Information Technology
Inside and Outside
- David Cyganski & John A. Orr

V. Bandwidth and Information Theory

11. Sampling of Audio Signals

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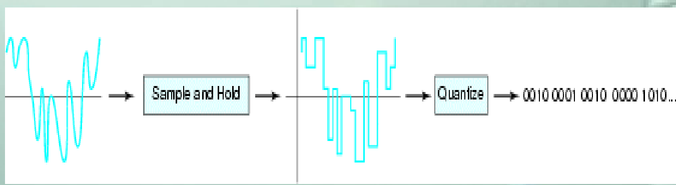
11. Sampling of Audio Signals

- Objectives:
  - the concept of sampling, where a signal that is continuous in time is observed only at periodic intervals;
  - the determination of an appropriate sampling rate (the Nyquist rate) based on the bandwidth of the signal, which guarantees that all of the information in the original analog signal is preserved; and
  - the way in which a signal may be exactly reconstructed from knowledge only of its samples.

11.1 Introduction

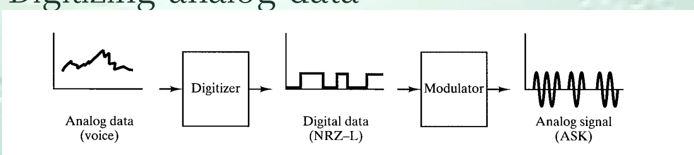
- **The digitization of audio signals : two steps**
  - The first step of this process, in which a continuous audio signal is made discrete in time, is called ***sampling***. This is because we choose to sample, or evaluate, the audio waveform at specific instants in time, rather than to attempt to represent its value for all moments of time.
  - The second step is then identical applicant **quantization** of the video samples; each audio sample is converted into a sequence of binary digits.

Figure 11.1: A block diagram of an audio sampling system.

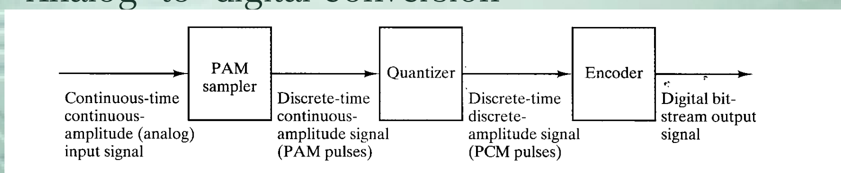


ANALOG DATA, DIGITAL SIGNALS

- Digitization : A process of converting analog data into digital data
- Digitizing analog data

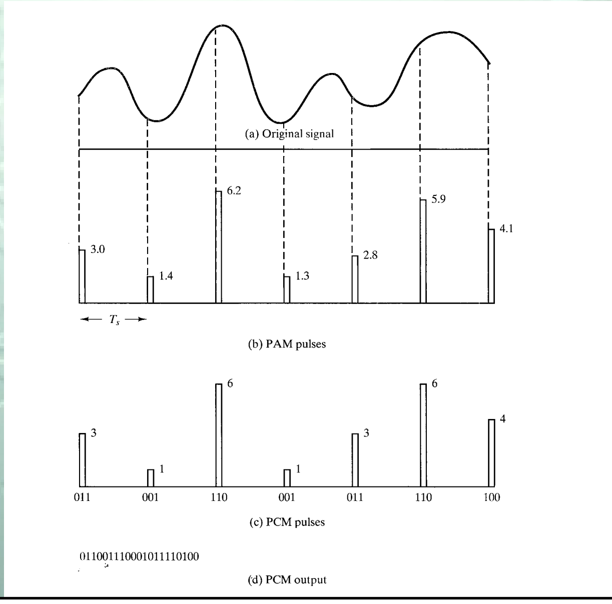


- Analog -to -digital conversion

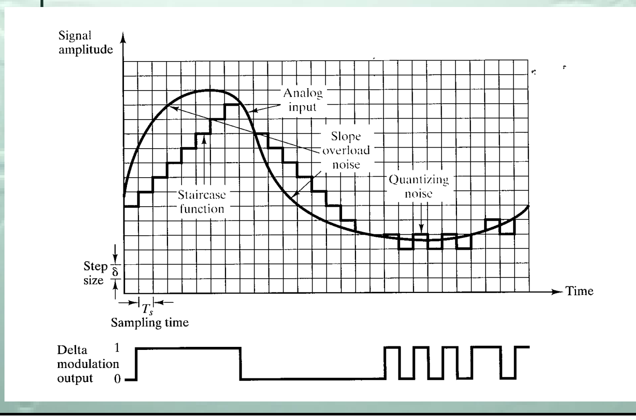


- PAM (Pulse Amplitude Modulation)
- PCM (Pulse Code Modulation) : based on sampling theorem

