DESC9115 2014 Written Review 2

Reverberation

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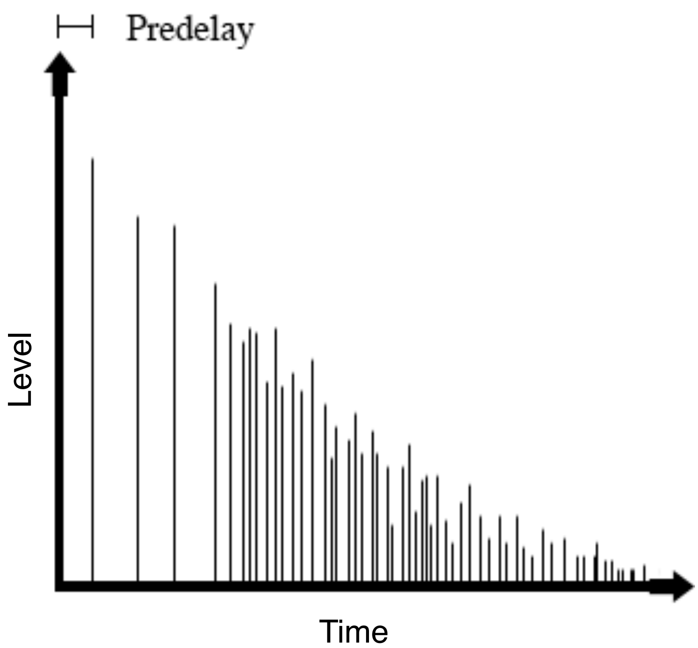
Reverberation

Reverberation is the reflections of a sound source which give a sense of space to a signal. Reverberation originated as a bi-product of the room in which music was played and its importance to musicality has varied over the ages. Over time people began to try and simulate this sound by using electronic reverberator units utilising DSP technology. The benefit of such a technology is that music can be recorded in a 'dead' room, with minimal reverberation time, and an artificial reverb may be applied to these recordings effectively recreating any desired space. There are two main approaches when designing an electronic reverb, artificial reverb and convolution based reverberation.

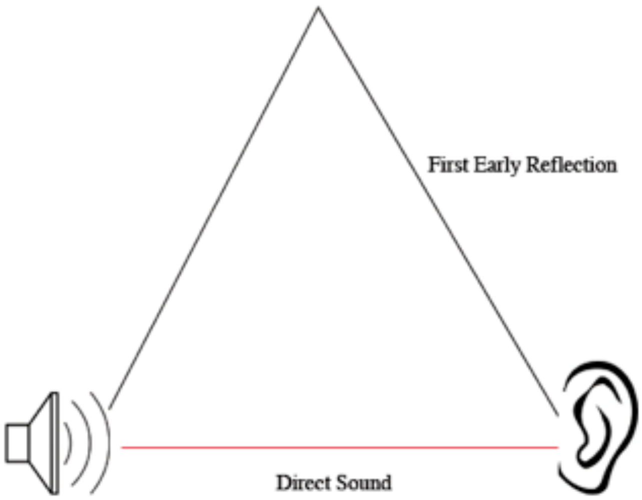
Reverberation can be divided into four main parameters these are; the initial dry signal, pre delay, early reflections and decay time. These parameters all contribute to a naturally sounding reverb and must be taken into account when designing a reverb.

Pre delay is the time between the initial dry signal and the onset of the early and late reflections. Sound on Sound (2000) suggest that pre-delay has an effect on the perceived position of the source within an artificial room. By decreasing pre delay the source is moved closer to the edges of the room and further away from the listening position. This parameter can be adjusted accordingly to help achieve the desired ambience. For example a vocal track can be made clearer and more intelligible by increasing the pre delay time.



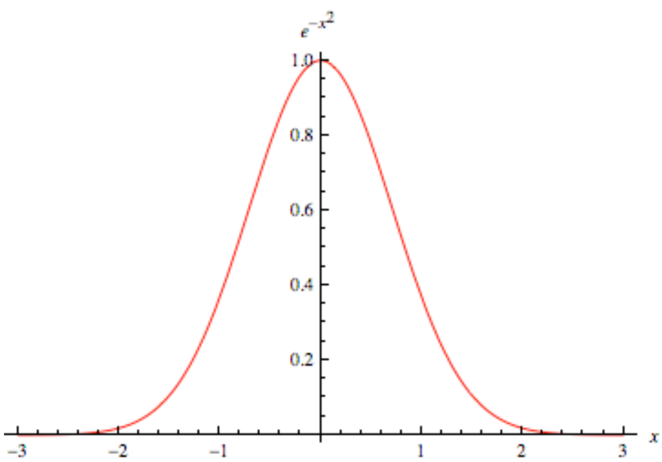
Pre delay

Early reflections occur after the initial sound has reflected off a surface one or two times. These early reflections give an indication to the size of the room. According to SAE (2002) early reflections become closer and quieter as they make more passes between the walls and the listener. Eventually these reflections are so close that they become part of the reverberant tail of the signal.



Early Reflections

One of the most important features of a reverberation is the decay time of the reverberant tail. This is caused by the reflections of a sound building up and becoming exponentially quieter as more and more reflections are heard. The rate of this buildup is proportional to the square root of the volume of the room Hass (2003). Jot (1997) shows that reverb time can be modeled as 'Gaussian exponentially decaying process'. It is important that there is an element of randomization to the decay as reverbs that use multiples as decay factors fall into rhythms which causes an unnatural sounding reverberant tail.



