# DATORRO'S NATURAL REVERBERATOR

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## ABSTRACT

Nowadays, we can find several types of reverberator music effects. The most popular brands are Strymon with the BlueSky reverberator; Boss with the RV-5 Reverb; DigiTech with the DigiVerb and HardWire with the RV-7.

Spring, plate and room are the types of algorithms that a user can choose to add reverb to the signal by means of a switch.

By using knobs the user can set the decay time of the reverberated signal, control the balance of the Dry and Wet signal, and filter some frequencies so they can decay more quickly than the others as well as set the starting time of the early reflections commonly known as Pre-Delay.

## 1. INTRODUCTION

Reverberation can be divided in the frequency spectrum in three sections: direct sound, early reflections and echo density also named late reverberation; Figure 1.

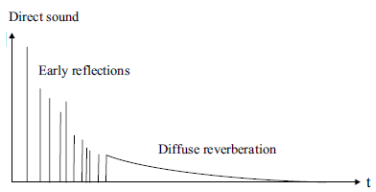



Figure 1. *Direct sound, early reflections and echo density in the frequency spectrum.*

The reverberation time is defined by W. C. Sabine as the time interval in which reverberation level drops by 60dB [1].

Direct sound is the sound that travels from the source to the listener. Early reflections are reflections that arrive between 50 and 80 ms after the direct signal. The time between direct sound and early reflections is named Pre-Delay or Initial Time Gap.

Considering sound pressure, the particle velocity of air is in phase with pressure for long distances and in quadrature for short distances. If the source is at long distance we consider the condition of propagation of plane waves. In short distances the air velocity and pressure are in quadrature and therefore energy doesn't radiate. Therefore, we will focus on plane waves.

The harmonic series of plane waves can be reproduced by a comb filter refer to Figure 2, by means of a feedback delay line.

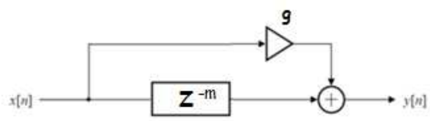



Figure 2. *Simple comb filter.*

The z transform of the comb filter is

$$H(z) = \frac{z^{-m}}{(1 - g^{z-m})}$$

In Dattorro's algorithm the input signal flows through a pre-delay followed by a low pass filter which defines the bandwidth. After this, four cascade all pass filters lead to a buildup of the echo density. Finally the signal enters a structure called 'Tank' where it recirculates indefinitely. The 'Tank' consists of two blocks which are cross-couple. Every block contains two all pass filters, two delays, a low pass and decay and damping coefficients for attenuation.

The purpose of the four all pass filters are to decorrelate the incoming signal before reaches the Tank [2].

The four decay coefficients determine the rate of the decay and the damping helps to decay certain frequencies more than others.

The z transform of the Delay is:

$$H(z) = z^{-k}$$

The z transform of the Lowpass Filter (Figure 3) is:

$$H(z) = \frac{1 - g}{1 - gz^{-1}}$$

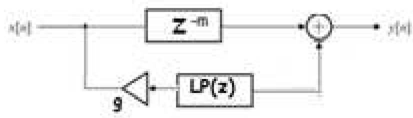



Figure 3. *Low pass filter inserted on the feedback of a comb filter*

The z transform of the Allpass Filter (Figure 4) is:

$$H(z) = \frac{g + z^{-k}}{1 + gz^{-k}}$$

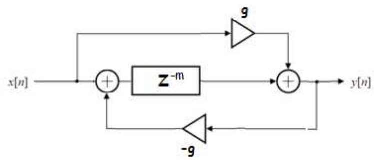



Figure 4. *All pass filter*

## 2. DESCRIPTION

The algorithm that adds to the signal the most natural reverberation sound is Dattorro's algorithm. This type of algorithm doesn't add color to the signal as the others does (spring and plate).

To give the user a more simple and natural way to add reverberation to the signal we get rid of the colour that spring and plate reverb add and the unnatural early reflections.

To give the user the freedom to manipulate the desire sound of a natural reverberation sound we keep the most essential knobs:

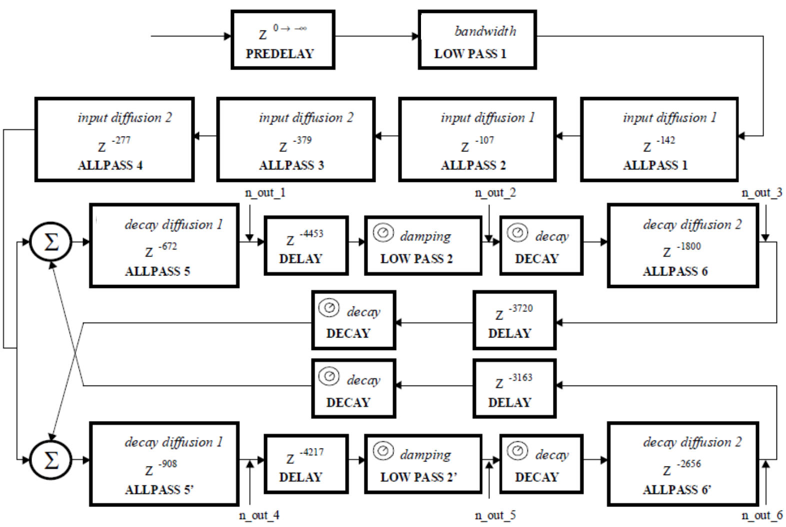
- Decay
- Mix
- Damping

Decay controls the decay time of the reverberated signal

Mix varies the balance of the Wet and Dry signal

Damping causes the High Frequencies to decay more quickly than the Low Frequencies.

