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Simulating Changing Distance in a Real-Time Audio Environment

Humans' perception of distance through hearing is not very sophisticated. With sound sources greater than one meter away, the localization blur phenomenon occurs, which makes it difficult to determine exactly how far away the source of a sound is. In virtual sound environments, then, the distance of an object is not treated as importantly as the position of the object relative to the listener and its azimuth. In the Dolby Atmos renderer, for example, the user is given only three options for distance: "Near," "Mid," and "Far." This works well for when the virtual environment is made to replicate an enclosed space, such as a room, but what about when an object is meant to sound like it is coming from far away? Say, for example, the user wants a virtual spaceship to fly from one horizon to the other, speeding past the listener. Atmos allows the mixer to recreate the motion of the spaceship with a moveable object, but the sense of the sound traveling through a vast landscape would be lost. This shortcoming in Atmos's design is the motivation for this project; we have created a Max patch that allows the user to input an audio file and select their desired distance and velocity in order to dynamically change an object's position over a large virtual distance.

When a sound travels over a long distance in open air, its timbral qualities are changed by the physical properties of the medium it travels through. In order to create the most true-to-life representation of open-air acoustics, this patch replicates a few of the many changes to the sound, such as attenuation due to air absorption, distance damping of high frequencies, and pitch changes due to the Doppler effect. All of these changes can be modeled using different formulae which are often complicated and use many variables. For the purposes of this project, we chose to model the general behavior of sound under these conditions and used our own ears and taste to fine tune the filters and controls.

To start, air absorption is the process through which a sound wave decreases in energy as it travels (Russell, 2016). This is due to friction between air molecules which are made more or less viscous depending on the temperature, air pressure, and humidity (Russell, 2016). The most precise calculations include many factors, even involving the molecular relaxation of the two most abundant elements in air: nitrogen and oxygen. All together, the total attenuation of the soundwave can be represented by an absorption coefficient a which is measured in decibels per 100 meters (Russell, 2016). In general, as the frequency of a sound increases, the attenuation coefficient increases logarithmically. This means that higher frequencies are attenuated much more than lower frequencies. This process is called damping (Russell, 2016). Another property of acoustics that affects the sound is the principle that a sound's pressure level will decreased by 6 dB every time the distance between the source and the listener is doubled (Sengpiel). This measure is relative to where the listener is and is not dependent on the frequency of the sound. On top of high frequencies being filtered out first, the overall intensity of the sound will decrease with an increase in distance.

Next, the Doppler effect changes the perceived frequency of a sound due to the motion of the source or the listener (Libretexts, 2022). The effect states that a source moving away from the listener causes a lower observed frequency and a source moving closer to the listener will have a higher frequency, with the maximum frequency being equal to what the sound source is emitting. For example, the pitch of a motorcycle or ambulance passing by a listener is observed to start low and reach the highest pitch once it is directly in front of the listener before decreasing in pitch as it drives away. This phenomenon can also be observed and calculated for a moving listener and stationary sound source (Libretexts, 2022).

We decided to implement this project in Max because of the ease with which audio signals are able to be filtered and manipulated, and the ability to change the input parameters in real-time is useful for live mixing. Ideally, this system could interface with Dolby Atmos, but that is out of the scope of this project. The high-level signal flow of the patch is as follows: user inputs are collected, calculations are made for the aforementioned environmental acoustic effects, and the audio is binauralized before being output. We have included multiple test flows so that users are able to easily understand how the audio output is changed with the parameters.

To start, the user must upload the audio or use one of the default Max sounds. They are given options to change the distance in meters as well as the velocity of the object in meters per second. A positive velocity means that the object is moving towards the listener and a negative velocity means it is moving away. There is also a dial for azimuth, which indicates at what degree in front of the listener is the sound source coming from. We found that the patch works well for distances up to 200 meters.

Next, the signal goes through the filtering process based on air absorption. Due to the complicated nature of calculating the attenuation of each frequency, we based our filter on the logarithmic attenuation curves created using the absorption calculator created by Russell (2016). We used a high shelf filter in order to simulate the dropping of high frequencies that occurs over distance. We take in the user's distance and scale it so that it corresponds with the correct amount of gain to add to the signal. The signal is then sent to the Doppler part of the patch. At this point, the observer frequency of the sound is computed first, taking in the velocity of the object and the frequency of the original sound. Then, the frequency of the output signal is gradually changed using a phasor until it meets the calculated observer frequency. The gain of the clip is then computed based on the distance; the inverse square law can be applied since the audio intensity is proportional to the inverse of distance squared.

Finally the signal is run through Spat, a tool created by Ircam for spatialization of sound signals in Max (Spat). We decided to binauralize the recording because it was easiest to test with the resources we had at hand. Ideally, the end users of this patch would have the option to listen to the output over loudspeakers as well. Spat is able to handle either case. For creating a binaural recording, Spat allows the user to upload an HRTF of their choice or use a default. Spat also takes the azimuth input information in real time and convolves the signal on the fly. The signal is split between left and right channels and finally sent to the output.

The patch was successful in providing a realistic-sounding, immersive audio that simulates a sound source moving from a great distance. The user is able to customize to the meter how far away an object is, with best results occurring up to 200 meters. The patch was demoed live over speakers with success despite the fact that the signal was binauralized. For a quick start option, we included a test signal path within the patch that allows the user to test the different input parameters automatically. The first changes the azimuth of the sound, essentially moving the virtual sound source from left to right. Next, there is an automated distance test that moves the object from in front of the listener to behind. Finally, there is a test that changes the velocity of the object as it is moving so the Doppler effect is exaggerated. These tests are good introductions into how the patch works, as the learning curve for using a new Max patch can be steep.

For future work, the first change to make would be making the calculations as accurate as possible. The process for calculating the damping of air is currently the least accurate. The actual formula for damping of high frequencies involves variables such as the relaxation frequencies for oxygen and nitrogen, the triple-point isothermal temperature, and the saturation vapor pressure. A subsequent version of this patch could include a basic mode (similar to the current system) and an advanced mode, in which the user is able to control all of the variables themself. Another important improvement that could be made would be to integrate this application with Atmos. The most plausible way to achieve this would be to set-up an interface with ProTools or another digital audio workstation that can be used for live recording. Once a recording has been made with the patch, a mixer would create an Atmos object with the same properties that will play the audio. Hopefully Dolby can provide users with finer control over the distance parameter of their sound objects. Until then, the process of creating virtual sound sources that are moving quickly and traveling great distances will remain impractical.

References

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