A VIRTUAL TUBE DELAY EFFECT

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ABSTRACT

A virtual tube delay effect based on the real-time simulation of acoustic wave propagation in a garden hose is presented. The paper describes the acoustic measurements conducted and the analysis of the sound propagation in long narrow tubes. The obtained impulse responses are used to design delay lines and digital filters, which simulate the propagation delay, losses, and reflections from the end of the tube which may be open, closed, or acoustically attenuated. A study on the reflection caused by a finite-length tube is described. The resulting system consists of a digital waveguide model and produces delay effects having a realistic low-pass filtering. A stereo delay effect plugin in PURE DATA¹ has been implemented and it is described here.

1. INTRODUCTION

Analog and digital delays are at the basis of several audio effects, including vibrato, flanger, chorus, echo, as well as spatial effects such as reverberation [1]. This paper investigates in particular the delay effects produced by a long narrow tube and presents a digital model of sound propagation in such a medium, including time delay, propagation losses, and end-reflections.

The first analog audio effect based on a narrow long tube was proposed in 1960 [2]. Olson and Bleazey presented a synthetic reverberator built with a tube, a loudspeaker, transducers, and a microphone delay unit in combination with a feedback system. A horn-loudspeaker coupled to a tube with three microphones located at different distances realized three different delays that, in conjunction with a positive feedback system, provided time spaced reverberation. The virtual tube delay effect plugin in PURE DATA¹ has been implemented and it is described here.

With low-order digital infinite impulse-response (IIR) filters based on the resistance and capacitance values of the filters [4]. Other delay-based system examples are the spring [5, 6, 7] and plate reverbs [8, 9]. Spring reverberation is an electromechanical effect based on metal springs [10]. A first simulation by measuring the response of a real spring reverberation unit and by using digital waveguide methods was proposed in [5]. Two other methods involving a finite difference scheme [6], and by using delay-network reverberation techniques [7] were later presented. Instead, plate reverberation uses steel plates under tension [11], and it can be simulated with finite difference methods [8] and by using a hybrid structure consisting of a short convolution section and a feedback delay network [9].

The reverberation and coloration caused by a long tube has also been shown to be a robust cue for the distance perception of a sound source [12]. In a recent study, a digital-waveguide-mesh model of a small tubular shape has been used to simulate distance in a virtual environment [13]. The virtual tube delay effect presented in this paper can also be employed for this application.

Digital waveguide modeling for wave propagation in cylindrical and conical instruments is often used [14, 15, 16]. A technique for estimating a waveguide model of wind instrument from acoustic tube measurements was also presented in [17].

The rest of the paper is organized as follows. An overview of the performed measurements is given in Sec. 2, while their analysis is presented in Sec. 3. Sections 4 and 5 describe the approach used to design the digital propagation and reflections filters, which are then compared to the measurements in Sec. 6 in order to provide an objective evaluation of the results. Section 7 presents and discusses the implementation of a real-time plugin in the PURE DATA environment. Finally, Sec. 8 concludes this paper.

Supplementary materials including the plugin, the externals for MAC OS X and LINUX, the source C++ file, and some dry sounds are available for download at https://github.com/RiccardoVib/VIRTUAL_TUBE_DELAY-EFFECT/.

2. ACOUSTIC TUBE DELAY MEASUREMENT

Three different tubes were used with an internal diameter of 1.2, 1.9 and 2.5 cm, respectively. The first tube was 8.8 m long and the other ones 25 m. The tube responses were measured with a logarithmic sine sweep that was played back to the tube with a full range loudspeaker. Figure 1 shows the equipment and the setup of the measurements.

The measurements were conducted in an anechoic chamber and in two modalities: closed end and open end. The goal of the first modality was to obtain a clean impulse response caused by propagation and losses without any reflections. Polyurethane and a metal plate were used to absorb and block reflections from the
tube end. The measurements with the open end were performed by using the acoustic pulse reflectometry technique [18] and required further analysis of the reflection behavior, as the impulse responses contained clearly observable repeating reflections. The polyurethane and the metal plate were chosen based on initial experiments to minimize the reflections from the end.

Ten holes were drilled 1 m apart along the length of the tube starting 2.5 cm from the loudspeaker end of the tube. Multiple measurements were made, recording the response of one hole at a time with a miniature microphone while blocking the others with moldable plastic material in order to avoid a “flute finger-hole effect” in the recordings. In addition, in order to record the cleanest possible impulse responses, the measurements were taken from hole positions drilled up to 10 m from the loudspeaker end of the tube to ensure at least 15 m in length to the opposite end (and a round-trip travel distance of 30 m before returning to the microphone). An exception was made with the 1.2 cm diameter tube, since it was only 8.8 m long.

