SCHOOL OF INNOVATIVE TECHNOLOGIES & ENGINEERING

Module Information Pack

BEng (Hons.) Telecommunications

BTEL v1.2

Wireless Communications

TELC2103

PART 1

Academic Year 2015/2016

A.	**Module Convenor:**	Mr. Rishi H. Heerasing
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B.	**Module Coordinator:**	Mr. Rishi H. Heerasing
C.	**Programme Coordinator:**	Mr. Rishi H. Heerasing
D.	**Programme Director:**	Dr. Vinaye Armoogum
E.	**Credits:**	6
F.	**Pre-required skills (if any):**	TELC 1101 or Equivalent
G.	**Teaching and Learning Strategies:**	30 x 4 Hrs sessions of lectures, tutorials and practicals
H.	**Academic Tutoring:**	None
I.	**Student Progress and Assessment:**	**50%** 3¼ Hours Unseen Exam & 50% Coursework

J. Summary of Module Content:

Overview of wireless systems, transmission basics, analogue and digital data transmission, bandwidth and channel capacity, transmission impairments, transmission media, analogue and digital modulation-demodulation, encoding-decoding techniques, error correction and detection, asynchronous and synchronous transmission, multiplexing, spread-spectrum techniques, wireless technologies, satellite communication, VSAT, PCS technology

K. Module Aims:

- To investigate the fundamentals of data communication and the problems that affects communication in relation to wireless networks.
- To investigate the various wireless communications standards, protocols, architectures, and transmission techniques currently available.
- To introduce the need for modulation, encoding, multiplexing, error detection and correction of data in wireless communication.
- To understand the need for spread-spectrum techniques in wireless communication.

L. Learning Outcomes:

By the end of this module, students are expected:

- Understand the basics behind reliable, effective, wireless communication, wireless LANs, and fixed wireless access systems and technologies.
- Understand the basics of data communication and the fundamental differences between different multiple-access strategies.

M. Two Semester Plan

Week	Topics Covered
1&2	Overview of Wireless Systems
3&4	Transmission Basics; Analog and Digital Data Transmission
5&6	Transmission Impairments. Nyquist formula and Shannon's Capacity Theorem
7&8	Wireless Transmission and Propagation. Antennas.
9&10	Encoding & Modulation: Digital Data > Digital Signal; Digital Data > Analog Signal
11&12	Encoding & Modulation: Analog Data > Digital Signal; Analog Data > Analog Signal
13&14	Asynchronous and Synchronous Transmission - Error Control; Multiplexing Techniques: FDM, Synchronous TDM; Statistical TDM; OFDM.
15&16	Spread-Spectrum Techniques
17&18	PCS & VSAT
19&20	Wireless Technologies: Bluetooth, WLAN, WiMAx, ZigBee
21&22	Introduction to Satellite Communication
23&24	Buffer
25&26	Buffer
27&28	Presentations
29&30	Revision

N. READING LIST

1. RECOMMENDED TEXTS (as per availability in the UTM Resource Centre):

- **Blake R. (2001) *Wireless Communication Technology*, Thomson Learning ***
- **Rappaport T.S. (2001) *Wireless Communications – Principles & Practice:* 2nd Ed., Prentice-Hall Publishing ***
- Goldsmith A. (2005) *Wireless Communications*, Cambridge University Press *

ж *You can download a copy of these books in e-book format on my Intr@web site at http://intraweb/~rh/download/network.html*

2. OTHER READING MATERIALS e.g. TEXTS/JOURNALS/ARTICLES/WEBSITES:

- Prentice-Hall Companion Website for *Wireless Communications -Principles* & Practice, 2nd Edition at http://authors.phptr.com/rappaport/

3. PAST EXAM PAPERS

- ○ Included in second part of MIP

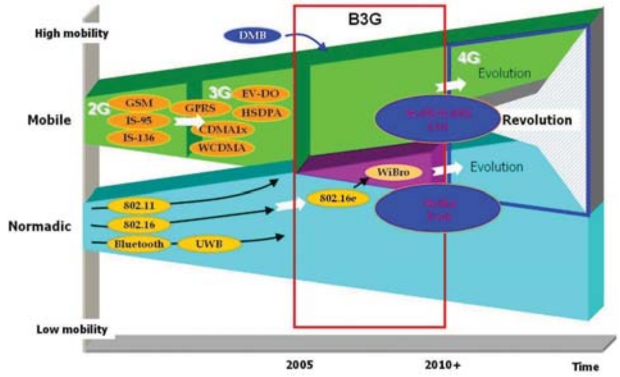
O. LECTURE NOTES

Lecture notes are available on the **INTR@WEB** site (campus-access only) at http://intraweb and clicking on my link or to access it directly at http://intraweb/~rh You can also access the site externally at the following address: http://www.rishiheerasing.net

