Audio: Rendering

CS 6334 Virtual Reality

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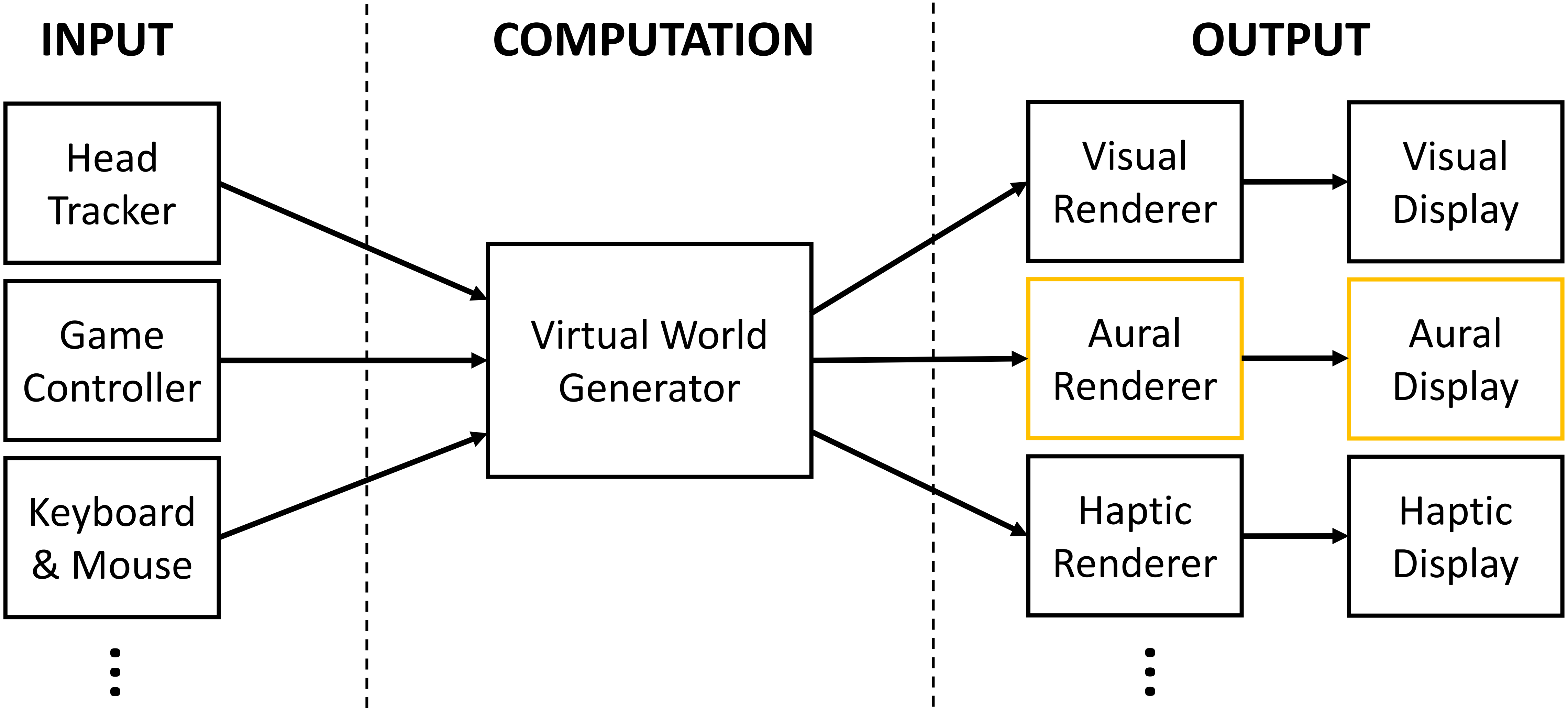
Some slides of this lecture are based on the Virtual Reality textbook by Steven LaValle

