S-72.1140 Transmission Methods in Telecommunication Systems (5 cr)

Introduction

S-72.1140 Transmission Methods in Telecommunication Systems (5 cr)

- Lectures take place on Tuesdays, 14-16, in hall S4
- Tutorials in Finnish on Wednesdays at 12-14 in hall S4 3.10., 10.10., 17.10. and 24.10.
- Tutorials in English on Wednesdays at 14 -16 in hall S5 3.10., 10.10., 17.10. and 24.10.
- Lecturer: Timo Korhonen, Otakaari 7, room 214, 2. Floor, tel. 09 451 2351
- Tutorial assistants: Seppo Saastamoinen (seppo.saastamoinen@hut.fi), Naser Tarhuni (ntarhuni@cc.hut.fi)
- Much of materials in printed (or downloadable) handouts
- References:
  – J. G. Proakis, et al: Contemporary Communication Systems using Matlab, II ed. (some theory and a lot of Matlab exercises)
  – A. B. Carlson: Communication Systems IV ed. (basic theory)
  – B. P. Lathi: Modern Digital and Analog Communication Systems III ed. (basic theory but more text than in Carlson's book)

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S-72.1140 Transmission Methods in Telecommunication Systems (5 cr)

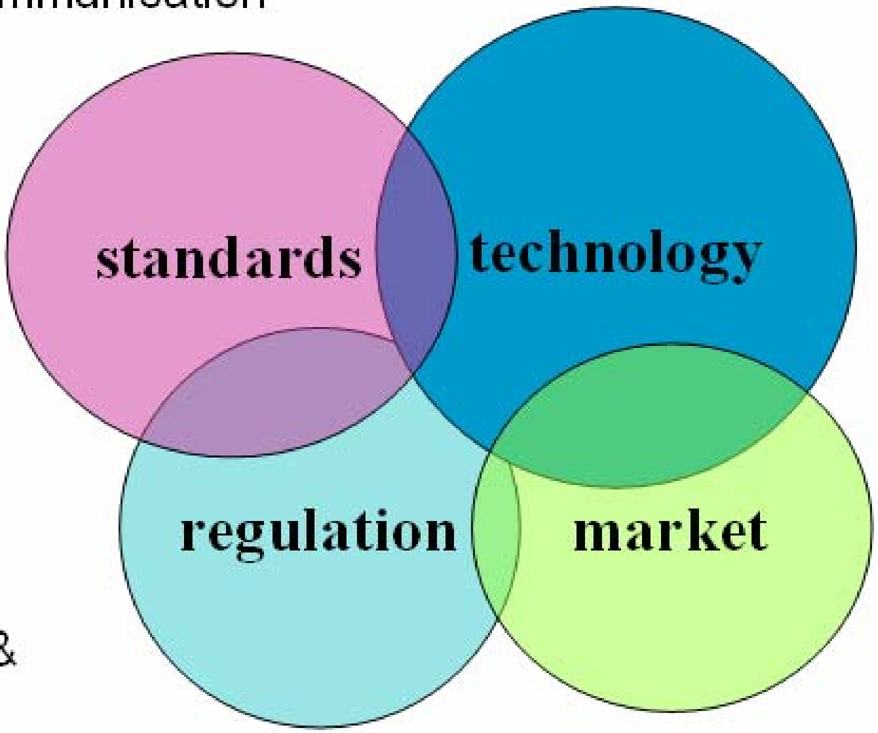
- References … cont.
  - Duck, Read: Data Communications and Computer Networks, II ed. (simple to follow)
  - A Leon-Garcia et al: Communication Networks, 2nd ed
- Grading: Closed-book exam. (Return of your Matlab exercises can increase your grade by the maximum of 0.6 and participation to tutorials by 0.4)
- NOTE: About 50% of exam based directly on tutorials
- Homepage: **http://www.comlab.hut.fi/studies/1140**
- **Follow the course homepage and Eeforum (http://tltpc12.hut.fi/cgi-bin/mwf/forum.pl) for the very latest course info!**

Overview

- Role of regulation and market
- Fundamental concepts
  - information
  - bandwidth
  - data rate
  - signal power and energy in time and frequency domain
- Telecommunication channels
- Telecommunication systems and their basic features
- Reassuring communication quality & correctness / channel adaptation
    - modulation
    - coding
      - source
      - channel

Network Products/Services and Market

- Requirements for successful telecommunication product/service development:
  - cost effective implementation
  - regulations must allow and support implementation
  - there must be a market
    - already existing
    - create a market!
  - for commercial implementation
    - designer should know & listen customers
    - designer must understand technology
    - there must be usability in design
    - output should be applicable/fashionable
    - outputs should be marketed appropriately



Ref: [6, Part II] pp. 43

Assignment

1. Commercial feasibility of telecommunication products is determined by standardization, technology, regulation and market – factors.

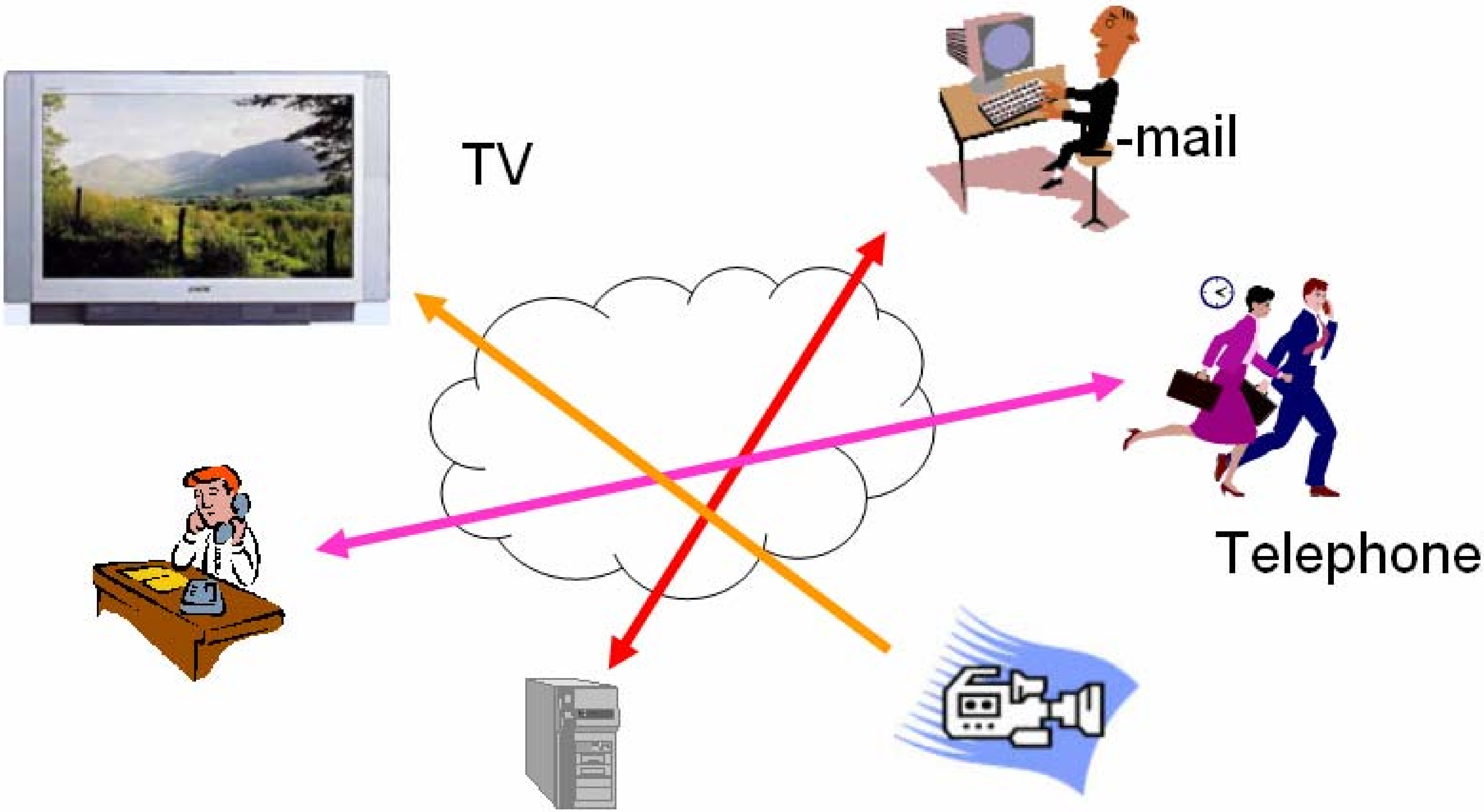
(i) List some relating standards and technologies!

