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ABSTRACT

The aim of this work is to delve into the concept of reverberant panning explored in the second laboratory report, for which a Matlab code was designed. In the aforementioned report, the requirement for a digital audio system that implemented reverberation as well as 3D panning through stereo channel output in the context of virtual simulation design was satisfied. Documentation provided to audio engineers regarding the function and explanation of the system was provided, however the scope of this review is to provide a broader understanding of the underlying digital audio principles, offering a clearer perspective of the basis of these two major spatial effects and the logic that governs them.

1. INTRODUCTION

In the context of sound design, three-dimensional audio-visual virtually simulated entertainment requires a highly realistic element of detail in order for it to successfully convince participants that this virtual experience was real. Despite the limitations of a dark room with seats, speakers, and a screen, and no other configuration, the auditory system can be the mystery tool in the creation of a sense of adventure, allowing one to transcend the physical and mental limitations of space and realism.

The trajectory of sound is one that takes many turns as soon as it is emitted from a sound source. It travels to the listener's ears and within a space. Direct sound is first heard, followed by early reflections of sound that have hit surrounding objects, and then late reflections coming from the whole room. The extent of the reflections (amount of reverberation) depends on factors such as size, volume and absorptive qualities of the room. Thus a sense of space is perceived.

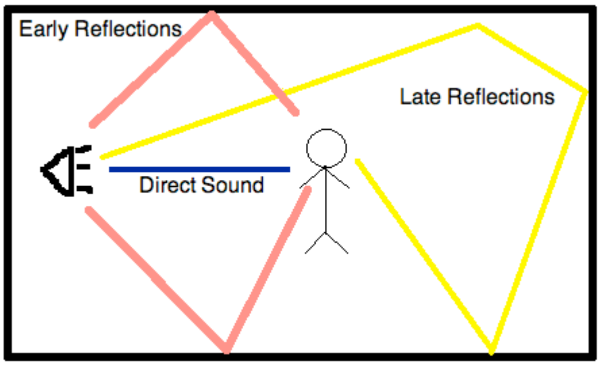
As with reverberation, sound also travels in relation to the human head. Ears on opposite sides of the head perceive sound and collect data for sound localisation. This means that information regarding a sound source's distance and direction in relation to one's position can be interpreted by the auditory system (Ballou 2008, pg. 1394).

REVERBERATION

1.1.Reverberation – Sound in three classes

Simulating the natural occurrences of sound in an acoustical environment presents the task of creating a virtual space for sound to reflect (Figure 1). In 1962, Manfred Schroeder conducted research in digital reverberation and founded a model for future studies in this area. It was discovered that sound reflections were classed in two divisions – those early and late. Early reflections typically reach the listener within 50 to 80 ms of the sound being emitted from the source, while late reflections arrive afterwards. The reason it takes reflections longer to reach the listener than direct sound is because they travel to a surface and reflect before being heard, meaning they have a greater distance to travel, whereas direct sound follows a straight path to our ears. This is why larger rooms generally have more reverberation.

Figure 1: Sound travels directly to the listener, followed by early, then late reflections.



1.2.Schroeder's Model

Initially, Schroeder designed a model for reverberation using a combination of all pass and comb filters. The input (x) passes through a set of four parallel comb filters before going through a pair of all-pass filters to create the output (y) (figure 2). Comb filters provide delayed versions of the input signal and combine it with the input to emulate the effect of reverberation. The all-pass filter has a flat frequency response and increases the echo density of the input, while simultaneously reducing the timbre of a signal. This creates an infinite impulse response (IIR) system whereby an exponential decay function is formed over time, causing the echoes to continue to decay but never to reach zero (Figure 3).

(Lee, Keun Sup 2010)(Bitzer, J et al 2006, pg. 6928).

