

MUMT 618: Week #2

1 Sound and Wave Phenomena

Wave motion involves the transfer of energy. The behavior of this energy transfer varies with the particular medium of transport and energy form. In general, energetic disturbances propagate in the form of waves. Mechanical waves travel in a material medium, such as a string or a membrane. Acoustic waves travel in fluids, such as air or water.

1.1 General Wave Properties

- Wave motion is initiated by an energetic disturbance that subsequently travels through a medium with a fixed velocity (for homogeneous media). This moving disturbance is referred to as a traveling wave.
- A wave propagates through a medium via internal cohesive forces, though the medium itself is not transported.
- The wave speed is determined by the mass (or mass density) and elastic modulus (or tension) of the medium in which it travels. A more “massy” material will have a lower propagation speed. A “stiffer” material will have a higher speed of propagation.
- *Longitudinal* Wave Motion: vibration of particles in the medium is along the same direction as the wave motion (longitudinal standing wave animation).
- *Transverse* Wave Motion: vibration of particles in the medium is perpendicular to the direction of wave motion.
- If a system is driven by a continuous sinusoidal disturbance of frequency f , periodic motion will result with a *wavelength* given by $\lambda = c/f$, where c is the wave speed of propagation. The wavelength λ represents the distance between successive, periodic movements of a medium.
- Low amplitude waves that interact (linearly) from different directions will pass through each other unchanged (referred to as linear superposition).

1.2 Sound Waves

- Sound waves are longitudinal disturbances that travel in a fluid.
- Sound involves time-dependent changes of density, with which are associated time-dependent changes of pressure, temperature and positions of the fluid particles [Fahy, 2001].
- Sound is generally associated with sound pressure variations in air within a frequency range from about 20 Hz to 20 kHz, which corresponds to the general range of the human auditory system. Vibrations of fluids outside this range are referred to as **infrasound** or **ultrasound**.
- Sound is often transmitted from one volume of fluid to another by means of audio-frequency waves in solid media (often referred to as *structure-borne sound*).

- In musical acoustics, sounds often result from the transmission of mechanical vibrations in solids into air pressure fluctuations (such as in stringed instruments).
- The speed of sound in air is approximately given by $c = 331.3 + 0.6t$ (meters / second), where t is the temperature of the air in degrees Celsius. A value of 345 meters / second is a good estimate at room temperature.

1.3 Wavefronts and Interference

- A very brief energetic disturbance of air pressure, for example from a sparkplug, will produce a spherical wavefront that propagates away from the source (at speed c) uniformly in all directions. This disturbance will arrive at a distance r from the source after a time delay from emission to reception of r/c .
- Due to the spreading of energy over an outwardly growing spherical surface area of $4\pi r^2$, the magnitude of the pressure disturbance decreases linearly with distance from the source (sound pressure level, which is proportional to the square root of energy, will decrease by 6 dB per doubling of distance).
- Wavefronts generated by multiple sources can superpose in various ways, depending on the arrangement of the sources and the time at which the disturbances are generated. The resulting wavefront can be visualized according to Huygens' principle.
- Sound wavefronts may sometimes be primarily planar, for example when generated in cylindrical tubes or when considering a spherical wavefront very far from its source. In such cases, the magnitude of the pressure disturbance will not decrease with distance (no spherical spreading), neglecting losses.
- Wave interference results from the linear superposition of multiple wave disturbances, which can result from multiple sources and/or reflections.
- When the interacting waves contain common frequencies and fixed phase relationships, constructive and destructive interference creates steady spatial patterns of high and low amplitude (resulting in standing wave patterns on a fixed string or acoustic resonances in a room).

