Lecture 1: Introduction to Digital Communication Systems

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1 Overview

A communication system, block diagram of which depicted in Fig. 1, typically consists of a *source* and a *destination*. The goal of a communication system is to deliver a certain message (text, audio, image, video, etc.) to the destination over a noisy physical channel (copper wire, optical fiber, EM wave radiation, etc.).

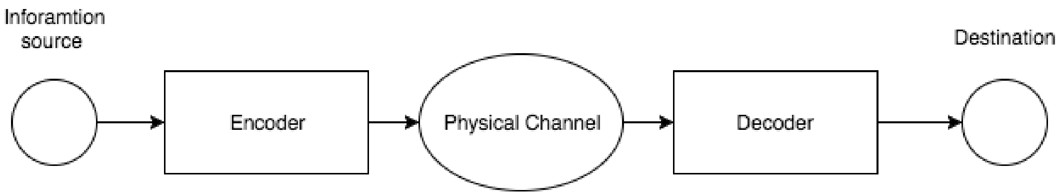

Figure 1: Basic Diagram of Communication System

Digital communication systems are communication systems that use digital sequence (typically binary digits, that is, bits) as an interface between the *source coding* part and the *channel coding* part. The block diagram is depicted in Fig. 2.

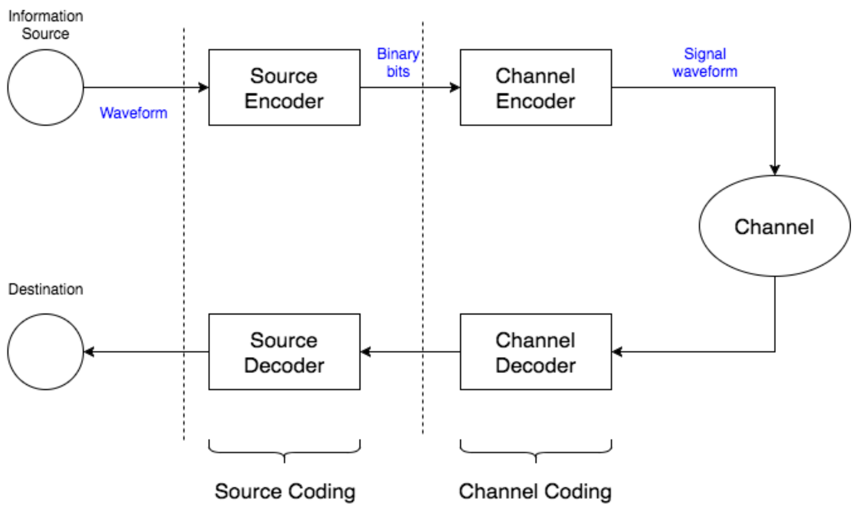

Figure 2: Basic Diagram of Digital Communication System

1. Source coding: The source encoder converts information waveforms to bits, while the decoder converts bits back to waveforms. The block diagram of source coding is depicted in Fig. 3.
2. Channel coding: The channel encoder converts bits to signal waveform, while the decoder converts received waveform back to bits. The block diagram of channel coding is depicted in Fig. 4.

Benefits of Separating Source & Channel with a binary interface:

- *Digital* Hardware - cheap, reliable, scalable
- Source / Channel Coding can be *independently* designed.
- Source-Channel Separation attains *optimal* transmission efficiency (Shannon's Theorem)