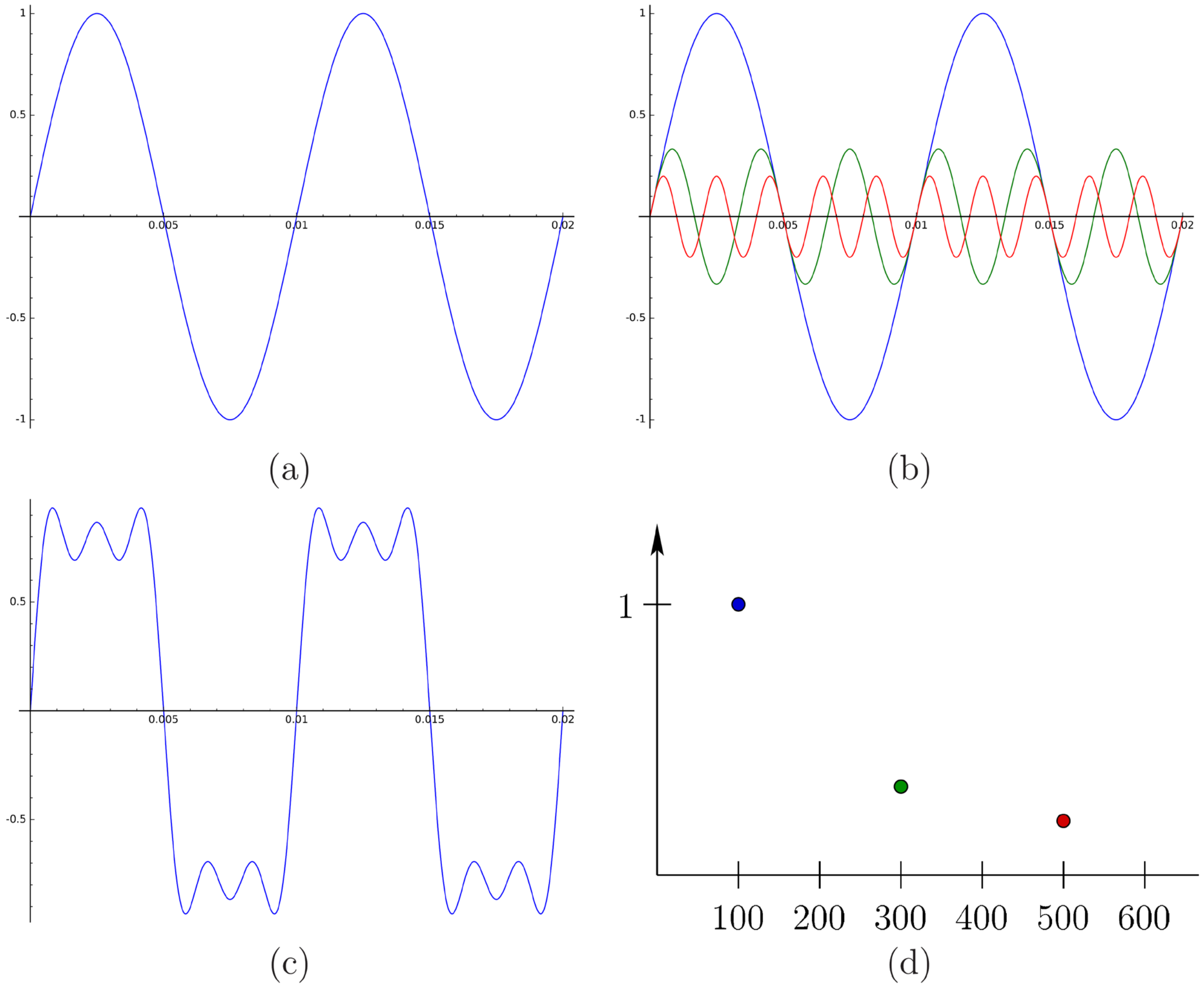
Review of VR Systems



Auditory Rendering

- Producing sounds for the virtual world
- Aural displays: speakers
- The generated sounds should be consistent with visual cues and with past auditory experiences in the real world

