Digital Communications

Chapter 1. Introduction

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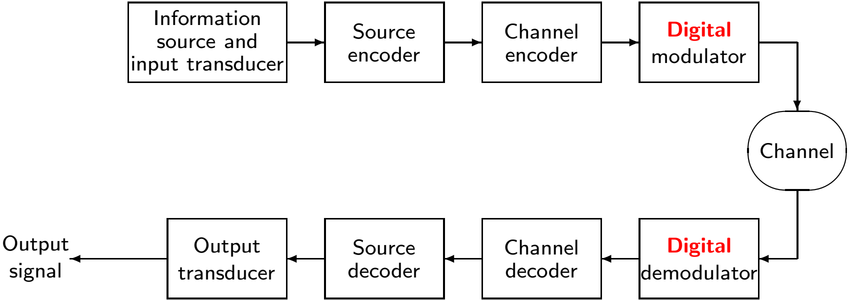
What we study in this course is:

- Theories of information transmission in digital form from one or more sources to one or more destinations.

1.1 Elements of digital communication system

Functional diagram of a digital communication system



Basic elements of a digital communication system

1.2 Comm channels and their characteristics

- **Physical channel media**
  - (magnetic-electrical signaled) Wireline channel
    - Telephone line, twisted-pair and coaxial cable, etc.
  - (modulated light beam) Fiber-optical channel
  - (antenna radiated) Wireless electromagnetic channel
    - ground-wave propagation, sky-wave propagation, line-of-sight (LOS) propagation, etc.
  - (multipath) Underwater acoustic channel
  - . . . etc.
- **Virtual channel**
  - Storage channel
    - Magnetic storage, CD, DVD, etc.

1.2 Comm channels and their characteristics

- **Channel impairments**
  - Thermal noise (additive noise)
  - Signal attenuation
  - Amplitude and phase distortion
  - Multi-path distortion

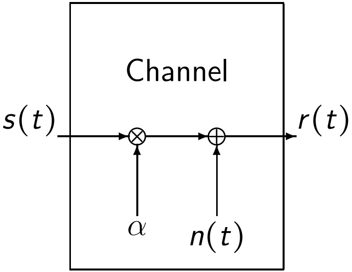
- **Limitations of channel usage**
  - Transmission power
  - Receiver sensitivity
  - Bandwidth
  - Transmission time

1.3 Math models for communication channels

Additive noise channel (with attenuation)

In studying these channels, a mathematical model is necessary.

$$r(t) = \alpha s(t) + n(t)$$

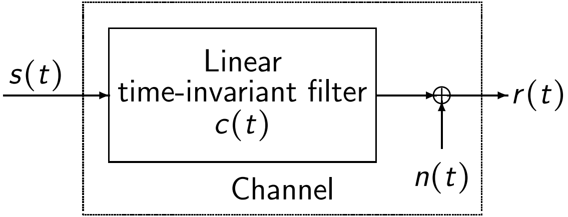


where

- α is the attenuation factor
- $s(t)$ is the transmitted signal
- $n(t)$ is the additive random noise (a random process, usually Gaussian)

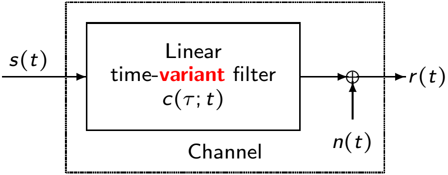
Linear filter channel with additive noise

To meet the specified bandwidth limitation



$$\begin{aligned} r(t) &= s(t) \star c(t) + n(t) \\ &= \int_{-\infty}^{\infty} c(\tau) s(t-\tau)\, d\tau + n(t) \end{aligned}$$

Linear time-variant (LTV) filter channel with additive noise

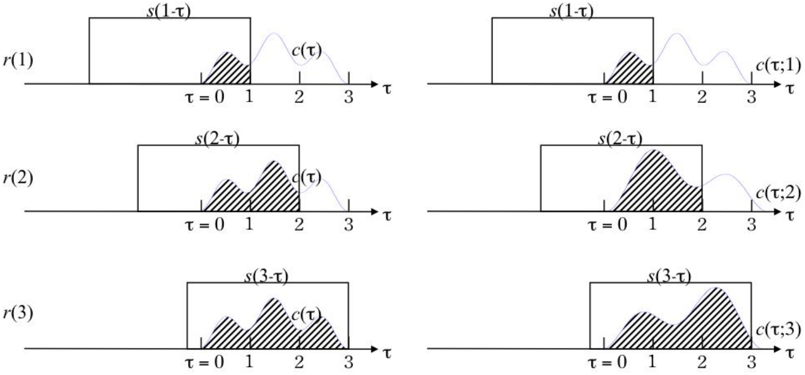


$$\begin{aligned} r(t) &= s(t) \star c(\tau;t) + n(t) \\ &= \int_{-\infty}^{\infty} c(\tau;t) s(t-\tau)\, d\tau + n(t) \end{aligned}$$

- τ is the argument for filtering.
- t is the argument for time-dependence.
- The time-invariant filter can be viewed as a special case of the time-variant filter.