## 3. METHOD




1. *By means of Software:*

First step, the initialization of all the parameters is made: decay_diffusion_1, decay_diffusion_2, input_diffusion_1, input_diffusion_2, bandwidth and delays.

Optimum diffusion for all-pass filters lies somewhere in a region closer to|0.5|. The preset values were determined by trial and error [2].

Second step, according to Fernando a Beltran [3], the transfer function of two or more filters in series is the product of the individual transfer functions. Therefore, he used the Matlab function 'conv' to compute the polynomial product of the numerators and the denominators of the transfer functions within the 'Tank'; for the numerators and denominators for both, left and right channel, as part of the initialization of the program.

The third step, based on the aforementioned is the calculation of the 6 first network nodes; the signal is calculated for the first node 'PREDELAY'; then the second node 'LOW PASS 1'; later the signal passes through node 3 '(All PASS 1), node 4 (All PASS 2), node 5 (All PASS 3) and node 6 (All PASS 4).

Just before entering into the 'Tank' the signal is filtered with the 'b' and the 'a' coefficients obtained from the third step.

The signal circulates one time in the 'Tank' calculating progressively the data for the left and the right channel of the stereo output.

Node 7 is 'All PASS 5'; node 8 is 'DELAY' ; node 9 is 'LOW PASS 2' which is modified by both knobs 'DECAY' & 'DAMPING'; node 10 is 'ALL PASS 6'; node 11 is 'DELAY'; node 12 is 'ALL PASS 5'' modified by the knob 'DECAY'; node 13 is 'DELAY'; node 14 is 'LOW PASS 2'' modified by both knobs 'DECAY' & 'DAMPING' and node 15 is 'ALL PASS 6''.

2. *By means of Hardware*

The knob 'MIX' will add the signals by an analog HW device.

## 4. RESULTS

So far, the code was developed in Matlab as well as the balance between the Wet and Dry signal that is meant to be made by Hardware.

The code can be found in reference [4] and the mix of the Wet and Dry signal in reference [5].

The final Software is intended to be programmed on a DSP.

## 5. FINAL PRODUCT

The final product will be a portable electronic device. Refer to Annex A.

## 6. AKNOWLEDGMENTS


I want to thank to my mother for gave me her unconditional support, as well as to all my friends from all over the world, especially to my Nora and Micha.

## 7. REFERENCES


[1] Artificial Reverberation: Comparing algorithms by using monaural analysis tools. Presented at the 121ts convention, 2006 October 5-8 San Francisco, CA, USA.

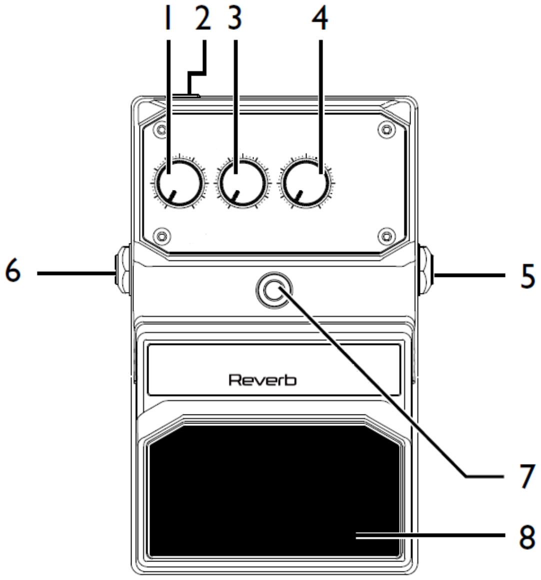
[2] J.Dattorro. “Effect Design - Part 1: Reverberator and other Filters,”J.Audio Eng. Soc. 45(19):660-684, 1997 September. Matlab Implementation of Reverberation Algorithms.

[3] Fernando A. Beltrán, José R. Beltrán, Nicolás Holzem, Adrian Gogu. Department of Electronic Engineering and Communications. University of Zaragoza (Spain). http://ftp.ipv6.funet.fi/pub/sci/audio/dafx/1999/www.tele.ntnu.no/akustikk/meetings/DAFx99/beltran.pdf

[4] LABORATORY REPORT 1: REVERBERATION
J. Dario R. Barbosa 311275087 (SID); Digital Audio Systems, DESC9115, Semester 1 2012; Graduate Program in Audio and Acoustics; Faculty of Architecture, Design and Planning, The University of Sydney

[5] DESC9115 2011 LAB REPORT 2; DIGITAL AUDIO SYSTEMS; Instructor: William L. Martens; Student: J. Dario R. Barbosa; SID: 311275087; Date: Monday, May 7, 2012.

**Annex A**




**1. Decay Knob**

**2. DC Adapter Jack**.

**3. Mix Knob**

**4. Damping Knob**

**5. Input 1 (Mono)**

**6. Output 1 (Mono)**

**7. Indicator LED - Lights when the effect is turned on.**

**8. Foot Switch - Turns the effect on and off.**