1.4 Reflection, Diffraction, Refraction and Doppler

- When a wave encounters a change in the material in which it propagates, wave reflection will occur at that boundary.
- The way in which waves reflect at a boundary is determined by boundary conditions, such as the size of the boundary in comparison to the incident wavelength, the relative smoothness of the surface, and/or the relationship between media properties on both sides of the boundary.
- If a reflecting surface is smooth over several wavelengths in all directions, the angle an incident wavefront makes with the surface is equal to the angle of reflection (referred to as *specular reflection*).
- On the other hand, if a reflecting surface is quite irregularly shaped, wavefronts will *scatter* in many different directions (sometimes referred to as *diffuse scattering*).
- Specular vs. diffuse scattering behaviours will depend on incident wavelengths. Low-frequency (large wavelength) sound wave components will reflect specularly from a choppy sea surface but high-frequency (short wavelength) components will scatter diffusely.
- Components of a wavefront with wavelengths much larger than the size of an obstacle will diffract or bend around the object almost as if it isn't there. For wave components with wavelengths shorter than the size of an obstacle, a "shadow" region will result behind the object.
- Diffraction is caused by the "removal" of some portion of the incident wavefront and can be qualitatively understood in terms of Huygens principle [Fahy, 2001].

- A sudden or progressive change in wave speed will produce a change in propagation direction or a “bending” of the waves. This is known as refraction.
- The Doppler effect occurs when a sound source and/or listener are moving relative to one another. When the source and listener move toward each other, the emitted sound is perceived by the listener to increase in frequency. When the source and listener move away from each other, the sound is heard to decrease in frequency.
- The perceived Doppler frequency shift results from an effective change in wave propagation speed due to the moving source and/or listener.

1.5 Simulating Sound Wave Propagation

- Sound waves travel at a speed of approximately 345 meters per second. As a result, there is a time delay for sound to travel from an emitting source to a listener some distance away. This is more obvious when the distance between the sound source and listener is large, for example when observing fireworks from a few kilometers away.
- The time delay that results from this finite speed of propagation can be implemented in a discrete-time simulation with a delay line.
- A distance d between source and listener will result in a time delay of d/c seconds (where c is the speed of sound propagation).
- The delay line length can be determined as $M = d/(cT_s)$, where $T_s = 1/f_s$ is the digital sample period (and f_s is the sampling rate in samples per second).
- Note that the quantity cT_s represents the distance traveled by sound in a single sample period, which is about 7 millimeters at a sample rate of 48000 Hz.
- In this way, we can simulate the propagation of traveling-waves of sound over a specified distance.
- To simulate damped traveling-waves, we should include terms that represent the loss experienced over the distance traveled per unit delay, as represented in Fig. 1.

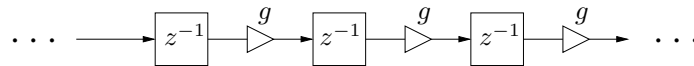


Figure 1: A damped traveling-wave simulator.

- For efficiency, distributed damping constants can be “commuted” (assuming linearity) and implemented at a few (or just one) discrete points in the system, as shown in Fig. 2.

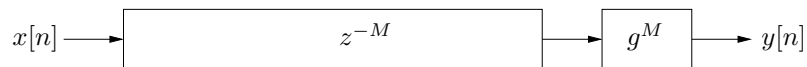


Figure 2: An efficient damped traveling-wave simulator (frequency-independent losses).

- In reality, these losses will be frequency dependent (typically more losses at higher frequencies) and thus more accurately represented with appropriately designed lowpass digital filters, as shown in Fig. 3



Figure 3: A damped, spherical traveling-wave simulator (frequency-dependent losses).

- As mentioned earlier, the amplitude of spherical pressure waves in air is proportional to $1/d$ (due to spherical spreading), where d is the distance of the wavefront from its source. Thus, when simulating the pressure of spherical wavefronts, an additional “spreading” factor should be added as shown in Fig. 3.

1.6 Wave Reflections and “Echo”

- If a traveling wave encounters a change in the physical properties of the medium through which it propagates, the wave will be perturbed where the change occurs. This perturbation generally involves some level of reflection, absorption and transmission at the boundary.
- For example, if a wavefront impinges on an ideally rigid surface, all of the wave energy will be reflected from the surface. However, if the surface is instead covered with a layer of absorbing material, only a portion of the wave energy will be reflected, with the remainder being trapped and damped within the material.
- The extent of such reflection can be characterized by a reflection coefficient (R), which specifies the ratio of reflected to incident wave energy. Materials that are very reflective will have a value of R close to 1, while R will be close to zero for materials that are very absorptive. In general, the reflection coefficient will be frequency dependent.
- As previously mentioned, wave reflection from surfaces will also depend on the shape of the surface. If an acoustic wave encounters a rigid wall that is flat over at least several wavelengths in all directions, the wavefront will experience specular reflection from that surface (equal angles of incidence and reflection). On the other hand, a wall that is very uneven will reflect a wavefront in many directions (“diffuse” scattering).
- Figure 4 illustrates a source-listener arrangement with “multipath” wave propagation. If the time delay between the arrival of the direct and reflected waves is greater than about 50 milliseconds, the reflected sound will be perceived as an echo.