Assume $n(t) = 0$ (noise-free).

LTI versus LTV



1.3 Math models for communication channels

LTV filter channel with additive noise

$c(\tau; t)$ usually has the form

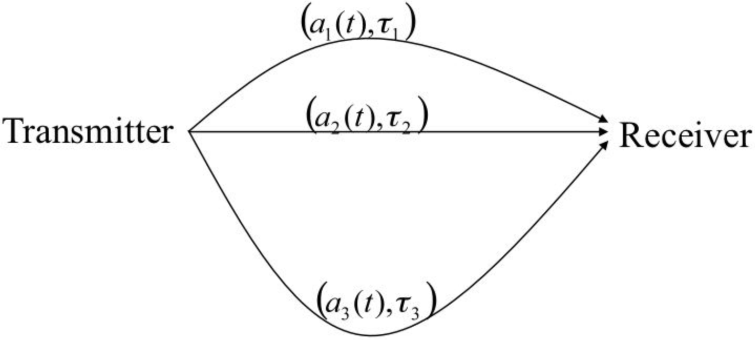
$$c(\tau; t) = \sum_{k=1}^{L} a_k(t)\delta(\tau - \tau_k)$$

where

- $\{a_k(t)\}_{k=1}^{L}$ represent the possibly time-varying attenuation factor for the L multipath propagation paths
- $\{\tau_k\}_{k=1}^{L}$ are the corresponding time delays.

Hence

$$r(t) = \sum_{k=1}^{L} a_k(t)s(t - \tau_k) + n(t)$$



$$r(t) = a_1(t)s(t-\tau_1) + a_2(t)s(t-\tau_2) + a_3(t)s(t-\tau_3) + n(t)$$

1.4 A historical perspective in the developement of digital communications

- Morse code (1837)
    - Variable-length binary code for telegraph
- Baudot code (1875)
    - Fixed-length binary code of length 5
- Nyquist (1924)
    - Determine the maximum signaling rate without intersymbol interference over, e.g., a telegraph channel

1.4 A historical perspective in the developement of digital communications - Nyquist rate

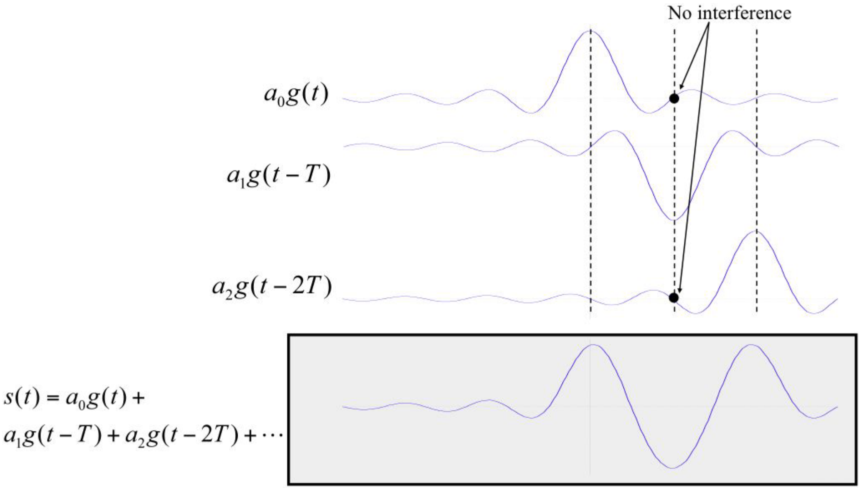
- Nyquist (1924)
  - Define basic pulse shape $g(t)$ that is bandlimited to W.