Figure 2: Multiple parallel comb filters

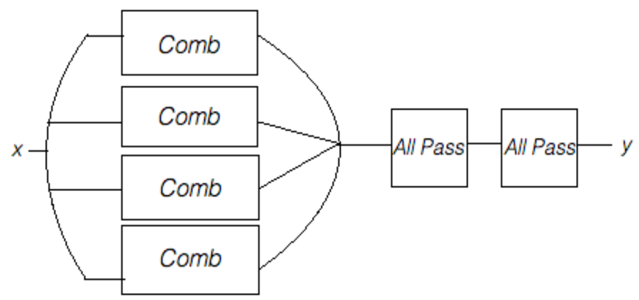
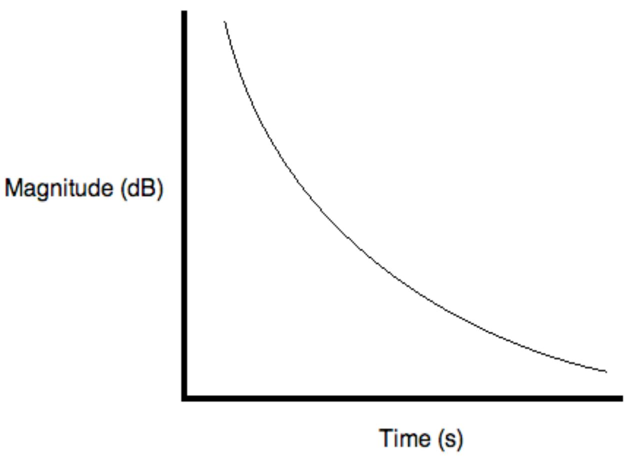


Figure 3: Exponential decay function of Schroeder's model based on an IIR(Infinite Impulse Response System).

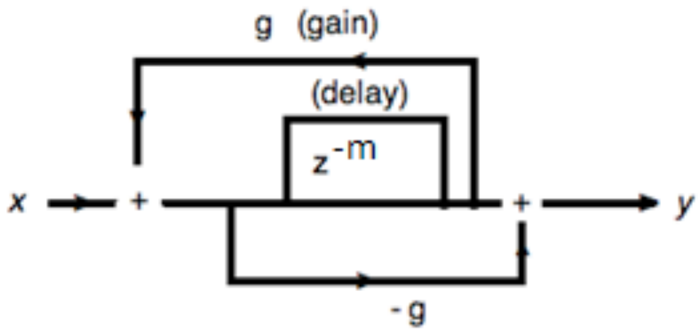


1.2.1. *Spatial Impression*

The use of a Schroeder function in creating artificial (virtual) reverb is of great significance, due to the fact that in his later model (1970), he accounts for early reflections through the implementation of a delay buffer (Figure 4)(Talbot-Smith 2013, ch. 4, pg. 4).

Early reflections are effective in enhancing spatial impression as concluded by Michael Barron, who conducted experiments in concert halls in the pursuit to find out the significance of early lateral reflections in spatial perception, and found that they reinforce the acoustics of a room while giving a perceived clarity to overall sound quality (as early reflections combine with direct sound because of the short time gap, giving a stronger sound), and creating a sense of presence in the acoustic environment. When creating a virtual space, making people believe they are in another space is challenging, but the Schroeder function in Revvvpan is inclusive of an initial delay, which is advantageous in creating spatial presence (Barron 1974). In a virtual simulation, the physical space will always be the same between the observer and the screen, but lateral reflections are known to refine sound localization and distance perception of a sound source beyond a virtual display's dimensions, illustrating an accurate virtual sound field (Adams 2008, pg. 32)(Zölzer 2002, pg. 175)

Figure 4: an initial delay creates the effect of early reflections typical of an acoustical environment



1.2.2. *Reverberation Time*

The Schroeder reverberator function incorporated in the *Revvvpan* digital audio system I created in my last laboratory report includes a value for reverberation time, RT60. This stems from the theory by Wallace Clement Sabine that sound takes a certain time to reduce by 60 dB. The decay time is affected by the absorption coefficient of materials in a room, its surface area and volume. (Figure 5). This value allows one to input any reverberation time without necessarily having to calculate a room's dimensions, as it is already built in the function's syntax (Long 2005, pg 305).

