**Doppler Effect**

1. **Introduction**

Doppler Effect is the change in frequency of a sound caused by the movement of the source producing the sound. The distance, velocity of the source movement and speed of sound in medium determines how the frequency changes. Distance is measured from the listener, when the source moves away from the listener there is a frequency shift to a lower frequency and when the source moves closer to the listener there is a frequency shift to a higher frequency. This can be shown by the equation given below

\[ fd = fs \left( 1 + \frac{Cs}{c} \right) \]  

(2.1)

The above shown formula describes the change in frequency with respect to change in speed of sound, frequency of source and velocity of the source, which is the effect called Doppler’s effect. ‘fd’ is the change in frequency, ‘fs’ is the source frequency, ‘c’ is the speed of sound and ‘Cs’ is the velocity of the source.

![Figurative representation of Doppler effect](image)

Figure 1, figurative representation of Doppler effect

Doppler shift can be reproduced using programming languages like ‘Matlab’. But usually the reproduction quality of the Doppler shift which is produced using the programming language is not anything similar to the Doppler shift experienced in the real life. This is mainly because there are number factors which are ignored while programming the Doppler shift reproduction.
program. This review focuses on how to reproduce virtual moving source causing Doppler shift more accurately as heard in the natural world.

2. Review

![Figure 3, Doppler Effect on multiple frequency.](image)

The above shown figure shows Doppler Effect on multiple frequencies. One can see there is a change in intensity in the middle part of the waveform shown in the figure. In this part it is assumed that the source sound is coming closer to the observer/listener and as the intensity reduces in the later part of the waveform source sound is moving away from the listener.

The below shown figure shows the change in intensity more accurately for the waveform shown in figure 3.

![Figure 4, figure shows the change in amplitude more accurately for the waveform in figure 3.](image)
Time shown in the figure above is shown from -5 to +5. This is actually apparent time i.e., -5 to 0 shows the time required for sound source to reach the listener/observer and the latter part i.e., time from 0 to 5 seconds shows time required for sound source to move away from the listener or observer as far away as it was initially. Here the speed of the sound source is considered to be constant and total time is considered to be 10 secs i.e., time required for sound source to come close to the observer position and goes past the observer to a particular known distance. The amplitude shown in the figure above is from 0 to 1.

The paper (John chowning, 1970) talks about some of the important problems of synthesizing a sound source in an imaginary space and for moving sound source. The author intends to figure out a technique to produce reasonably convincing spatial image.

Localization cues

In an enclosed space such as a hall there are two types of information that is required in order to localize sound. Angular localization is with respect to the listener and distance between listener and source of sound.

The angle of the source of sound with respect to the listener/observer is important. Whenever the source of sound is not center focused there is a delay in arrival time at the two ears. There may be also a pressure level difference at high frequencies. This is mainly because the wave length of the high frequency and also because high frequency waves highly directional.

As the sound source comes closer to the listener/observer the direct sound energy is received more than indirect sound energy whereas when the sound source moves away from the listener/observer indirect sound source is more evident than direct sound source. When distance from the observer and source of sound is more there can be a loss found in the low frequency components of the sound produced.

For the simulation of the distance cue in the virtual environment, direct and reverberant signals are synthesized and controlled in such a way that with increase in distance, intensity of the reverberant signal increases and direct signal intensity reduces. Hence direct signal is inversely proportional to the distance.

The Doppler sift depends on change in velocity. If ‘D’ represents distance in feet and ‘t’ represents time. The change in frequency due to the phenomena called Doppler is proportional to the ‘dD/dt’ which is change in frequency per change in time.

In the paper (Mark A. Ericson, 2007) roles of change in frequency and change in intensity for the perception of moving sound sources are discussed. A number of practical tests are conducted to
in order to find out how a person listening to a moving sound source judges change in speed of sound based on the change in frequency and intensity. The paper discusses about the importance of air absorption in the perception of speed of sound source. The parameters for air absorption include humidity, temperature, and atmospheric pressure. Absorption of the sound is frequency depended and hence high frequency energy is attenuated more than low frequency energy.