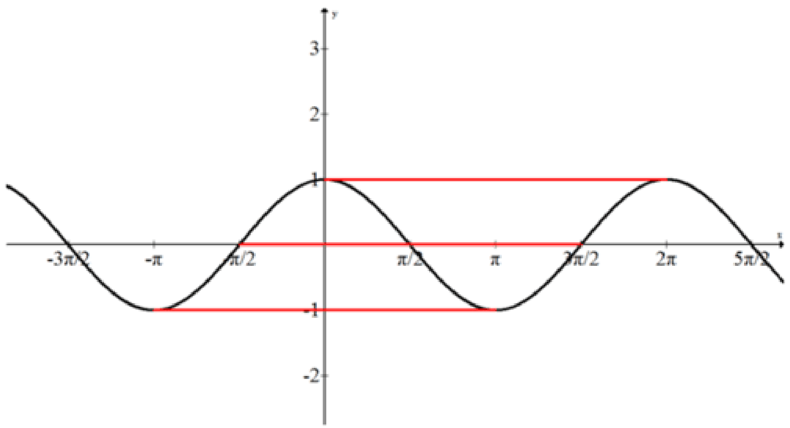
Spectral Decomposition



Fourier analysis: any periodic function can be decomposed into sinusoids

Frequency of Sinusoidal Functions

- Period: length of a complete cycle



- Frequency: number of cycles in 2π

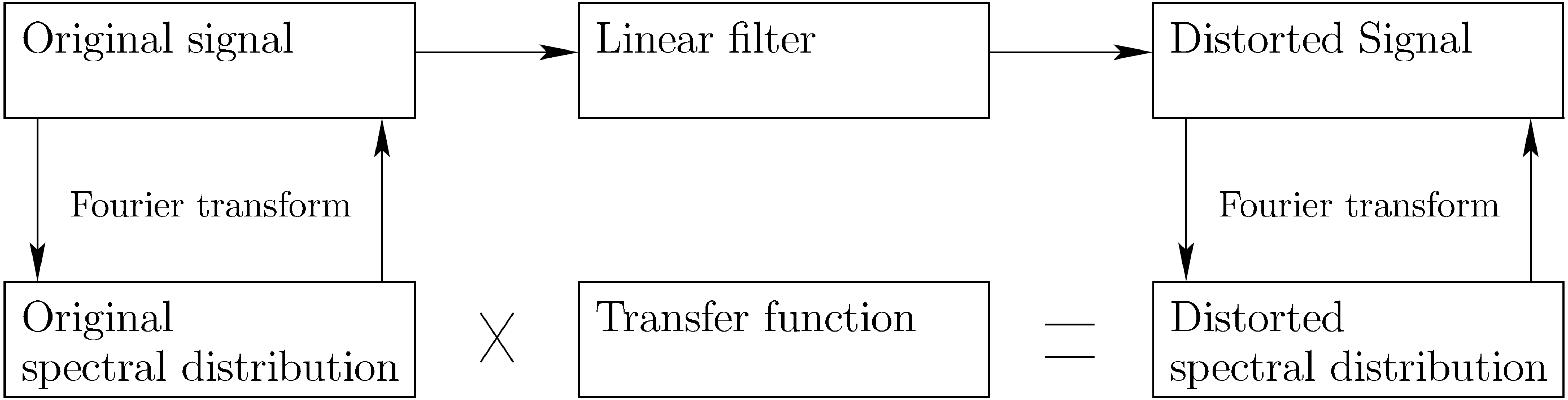
$$\text{period} = \frac{2\pi}{\text{frequency}}$$

$$f(x) = \sin x$$

$$f(x) = \sin \frac{1}{2} x$$

$$f(x) = \sin bx \qquad f(x) = \sin 2\pi f x$$

Signal Processing



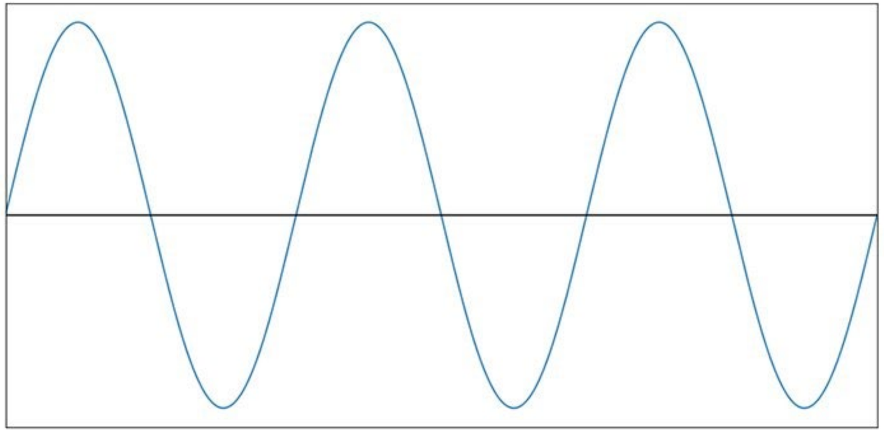
Sampling Rate

- Continuous-time signal $x(t)$
- Discrete-time signal, how computers process signals
- Sampling interval Δt
- Sampling rate (sampling frequency) Hz $1/\Delta t$
  - 1000Hz sampling rate, Δt is 1ms
  - How many samples per second