(ii) What is the difference of standardization and regulation?

(iii) How the effect of market can be quantified?

Telecommunications

- Digital transmission enables networks to support many services



Transmitting Information by Telecommunication Systems

- Telecommunication systems utilize varying **currents, voltages** (eg. varying electromagnetic fields) to store and convey information
- Telecommunication systems are dedicated to **transporting information** for instance from point to point (unicast) or point to multi-point (multicast) using links and networks
- Telecommunication messages are transmitted via various **medium** (or **media**) as by
  - copper wires (twisted cable pairs, coaxial cables...
  - Microwave beams and wave guides
  - optical fibers and free-space radiators
- Signal is **adapted to the transmission** and medium by *modulation and coding* (=adaptation to physical transmission)
- Modulation/coding method must be selected as dictated by
  - medium
  - information sources (statistics, rate ... - QoS point of view)

Questions of Interest

- How long will it take to transmit a message?
  - How many bits are in the message (text, image)?
  - How fast does the network/system transfer information?
- Can a network/system handle a voice (video) call?
  - How many bits/second does voice/video require? At what quality?
- How long will it take to transmit a message without errors?
  - How are errors introduced?
  - How are errors detected and corrected?
- What transmission speed is possible over radio, copper cables, fiber, infrared, …?

Telecommunication Systems Transmit Information -What is Information?

- Consider two sentences, which one carries more information?
  - The sun will rise tomorrow.
  - There will be a tornado tomorrow.
- The measure of information is its probability: If an event x_i has the probability $P(x_i) = P_i$, it has the **self-information**

$$I_i \equiv -\log_b P_i = \log_b \frac{1}{P_i}$$

- Note that self-information is 0 for $P_i = 1$. The properties of self-information can be summarized as

$$\begin{cases} I_i \geq 0, 0 \leq P_i \leq 1 \\ I_i > I_j, P_i < P_j \end{cases} \qquad \begin{cases} P_i \to 0, I_i \to \infty \\ I_{ij} = \log_b \dfrac{1}{P_i P_j} = \log_b P_i^{-1} + \log_b P_{ij}^{-1} = I_i + I_j \end{cases}$$

- Note that for binary, equal probable symbols

$$P(x_1) = P(x_2) = 1/2, I_1 = I_2 = \log_2 2 = 1\text{bit}$$

Ref: [2] Chapter 16.1

Assignment

2. The average amount of information / time unit for a source emitting M kind of symbols is called as source entropy

$$H(P) = \sum_{i=1}^{M} P_i I_i = \sum_{i=1}^{M} P_i \log_2 \frac{1}{P_i} \qquad \text{bits/symbol,}$$

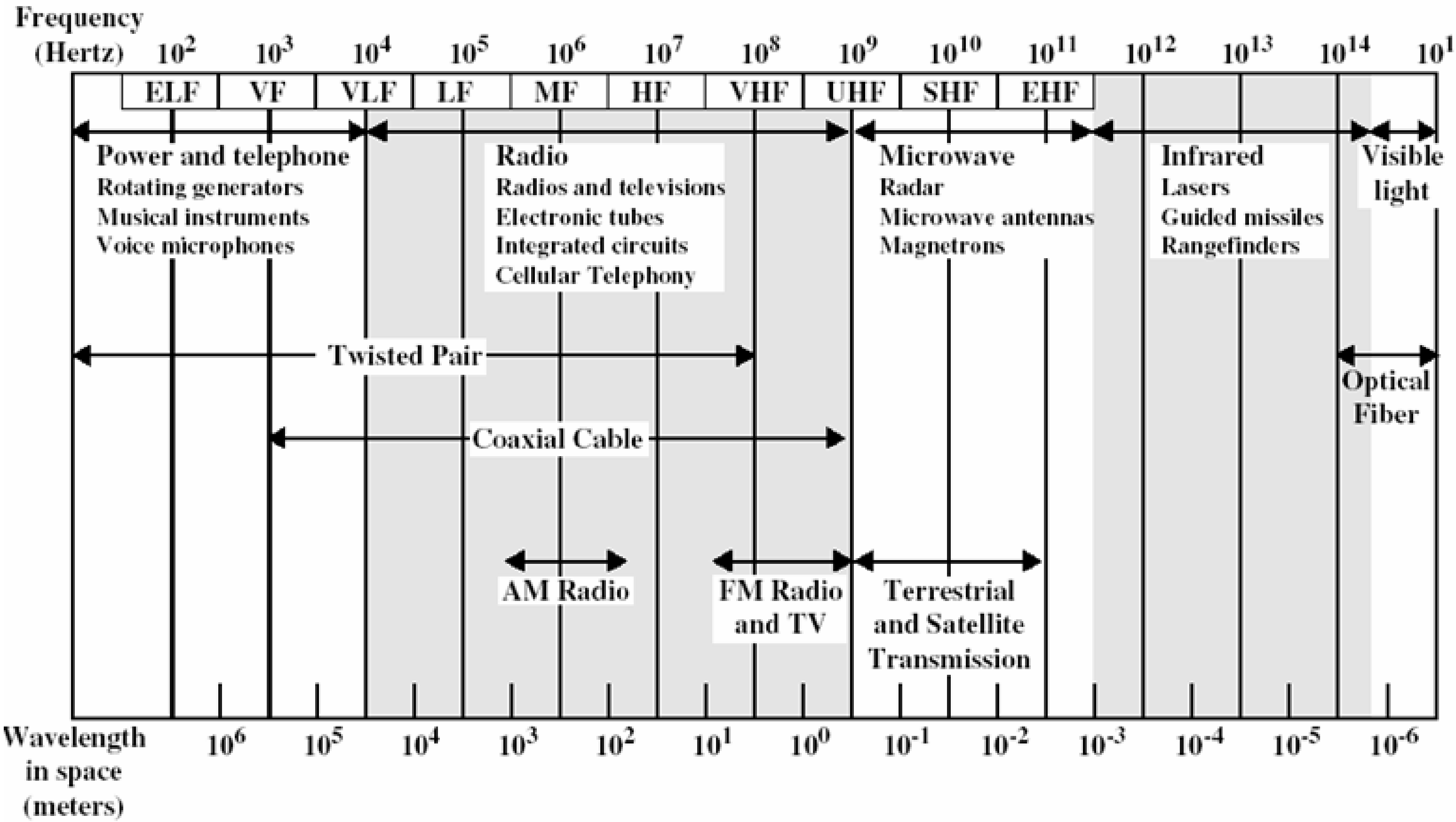
where the symbol probabilities P_i are defined such that $\sum P_i = 1$.

How much information is convoyed by M symbols having alphabet probabilities of

(i) {1/2, 1/2}
(ii) {1/4, 3/4}
(iii) {1/3, 1/3, 1/3}
(iv) {1/2, 1/4, 1/4}

Conclusions?

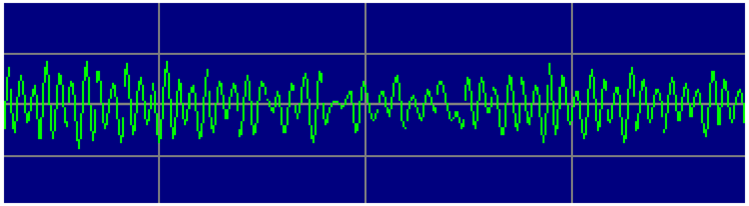
Mediums and Electromagnetic Spectra [5]



ELF = Extremely low frequency
VF = Voice frequency
VLF = Very low frequency
LF = Low frequency
MF = Medium frequency
HF = High frequency
VHF = Very high frequency
UHF = Ultrahigh frequency
SHF = Superhigh frequency
EHF = Extremely high frequency

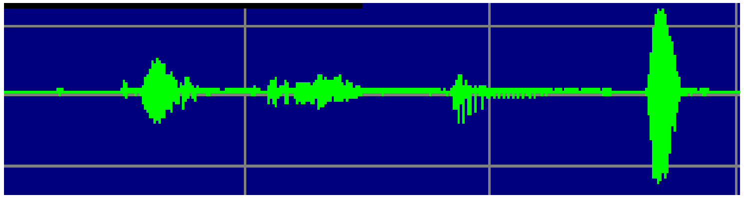
Analog Signals

- A set of voice tones:



  – Several tones superimposed (added)
  – Tones can not be separated from the time domain representation
  – **Frequency components** can be separated from frequency domain representation

- *"This is some speech"*



  – Bursts
  – Amplitude varies
  – Frequency (phase) varies
  – Many other practical sources are bursty as
    - video signals
    - Ethernet data packets