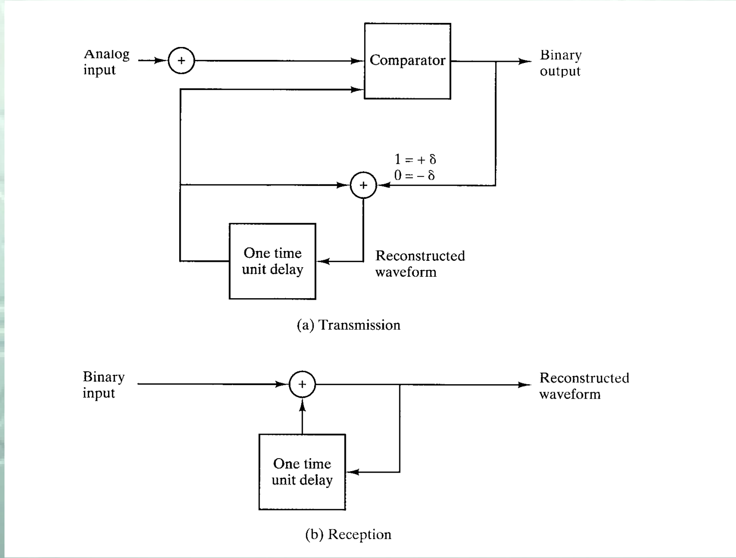
- Pulse -code modulation



- Delta Modulation (DM)
  - Improvement of PCM
  - Analog input is approximated by a staircase function that moves up or down by one quantization level (δ) at each sampling interval (Ts).
  - Example of delta modulation



- Delta Modulation



(a) Transmission

(b) Reception

Vocoding

- Vioce/UnVoice/Silence(Noise)
- Voice → Pitch, Amplitude, AMDF, Formant
- Unvoice → Noise -like generation
- Synthesis → V/UV information

11.2 Sampling an Audio Signal

- **The digitization of audio signals : two steps**
  - The first step of this process, in which a continuous audio signal is made discrete in time, is called ***sampling***. This is because we choose to sample, or evaluate, the audio waveform at specific instants in time, rather than to attempt to represent its value for all moments of time.
  - The second step is then identical applicant **quantization** of the video samples; each audio sample is converted into a sequence of binary digits.

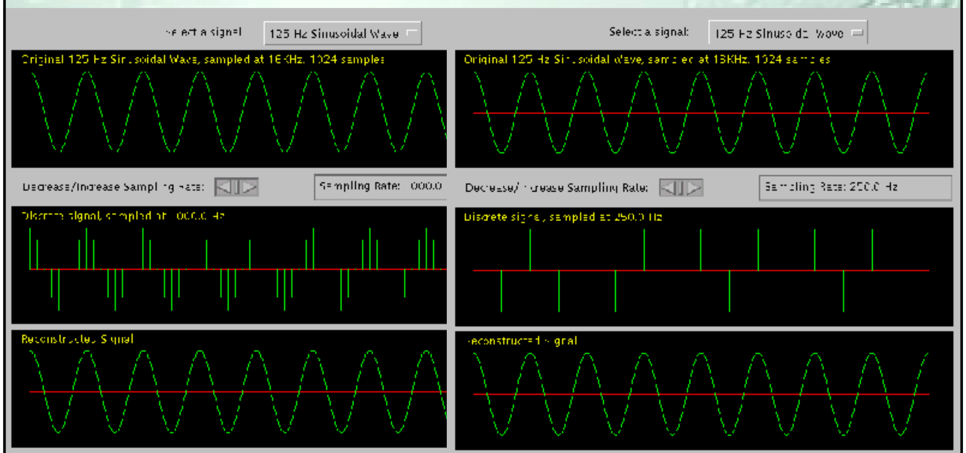
11.2.1 Sampling Intervals and Sampling Frequency

- The location of each vertical line in the center graph indicates the **time** at which the input signal was ``looked" at.
- The height of each line represents the **amplitude** of the input signal at that time. It is important to note that all of the other information between sampling points on the input signal is discarded!