  - One wishes to transmit $\{-1, 1\}$ signals in terms of $g(t)$, or equivalently, one wishes to transmit a_0, a_1, a_2, ... in $\{-1, 1\}$ in terms of $s(t)$ defined as

$$s(t) = a_0 g(t) + a_1 g(t - T) + a_2 g(t - 2T) + \cdots$$

1.4 A historical perspective in the developement of digital communications - Nyquist rate

Example. $(a_0, a_1, a_2, \ldots) = (+1, -1, +1, \ldots)$.



1.4 A historical perspective in the developement of digital communications - Nyquist rate

- Question that Nyquist shoots for:
  - What is the maximum rate that the data can be transmitted under the constraint that $g(t)$ causes no intersymbol interference (at the sampling instances)?

    Answer : $2W$ pulses/second. (Not $2W$ bits/second!)

  - What $g(t)$ can achieve this rate?
    Answer :
    $$g(t) = \frac{\sin(2\pi Wt)}{2\pi Wt}$$
  - Conclusion:
    - A bandlimited-to-W basic pulse shape signal (or symbol) can convey at most $2W$ pulses/second (or symbols/second) without introducing inter-pulse (or inter-symbol) interference.

1.4 A historical perspective in the developement of digital communications - Sampling theorem

- Shannon (1948)
  - Sampling theorem
    - A signal of bandwidth W can be reconstructed from samples taken at the Nyquist rate ($= 2W$ samples/second) using the interpolation formula

$$s(t) = \sum_{n=-\infty}^{\infty} s\left(\frac{n}{2W}\right) \times \left(\frac{\sin[2\pi W(t - n/(2W))]}{2\pi W(t - n/(2W))}\right).$$

1.4 A historical perspective in the developement of digital communications - Shannon's channel coding theorem

- Channel capacity of additive white Gaussian noise

$$C = W \log_2 \left(1 + \frac{P}{WN_0}\right) \quad \text{bits/second}$$

where $\begin{cases} W \text{ is the bandwidth of the bandlimited channel,} \\ P \text{ is the average transmitted power,} \\ N_0 \text{ is single-sided noise power per hertz.} \end{cases}$

- Shannon's channel coding theorem
  - Let R be the information rate of the source. Then
    - if $R < C$, it is theoretically possible to achieve reliable (asymptotically error-free) transmission by appropriate coding;
    - if $R > C$, reliable transmission is impossible.

  This gives birth to a new field named Information Theory.

1.4 A historical perspective in the developement of digital communications

- Other important contributions
  - Harvey (1928), based on Nyquists result, concluded that
    - a maximum reliably transmitted data rate exists for a bandlimited channel under maximum transmitted signal amplitude constraint and minimum transmitted signal amplitude resolution constraint.
  - Kolmogorov (1939) and Wiener (1942)
    - Optimum linear (Kolmogorov-Wiener) filter whose output is the best mean-square approximation to the desired signal $s(t)$ in presence of additive noise
  - Kotenikov (1947), Wozencraft and Jacobs (1965)
    - Use geometric approach to analyze various coherent digital communication systems.
  - Hamming (1950)
    - Hamming codes

1.4 A historical perspective in the developement of digital communications

- Other important contributions (Continue)
  - Muller (1954), Reed (1954), Reed and Solomon (1960), Bose and Ray-Chaudhuri (1960), and Goppa (1970,1971)
    - New block codes, such as Reed-Solomon codes, Bose-Chaudhuri-Hocquenghem (BCH) codes and Goppa codes.
  - Forney (1966)
    - Concatenated codes
  - Chien (1964), Berlekamp (1968)
    - Berlekamp-Massey BCH-code decoding algorithm

1.4 A historical perspective in the developement of digital communications

- Other important contributions (Continue)
  - Wozencraft and Reiffen (1961), Fano (1963), Zigangirov (1966), Jelinek (1969), Forney (1970, 1972, 1974) and Viterbi (1967, 1971)
    - Convolusional code and its decoding
  - Ungerboeck (1982), Forney *et al.* (1984), Wei (1987)
    - Trellis-coded modulation
  - Ziv and Lempel (1977, 1978) and Linde *et al.* (1980)
    - Source encoding and decoding algorithms, such as Lempel-Ziv code
  - Berrou *et al.* (1993)
    - Turbo code and iterative decoding

1.4 A historical perspective in the developement of digital communications

- Other important contributions (Continue)
  - Gallager (1963), Davey and Mackay (1998)
    - Low-density parity-check (LDPC) code and the sum-product decoding algorithm

What you learn from Chapter 1

- Mathematical models of
  - time-variant and time-invariant additive noise channels
  - multipath channels
- Nyquist rates and Sampling theorem