- Often analog sources are digitized for transmission that carries several benefits as
  – error correction & detection
  – easy multiplexing
  – easy adaptivity

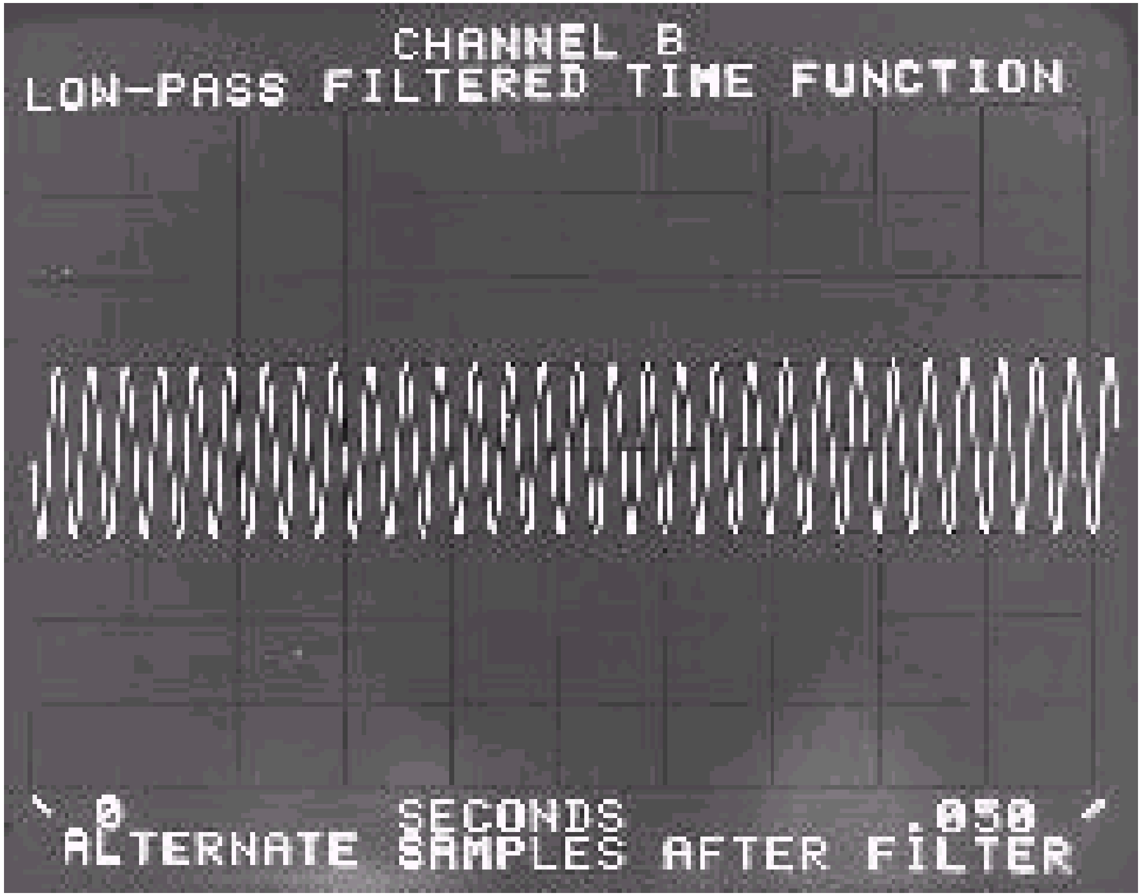
Classification of Signals

- Deterministic signals
- Random signals; pure or pseudo-random
- Energy signals; pulses
- Power signal; periodic
- Continuous time - discrete time:

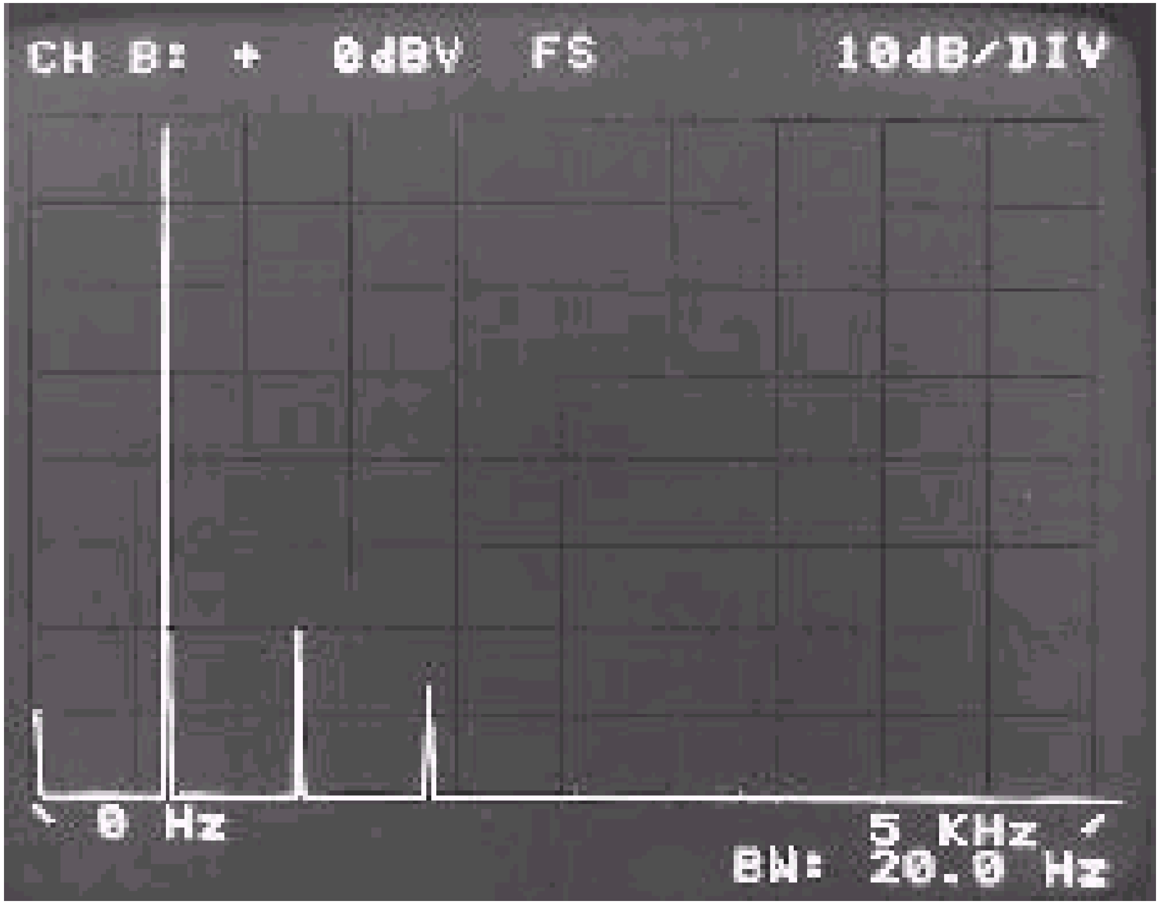
$$x(t), x[n] = x(nT_S), X(f), X[k]$$

- Analog - digital
- Real - complex
- Time variable (Average power) - constant (DC-level)

Time Domain Representation Can Only Seldom Reveal Small Signal Impairments

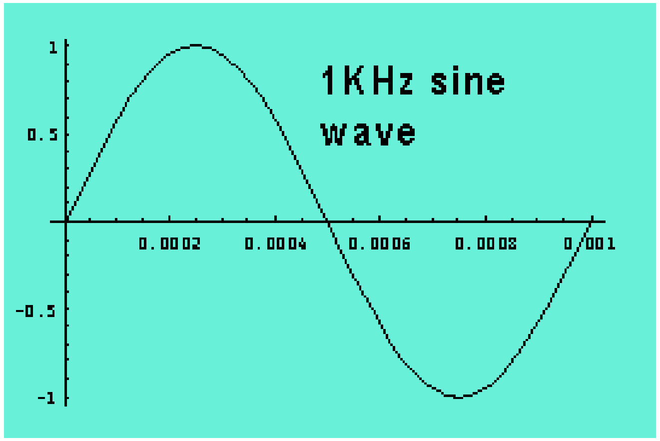


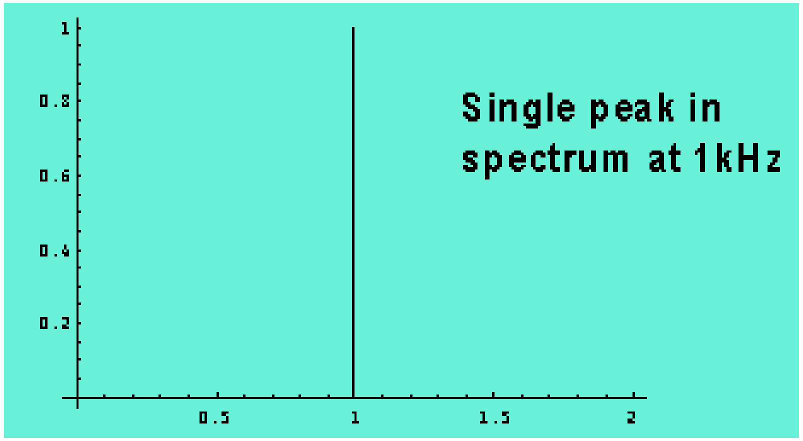
Frequency Domain Representation of the Same Signal Reveals More!