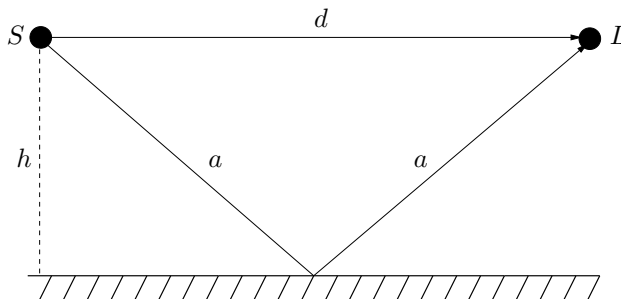


Figure 4: A source-listener arrangement with an “echo” or “floor bounce” propagation delay.

- The system of Fig. 5 (top) provides a signal processing block diagram to simulate the sound wave propagation of Fig. 4 using digital delay lines. The scale factors g_d and g_r account for losses over the respective direct and reflected paths due to the combined effects of air absorption and spherical

spreading. If the floor had a reflection coefficient less than one, this could also be included in the g_r factor.

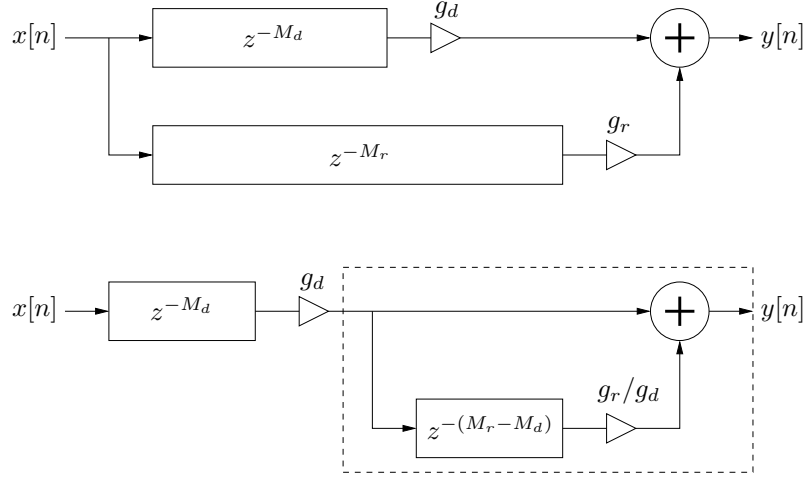


Figure 5: Floor reflection block diagrams.

- The delay common to the two paths can be pulled out and implemented separately, as illustrated in the lower part of Fig. 5. In this case, the length of the delay line for the reflected path must be adjusted by subtracting from it the common delay length and its attenuation factor appropriately scaled.
- $M_r - M_d$ is found as:

$$M_r - M_d = \frac{2a - d}{cT_s} = \frac{2\sqrt{h^2 + (d/2)^2} - d}{cT_s}.$$

- If the gain factors were only used to simulate spherical propagation scaling, the scaling for the reflected path in the lower plot of Fig. 5 would be calculated relative to d as:

$$g_r/g_d = \frac{1/2a}{1/d} = \frac{d}{2a} = \frac{1}{\sqrt{1 + (2h/d)^2}}.$$

- The portion of the signal processing block diagram within the dashed lines at the bottom of Fig. 5 is a feedforward comb filter. This simulated system results in a feedforward filter structure because none of the propagated sound returns to the listener.
- Depending on the distance between the direct and reflected paths, certain frequency components in the sound will be destructively cancelled at the listener position, corresponding to the notches in the frequency response of the feedforward comb filter. In this way, the feedforward comb filter is a computational physical model of a source-listener arrangement involving a direct and single reflected path.
- If the source-listener arrangement involved two parallel walls with the source and listener located in between, the corresponding wave simulation block diagram would include a *feedback* comb filter, which would model a series of echoes, exponentially decaying and uniformly spaced in time.

