Designing Multi-Channel Reverberators

Introduction

We present here a methodology for the design of digital reverberators. This approach to designing digital reverberators has been used with success in a number of previous works. In this approach, the reverbator is modeled as a delay line with feedback. The feedback is chosen to give a desired impulse response.

Overview

In designing reverberators, we must attempt to account for both the early part of the reverberation and the late part of the reverberation. It is very difficult to model exactly the long-term behavior of sound in an acoustic system. However, the early part of the reverberation can be approximated by a simple delay line model. The late part of the reverberation can be modeled by using a digital filter to simulate the early decay of the reverbation.

Model of the General Delay Network

A network of delays may be described as an instance of the network shown in (1), where system equation is usually written:

\[ y = - \alpha x + y_0 \]  (1)

where \( y \) is the output, \( x \) is the input, \( \alpha \) is the delay, and \( y_0 \) is the initial condition.

Both of these methods are used in our designs. First, we create a digital acoustic network where each delay unit is assigned to a speaker. The output of a delay unit feeds a speaker.

\[ T = CT + CV \]  (2)

where \( T \) is a vector whose components are the


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The form of the network is shown in Fig. 2. The network is stable if the transfer function at each node is a stable transfer function. In general, we consider the stability and time constant of the closed loop network shown. Additional elements with zero time constant are useful here for eliminating steady state as well as for controlling the direction and magnitude of the transient response.
The simplest example of a linear network is the matching which gives perfect equality between two networks. A linear network can be described by a matrix equation:

\[ \mathbf{U} = \mathbf{A} \mathbf{X} \]

for any \( \mathbf{U} \). Since the products of two linear networks in a given order, the responses of the above forms can be applied repeatedly to any number of pairs of devices, then passing more complex networks.

Equation (6) may now be rewritten as

\[ \mathbf{X} = \mathbf{A}^{-1} \mathbf{U} \]

or in an equivalent form for a network. Since \( \mathbf{A} \) is a square matrix, the determinant is zero, and the matrix is singular. Therefore, a solution may be found by multiplying \( \mathbf{U} \) by the inverse of \( \mathbf{A} \).

\[ \mathbf{X} = \mathbf{A}^{-1} \mathbf{U} \]

A Four-Channel Network

For some of the examples of this discussion, we will consider the electrical delay network given in Fig. 3, which is made up of a four-channel path-echo network where the echoes are summed at the output of a single. The feedback equations for this system are:

\[ \mathbf{U} = \begin{bmatrix} 1 & 1 & 0 & 0 \\ 0 & 0 & 1 & 0 \\ 0 & 0 & 0 & 1 \\ 0 & 1 & 0 & 0 \end{bmatrix} \]

where

\[ \mathbf{U} = \begin{bmatrix} U_1 \\ U_2 \\ U_3 \\ U_4 \end{bmatrix} \]

A source signal may be introduced as one input to this network, such as the red of sound of the delay signal. An advantage of this delay network is that the system can be linearized by exploiting the fact that the slope for the input signal is constant and the first and second channels of the delay are equally spaced in time. This means that the second channel is an equal delay of the input signal, the third channel is a delay twice that of the second channel, the fourth channel is a delay of the second channel, and the fifth channel is a delay of the third channel. This behavior is an additional input of source position and of the physical stimulus of the stimulated space.

Definition of Early Response

Much research has been done to determine the importance of the early responses in the perception of the overall timbre. The early responses are complex sinusoids or complex sinusoids plus a carrier signal. The carrier signal is usually much less noticeable than that of the equivalent number of each filter in parallel.
and
\[ v = \lambda / (2 \pi f) \]

where \( \lambda \) is the wavelength, \( f \) is the frequency, \( v \) is the speed of light in a vacuum, \( \pi \) is the constant pi, and \( 2 \) is the factor representing the two-way travel of the wave. \( v \) is the speed of light in a vacuum, which is approximately 3 x 10^8 m/s.

In an electrical system, the characteristic impedance of the line is determined by the ratio of voltage to current. This ratio is denoted by \( Z \) and is given by
\[ Z = \frac{V}{I} \]

The choice of values for \( Z \) in the above relationship is critical and should be carefully selected to ensure the system is stable and efficient. This is particularly important when dealing with transmission lines, where the impedance of the line is often different from the characteristic value of the source.

Absorption coefficients vary with the frequency and type of material, and can be calculated using the empirical formula
\[ a = 10^{-13} \frac{f^2}{\lambda^2} \]

where \( a \) is the absorption coefficient, \( f \) is the frequency, and \( \lambda \) is the wavelength. The formula is valid for frequencies below 1 MHz and wavelengths above 3000 m.

Long-Term Response

The choice of the delay length is critical to the performance of the system. A single guide, the length determines the (1) time between successive events, which is directly related to the system's output. The delay length is also influenced by the system's runtime, where longer delays give more accurate results in the long run.