The impulse response of the system was computed using Farina’s method, convolving the recorded signal with the time-inverted logarithmic sweep [19]. The input signal was 3 s long and with an amplitude of $-41$ dB, chosen after several experiments in order to find a trade-off between the signal-to-noise ratio and the harmonic distortion. The average SNR in the measurements ranged from 50 dB (for the narrowest tube) up to 40 dB (in the largest one).

Finally, the measurements were performed with a sample rate of 44.1 kHz.

3. TUBE DELAY ANALYSIS

Figure 2 shows two example impulse responses collected in the open-end mode. The main spike of the impulse response followed by some ripple, identified with circles, and reflections can be seen. The ripple is due to the holes along the tube. The holes could not be filled completely, and the resulting cavities created small reflections.

Due to the finite length of the tube, the microphone recordings contain reflections from both ends. The waves propagating through the tube are reflected at the open end and, coming back, they are reflected again from the loudspeaker. Reflections appear in pairs repeated in time and progressively attenuated along the response. The location of the impulses can also be seen to differ between the two measurements in Fig. 2 due to the increased distance of the microphone from the loudspeaker.

3.1. Impulse Response Analysis

The measured responses were windowed in time to remove harmonic distortion components and unwanted reflections. A processed impulse response is presented in Fig. 3. The frequency response exhibits losses in the high end of the spectrum caused by propagation losses through the tube. There are also some losses in the low frequencies caused by the windowing. Significant attenuations of 20 dB or more appear above about 300 Hz.

As expected, spectral analysis of the windowed responses exhibits highly attenuated behavior at very high frequencies, as seen in the example in Fig. 3. This can be caused in part by the effect of non-planar wave propagation above the cutoff of planar waves. The behavior of the spectrum in the extreme high end is very noisy and, thus, unreliable.

The group delay was also computed. It showed an approximately flat response, indicating no time delay between the various sinusoidal components of the signal. This suggests that a delay line is suitable for simulating the propagation delay.

Figure 4 (left) shows the impulse responses recorded at three different holes. The time delay and the propagation loss can be
In addition, our measurements show that energy losses at high frequencies depend on the tube diameter. This behavior can be observed here. Their corresponding frequency contents are shown in Fig. 4 (right), and they reveal an increase of the attenuation with the increasing distance traveled and more significant losses at high frequencies when compared to low frequencies.

In addition, our measurements show that energy losses at high frequencies depend on the tube diameter. This behavior can be observed here. Their corresponding frequency contents are shown in Fig. 4 (right), and they reveal an increase of the attenuation with the increasing distance traveled and more significant losses at high frequencies when compared to low frequencies.

The farthest hole is at 15.75 m from the loudspeaker. The negative reflection and the positive one can be clearly seen. The gap between reflections depends on the position of the microphone which recorded them. The positive one can be clearly seen. The gap between reflections depends on the position of the microphone which recorded them.

The reflections were windowed as well. The analysis shows that energy exhibits losses in the high end of the spectrum and, instead, it is concentrated in the low frequencies. Figure 6 shows the windowed reflection result at the tube end together with its spectrum. From 300 Hz up to 1.5 kHz the spectrum exhibits a steep slope and above that extreme low energy values. The inverted pressure pulse due the open end can also be noticed.

Figure 2 shows the behavior of the reflections at the closest and the farthest hole to the loudspeaker. The negative reflection and the positive one can be clearly seen. The gap between reflections depends on the position of the microphone which recorded them. The farthest hole is 9.25 m from the loudspeaker and 15.75 m from the open end, which means a longer distance for the reflections to meet the microphone.

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The reflection effect of the open end. The distance traveled by the reflection pulse had traveled. This simulation could be compared with the reflection captured by the microphone (dash-dot line), and the simulated spectrum as it should be without the open-end effect (solid line): 1.2-cm (top), 1.9-cm (middle) and 2.5-cm (bottom) diameter tubes.

Increasing diameters result in a steeper shape, but with smaller attenuation. The responses below 300 Hz, despite some oscillations, are very similar to each other near 0 dB. Attenuation is noticeable above 300 Hz and becomes more significant around 1 kHz.

Since the first modes of the tubes are at 8054 Hz, 10598, and 16780 Hz, the results above these frequencies are unreliable. For this reason, the responses above these frequencies were not considered, and a continuous slope for the frequencies larger than 10 kHz in the design of the filters was taken.