2 Delay-Based Effects

A variety of common audio effects are implemented using variable-length delay lines. In this section, we analyze several of these techniques, many of which can be seen as simulating physical or non-physical variations of sound sources and/or receivers. Note that a discussion of artificial reverberation will be “delayed” to a later section.

2.1 Flanging

- Flanging involves the summing together of a signal and a time-varying delayed version of itself.
- The input-output relationship for a flanger is given by:

$$y[n] = x[n] + gx[n - M[n]],$$

where $M[n]$ is the time-varying length of a delay line and g is the “depth” of the flanging effect. A flanger block diagram is shown in Fig. 6.

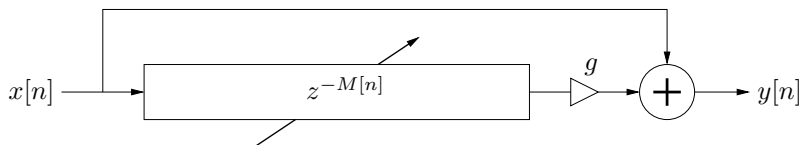


Figure 6: A digital flanger block diagram.

- Because the delay-line length, $M[n]$, must change continuously and smoothly through time, it is necessary to make use of an interpolating delay line.
- At any instant in time, the flanger is equivalent to a feedforward comb filter, which has a frequency response as shown in Fig. 7.

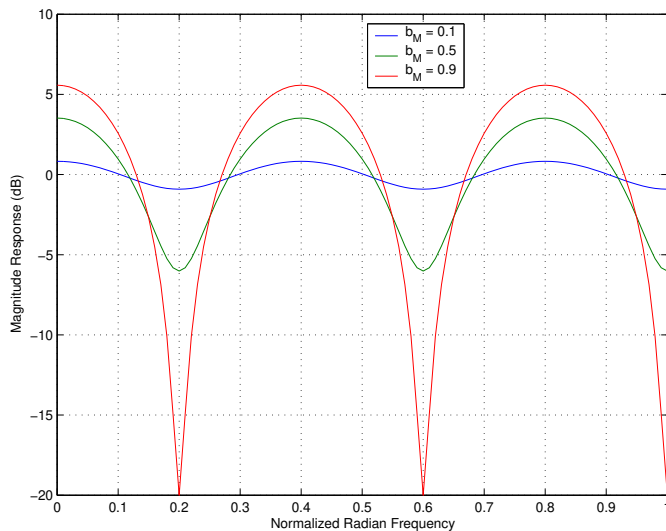


Figure 7: Magnitude response of a feedforward comb filter with $M = 5$, $b_0 = 1$, and $g = b_M = 0.1, 0.5$, and 0.9 .

- For $g > 0$, there are M peaks in the frequency response, centered about the frequencies $\omega_k = 2\pi k/M, k = 0, 1, \dots, M - 1$. Between these peaks, there are M notches at intervals of f_s/M Hz.
- As M changes over time, the peaks and notches of the comb response are compressed and expanded. The spectrum of a sound passing through the flanger is thus accentuated and deaccentuated by frequency region in a time-varying manner.
- The delay-line length of a flanger is typically modulated by a low-frequency oscillator (LFO). Oscillator waveforms are typically sinusoidal, triangular, or exponential.

- For a sinusoidally varied delay,

$$M[n] = M_0 \cdot (1 + A \sin[2\pi f n T_s]),$$

where f is the flanger “rate” in Hz, A is the “excursion” (maximum delay swing), M_0 is the average delay-line length that controls the average notch density, and T_s is the sample period.

- For values of $-1 \leq g \leq 0$, the peaks and notches of the comb filter trade places. In practice, g is normally constrained to the interval $[0, 1]$ and the option of sign inversion is provided by a “phase inversion” switch.
- In the inverted mode, a notch is located at zero frequency. As a result, bass response will likely be weakened.
- Some flangers also implement feedback (in addition to the delayed feedforward path), which introduces spectral peaks and notches as previously described for feedback comb filters.
- The following MSP patch implements a flanger.