11.2 Sampling an Audio Signal(2)

Figure 11.2: The effects of sampling a 125 Hz sinusoidal wave at 1 kHz.

Figure 11.3: The effects of sampling a 125 Hz sinusoidal wave at 250 Hz.



11.2 Sampling an Audio Signal(3)

- **Fig. 11.2** ➔ A sampling rate of 1 kHz **($f_s > 2f_m$)**
- **Fig. 11.3** ➔ The sampling rate is the minimum value that provides complete information about the input signal. That rate is twice the frequency of the signal, or 250 Hz. **($f_s = 2f_m$)**
- **Fig. 11.4** ➔ the effect of a sampling rate that is too low, in this case 187.5 Hz ➔ **"Aliasing" ($f_s < 2f_m$)**

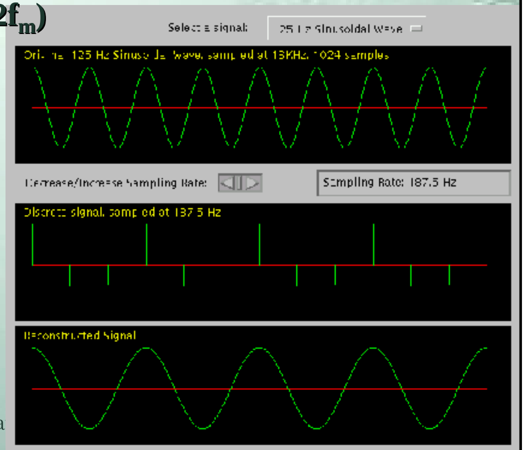
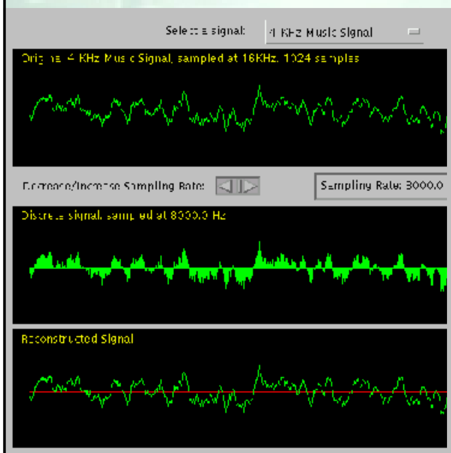


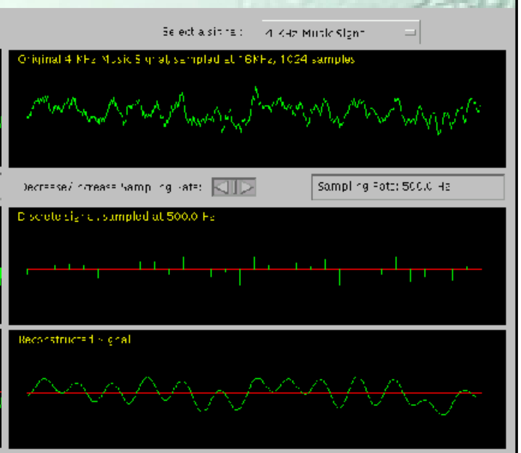
Figure 11.4: The effects of sampling a 125 Hz sinusoidal wave at 187.5 Hz.Note the erroneous reconstructed signal!

11.2 Sampling an Audio Signal(4)

Figure 11.5: The effects of sampling a music signal at a **sufficient rate (8000 Hz.).**

Figure 11.6: The effects of sampling a music signal at an **insufficient rate (500 Hz.).**





11.2 Sampling an Audio Signal(5)

- Assumption : **Sample** a signal ***uniformly***--that is, with sampling instants evenly spaced at regular intervals of time
- The ***sampling interval*** ➔ the amount of time separating the samples
- The ***sampling rate*** => the number of samples taken per second
- ***Sampling frequency***
- A sampling rate **in hertz**, the interpretation is ***samples per second***.

