Digital Communications

Chapter 1. Introduction

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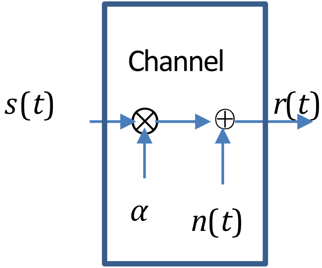
Additive noise channel (with attenuation)

In studying these channels, a mathematical model is necessary.

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



where

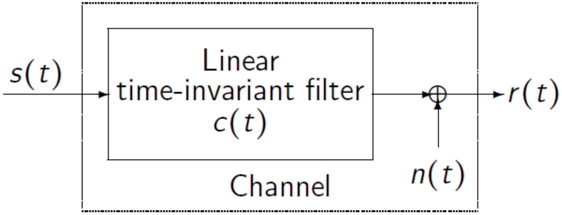
- α is the attenuation factor
- $s(t)$ is the transmitted signal
- $n(t)$ is the additive random noise (a random process, usually Gaussian)

1.3 Math models for communication channels

Linear filter channel with additive noise

To meet the specified bandwidth limitation

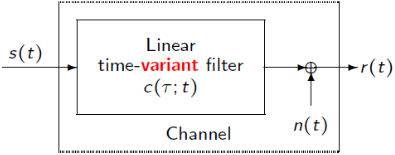
Wireline telephone channel



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

1.3 Math models for communication channels

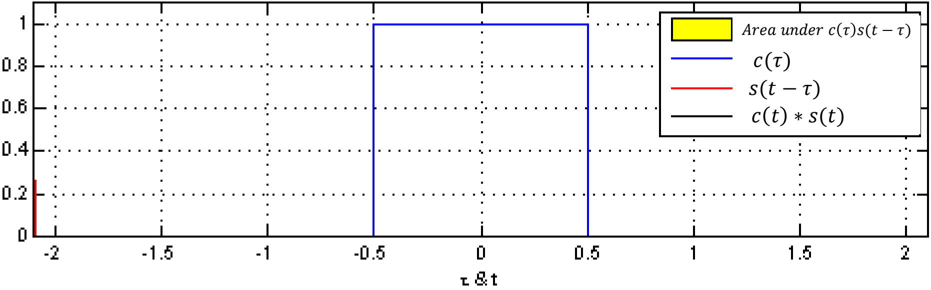
Linear time-variant (LTV) filter channel with additive noise

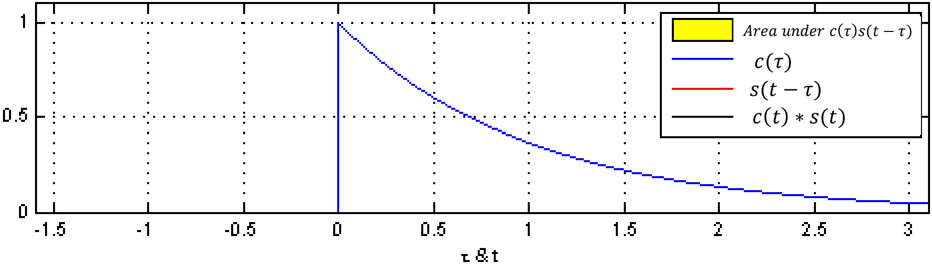


- Underwater acoustic
- Ionospheric radio (Freq. <30M)
- Mobile cellular radio

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

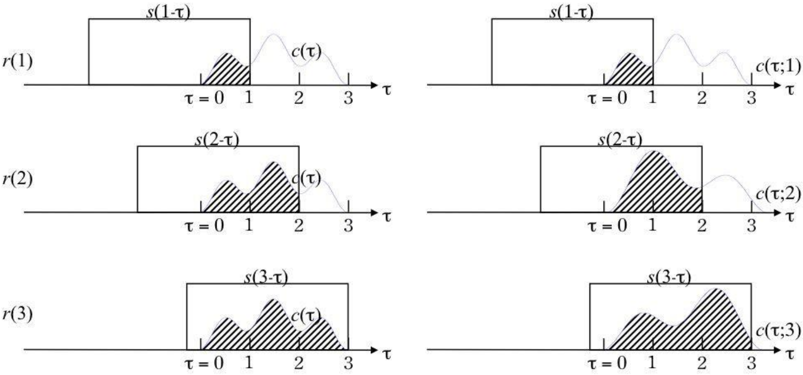
- τ is the argument for filtering. “age ” variable
- t is the argument for time-dependence.
- The time-invariant filter can be viewed as a special case of the time-variant filter.