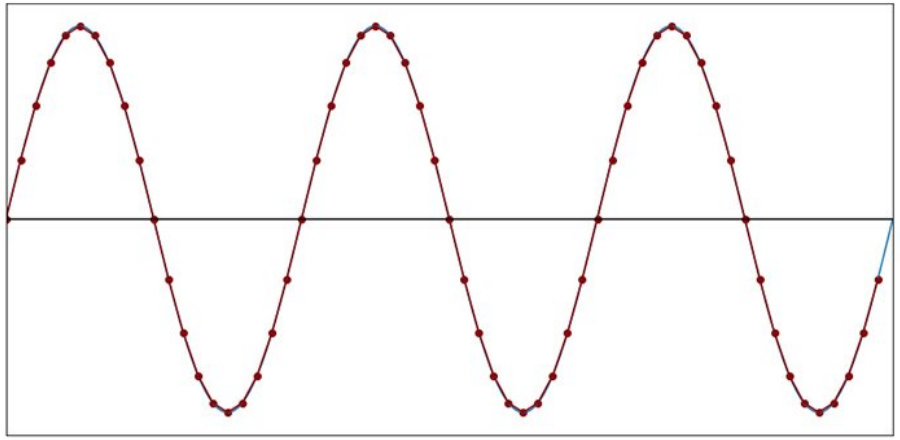
Nyquist–Shannon Sampling Theorem

- The sampling rate should be at least **two times** the highest frequency component in the signal
- The highest frequency for audio is 20,000 Hz, sampling rate at least 40,000 Hz
- Sampling rate of CDs and DVDs: 44,100 Hz and 48,000 Hz
- kth sample $x[k] = x(k\Delta t)$

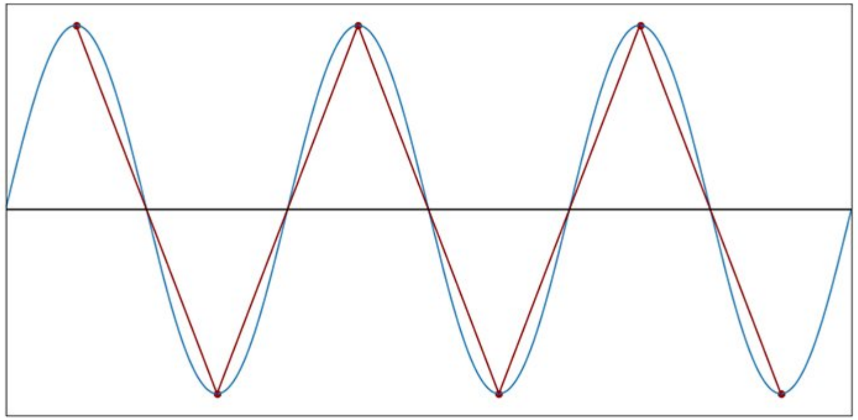
Nyquist–Shannon Sampling Theorem



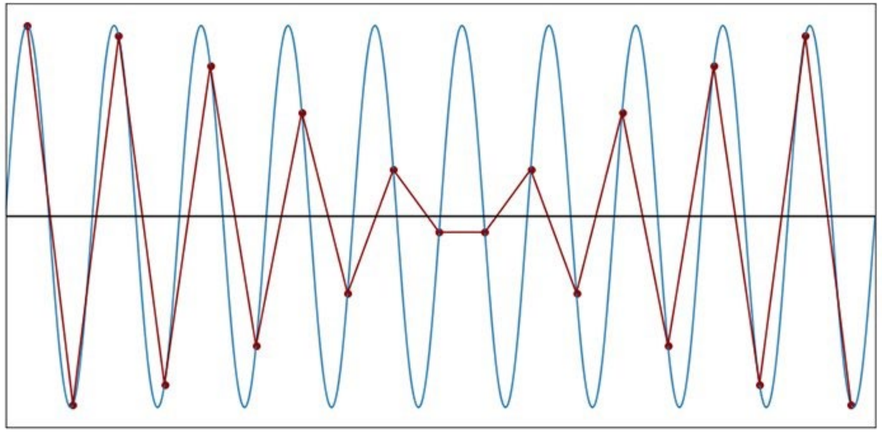
Continuous-time signal



20 samples per cycle



2 samples per cycle



1.9 samples per cycle

https://www.allaboutcircuits.com/technical-articles/nyquist-shannon-theorem-understanding-sampled-systems/