Based on the above considerations, the spectrum can be assumed to have a low-pass shape. Increasing the tube diameter decreases the spectral slope and increases the cutoff frequency.

4.1. Reflections from the End of the Tube

In order to understand the effect of the open end on the responses, a different approach was chosen. Using the acquired information, the impulse response measured at the farthest hole from the loudspeaker was filtered with the filter approximating an appropriate power of the 1 m segment shape of Fig. 7. The filter design procedure will be described in Sec. 5. The aim here was to simulate the losses of the same distance that the reflected pulse had traveled. This simulation could be compared with the reflection, separating the reflection effect of the open end. The distance traveled by the reflection was computed and used to build the filter, accounting for the approximation error which becomes significant for long distances.

Figure 8 shows the spectrum of the reflection captured by the microphone and the simulated spectrum as it should be without the open-end effect. A slight attenuation can be seen below 100 Hz, and a stronger one up to 1 kHz. Since the impulse travels along the whole tube before reaching the open end, it has very low energy above 3 kHz and the recorded reflection is superimposed by the noise. When the impulse crosses the boundary at open end, the pressure wave hits the outside air, at atmospheric pressure, creating a compression wave heading back down the tube with some energy left.

Using the filter designed for the tube model, the effect of the reflection \( R \) due the open end was obtained:

\[
R = \frac{H_{\text{ref}}(f)}{H_{\text{sim}}(f)}, \tag{2}
\]

where \( H_{\text{sim}}(f) \) is the spectrum of the response without the open end effect simulated with the approach described above using the same distance traveled by the corresponding windowed reflection \( H_{\text{ref}}(f) \). This allows for the estimation of how the reflection affects the spectrum. Equation (2) was estimated for each measure where the reflections were isolated enough and could be windowed. Finally, the average for each tube size was computed. The shapes shown in Fig. 9 summarize the results.

The results show that the attenuation depends on the diameter of the tube, starting with a low value increasing above 100 Hz. The attenuation becomes smaller at higher frequencies because of the noise level.

5. FILTER DESIGN

This section describes the design of the filters simulating the sound propagation through the tube and the reflection effect by the open end.
end. For each of these effects, the average filters previously computed and summarized in Figs. 7 and 9 were used as target shapes to be approximated with low-order filters. Then, a unique form to interpolate between the different diameters values was found.

### 5.1. Propagation Filter

Given the simple shapes of these filters (see Fig. 7), attempts were made to find a low-order filter simulating their behavior. Keeping the three averages as targets, three parametric filters were computed, approximating the shape in order to minimize audible errors.

A cascade of two high-shelving filters and one low-pass filter was built, resulting in a 5th-order parametric filter. The high-shelving filters were used to approximate the shape from 300 Hz to 3 kHz, while the low-pass filter was needed to cut the high end of the spectrum.

Since the three target shapes behave very similarly at low frequencies, the filters have the same behavior until 300 Hz with a slight attenuation depending on the diameter of the tube. The significant variations are in the range above 1 kHz, where different attenuations and cut-offs can be seen. The cut-off frequencies for the three target shapes are 4062, 5950, and 7015 Hz, respectively.

Figure 10 shows the different filters designed for the three diameter tubes to be compared with those in Fig. 7. With these low-order filters, a tube with arbitrary length can be simulated. Moreover, interpolating between the three filters allows to simulate different diameters sizes.

Since a cascade is an inefficient approach to produce tubes longer than 1 m, an approximation was found. Starting from the filter computed for the 1.2 cm tube, all the parameters of the three basic filters composing it were gradually varied in a linear way to achieve an approximated filter for longer lengths. A cascade of two 1st-order low-pass filter replaced the simple 1st-order one, resulting in a 6th-order parametric filter. A good approximation up to 30 m (which is sufficient for the purpose of the audio effect) was obtained with an error smaller than 0.6 dB. In addition, with this method a better accuracy creating the tube can be achieved. Instead of 1 m as the incremental step, a finer control, like 1 cm, can be implemented. Figure 11 shows the approximation for 30 m. The designed filter follows accurately the general shape except for a critical range between 300 Hz and 1 kHz. In the case of 30 m tube, the maximum error is 0.57 dB.

After obtaining an accurate approximation of frequency attenuations due to propagation in the tube, the final filter was obtained by using a delay line that simulates the propagation delay and is connected in series with the previously discussed filter.