Note: ***Downloads*** *Section is only available on campus and the notes are in .pdf format so you will need Adobe Acrobat® Reader to view them. This reader can also be obtained from the* ***Downloads*** *Section.*

Wireless Communications

Overview of Wireless Systems



Outline

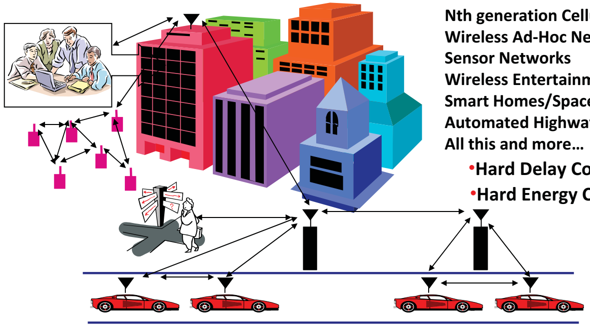
- The Wireless Vision
- Technical Challenges
- Current Wireless Systems
- Emerging Wireless Systems
- Spectrum Regulation
- Standards

Wireless History

- **Ancient Systems: Smoke Signals, Carrier Pigeons, …**
- **Radio invented in the 1880s by Marconi**
- **Many sophisticated military radio systems were developed during and after WW2**
- **Cellular has enjoyed exponential growth since 1988, with almost 3 billion users worldwide today**
  - **Ignited the wireless revolution**
  - **Voice, data, and multimedia becoming ubiquitous**
  - **Use in third world countries growing rapidly**
- **WiFi also enjoying tremendous success and growth**
  - **Wide area networks (e.g. Wimax) and short-range systems other than Bluetooth (e.g. UWB) less successful**

Future Wireless Communication

Ubiquitous Communication Among People and Devices



Wireless Internet access
Nth generation Cellular
Wireless Ad-Hoc Networks
Sensor Networks
Wireless Entertainment
Smart Homes/Spaces
Automated Highways
All this and more…

- **Hard Delay Constraints**
- **Hard Energy Constraints**

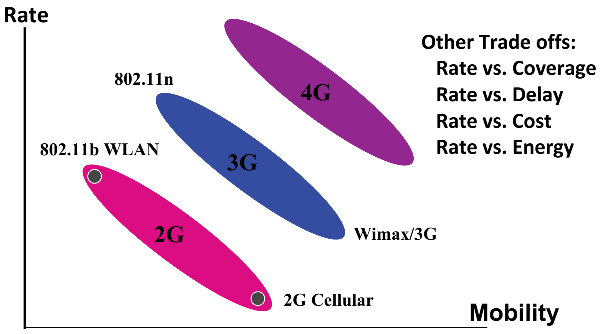
Design Challenges

- Wireless channels are a difficult and capacity-limited broadcast communications medium
- Traffic patterns, user locations, and network conditions are constantly changing
- Applications are heterogeneous with hard constraints that must be met by the network
- Energy and delay constraints change design principles across all layers of the protocol stack

Evolution of the Current Systems

- Wireless systems today
  - 3G Cellular: ~200-300 Kbps.
  - WLANs: ~450 Mbps (and growing).
- Next Generation is in the works
  - 4G Cellular: Likely OFDM/MIMO
  - 4G WLANs: Wide open, 3G just being finalized.
- Technology Enhancements
  - Hardware: Better batteries, circuits and processors.
  - Link: Antennas, modulation, coding, adaptivity, DSP, BW.
  - Network: Not much: more efficient algorithms & ACK
  - Application: Soft and adaptive QoS.

Future Generations



Other Trade offs:
Rate vs. Coverage
Rate vs. Delay
Rate vs. Cost
Rate vs. Energy

Fundamental Design Breakthroughs Needed

Multimedia Requirements

	Voice	Data	Video
Delay	<100ms	-	<100ms
Packet Loss	<1%	0	<1%
BER	10-3	10-6	10-6
Data Rate	8-32 Kbps	1-100 Mbps	1-20 Mbps
Traffic	Continuous	Bursty	Continuous

One-size-fits-all protocols and design do not work well

Wired networks use this approach, with poor results

Quality of Service (QOS)

- QoS refers to the requirements associated with a given application, typically rate and delay requirements.
- It is hard to make a one-size-fits all network that supports requirements of different applications.
- Wired networks often use this approach with poor results, and they have much higher data rates and better reliability than wireless.
- QoS for all applications requires a cross-layer design approach.