Linear Filters

- A filter is a transformation that maps one signal to another

$$x(t) \longrightarrow \boxed{\text{Filter}} \longrightarrow F(x(t))$$

- Linear filters
  - Additivity $F(x + x') = F(x) + F(x')$
  - Homogeneity $cF(x) = F(cx)$
  - A general form

$$y[k] = c_0 x[k] + c_1 x[k-1] + c_2 x[k-2] + c_3 x[k-3] + \cdots + c_n x[k-n]$$

Examples of Linear Filters

- Moving average

$$y[k] = \frac{1}{3}x[k] + \frac{1}{3}x[k-1] + \frac{1}{3}x[k-2]$$

- Exponential Smoothing (exponentially weighted moving average)

$$y[k] = \frac{1}{2}x[k] + \frac{1}{4}x[k-1] + \frac{1}{8}x[k-2] + \frac{1}{16}x[k-3]$$

Nonlinear Filters

- Any filter that does not follow the following form

$$y[k] = c_0 x[k] + c_1 x[k-1] + c_2 x[k-2] + c_3 x[k-3] + \cdots + c_n x[k-n]$$

- Human auditory system is almost a linear filter, but contains nonlinear behaviors

Fourier Analysis

- Fourier transform for discrete-time systems

$$X(f) = \sum_{k=-\infty}^{\infty} x[k] e^{-i2\pi f k}$$

frequency

Spectral distribution: a function of the frequency

Euler's formula

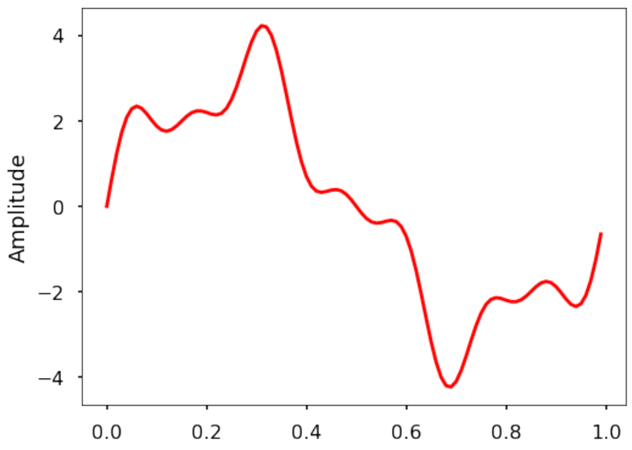
$$e^{-i2\pi f k} = \cos(-2\pi f k) + i \sin(-2\pi f k) \qquad i = \sqrt{-1}$$