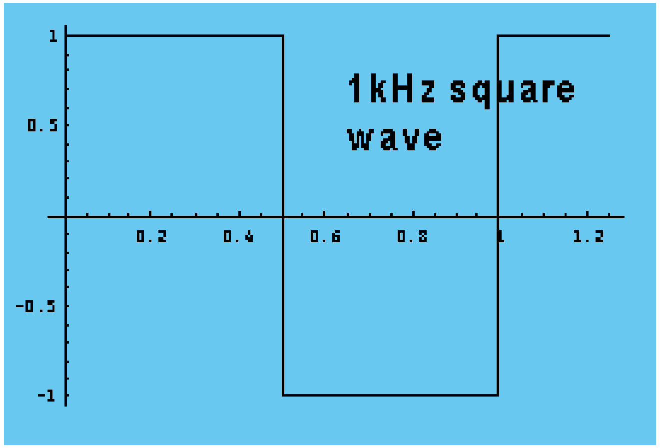


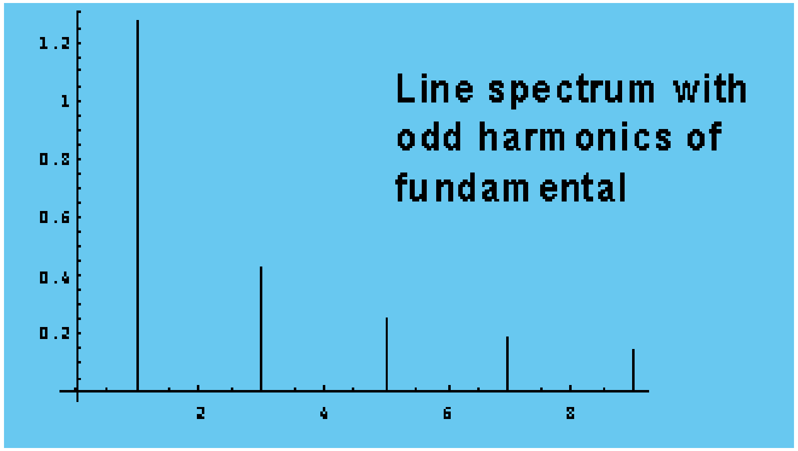
Examples of Signal Spectra

- All finite signals have spectra that can be determined via Fourier transformation (pulses) or Fourier series (periodic signals)







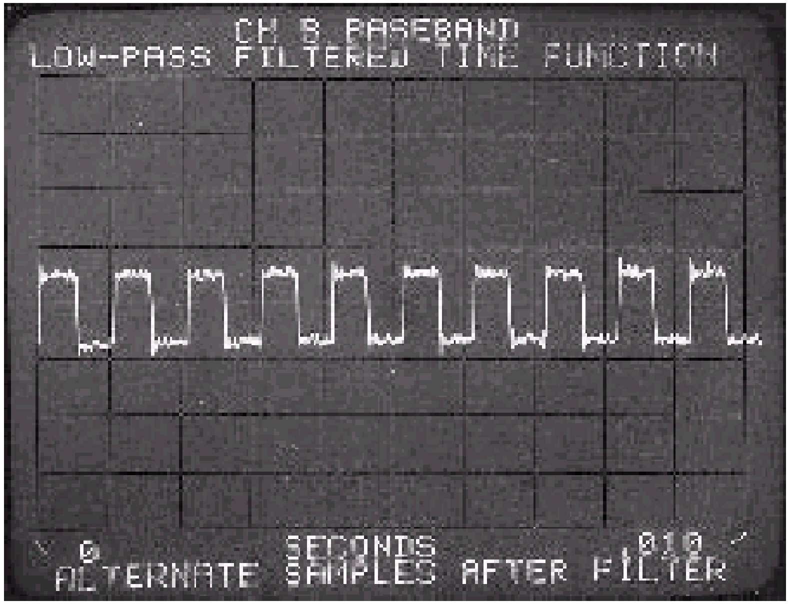


Ref: [2] Chapter 2

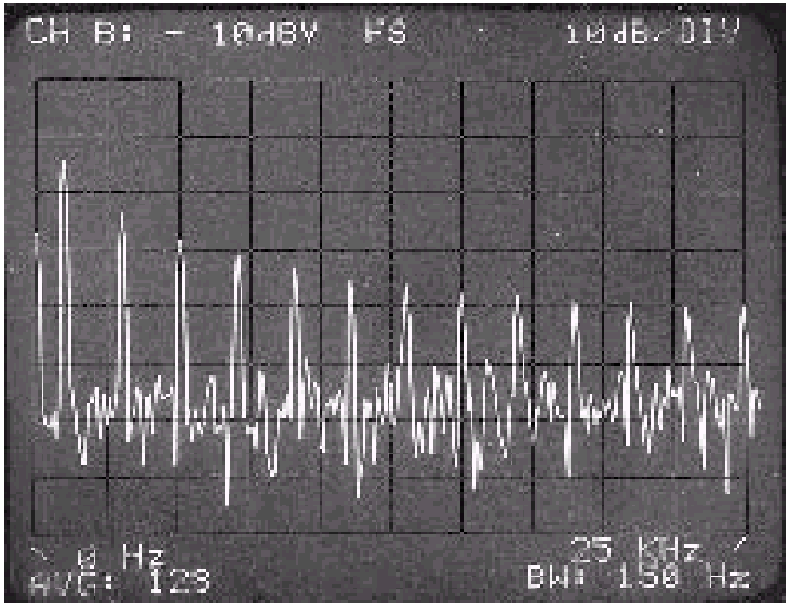
Noise and Interference

- In practical communication systems signals are blurred by noise and interference:

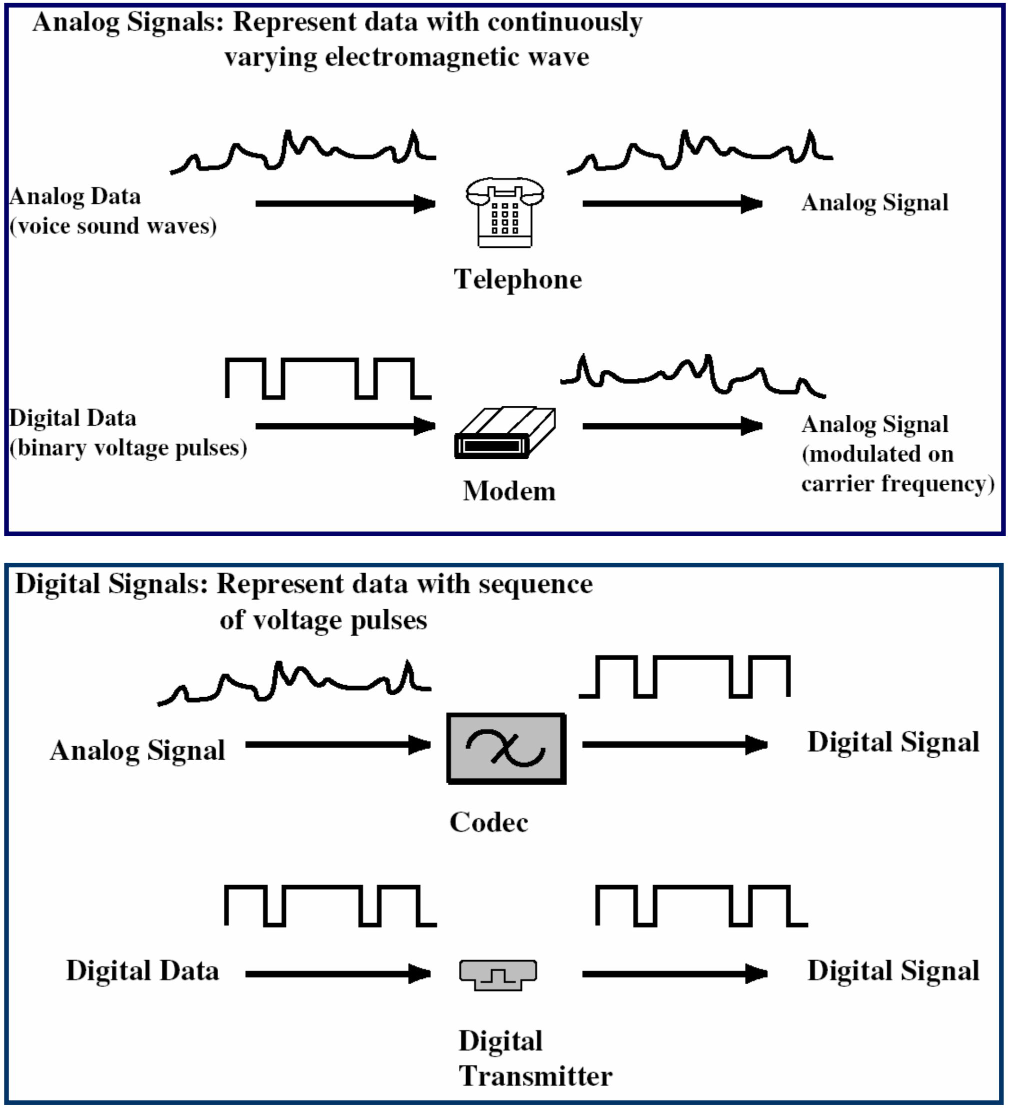
Time domain



Frequency domain



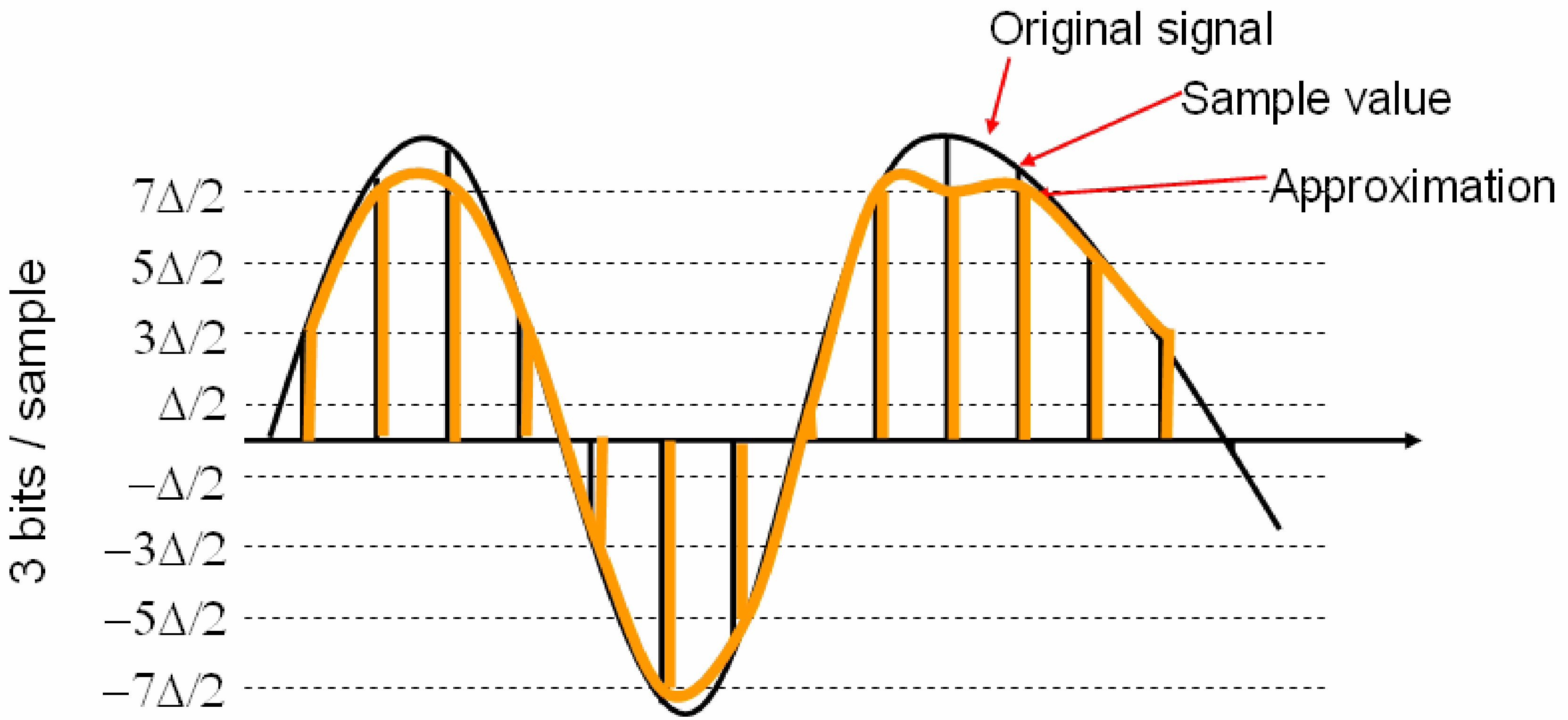
Analog and Digital Signals [5]



Why the signals in the figure are analog or digital?

Digitization of Analog Signal – Why?

- Sample analog signal in time and amplitude
- Find closest approximation



R_s = Bit rate = # bits/sample x # samples/second

Ref: [7]

Bit Rate of Digitized Signal

- Bandwidth W_s Hertz: how fast the signal changes
  - Higher bandwidth → more frequent samples
  - Minimum sampling rate = 2 x W_s

- Representation accuracy: range of approximation error
  - Higher accuracy

  → smaller spacing between approximation values

  → more bits per sample

Example: Voice & Audio

Telephone voice

- W_s = 4 kHz → 8000 samples/sec
- 8 bits/sample
- R_s=8 x 8000 = 64 kbps
- Cellular phones use more powerful compression algorithms: 8-12 kbps

CD Audio

- W_s = 22 kHz → 44000 samples/sec
- 16 bits/sample
- R_s=16 x 44000= 704 kbps per audio channel
- MP3 uses more powerful compression algorithms: 50 kbps per audio channel

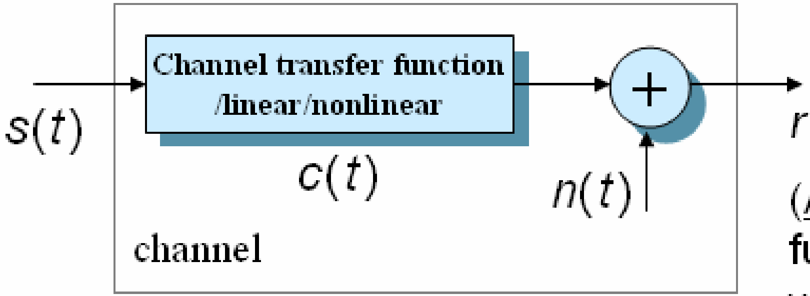
Assignment

3. Assume a two-bit word is transmitted with equal probability digits and a bit error probability of $\alpha = 1/4$. Determine what is the probability to have a bit error in a word by

(i) listing all the cases

(ii) by using definition of binomial distribution: $P(n,k,\alpha) = \dfrac{n!}{k!(n-k)!}\alpha^k(1-\alpha)^{n-k}$

