FINAL REVIEW

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Overview

Our company is called Data Dynamics, we've been designing digital effects processors for roughly 3 years and would like to introduce to you a new digital delay and reverb unit the D16 reverberation processor to which our research and development team has come to create a new standard in digital signal processing which is suitable for live sound or for studio use. For example, if you want to match the phase of the mic and DI signals while recording, the simplest solution is to artificially delay the DI signal as well, in order to match its phase with that of the already-delayed mic signal.
We have developed this algorithm with focus on real-time performance, which is widely used on many modern recording systems as plugins or electro-acoustic devices simulating reverberation of a room with changeable parameters.
The unit offers mono operation and that the reverb treatments it offers are the ones most likely to get used in a live context for use with little or nothing in the way of special effects, while the operating system is very obvious that users are unlikely to spend much time reading the manual, though this will be provided with a Matlab coded script at the end. We have given it a lot of thought and effort in aiming for the best sound quality without the big price of other competitors offering

Description

An electroacoustic enhancement system or a reverberation enhancement system is a system that is used to alter the sound field in a space using microphones, loudspeakers and electronic circuits. Reverberation enhancement systems have been used in concert halls and multipurpose halls either to correct for inferior acoustic design or to provide means to change the acoustical properties of the hall. They are used in halls that don't naturally have enough reverberant energy. It's also a great tool for sound engineers, live sound engineers and mastering engineers to recreate the acoustics of a room or a favorite concert hall that they have visited.

In essence, it all starts with a sound in a room. Pressure causes sound waves to travel in all directions, rapidly reaching out to walls and ceiling where its being absorbed and reflected. This also called the transfer function. They are present in all electroacoustic enhancement systems as depicted in Fig. 1. The ideal way in which to physically simulate reverberation of a room is to combine it with binaural impulse response of the subject room and outputting it with reverberation by convolution.

But what if the room doesn't exist? This makes things harder for the system designer and may very well be a costly exercise. Initially, we need to have an idea of how we want this room to sound, so given that basis we need a detailed sketch of the physical geometry of the room, materials, dimensions, source and receiver. The next step when we have all these parameters is to apply wave propagation and see how the waves interacts within the environment and in turn by using finite impulse response filters (FIR), will get a rendering of the room reverberation. This exercise is useful when in comparison between the physical dimension of a room and the computational results.

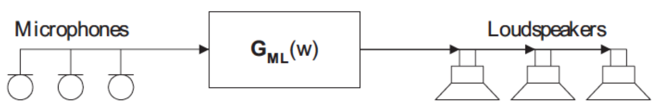


Figure 1: Transfer function of a reverberation enhancement system.

The impulse response diagram depicted in Fig. 2 comprises of peaks, which are discrete while it almost exponentially decreases. By dissecting this impulse response in three parts, we are able to model it. The model will consist of a direct signal followed by early reflections and late/diffuse reverberations.

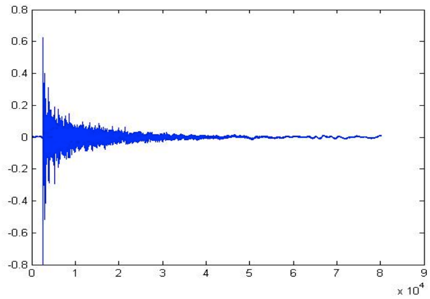
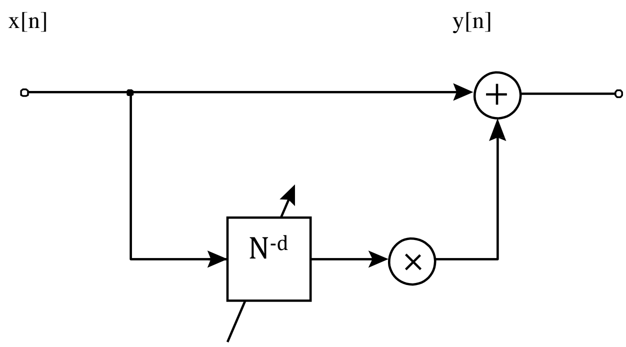


Figure 2: Impulse response of a room

This tells us that a reverberation algorithm can essentially be divided in two parts. This would include a number of discrete finite echoes that are found in the original impulse response with the second part an exponentially decreasing echo density. With this in mind, these calculations can be performed by using geometric techniques such as beam tracing or ray tracing which has gained popular approach for its use in real-time applications due to is efficient algorithm

Specification

Our thought process on this reverb algorithm was to use an enhancement system that can operate on the whole audio range or on a narrow band. There are also systems with no additional reverberation algorithm in. The choice of bandwidth and number of channels is application dependent. In this design, our engineers concentrate on a system that is a wideband single channel system containing additional gain. The block diagram below of the applied reverberation algorithm is depicted in Fig. 3



The architecture of this design is quite simple. A user input signal will enter through x[n] then combining it with delayed version of the original signal [N^{-d}]. The user will then have the option to either gain (x) the signal before it reaches the output stage or so choose to leave it at unity gain. The limitation to the user variable is that the delay time is limited to 0 to 1 seconds and gain is limited from 1 to 3 seconds. We found that this is sufficient enough to accomplish the needs of a live sound application and also the possibility of studio applications aswell. We have also implemented a Matlab coded script for you to assess before implementing this system to hardware. Full Matlab script will be included at the end of this report.

Implementation

Although the number of editable parameters may seem small, the algorithms are so well designed that for much of the time you may find your desired parameter with little effort. Matter of fact, it is very hard to make this reverb sound bad. We have tested this algorithm and software to be very reliable. Implementation of this software have referenced with previous Lab 2 report.

Delay2.m

Analysis of a delay filter with user definable variables within the delay time and gain and graphical representation.

Syntax

Outwave = Delay2 (WavFN, DTsec, Gain)

Description

Input signal is to be imported for analysis and can only be used within the function. In this case, the user is to input specific values as there will be no default values.

WavFN	=	Signal to be processes
DTsec	=	Delay time of the second signal in seconds
Gain	=	Amount of gain added to the second signal

Input variables:

- Delay time = 0 - 1
- Gain = 1 – 3

Output

Delay2 will output wave in three parameters. The input wave file, an audible playback and a graphical plot showing the original, combine and delayed wav file.

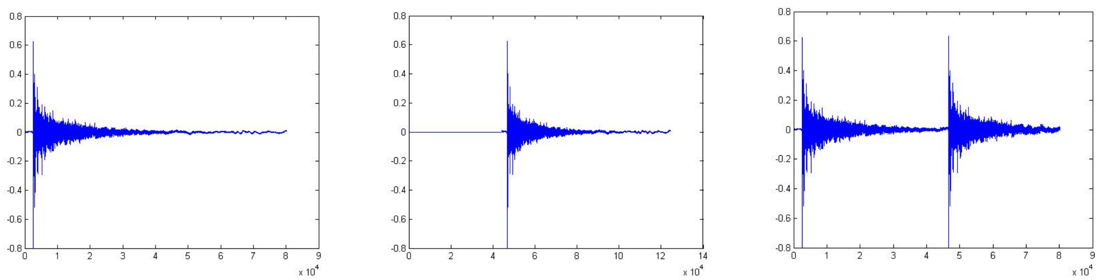


Figure 1 – Output plots

Example

To create a 1 second delay with no gain from original signal;

Outwave = Delay2 ('Impulse_Resp_A_Mono.wav',1,1)

Audible and graphical results will be initiated.

Diagnostic

Gain values lower than 1 will have no effect on delay filter.