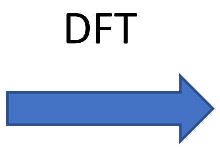
Discrete Fourier Transform

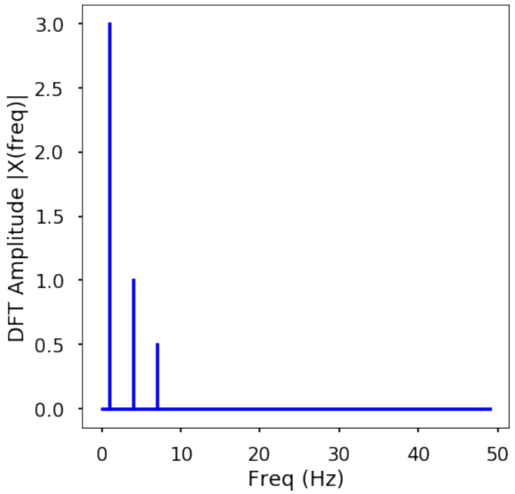
- N samples of a periodic signal

$$X[n] = \sum_{k=0}^{N-1} x[k] e^{-2\pi n k / N}$$

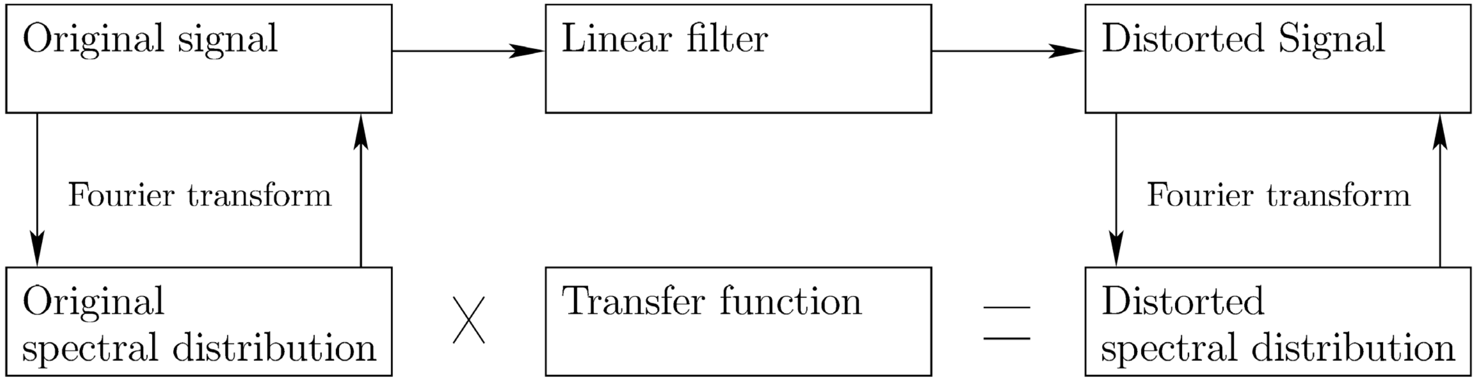
frequency $n \in [0, \ldots, N-1]$







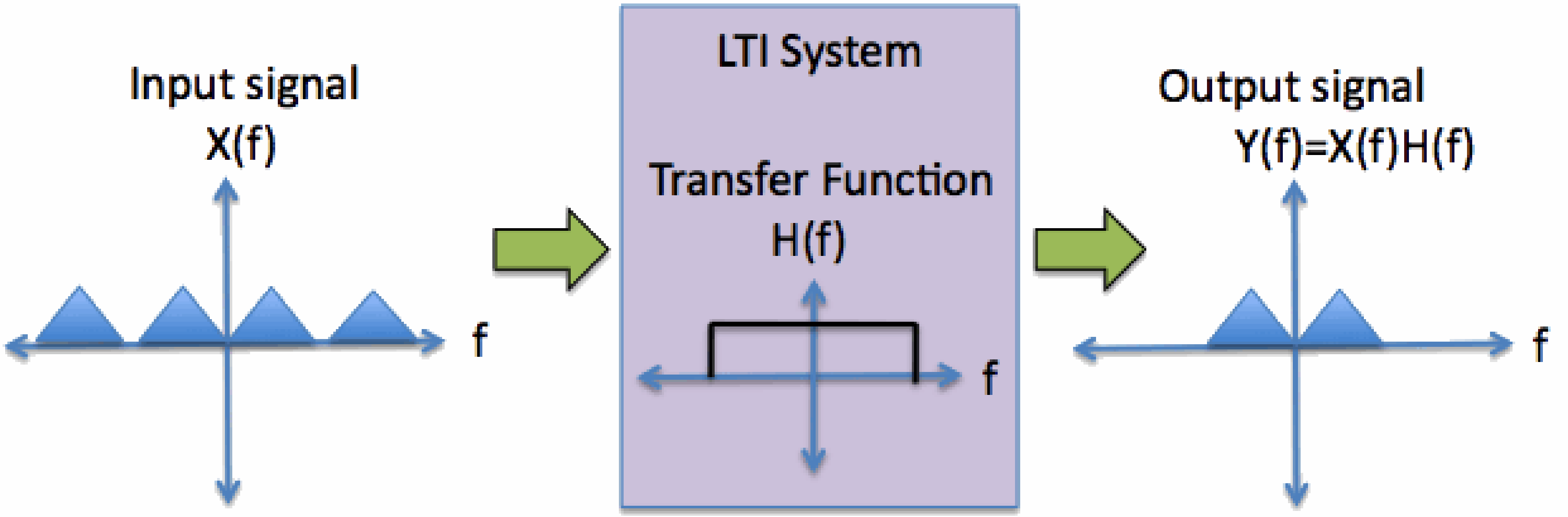
Transfer Function



- A linear filter can be designed to modify the spectral distribution
  - Amplify some frequencies, while suppressing others

- Applying a transfer function
  - Transforming the original signal using the Fourier transform
  - Multiplying the transfer function
  - Applying the inverse Fourier transform