Modeling Transmission Channels



Channel impulse response

$$r(t) = s(t) \otimes c(t) + n(t)$$

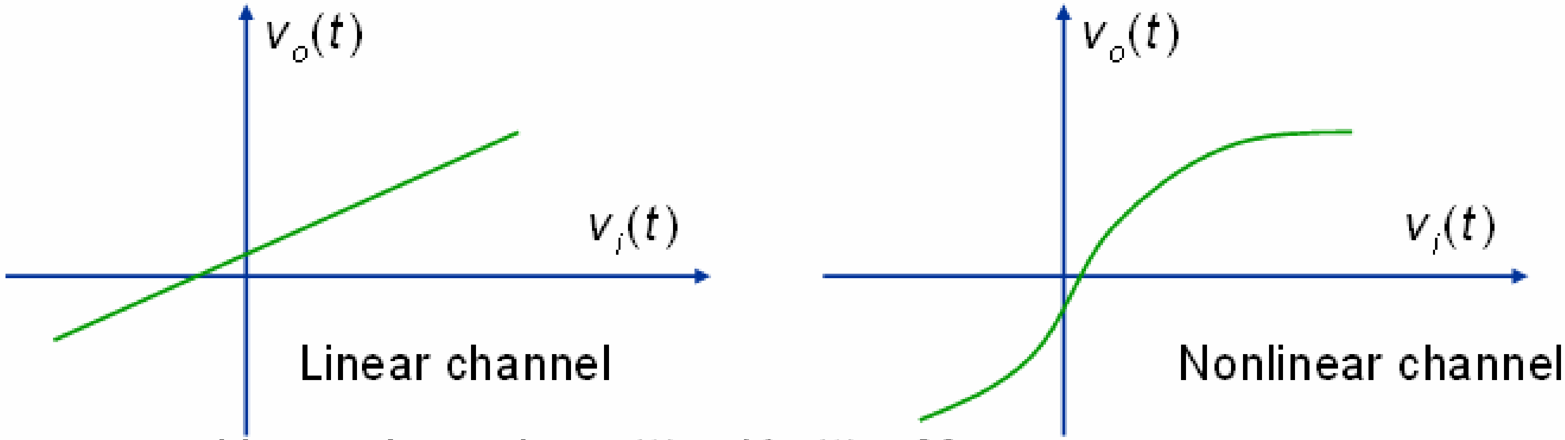
(AWGN channel (usually transfer function is linear) and *n(t)* is Gaussian, white noise)

- Information is always transmitted in channels as radio path (wireless cellular channel, microwave link, satellite link) or in wireline channels as coaxial cable, fiber optic cable or wave guide. Note that information storage is also a transmission channel
- Most common channels we discuss are **linear** Additive, White Gaussian Noise (AWGN) channels or linear, fading channels
- Note that the AWGN channel output is convolution of channel impulse response *c(t)* and channel input signal *s(t)* and has the noise term *n(t)* as additive component:

$$r(t) = s \otimes c(t) + n(t) = \int_u s(t)c(\tau - t)dt + n(t)$$

(*u*: where integrand exists)

Linear and Nonlinear Channels



- Linear channels: $v_0(t) = Kv_i(t) + M$
  - generate never new frequency components
  - characterized by transfer function
- Non-linear systems:
  - characterized by transfer characteristics
- Note: Often non-linearity in transmission is generated by transmitter or receiver, not by the channel itself
- Non-linear systems can generate new frequency components, example:

$$v_o(t) = a_o + \sum_{u=1}^{N} a_u v_i^u(t) \text{ with } v_i(t) = \sin(\omega t), N = 2$$

produces $v_o(t) = a_0 + a_1 \sin(\omega t) + a_2 / 2(1 - \cos(2\omega t))$

Assignment

4. Show that

$$v_o(t) = a_o + \sum_{u=1}^{N} a_u v_i^u(t)$$

with

$$v_i(t) = \sin(\omega t), N = 2$$

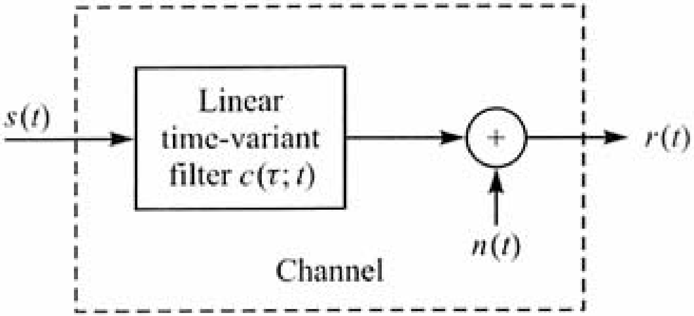
produces

$$v_o(t) = a_0 + a_1 \sin(\omega t) + a_2 / 2(1 - \cos(2\omega t))$$

by using the formulas

$$\begin{cases} \sin\alpha \sin\beta = 1/2\cos(\alpha-\beta) - 1/2\cos(\alpha+\beta) \\ \cos\alpha \cos\beta = 1/2\cos(\alpha-\beta) + 1/2\cos(\alpha+\beta) \\ \sin\alpha \cos\beta = 1/2\sin(\alpha-\beta) + 1/2\sin(\alpha+\beta) \end{cases}, \quad \begin{cases} \cos^2\alpha = (1+\cos 2\alpha)/2 \\ \cos^3\alpha = (3\cos\alpha + \cos 3\alpha)/4 \end{cases}$$

Time-variable Channel



- Most information channels are time-variable (fading) channels: cable, microwave link, cellular channel. Received signal is

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

- In frequency domain, (in differential time instant) there exists a frequency response $C(\tau_1;f) \approx C_1(f)$ and for this instance we may write

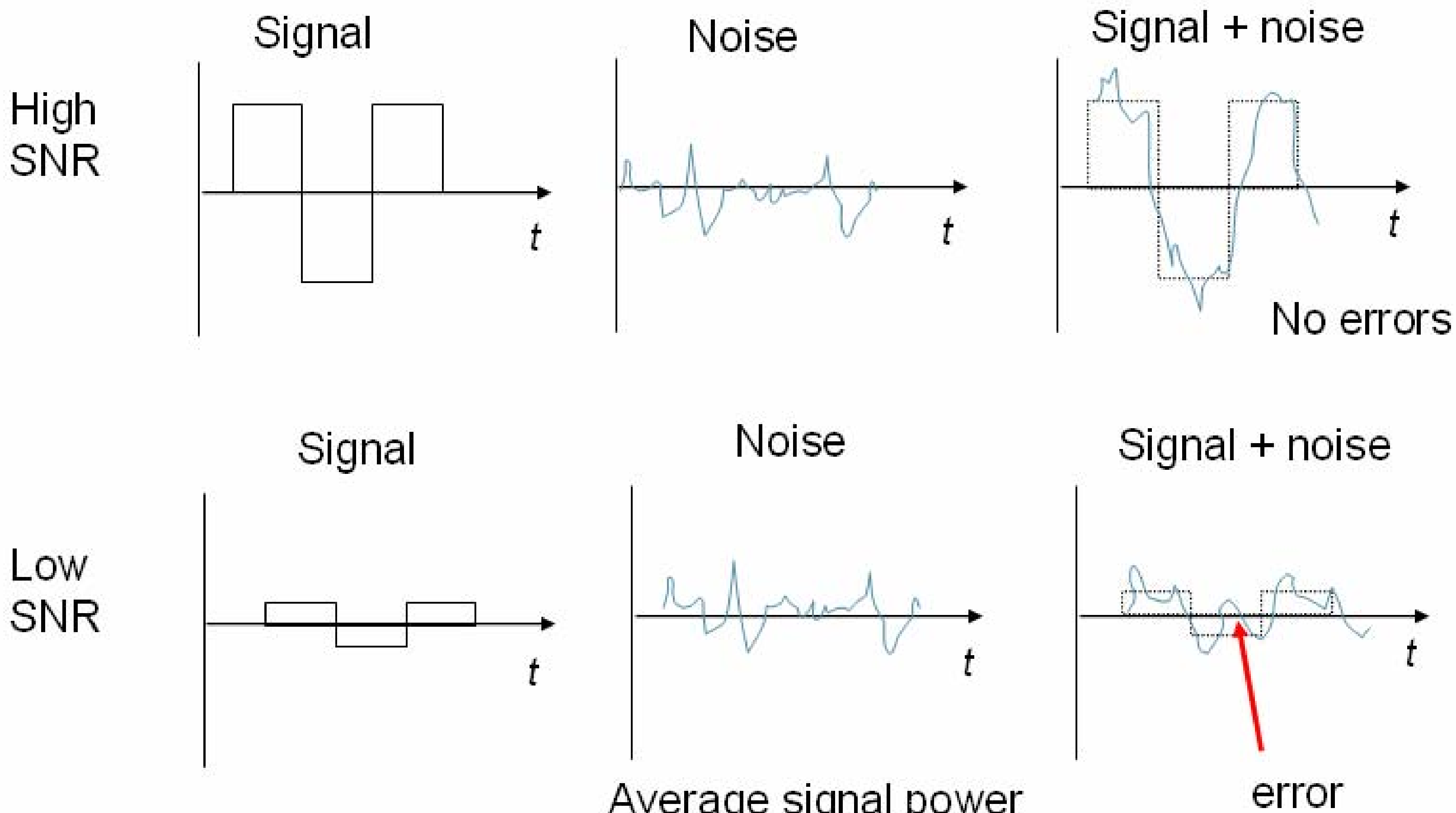
$$R_1(f) = N(f) + S(f)C_1(f)$$

- Channel variations / transmission errors compensated at the receiver:
  - equalization flattens frequency response (tapped delay line, decision feedback equalizer (DFE))
  - equalization assisted by channel estimation
  - channel errors can be compensated by channel coding (block and convolutional codes)

Noise & Reliable Communications

- All physical systems have noise
  - Electrons always vibrate (at non-zero temperature)
  - Motion of electrons induces noise
- Presence of noise limits accuracy of measurement of received signal amplitude
- Errors occur if signal separation is comparable to noise level
- Bit Error Rate (BER) increases with decreasing signal-to-noise ratio
- Noise places a limit on how many amplitude levels can be used for pulse transmission

Signal-to-Noise Ratio



$$\text{SNR} = \frac{\text{Average signal power}}{\text{Average noise power}}$$

$$\text{SNR (dB)} = 10 \log_{10} \text{SNR}$$

Assignment

5. SNR is to be determined for a system with reception bandwidth of 100 kHz with noise spectral density of 10^{-6} W/Hz and signal power of 10 W.