Figure 12 shows the three parametric filters in cascade and the delay line composing the system. The system can be described mathematically as follows:

\[
H_{\text{tube}}(z) = g z^{-M} H_{\text{HS1}}(z) H_{\text{HS2}}(z) H_{\text{LP}}(z),
\]

where \( g \) is a gain factor, \( z^{-M} \) is the delay line of \( M \) samples, \( H_{\text{HS1}}(z) \) and \( H_{\text{HS2}}(z) \) are 2nd-order IIR high-shelving filters, and \( H_{\text{LP}}(z) \) is a 1st-order IIR low-pass filter.

The coefficients of the high-shelving and low-pass filters were computed with the usual formulas of the 1st-order and 2nd-order filters [20]. Three different IIR filters were designed, one for each tube diameter (1.2, 1.9, 2.5 cm), giving the possibility to approximate the different behaviors by controlling the shape with the cut-off frequencies of the designed IIR digital filter.

In order to control the filter behavior as a function of the diameter of the simulated tube, the cut-off frequencies of all the filters and the gain factor \( g \) are linearly varied while the gains (dB) and the quality factors of the two high-shelving filters are kept fixed. Table 1 reports these latter values while Table 2 summarizes the filter cut-off frequencies and the gain factor for each tube diameter. Starting from these values, an interpolation was made with a granularity of 1 mm.
Table 1: Propagation filter: gain and quality factor values for the two high-shelving filters.

<table>
<thead>
<tr>
<th>Type of filter</th>
<th>G[dB]</th>
<th>Q</th>
</tr>
</thead>
<tbody>
<tr>
<td>HS1</td>
<td>-1</td>
<td>0.65</td>
</tr>
<tr>
<td>HS2</td>
<td>-0.9</td>
<td>0.5</td>
</tr>
</tbody>
</table>

Table 2: Propagation filter: cut-off frequencies of low-pass and high-shelving filters and overall gain for the three tube diameters.

<table>
<thead>
<tr>
<th>Type of filter</th>
<th>( f_{HS1} ) [Hz]</th>
<th>( f_{HS2} ) [Hz]</th>
<th>( f_{LP} ) [Hz]</th>
<th>g</th>
</tr>
</thead>
<tbody>
<tr>
<td>1.2 cm</td>
<td>1200</td>
<td>1500</td>
<td>9500</td>
<td>0.85</td>
</tr>
<tr>
<td>1.9 cm</td>
<td>900</td>
<td>7000</td>
<td>10200</td>
<td>0.87</td>
</tr>
<tr>
<td>2.5 cm</td>
<td>900</td>
<td>7000</td>
<td>11000</td>
<td>0.90</td>
</tr>
</tbody>
</table>

Figure 13: Block diagram of the reflection simulation.

5.2. Reflections

The block scheme in Fig. 13 shows the approach used to simulate the reflection. The delayed input is first filtered with the filter \( H_{ref}(z) \) that approximates the losses given by the open end reflection, and the output is fed to the filter \( H_{tube}(z) \) that simulates the losses caused by sound propagation in the tube. The computed reflection is finally added to the delayed sound resulting from unperturbed propagation in the tube.

The measured reflections have extremely low values in the high end of the spectrum (above 3 kHz) because of the long distance traveled. The simulation produces lower values in the high frequency region than the measured values. The extremely low values superimposed by noise produce unreliable results in this region of the spectrum. Since a steeper shape in the high frequency side due to high frequencies losses were expected, an approximation of the spectrum. Since a steeper shape in the high frequency side due to high frequencies losses were expected, an approximation of the differences found with a continuous slope was done.

In order to approximate \( H_{ref}(z) \), a cascade of a 2\textsuperscript{nd}-order high-shelving filter and a 1\textsuperscript{st}-order low-pass was chosen. Similarly to the propagation filter, by controlling the quality factors, the gains, and the cut-off frequencies, we were able to perform a linear interpolation between different diameters. An additional gain factor \( g_{ult} \) was introduced to control the scale for the different sizes. Table 3 summarizes the parameters values of the different filters.

6. COMPARISON

In this section, a comparison between the designed filters and the measurements is performed. The accuracy of the design is discussed, presenting the maximum approximation error in the frequency range of interest. Considering that the frequencies above 10 kHz are unreliable, as discussed in Sec. 4, the comparison refers the range between 20 Hz and 10 kHz.

Table 3: Reflection filter: parameters of the low-pass and high-shelving filter and overall gain for the three tube diameters.