Transfer Function



$$X(f) = \Im\{x(t)\}$$

$$H(f) = \Im\{h(t)\}$$

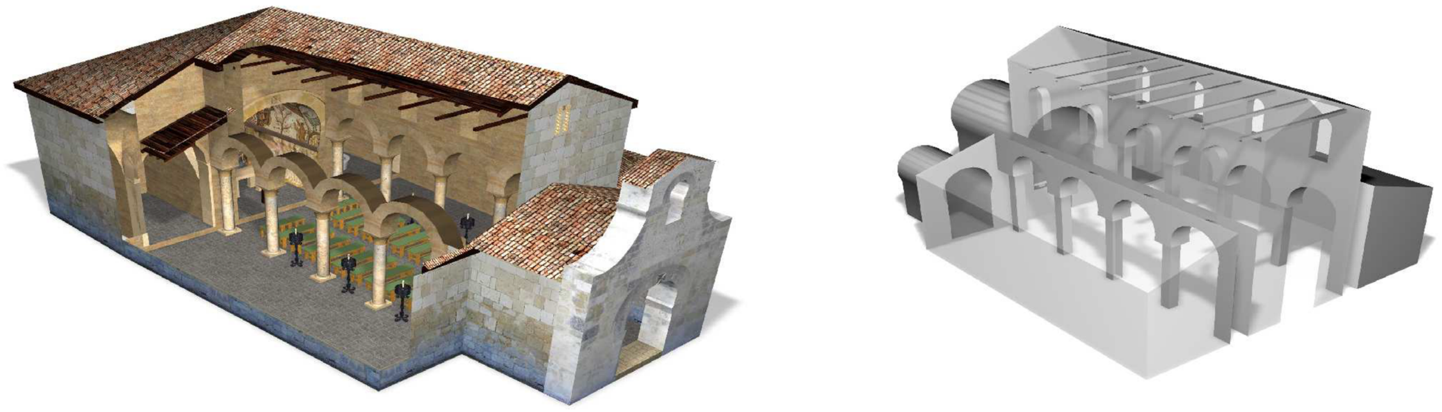
$$Y(f) = X(f)H(f)$$

$$y(t) = \Im^{-1}\{Y(f)\}$$

www.thefouriertransform.com

Acoustic Modeling

- The same geometry model can be used for both visual modeling and auditory modeling
  - E.g., walls can reflect lights and sound waves



(a) (b)

The acoustic model needs to have a spatial resolution of only 0.5m

Figure 11.13: An audio model is much simpler. (From Pelzer, Aspock, Schroder, and Vorlander, 2014, [253])

Acoustic Modeling

- Sound source in the virtual environment
  - White noise, equal weight of all frequencies in the audible spectrum
  - Interesting sounds, high concentration among specific frequencies
- Sound reflection (depends on wavelength)
  - Specular reflection for a large, smooth, flat surface
  - Diffuse reflection for smaller objects, surface with repeated structures (difficult to characterize for sounds)

Propagation of Sounds

- Method 1: simulating the physics as accurately as possible
  - When waves are large relative to objects in the environment
  - Low frequency, detailed environment

- Method 2: Shooting visibility rays and characterize the dominant interactions between sound sources, surfaces, and ears
  - Higher frequency, simpler model

Numerical Wave Propagation

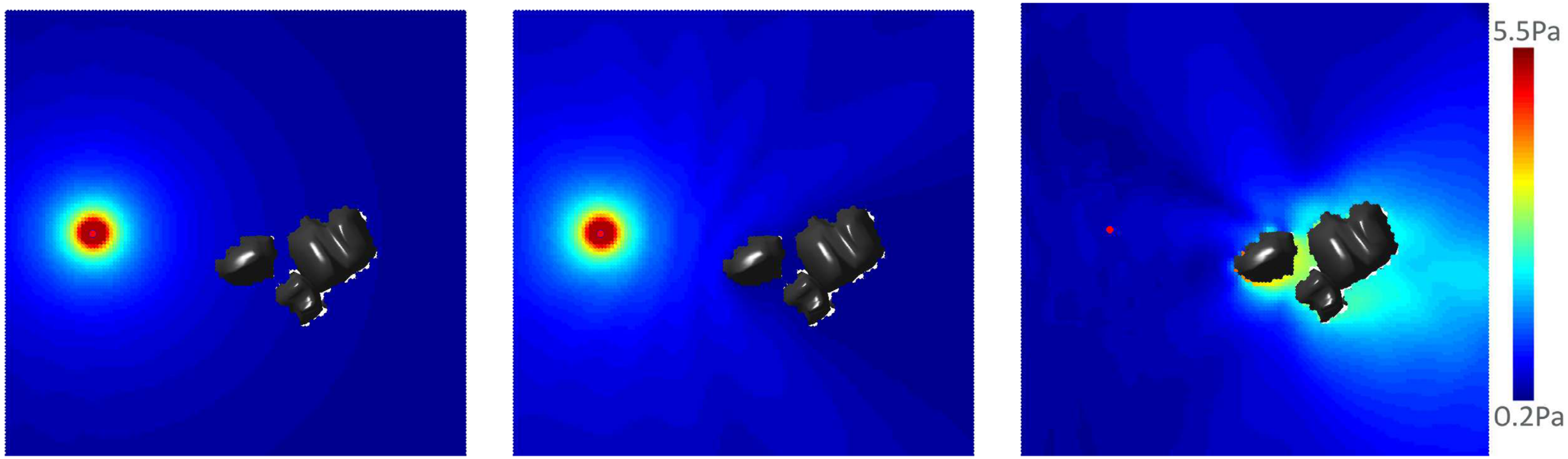
- Helmholtz wave equation
  - Constraints at every point in R^3 in terms of partial derivatives of the pressure function