Gaussian Function

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Artificial Reverberation

It is impossible to realise a fully accurate digital reverberation simulation due to the complexity of natural impulse responses and the multi-faceted nature of the directionality of a naturally reverberated sound (Griesinger, 1989).

Schroeder made the first problem of the complexity of natural impulse responses much easier to simulate with his groundbreaking development in reverberation.Schroeder (1961) found that existing electronic reverbs suffered from two major problems. The amplitude frequency response of reverbs was not flat leading to colouration and the number of echoes per second for electronic reverb units was too low causing fluttering effects.

These two issues of coloration and echo density go hand in hand. When using a single delay line to produce reverb, the resulting impulse response is strongly coloured due to comb filtering effects (Griesinger, 1989). Another problem with using only a single delay line is that the echo density can not practically be raised to high enough levels to create a natural sounding reverb without flutter. In order to produce high echo density, many delay units could be added together, although the colouration imparted by such a set up would greatly affect the quality of the output sound. Shroeder (1961) realised this and came up with a solution to increase echo density without contributing to colouration of the signal.H is solution was to use an allpass filter to ensure that all frequencies are passed through the reverb unit without gain or loss. Such a filter may be employed by using a recursive comb (IIR) filter in combination with a non-recursive (FIR) comb filter in order to cancel out the colouration added by the IIR filter.

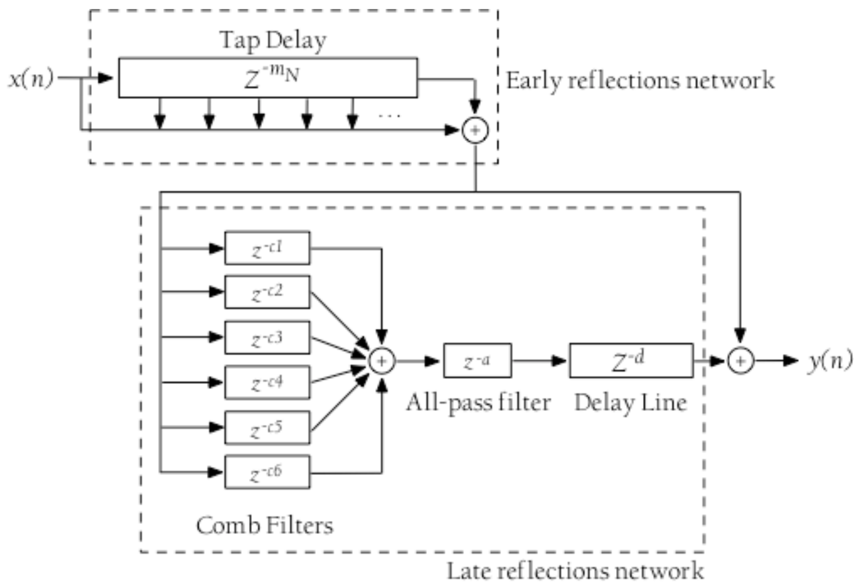

Diagram of Schroeder reverb from christianfloisand.wordpress.com

By using an all-pass filter the phase of the signal is being changed, this is particularly noticeable at high frequencies and this creates diffusion. A higher order allpass filter will have a greater shift in phase resulting in longer delays and a wider sounding reverb.

The second problem of directionality is much harder to tackle, however attempts have been made to simulate the directionality of reverberation using HRTF models. Adriaensen (2006) suggests that as well as providing information on the acoustic space, early reflections play a large role in determining the directionality of a sound. The correct pattern of early reflections can help to reinforce other localisation tools such as inter-aural time differences (ITD) and inter-aural level differences (ILD). One problem with using HRTF for modeling directionality is that this method is much more effective when heard through headphones rather than speakers. This is because it is difficult to account for the directionality imparted by the speakers themselves. Interaural crosstalk cancellation (ICC) can be used in conjunction with HRTF's in order to have them created by stereo speakers. A method for this is described by Kraemer (2001). HRTF's are also dependent on the physical attributes of the listener including ear canal length and shape, as well as head and torso shape.

Convolution

A more accurate simulation of the reverberation of a room can also be modeled by obtaining the impulse response of a room. An input signal is then convolved with this impulse response in order to apply the reverberation of that particular room onto the input signal (Griesinger, 1989). This process can replicate exactly the sound of a room at the position in which the impulse was recorded. While convolution is an effective way of replicating the sound of a room, there a some inherent problems with this type of reverberation. One major problem that Griesinger (1989) points out is that the impulse response will change as the position of the source changes within a room. This means that in order to accurately replicate the reverberation of a particular space a different impulse response would have to be applied to each instrument according to its position in the room.

$$x[n]*h[n]=y[n]$$

Convolution equation from Smith (1997) pg. 109

Conclusion

While there has been much research into the production of reverberation using DSP, there are still some major concerns in the design of reverberators. Schroeder (1961) and Griesinger (1989) addressed some of these problems that had a significant effect on the colouration of a signal and by using an all pass filter they ensured that unity gain was applied over all frequencies. While this area has been extensively researched, there is still some room to research into methods that will provide accurate directional information about reverberant signals in an artificial space. HRTFs can be used to model directionality, but are dependent on the listeners physical attributes and so can not be relied upon. Further research into this area could greatly improve the quality of reverberation and produce a more natural sounding reverb.

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