To reproduce Doppler shift effect one can include air absorption factor in order to sound more natural. The equation given mentioned in the paper (Mark A. Ericson, 2007) describes air absorption.

\[ I(t) = 8.69 \times \alpha \times (r_2 - r_1) \quad (1) \]

Here \( \alpha \) is the absorption coefficient, \( r_2 \) is the reference distance from an arbitrary location and \( r_1 \) is the distance from the observer and sound source. The result we get \( I(t) \) is given as a function of time for that particular time. Absorption coefficient \( \alpha \) should empirically measured because \( \alpha \) is affect by a number of physical properties.

Another way of representing moving sound source to sound naturally in a virtual environment is to use HRTFs (Head related transfer functions). HRTFs describe the path for the sound from the sound source to each of the human ear in terms of ITD and ILD. i.e., ‘inter aural time difference’ and ‘inter aural level difference’, use of binaural technology which is based on binaural synthesis. Binaural synthesis employs impulse responses from audio source to the human ear drum. Binaural synthesis helps in modeling closed enclosure for binaural room simulations. This helps in accurately reproducing sound in the virtual environments such as virtual spaces such as rooms. The paper (Christos Tsakostas and Andreas Floros, 2007) talks about Real-time Spatial Representation of Moving Sound Sources. To represent moving sound source HRTFs are used.

The paper (Holger Strauss, 1998) is about implementing Doppler shift effect in virtual auditory environment. The figure shown below is a figure from the paper (Holger Strauss, 1998).The below shown figure is an auralization unit. The auralization unit helps in simulating virtual sound source with the second order reflection.
Pitch shifter algorithm can be made i.e., the change in frequency according to movement of the source sound. The frequency shift can be found using the Doppler shift equation defined in equation (1). A pitch shifter algorithm followed by the use of the auralization unit can help calculated frequency changes audible but this approach does not reproduce the Doppler shift sound natural. The model described above does the Doppler effect simulation but it does not sound natural because in the real world the change in position and/or velocity of the sound source does not immediately affect the listener or observer experience. There are also a number of assumptions made in this model such as no wind is present while the source sound is moving. Source sound’s speed is considered to lesser the speed of sound in the medium.

There may be also some signal processing issues while implementing Doppler shift effect such as issues with the sampling frequency. Synchronous sampling frequency change and asynchronous frequency conversion can be easily distinguished. Synchronous sampling frequency convertors have the output frequency locked to the input frequency by a number (a factor). But asynchronous sampling frequency convertors have irrational relation between input and output sampling frequency. While implementing Doppler shifts asynchronous sampling frequency convertors are required. But one should use a synchronous sampling frequency convertor to increase the sampling frequency before using the asynchronous sampling frequency convertor.

Another issue is the computational cost, for research purposes computational cost is not much of a problem. Taking psychoacoustics into account one can do much optimization for signal processing. Some of the simplifications that can be applied to the signal processing are as
follows. 1) The amount of frequency shift. 2) Doppler shift not applied to smaller frequencies since it is not audible to human ear. 3) High frequency interpolation requirements not required because of the absorption that takes place due to reflection absorbs high frequency in the natural environment. 4) In order to avoid interpolation filters constant delay time should be rounded off to nearest integer values.

Doppler shift effect can be done real time if the computation time is reduced. This can be done by implementing “intelligent” algorithms. The simplification is discussed above from the paper (Holger Strauss, 1998) help in reducing the computation time.

Doppler Effect can be implemented real time using wave field synthesis. These are discussed in the papers Real-time Reproduction of Moving SoundSources by (Michele Gasparini, Paolo Peretti, 2011), (Jens Ahrens and Sascha Spors, 2008), (Christos Tsakostas and Andreas Floros, 2007), (Andreas Franck, 2007). Physical reconstruction of wave field is used to reproduce sound this is the basis for wave field synthesis. Spatial reproduction of sound is not based on a particular area such as a sweet spot but spread out over extended listening area.

3. Conclusion

Since the Doppler Effect is a spatial effect, wave field synthesis can be used for reproducing the Doppler Effect in the virtual environment. Using the Wave field synthesis number programs are written in order to reproduce the effect in the virtual environment real time. Using HRTFs one can simulate virtual environments so that Doppler Effect may sound more natural.

4. Reference