$$\nabla^2 p + \frac{\omega^2}{s^2} p = 0 \qquad \omega = 2\pi f$$

Laplacian operator sound pressure sound speed

$$\nabla^2 = \frac{\partial^2}{\partial x^2} + \frac{\partial^2}{\partial y^2} + \frac{\partial^2}{\partial z^2}$$

Numerical Wave Propagation



(a) (b) (c)

Figure 11.14: Computed results for sound propagation by numerically solving the Helmholtz wave equation (taken from [212]): (a) The pressure magnitude before obstacle interaction is considered. (b) The pressure after taking into account scattering. (c) The scattering component, which is the pressure from (b) minus the pressure from (a).

Visibility-based Wave Propagation

- Paths of sound rays that emanate from the source and bounce between obstacles

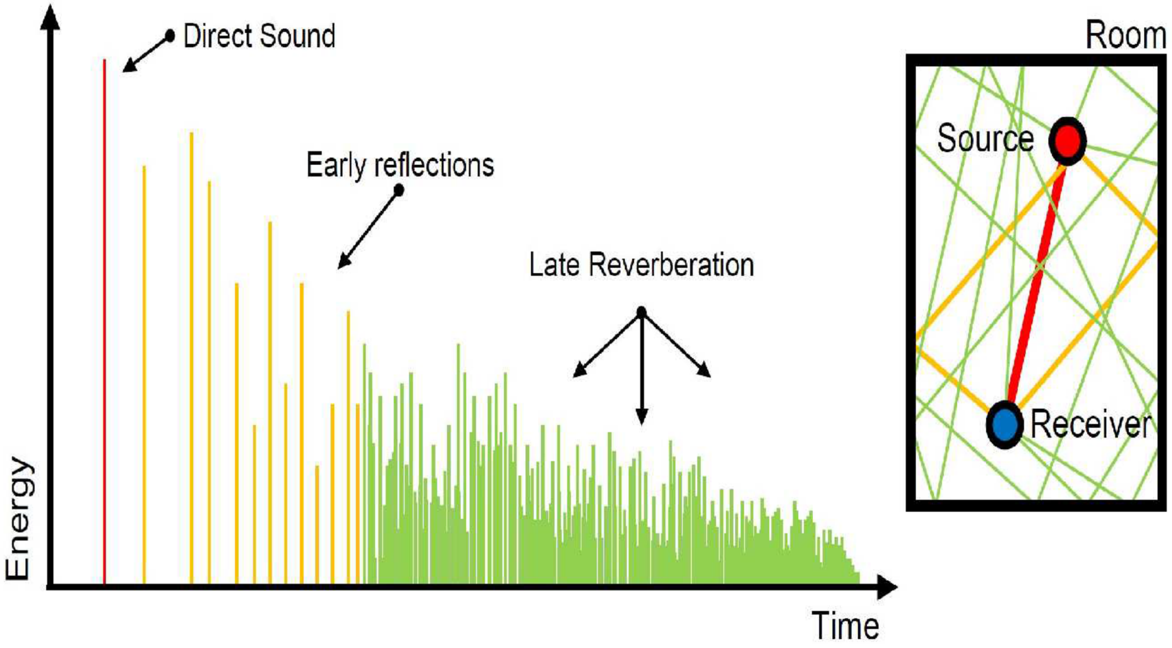


Figure 11.15: Reverberations. (From Pelzer, Aspock, Schroder, and Vorlander, 2014, [253])

Entering the Ear

- A virtual microphone positioned in the virtual world captures the simulated sound waves

- Convert into audio output through a speaker in front of the ear

- ILD and ITD can be simulated by accounting for both ears
  - Interaural Level Difference (ILD), Interaural Time Difference (ITD)
  - Need to model the physical head in the virtual world
  - Head related transfer function (HRTF)

Tracking the Ears

- If the user turns her head, the sound should be adjusted accordingly
- Perception of stationary for sounds
  - Fixed sound source should be perceived as fixed
- Tracking the ear poses to determine the "viewpoint" for sounds

Further Reading

- Section 11.4, Virtual Reality, Steven LaValle