

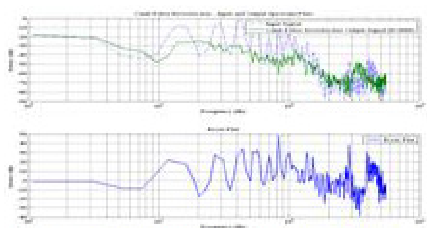


Fig. 6. Specturm Plot and Error Plot of Comb Reverberator

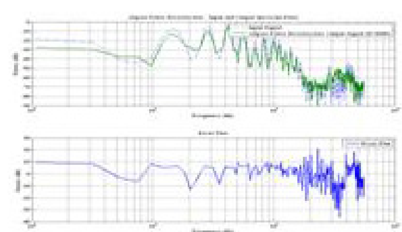


Fig. 7. Specturm Plot and Error Plot of Allpass Reverberator

Error Signal Power (dB) = Input Signal power (dB) – Output Signal Power (dB) **(1)**

The magnitude of Error Signal obtained from Equation 1, indicates the amount of power variation between input and output signal. We refer to this as Error Plot. Error Plot is simply the power difference between input and output power levels for all frequencies at a particular time index. Fig. 5, Fig. 6 and Fig. 7 represents the Single Spectrum Plots and Error Plots. Ideally, Equation 1 will yield 0 dB values if the listener perceives the sound signal with the same magnitude as that of the original signal without any reflections and decay in signal magnitude. But in all practical cases, we get some difference in values due to sound reflections and decay in signal magnitude. We refer to this as Error Value. In Echo spectrogram, the reflected signals will have considerable magnitude when compared to direct sound. But in Reverberation Spectrogram the amplitude of reflected signals decays with time. In both Reverberator models the entire spectrum gets shifted with increase in delay parameter but in Allpass Reverberator some spectrum is fixed due to feed-forward path which produces natural reverberation.From Figure 5 we can observe that Echo Error plot has almost 0 dB values for frequencies less than 10 KHz. But from Figure 6 and Figure 7 we observe that Reverberation Error plots has almost 0 dB values for frequencies less than 1 KHz. It must be reduced to 0 dB for better audio signal perception.

4 Conclusion

This paper provides clear distinction between Echo and Reverberation using Spectrogram and Single Spectrum Plot in a natural concert hall. From simulation analysis we conclude that sound signal perception can be enhanced if input and output signal power variations match together even with variations in delay parameter. This paper also lays the foundation for further research work in the field of room acoustics to increase the quality of sound signal perceived by the listener at any point in the room by getting the values of Error Plot to almost 0 dB for all frequencies with varying delay parameters. This methodology can be used to improve acoustic design.

References

1. Schroeder, M.R.: Natural Sounding Artificial Reverberation. JAES 10(3), 219–223 (1962)
2. Dornean, I., Topa, M., Kirei, B.S.: Digital Implementation of Artificial Reverberation Algorithms. Acta Technica Napocensis 49(4) (2008)
3. Beltrán, F.A., Beltrán, J.R., Holzem, N., Gogu, A.: Matlab Implementation of Reverberation Algorithms. Journal of New Music Research 31(2), 153–161 (2002)
4. Moorer, J.A.: About this Reverberation Business. Computer Music Journal 3(2) (1978)
5. Mitra, S.K.: Digital Signal Processing, 3rd edn. Tata McGraw-Hill Companies (2006)
6. Spectrogram Version 5.0, `http://smedor.com/gram50.zip`