2.2 Doppler Effect

- The Doppler effect can be used to enhance the realism of simulated moving sound sources.
- The amount of frequency shift is given by:

$$f_l = f_s \frac{c + v_{ls}}{c - v_{sl}},$$

where f_s is the frequency of the source at rest, f_l is the frequency perceived by the listener, v_{ls} is the speed of the listener toward the source (zero if not moving), v_{sl} is the speed of the source toward the listener (zero if not moving), and c is the sound speed.

- We can simulate time-varying source and listener velocities using a time-varying digital delay line with separate read and write pointers. The write pointer corresponds to the source signal and the read pointer corresponds to the listener. In a delay line implementation, the read pointer should follow behind the write pointer. Thus, if the source position is moving toward the listener, the write pointer increment should be changed from 1 to $1 - v_{sl}/c$. Likewise, if the listener is moving toward the source, the read pointer increment should be changed from 1 to $1 + v_{ls}/c$.
- Interpolated reads from a delay line (fractional delay lengths) were previously discussed. Interpolated writes are referred to as *de-interpolation*.
- Because Doppler shift is dependent only on the *relative* motion of a source and listener and because the de-interpolation process is generally more complicated to implement than interpolation, it is best to change only the read pointer increment when possible.
- A continuously varying delay can be implemented with a “growth parameter” $g = -v_{ls}/c$ such that the read pointer is incremented by $1 + g$ at each time step.
- Multiple read pointers can simulate multiple moving listeners.
- Multiple write pointers can simulate multiple moving sources.
- It is not possible to use a single delay line to simulate both multiple moving listeners and multiple moving sources. In this case, the number of necessary delay lines is the lesser of the number of listeners or sources.

2.3 Chorus Effect

- The simulation of many apparent similar sound sources occurring nearly in unison from a single, actual source is referred to as a chorus effect.
- The chorus effect is achieved by applying various independent modifications to multiple copies of a given input signal and summing them together to create the output signal.
- Signal modifications can include delay, frequency shift, and/or amplitude changes.
- Time-varying delay lines can simulate changes in delay and frequency shift.
- A single multi-tap interpolating delay line with modulated output tap locations can efficiently and effectively implement the chorus effect.
- The following MSP patch implements a chorus effect. Better results can be achieved with more “copies” of the input signal.
- The `rand~` MSP object produces interpolated noise at a subsampled rate, as specified by the input argument. This provides a noise like signal with energy only at frequencies below this rate.

2.4 Pitch Shifting

- The Doppler effect causes a source signal to appear as though it has been pitch shifted. Pitch shifting of an input signal can thus be implemented with time-varying delay lines as described above.
- Because the read pointer of a pitch shifter is incremented at a constant non-integer rate, the read pointer will eventually “catch up to” or “fall back into” the write pointer location (see Fig. 8). To avoid discontinuity issues caused when the read and write pointers cross, a multiple read pointer cross-fade system can be used.

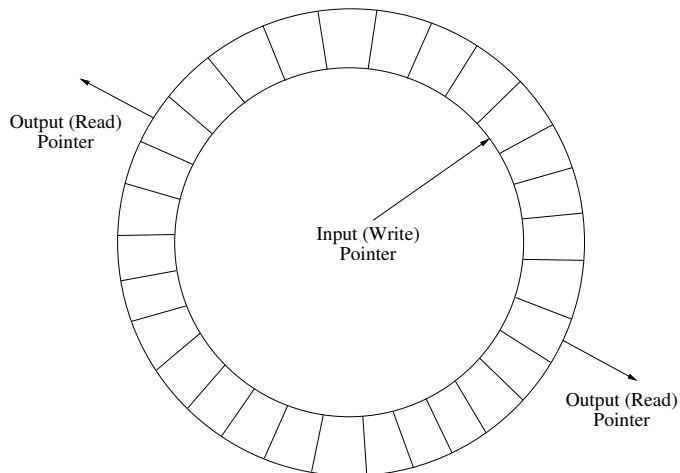


Figure 8: Circular delay line (buffer) arrangement for a basic pitch shifting algorithm.

References

F. Fahy. *Foundations of Engineering Acoustics*. Elsevier Academic Press, London, UK, 2001.