Area under $c(\tau)s(t-\tau)$
$c(\tau)$
$s(t-\tau)$
$c(t)*s(t)$
τ & t


Area under $c(\tau)s(t-\tau)$
$c(\tau)$
$s(t-\tau)$
$c(t)*s(t)$
τ & t

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

LTI versus LTV



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

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

where

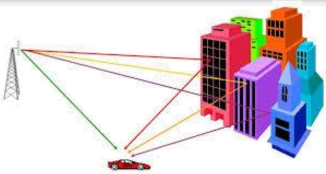
- $\{a_k(t)\}_{k=1}^{L}$ represent the possibly time-varying attenuation factor for the L multipath propagation paths
- $\{\tau_k\}_{k=1}^{L}$ are the corresponding time delays.

Hence

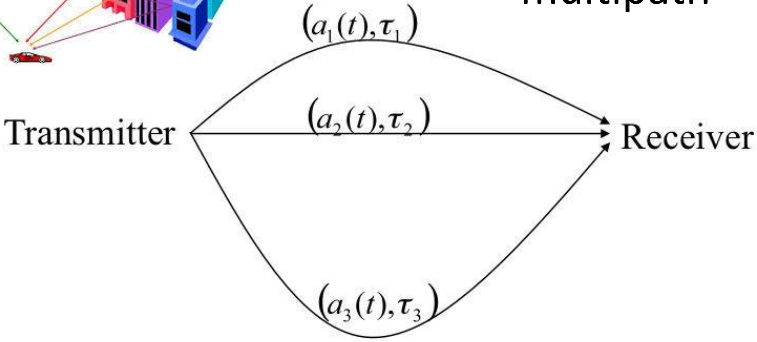
$$r(t) = \sum_{k=1}^{L} a_k(t)s(t - \tau_k) + n(t)$$

1.3 Math models for communication channels

Time varying multipath fading channel



multipath



Three models

- Linear Time-Invariant Filter Channel (AWGN)

$$r(t) = s(t) * c(t) + n(t)$$

- Linear Time-Variant Filter Channel

$$r(t) = s(t) * c(t;\tau) + n(t)$$

- Multipath propagation

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

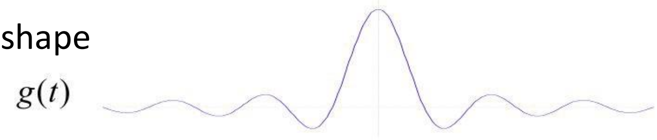
1.4 A historical perspective in the developement of digital communications

- Morse code (1837)
  - Variable-length binary code for telegraph
    - More frequent word uses shorter length, less frequent word uses longer length.
    - Dashes and dots
- Baudot code (1875)
  - Fixed-length binary code of length 5
  - Mark and space
- Nyquist (1924)
  - Determine the maximum signaling rate without inter-symbol 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.

Basic pulse shape

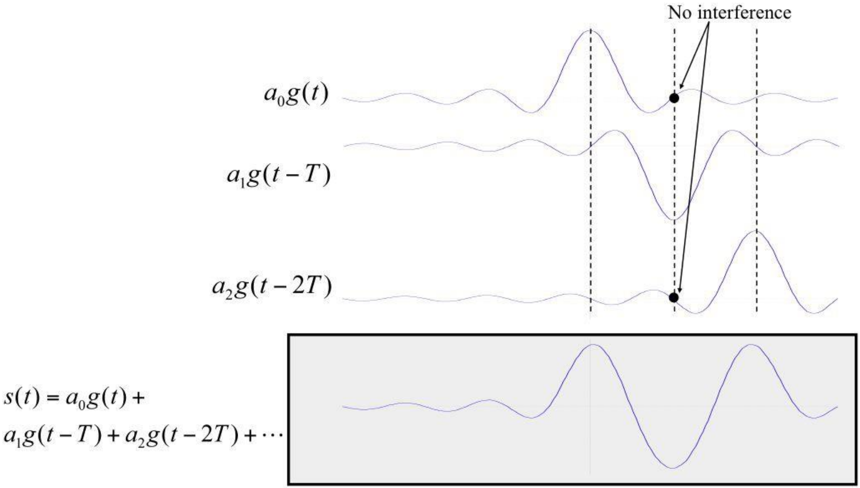


- One wishes to transmit $\{-1, 1\}$ signals in terms of $g(t)$, or equivalently, one wishes to transmit $a_0, a_1, a_2, \ldots$ in $\{-1, 1\}$ in terms of $s(t)$ defined as

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

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)?