Choosing the appropriate delay length in the system is crucial. However, this is a common practice in many systems. Schneider argues using longer operating times of 0.1-1.1 days (1972). Our largest delay length was typically above 10% to 50% of the longer delays, which may be due to the use of different systems or differences in the physical characteristics of the systems.

Another important part of the long-term response is the decay of the exponential effect of an absorption. To studies this effect, we measure absorption rates at the respective time delays. The absorption rates are then used to calculate the delay length of each delay (Nier 1978).

Additional Modifications

A richer and more realistic modeling must be achieved by predicting the system's characteristics. This is done by using a combination of the above techniques and empirical data. The model must be able to predict the system's behavior accurately.

In conclusion, the relationships and absorption rates ultimately lead to the system's output. The system's output is then used to determine the delay length of each delay.
Table 1. Finding the filter characteristics in a real wall reflection

<table>
<thead>
<tr>
<th>Domain of measurement</th>
<th>Frequency (Hz)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>120</td>
</tr>
<tr>
<td>Slope absorption coefficients for local surfaces from Russell [10]</td>
<td>0.02</td>
</tr>
<tr>
<td>Reflected amplitude increment/delay time</td>
<td>0.08</td>
</tr>
<tr>
<td>Filter, using absorption of 0.5</td>
<td>1</td>
</tr>
</tbody>
</table>

chosen to match that of the wall absorption is insufficient to model this behavior. The resulting factors are found in the used as the amplitude coefficients. As can be seen in the table, the reflective coefficients were lower than those of the filter. For example, at 4000 Hz the amplitude coefficient is 0.90, and the amplitude coefficient becomes 0.80 as before.

The temporal behavior of reflections from 510-
A 3.2 m long the 2.2 m wide and 3 ft high walls, with window openings larger than the dimensions of the filter, very little sound is reflected, and it is reduced in all dimensions. Based on window openings that the window is placed in front of such a window, the amplitude coefficient is 0.80 and the amplitude coefficient becomes 0.80 as before.

We may conveniently express this behavior into two ways. At a location on the wall of either type would be a window opening, it will reflect the incident sound, and a second sound, and so forth. If by the background reflecting rate, we choose the half power point of the highest lines at above 1/10,000, we can easily extended to allow the determination of sound sources in the structure.

Note that a similar problem exists in measuring the quality of the sound without considering the reverberation.

\[ \text{center_freq}(f) = \frac{V(f) \cdot \text{absorption}}{0.5 \cdot \text{center_freq}} \]

and it is a force: \[ 1 \over \sqrt{3} \]
The image contains a diagram and text discussing a topic. The diagram includes a figure with labels and a small text box that reads: "The smoother design contains several optimal benefits that affect the roughness and the overall performance of the design."

The text continues: "A smoother design can be produced using the present design methodology. The BOOM proposes targets to achieve the desired performance of a triangular array, a concept for reducing the roughness and improving the performance."
plotted during the day. This phenomenon is caused by the "cocktail" of a small number of paths in the frequency range that cause field enhancement. This is caused by the interaction of many different waves, and for every enhancement we have tried, and for every distance, it can happen, it is enough for the peak density to be so great that the beating is stronger than the direct term. As a result, the peak enhancement of the signal is a result of the beating of the signal and the direct term. However, we are unable to think of any daily or seasonal results to explain the result of our trial of the signal to the waves. We have an answer when the signal is clear, but a range of frequencies when the signal is a poor wave. A good step in the correlation design problem is to understand the scattering mechanism. We employ finite element analysis in the EEG program to infer to match it precisely. A computer simulation of the using relations of the early reflections is the problem we have been investigating recently. The problem is that the analyses are very free from assumptions and it should be possible to understand what is going on in the scattering mechanisms on a variety of waves with many different wave directions. We also found that variations of the early response can be used to the advantage of various signals, but to define the problem, we have a model of describing these relationships up to perfectly a scattered term. Some important results among these cases can be obtained by the use of the model of the frequency range of the brain in real space and help in choosing delay matrices for convoluted with many characteristics. We also find that the value of the early responses in the frequency range of the brain can be used to model the signal to the direct term. These could be used to define the frequency range of the brain in real space and help in choosing delay matrices for convoluted with many characteristics.

Conclusions

A comprehensive model for the design and representation of correlations developed in this paper presents some of the favorable properties of correlation design in the way we have outlined.

1. The correlation response is spatially scattered in the frequency range of the brain.
2. Several distinct input sources can be added without significantly distorting the brain in any significant way.
3. A significant portion of the early response characteristics can be modeled.
4. When a mean signal is dominant, there is an increase in computational efficiency.

The methods we have proposed are flexible enough to provide the design of multiple relative responses with an arbitrary number of wave directions.

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References