Determine the respective SNR in dB.

Shannon Channel Capacity

$$C = B_T \log_2 (1 + SNR) \text{ bps}$$

- Arbitrarily reliable communications is possible if the transmission rate $R < C$.
- If $R > C$, then arbitrarily reliable communications is not possible.
- "Arbitrarily reliable" means the BER can be made arbitrarily small through sufficiently complex coding.
- C can be used as a measure of how close a system design is to the best achievable performance.
- Bandwidth B_T & SNR determine C

Example

- Find the Shannon channel capacity for a telephone channel with B_T = 3400 Hz and *SNR* = 10000

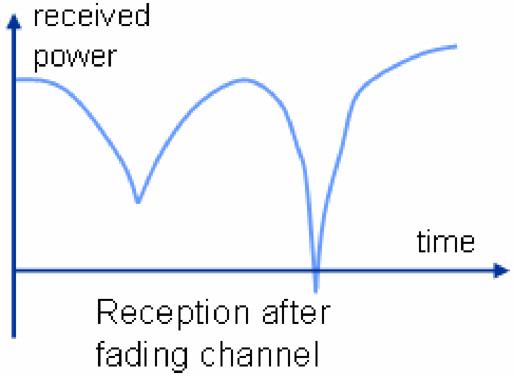
$$C = 3400 \log_2 (1 + 10000)$$
$$= 3400 \log_{10} (10001)/\log_{10} 2 = 45200 \text{ bps}$$

Note that *SNR* = 10000 corresponds to

$$SNR\ (\text{dB}) = 10 \log_{10}(10001) = 40 \text{ dB}$$

Interleaving

- In fading channels, received data can experience burst errors that destroy large number of consecutive bits. This is harmful in channel coding
- Interleaving distributes **burst errors** along data stream
- A problem of interleaving is introduced extra delay
- Example below shows block interleaving:



Received interleaved data: 1 0 0 0 1 1 1 0 1 0 1 1 1 0 0 0 1 1 0 0 1

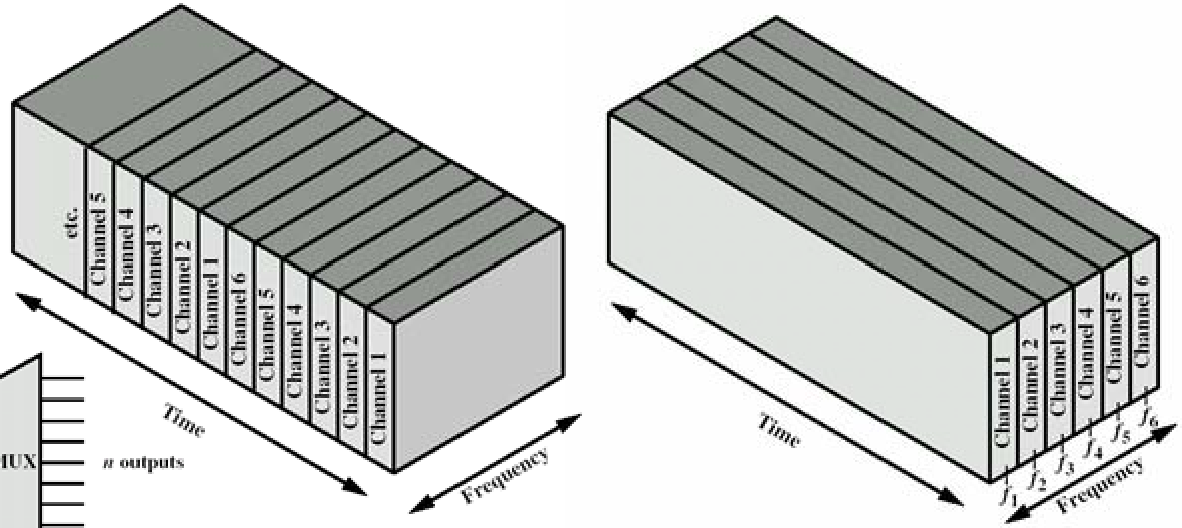
Block deinterleaving :

1 0 0 0 1 1 1
0 1 0 1 1 1 0
0 0 1 1 0 0 1

Recovered data: 1 0 0 0 1 0 0 0 1 0 1 1 1 1 0 1 1 0 1 0 1

Multiplexing

Multiplexing [5]

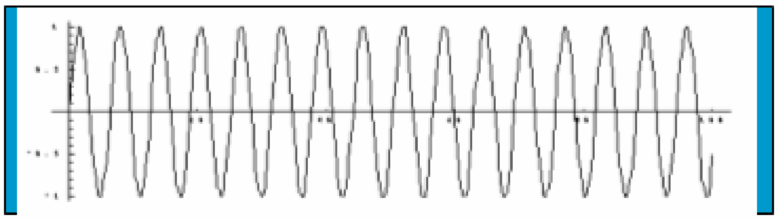


FDMA and TDMA multiplexing [5]

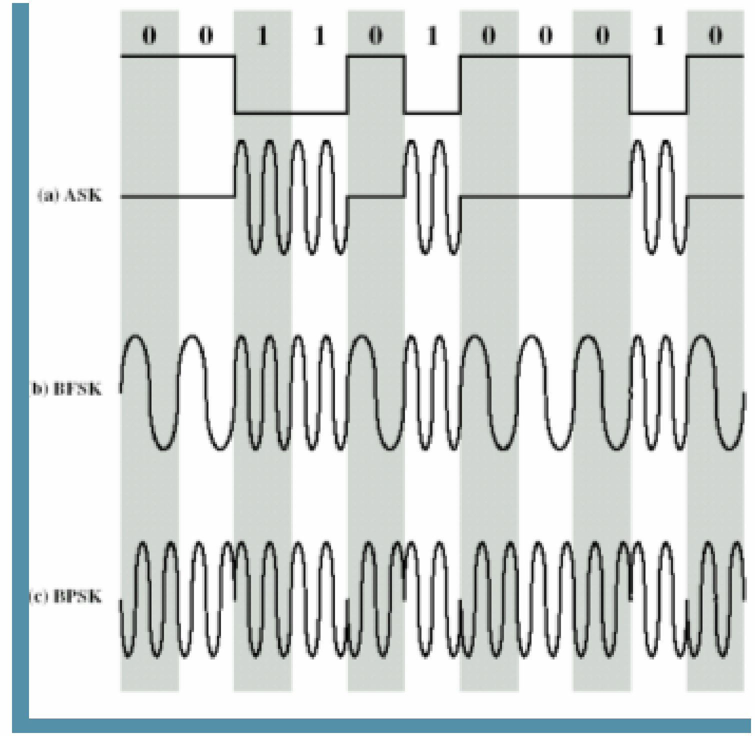
- Multiple information channels are transported by using **multiplexing**
- In **multiple access,** same channel is used to transmit multiple messages to different users
- **Fixed multiple access (originally for circuit switched networks)**:
  - **TDMA** (time division multiple access), users occupy different time slots
  - **FDMA** (frequency division multiple access), users occupy different frequency bands
  - **CDMA** (code division multiple access), users occupy the same frequency band but modulate their messages with different codes
- **Statistical multiple access (packet networks), example:**
  - **ALOHA**: Station send a packet and waits for acknowledgement (ACK) for the maximum time of round trip delay. If ACK not received (collision), send again!