- 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

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

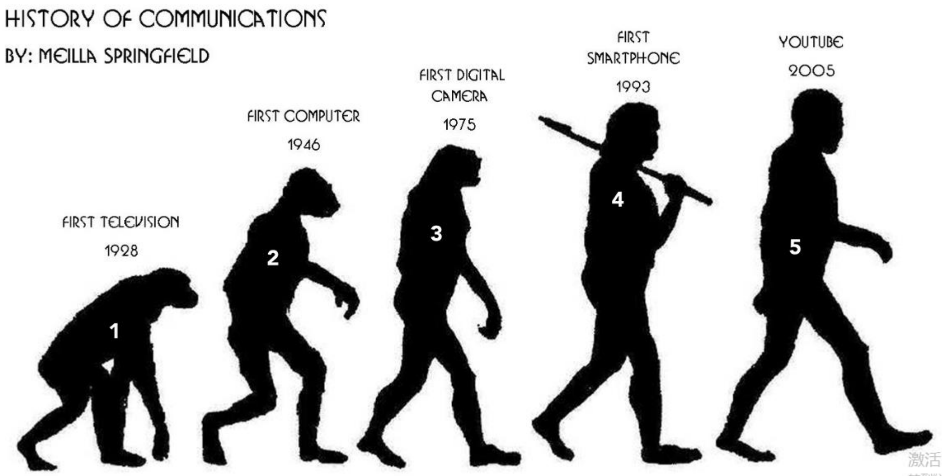
W is the bandwidth of the bandlimited channel,

where P is the average transmitted power,

N_0 is single-sided noise power per hertz.

- 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.


HISTORY OF COMMUNICATIONS
BY: MEILLA SPRINGFIELD
FIRST TELEVISION
1928
FIRST COMPUTER
1946
FIRST DIGITAL CAMERA
1975
FIRST SMARTPHONE
1993
YOUTUBE
2005
1
2
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1.4 A historical perspective in the developement of digital communications

- Other important contributions
  - Hartley (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.

He later worked at Bell Laboratories. He performed research on repeaters and voice and carrier transmission and formulated the law "that the total amount of information that can be transmitted is proportional to frequency range transmitted and the time of the transmission." His 1928 paper is considered as "the single most important prerequisite" for Shannon's theory of information.[2] After about 10 years of illness he returned to Bell Labs in 1939 as a consultant.

- Other important contributions
  - 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.
    - Kolmogorov： In his study of stochastic processes, especially Markov processes, Kolmogorov and the British mathematician Sydney Chapman independently developed the pivotal set of equations in the field, which have been given the name of the Chapman–Kolmogorov equations.
    - Wiener was an American mathematician and philosopher. He was a professor of mathematics at the Massachusetts Institute of Technology (MIT). A child prodigy, Wiener later became an early researcher in stochastic and mathematical noise processes, contributing work relevant to electronic engineering, electronic communication, and control systems.

- Other important contributions
  - Kotenikov (1947), Wozencraft and Jacobs (1965)
    - Use geometric approach to analyze various coherent digital communication systems.
    - Kotenikov was an information theory and radar astronomy pioneer from the Soviet Union.
  - Hamming (1950)
    - Hamming codes： combat the detrimental effects of the channel
    - Hamming codes can detect up to two-bit errors or correct one-bit errors without detection of uncorrected errors. By contrast, the simple parity code cannot correct errors, and can detect only an odd number of bits in error. Hamming codes are perfect codes, that is, they achieve the highest possible rate for codes with their block length and minimum distance of three.[1] Richard W. Hamming invented Hamming codes in 1950 as a way of automatically correcting errors introduced by punched card readers. In his original paper, Hamming elaborated his general idea, but specifically focused on the Hamming(7,4) code which adds three parity bits to four bits of data.[2]

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 code
    - the sum-product decoding algorithm
- Gallager's 1960 Sc.D. thesis, on low-density parity-check codes, was published by the MIT Press as a monograph in 1963.[4] The codes, which remained useful over 50 years, are sometimes called "Gallager codes".[5] An abbreviated version appeared in January 1962 in the IRE Transactions on Information Theory and was republished in the 1974 IEEE Press volume, Key Papers in The Development of Information Theory, edited by Elwyn Berlekamp. This paper won an IEEE Information Theory Society Golden-Jubilee Paper Award in 1998 and its subject matter is a very active area of research today. Gallager's January 1965 paper in the IEEE Transactions on Information Theory, "A Simple Derivation of the Coding Theorem and some Applications", won the 1966 IEEE W.R.G. Baker Award "for the most outstanding paper, reporting original work, in the Transactions, Journals and Magazines of the IEEE Societies, or in the Proceedings of the IEEE"[6] and also won another IEEE Information Theory Society Golden-Jubilee Paper Award in 1998. His book, Information Theory and Reliable Communication, Wiley 1968, placed Information Theory on a sound mathematical foundation and is still considered by many as the standard textbook on information theory.

- https://marconisociety.org/fellows/

What you learn from Chapter 1

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