Figure 5

(V is room volume, S is surface area, and *a* is the absorption coefficient of the room)

$$RT_{60} = 0.16\ V/Sa$$

<u>*STEREO PANNING*</u>

1.2.3. *Stereo Panning – Stereo Output*

For a virtual world to be recreated, sound cannot be limited to one audio output channel. An important part of virtual simulation is the fact that it involves movement. A space is created, and by aid of sounds, mobile seating platforms, and 3D animations being projected on a screen, the aim is to remove a listener from their physical space and take them on an imaginative journey. In real life, more than one sound source can exist in an environment, and if so, not all sounds come from the same direction. This is why stereo is the first step in creating a radius of realism surrounding the listener. An ideal setup endeavours to place two speakers on both sides of a person's head, the same distance apart, and positioned at the same angle so that the sound emitted from each speaker is equal, given that there is no central speaker directly behind the head.

1.3. Amplitude Panning

When using a left and right channel for stereo sound, panning can be done by modifying the sound amplitude coming from each speaker, causing an imbalance to give the illusion that either the sound source, or the listener has moved (making it suitable to virtual rides where seating platforms are moving to correspond with visual effects, enhancing this sensation). This is known as pair-wise amplitude panning and is based on a sine-cosine law by engineer Alan Blumlein (Neoran 2000). The importance of centralising the listener in the stereo layout may seem unjustified however if one speaker is closer than another, this may cause sounds coming from the other speaker to be ignored despite volume levels. This is called the precedence effect and is not desirable when using two speakers to act as the 'ears' of the protagonist in a virtual simulator (the human head being central to both) (Zölzer 2002, pg.141).

1.4. Sine-Cosine Law

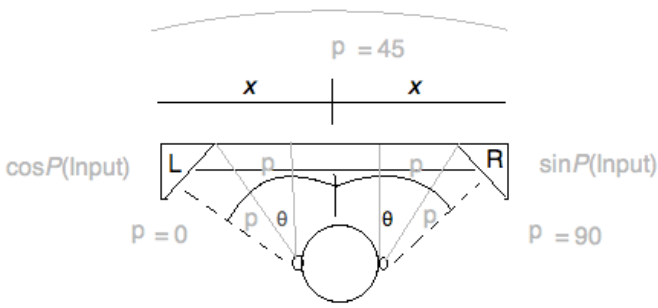
The output of both channels is calculated through the sine-cosine law that, for one channel, multiplies the input signal by sine of an angle (which is set as a control value in the digital audio system, such as Matlab) and cosine of another angle for the output signal of the second channel. As used commonly, the angle of 90 degrees marks the boundaries of the azimuth or lateral plane (horizontal axis), 0 degrees being the position of the left speaker, 45 degrees marking the centre (this is dubbed the phantom centre in a two-channel configuration as there is no speaker here), and 90 degrees at the right speaker. An advantage of using a rotation matrix is that it involves no fading of sound, but instead steers the sound in any direction across the azimuth without altering loudness. This in return, avoids the phantom centre from being exposed, and gives the illusion of a full sound even when only two speakers are being used. (Griesinger 2002)

Figure 6

(When L=left output, R =right output)

L=(cos$P$) (Input)

R=(sin$P$) (Input)



1.5. Digital Signal Processing - Panning

As the recent laboratory report required the creation of a digital audio system addressing a certain problem, being the requirement for a system which incorporates reverberant and panned output, the solution was to be coded into Matlab. For a concept such as the sine-cosine law to be understood by a digital audio processing workstation, the code was translated into a rotation matrix. A rotation matrix, *A*, transfers sound between channels based on the constant value, *p* (Figure 6). Sound input is divided into segments, *S*, for processing, as demonstrated in the algorithm below. (Figure 7)(Zölzer 2002, pg. 175)

Figure 7

$$A_p S = \begin{bmatrix} \cos p \ \sin p \\ -\sin p \ \cos p \end{bmatrix}$$

2. CONCLUSION

A virtual realm is designed to provide a mental and physical experience, which transcends the boundaries of space and reality, offering the opportunity to venture in a controlled space where variables are pre-decided. Since this realm is designed for human senses such as the visual and auditory system, virtual characteristics must be aligned with expectations, whether in sound or in vision. Spatial audio effects increase the sense of realism and follow the scientific logic which humans face every day such as sound localization, direction, and reverberation and sound movement. The creation of the *Revvvpan* digital audio system took these factors into account and focused on creating spatial impression, linearity and most importantly, credibility in audio-visual three dimensional simulation rides, where spaces are constantly changing.

3. REFERENCES

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