Unmodulated and Modulated Sinusoidals

- The unmodulated sinusoidal wave is parameterized by constant amplitude, frequency and phase



unmodulated sinusoidal



some digital carriers [5]

- In unmodulated sinusoidal all parameters known, convoys no information!
- Mathematically and experimentally convenient formulation whose ***parameterization*** by variables enables presenting all carrier wave modulation formats by

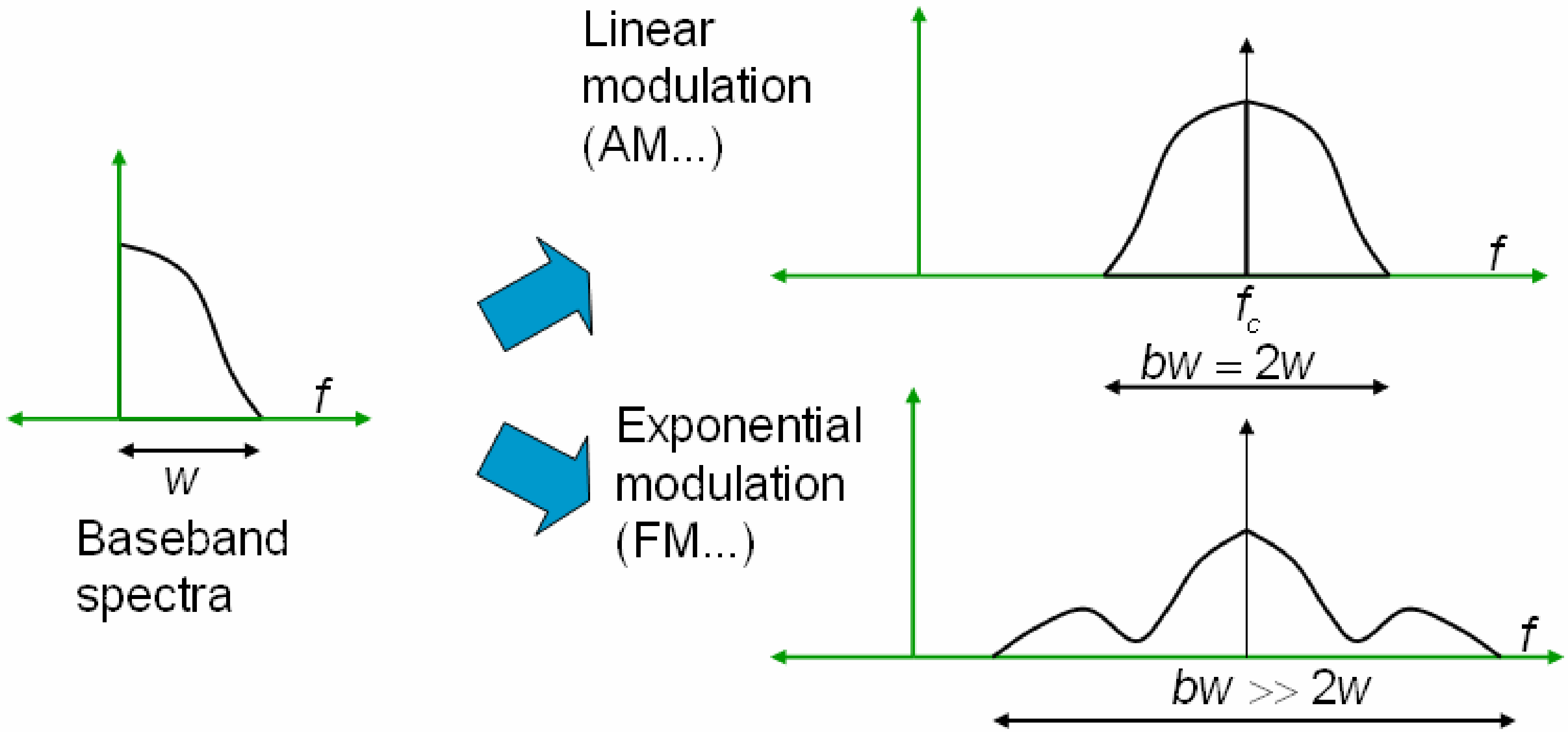
$$x(t) = A(t)\cos\left[\omega_c t + \phi(t)\right]$$

Amplitude modulation (AM)...,
Amplitude Shift Keying (ASK)...

Carrier-term

Frequency modulation (FM),
Frequency/Phase Shift Keying (FSK,PSK)...

Baseband and Carrier Wave (CW) Systems



- Figures show baseband message transfer by linear (AM) and exponential modulation (FM)
- In linear modulation, transmission bandwidth is always below or equal to $2W$ (W: message bandwidth)
- Non-linear (angle modulation) spreads message on much larger transmission bandwidth that 2W

Which Modulation Method to Apply?

- Modulation is done to enable the usage of medium for transmission. Thus the modulation method is selected based on
    - Message to be transmitted (source) as
        - voice/video (analog source)
        - data (digital source, machine-to-machine communications)
        - traffic statistics: continuous / bursty traffic
    - Allowed delay
    - Medium that is to be used
    - Networking type as
        - cellular wireless networks (GSM, AMPS*)
        - RF-LANs (802.11b Wi-Fi, HiperLAN /2)
        - wire-line local area networks (Ethernet LANs)
        - public switched telephone network (PSTN)

Channel determines modulation method

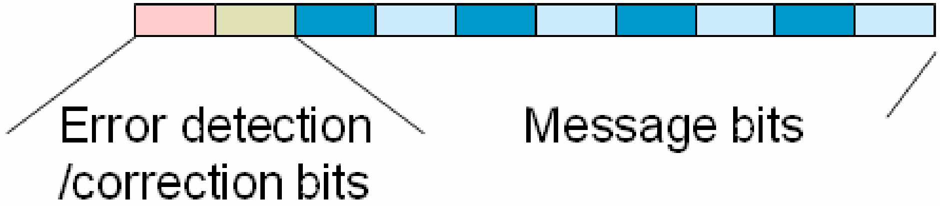
*Advanced Mobile Phone Service

Coding

- Channel coding is done ...
  - For **detection and/or correction of errors** produced by the channel (as block and convolutional coding) by
    - noise
    - interference
    - distortion
      - linear
      - nonlinear
  - To alleviate **synchronization** problems (as Manchester coding)
  - To alleviate **detection** problems (as differential coding)
  - To enable **secrecy and security** (as scrambling or ciphering)
- Channel coding principles:
  - **ARQ** (Automatic Repeat Request) as go-back-N ARQ
  - **FEC** (Forward Error Correction) as block & convolutional coding

Coding

- Coding is classified to two flavors
  - *source coding*: makes transmitted bits equal probable - maximizes channel capacity
  - *channel coding*: protects message & adapts it to channel
- Channel coding means adding extra bits for message for error detection and/or correction
- In **systematic coding** message bits remain the same in coded word:



- In coded systems **soft decision** can be used that calculates the distance of the received code word to the allowed code words for instance by using a least-square metric

Bits, Bytes and Words

- Bit: number with value 0 or 1
  - *n* bits: digital representation for 0, 1, … , 2^n
  - Byte or Octet, *n* = 8 bits
  - Computer word, *n* = 16, 32, or 64 bits
- *n* bits allows enumeration of 2^n possibilities (as for instance levels)
  - *n*-bit field in a header
  - *n*-bit representation of a voice sample
  - Message word consisting of n bits
- *The number of bits required to represent a message is a way to measure its information content*
  - More bits → More content?

Transmission Delay

- *L* number of bits in message
- *R* bps speed (signaling rate) of digital transmission system
- *L/R* time to transmit the information
- t_{prop} time for signal to propagate across medium
- *d* distance in meters
- *c* speed of light (3×10^8 m/s in vacuum)

$$Delay = t_{prop} + L/R = d/c + L/R \text{ seconds}$$

Use data compression to reduce L
Use higher speed modem to increase R
Place server closer to reduce d

Conclusions

- Telecommunication systems divided into
  - transmitters, channels, receivers
- Understanding of source statistics is important
  - Fixed multiple access for bulk data
  - Statistical multiplexing for demanding sources and networks
- Channels can be linear or non-linear. Non-linear channels generally more demanding due to introduced extra frequency components
- Coding is used to protect message in channels (channel coding) and to compress source information (source coding)
- Modulation is used to carry messages in carrier wave systems - Selection of modulation method effects
  - reception sensitivity
  - transmission bandwidth
  - applicability in networking applications

References

1 A. Burr: Modulation & Coding

2 A.B. Carlson: Communication Systems (4th ed)

4 Ahlin, Zhanders: Principles of Wireless Communications

5 W. Stallings: Wireless Communications and Networks

6 Telia, Ericsson: Understanding Telecommunications, Part I-II (Studentlitteratur)

7 A Leon-Garcia et al: Communication Networks, Mc GrawHill, Instructor's slide set