$$f_s = 1/\ T_s, \quad T_s = 1/\ f_s$$

where f_s is measured in hertz, and T_s is measured in seconds

11.2 Sampling an Audio Signal(6)

11.2.2 The Minimum Sampling Frequency

- **Harry Nyquist** and **Claude Shannon** developed a mathematical framework to determine how often a signal should be sampled. Their result, known as the ***sampling theorem***, states that: **IN ORDER TO BE PERFECTLY REPRESENTED BY ITS SAMPLES, A SIGNAL MUST BE SAMPLED AT A SAMPLING RATE EQUAL TO AT LEAST TWICE ITS HIGHEST FREQUENCY COMPONENT.**
- For our 125 Hz audio signal,we see that we should sample it at a rate which is **at least** $f_s = (2) \times (125) = 250$ Hz or 250 samples per second.
  - Because the **minimum sampling rate** for this signal is **250 Hz**,
    - ✓ The samples in **Figure 11.4, (@187.5 Hz)➔ inadequate**.
    - ✓ The samples in **Figure 11.2, (@1 kHz)➔ more than sufficient**.
    - ✓ The samples in **Figure 11.3, (@250Hz)➔ just right.**

11.3 Reconstructing Audio from Samples

- Fig 11.7 : The **bandwidth** is **2,500 Hz**, so a **sample rate** of **5,000 Hz is required** to represent it.

Figure 11.7: A short section of a speech waveform.

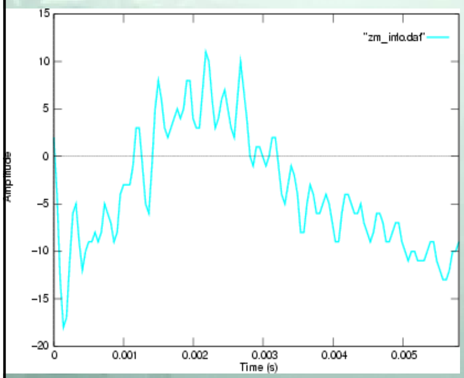
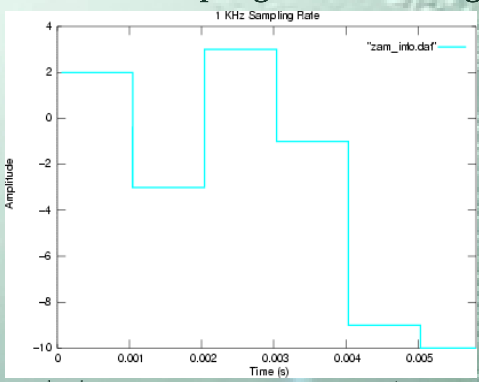


Figure 11.8: The reconstructed speech waveform, based on a **1 KHz sampling rate=>Aliasing**.



11.3 Reconstructing Audio from Samples(2)

- The signal maintains each sample value until the next sample occurs, is called ***zero-order hold*** reconstruction.

Figure 11.9: The reconstructed speech waveform, based on a **5 KHz sampling rate**.

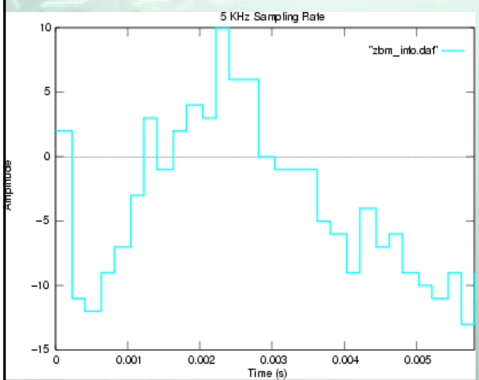
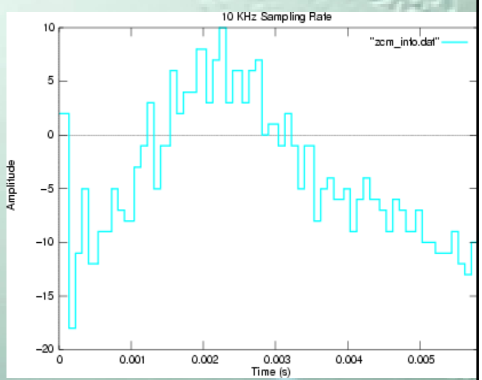


Figure 11.10: The reconstructed speech waveform, based on a **10 kHz sampling rate**.



11.3 Reconstructing Audio from Samples(3)

11.3.1. Oversampling

- **Undersampling** : Sampling at a rate lower than the minimum rate required ➔ dangerous to reconstruction
- **Oversampling** : Sampling at a rate higher than the minimum rate required ➔ good thing
- The faster we sample, the easier it is to reconstruct the original signal in a simple and inexpensive way.