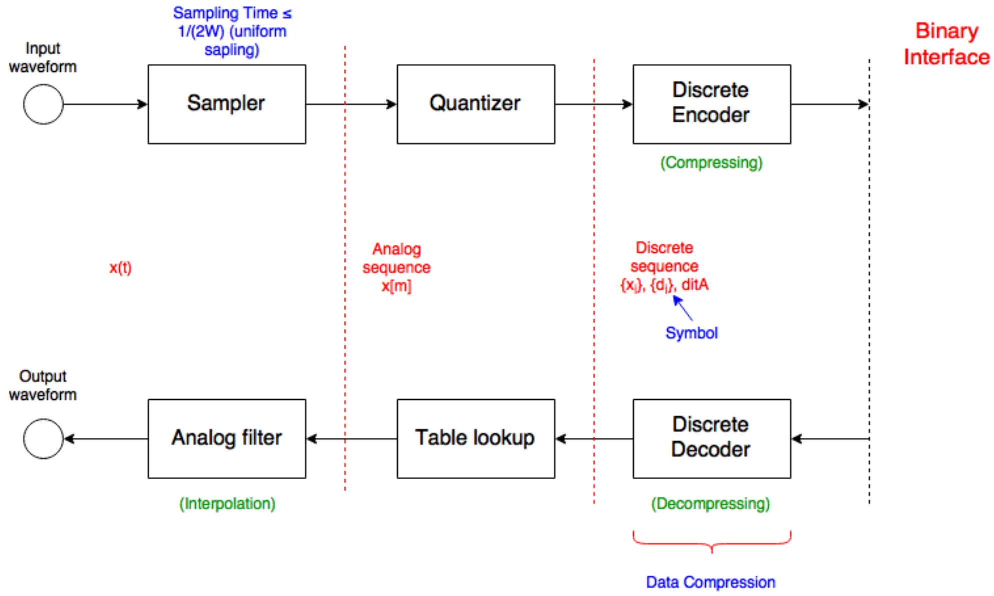

Figure 3: Basic Diagram of Source Coding

1.1 Source Coding

Source encoding aims to convert information waveforms (text, audio, image, video, etc.) into bits, the universal currency of information in the digital world. The three major steps are:

- Sampling: convert the continuous-time analog waveform to discrete-time sequence (but still continuous-valued).
- Quantization: convert each continuous-valued symbol to discrete-valued representatives.
- Data compression: remove the redundancy in the data and generate roughly i.i.d. uniformly distributed bits.

Source decoding does the reverse of encoding.

In this course, we focus on the channel coding part and will not go into details of source coding.

1.2 Channel Coding

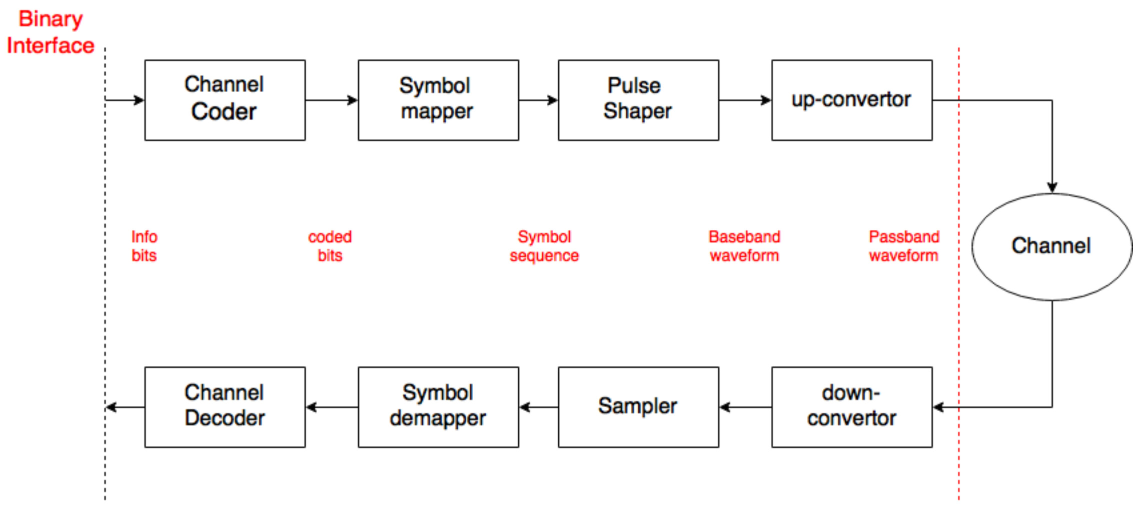

Figure 4: Basic Diagram of Channel Coding

Channel encoding aims to convert information bits into passband waveformsbits, the universal currency of information in the digital world. The four major steps are:

- Error correcting codes: introduce redundancy into the information bits and produce longer coded bits.