<table>
<thead>
<tr>
<th>Type of filter</th>
<th>1.2 cm</th>
<th>1.9 cm</th>
<th>2.5 cm</th>
</tr>
</thead>
<tbody>
<tr>
<td>( f_{LP} ) [Hz]</td>
<td>100</td>
<td>250</td>
<td>600</td>
</tr>
<tr>
<td>( f_{HS} ) [Hz]</td>
<td>500</td>
<td>225</td>
<td>160</td>
</tr>
<tr>
<td>( G_{HS} ) [dB]</td>
<td>-14</td>
<td>-14</td>
<td>-10</td>
</tr>
<tr>
<td>( Q_{HS} )</td>
<td>0.35</td>
<td>0.4</td>
<td>0.60</td>
</tr>
<tr>
<td>( g_{ult} )</td>
<td>0.4225</td>
<td>0.5180</td>
<td>0.9</td>
</tr>
</tbody>
</table>

Figure 14 shows the three designed propagation filters compared with the results obtained from the measurements. The filter approximating the 1.2-cm tube has a maximum error of 0.97 dB, which is mainly due to the shelf filter having a flat magnitude response at low frequencies instead of the declining slope of the measured response as shown in the top of Fig. 14. This way, a good approximation at high frequencies is obtained, which is considered to be more important that the response below 100 Hz.

The 1.9-cm filter presents a maximum error of 0.5 dB in the lowest part of the frequency range. The fit becomes very accurate at higher frequencies as seen in Fig. 14 (middle). The error is 0.31 dB at 60 Hz and decreases close to zero at frequencies above 100 Hz.

The third filter is shown in Fig. 14 (bottom) that, with the exception of an anomaly at about 1900 Hz, also fits the target shape with good accuracy. It has a maximum error of 0.5 dB at 0.184 Hz, and an error smaller than 0.3 dB in the rest of the frequency range.

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6.2. Reflection Filter

Figure 15 shows the difference between the three designed reflection filters and the simulation results. In this case, the range between 20 and 500 Hz is significant for the comparison as discussed in Sec. 4.1.

The filter for the 1.2-cm diameter tube, shown in the top of
corresponding to a temperature of $m/s$.

The diameter of the two tubes is always the same. The length of each tube can be set by the user and determines the desired delay. The plugin offers the possibility to create virtual tubes up to 30 m long in default mode, and 40 m long tubes in the reflection mode. These maximums correspond to a delay of 87 ms and a reflection coming after 29 ms. Figure 17 shows a screenshot of the plugin implemented in PURE DATA.

7. IMPLEMENTATION

The implementation was written in C++ as an external library for PURE DATA, an open-source real-time environment for audio processing. The stereo plugin, working at sample rate 44.1 kHz, simulates the wave propagation in a narrow tube and produces associated audio effects. It creates two virtual tubes, one for each channel. The diameter of the two tubes is always the same. The length of each tube can be set by the user and determines the desired delay in milliseconds. The speed of sound is assumed to be 345 m/s corresponding to a temperature of 23°C.

In addition, it is possible to control the volume of the delayed sound and the ratio of the dry and the wet signals in the output. The filter simulates the tube length for each virtual tube. A gain parameter was measured and analyzed. Studying and elaborating the recorded tube responses, a virtual tube model was developed and a digital IIR filter controlling the length and the diameter of the virtual tube was designed with a negligible error. From the analysis of the measurements, a parametric filter was designed in which the tube diameter and length can be continuously varied. Because of the simplicity of the magnitude response shapes, a cascade of filters and a low-pass filter was sufficient for approximating the behavior correctly. In addition, an analysis on the reflection due to the open end of the tube was conducted, and a filter approximating it was added in the model. Finally, a stereo delay effect plugin in PURE DATA was presented describing the design specifications.

8. CONCLUSION

A simulation of a tube delay effect was proposed in this paper. Acoustic wave propagation in garden hoses of three different diameter was measured and analyzed. Studying and elaborating the recorded tube responses, a virtual tube model was developed and a digital IIR filter controlling the length and the diameter of the virtual tube was designed with a negligible error. From the analysis of the measurements, a parametric filter was designed in which the tube diameter and length can be continuously varied. Because of the simplicity of the magnitude response shapes, a cascade of filters and a low-pass filter was sufficient for approximating the behavior correctly. In addition, an analysis on the reflection due to the open end of the tube was conducted, and a filter approximating it was added in the model. Finally, a stereo delay effect plugin in PURE DATA was presented describing the design specifications.

9. ACKNOWLEDGMENT

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10. REFERENCES

Figure 17: The virtual tube delay effect plugin in PURE DATA.