  Remark Example of "error correction":

  - Repetition code: each bit repeat N times
    Channel noise: flip the bit n.p. p, w.p.$(1-p)$ remain the same.
  - Bit Error happens when there are more than $\frac{N}{2}$ bit flips

$$\Pr\{\text{error}\} = 1 - \sum_{i=0}^{\lfloor \frac{N}{2} \rfloor} \binom{N}{i} p^i (1-p)^{n-i}$$

- Symbol mapping: map the coded bits to *constellation points*, each of which is a complex symbol.
- Pulse shaping: modulate the symbol to suitable *baseband* waveforms. There are some specific conditions needed to be satisfied, which will be discussed in later lectures.
- Up conversion: convert the baseband waveform to *passband* waveform, so that the effective frequency band follows the constraints from the physical world.

Channel decoding does the reverse of encoding.

2 From Analog to Digital and Back

From the above discussion, two key elements in digital communication systems can be recognized:

1. Conversion between (discrete-time) sequences and (continuous-time) waveforms

$$\begin{cases} \text{source coding: sampling/interpolation filter} \\ \text{channel coding: pulse shaping/sampling} \end{cases}$$

2. Conversion between bits and symbol sequences

$$\begin{cases} \text{source coding: quantization/table lookup} \\ \text{channel coding: symbol mapper/demapper} \end{cases}$$

You can find the similarity and differences between source and channel coding.

2.1 Conversion between Sequences & Waveforms via Orthogonal Expansion

Recall: we have learned two approaches in signals and systems

- For time-limited signals $x(t) : 0 \le t \le T$: Fourier Series
- For band-limited signals $x(f) : -W \le f \le W$: Sampling Theorem (Nyquist–Shannon sampling theorem)

	分解 Analysis ($x(t) \to x[m]$)	合成 Sythesis ($x[m] \to x(t)$)
Fourier Series	$x[m] = \frac{1}{T}\int_T x(t) e^{-\frac{j2\pi m}{T}t} \mathrm{d}t$	$x(t) = \sum_{m=-\infty}^{\infty} x[m] e^{j\frac{2\pi m}{T}t}$
Sampling Theorem	$x[m] = x(m \cdot \frac{1}{2W})$	$x(t) = \sum_{m=-\infty}^{\infty} x[m] \cdot \text{sinc}(2Wt - m)$

where

$$\text{sinc}(t) \triangleq \frac{\sin(\pi t)}{\pi t}$$

Signal Space Point of View

- Synthesis:

$$\{x[m]\} \rightarrow \boxed{\{\phi_m\}} \rightarrow x(t) = \sum_{m=-\infty}^{\infty} x[m] \cdot \phi_m(t)$$

From a linear algebra point of view,

$$\text{vector} \longleftrightarrow \text{waveform}$$
$$\text{basis} \longleftrightarrow \{\phi_m(t)\}$$

The basis of the functional space:

$$\text{Time-limited signals:} \quad \Phi = \{\phi_m(t)\} = \{e^{j\frac{2\pi m}{T}t}, m \in \mathbb{Z}\}$$
$$\text{band-limited signals:} \quad \Phi = \{\text{sinc}(2Wt - m), m \in \mathbb{Z}\}$$

In practice: one usually choose $\phi_m(t) = p(t - mT)$, where

$$p(\cdot) : \text{“Pulse” in Pulse Shaping}$$
$$x(t) = \sum_{m=-\infty}^{+\infty} x[m]p(t - mT)$$

Example

1. Sinc pulse: $p(t) = \text{sinc}(2Wt), W = \dfrac{1}{T}$
2. Raised cosine: $p(t) = \text{sinc}\dfrac{t}{T} \cdot \dfrac{\cos(\frac{\pi\beta t}{T})}{1 - \frac{4\beta t^2}{T^2}}$

- Analysis (Decomposition)

$$x(t) \rightarrow \boxed{\phi_m(t)} \rightarrow x[m]$$

If $\{\phi_m\}$ forms an orthnormal basis, then $x[m] =< x(t), \phi_m(t) >$.

  - Orthonormal:

$$< \phi_i, \phi_j >= \begin{cases} 0 & \text{if } i \neq j \\ 1 & \text{if } i = j \end{cases}$$

  - In the signal space, inner product is defined by “integral”.

$$< \phi_i, \phi_j > \triangleq \int_{-\infty}^{\infty} \phi_i(\tau)\phi_j^*(\tau)\mathrm{d}\tau$$