Cross-Layer Design

- Application
- Network
- Access
- Link
- Hardware

Delay Constraints
Rate Constraints
Energy Constraints

Adapt across design layers
Reduce uncertainty through scheduling
Provide robustness via diversity

Cross-Layer Techniques

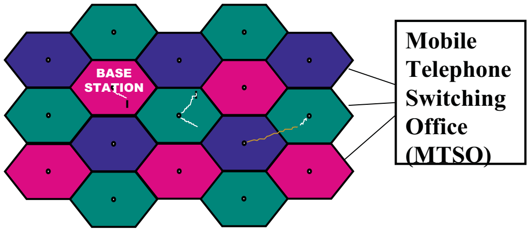
- Adaptive techniques
  - Link, MAC, network, and application adaptation
  - Resource management and allocation (power control)
- Diversity techniques
  - Link diversity (antennas, channels, etc.)
  - Access diversity/ Route diversity
  - Application diversity
  - Content location/server diversity
- Scheduling
  - Application scheduling/data prioritization
  - Resource reservation
  - Access scheduling

Current Wireless Systems

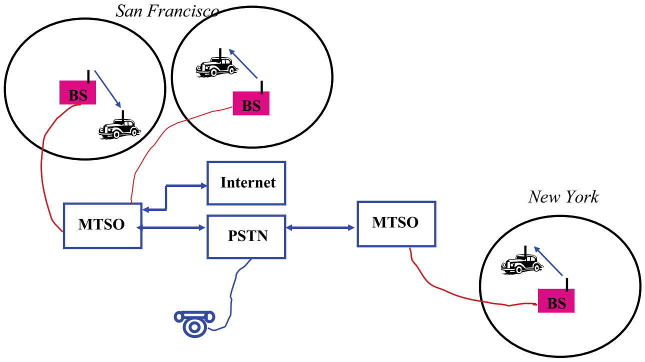
- Cellular Systems
- Wireless LANs
- WiMAX
- Satellite Systems
- Paging Systems
- Bluetooth
- Ultra-WideBand radios
- Zigbee radios

Cellular System

- Geographic region divided into cells
- Frequency/time slots/codes/ reused at spatially-separated locations.
- Co-channel interference between same colour cells.
- Base stations/MTSOs coordinate hand-off and control functions
- Shrinking cell size increases capacity, as well as networking burden



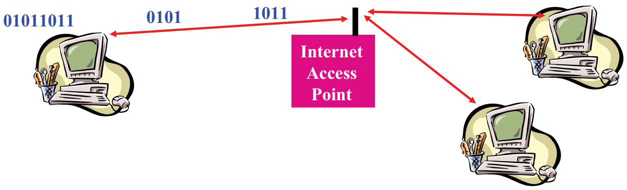
Cellular Phone Networks



3G Cellular Design

- Data is bursty, whereas voice is continuous
  - Typically require different access and routing strategies
- 3G "widens the data pipe":
  - 384 Kbps (802.11n has 100s of Mbps).
  - Standard based on Wideband CDMA
  - Packet-based switching for both voice and data
- 3G cellular popular in Asia and Europe
- Evolution of existing systems in US (2.5G++)
  - GSM+EDGE, IS-95(CDMA)+HDR
  - 100 Kbps may be enough
  - Dual phone (2/3G+Wifi) use growing (iPhone, etc)
- *What is beyond 3G?*

Wireless Local Network



- **WLANs connect "local" computers (100m range)**
- **Breaks data into packets**
- **Channel access is shared (random access)**
- **Backbone Internet provides best-effort service**
  - **Poor performance in some applications (e.g. video)**

WLAN Standards

- 802.11b (Old – 1990s)
  - Standard for 2.4GHz ISM band (80 MHz)
  - Direct sequence spread spectrum (DSSS)
  - Speeds of 11 Mbps, approx. 500 ft range
- 802.11a/g (Middle Age– mid-late 1990s)
  - Standard for 5GHz NII band (300 MHz)
  - OFDM in 20 MHz with adaptive rate/codes
  - Speeds of 54 Mbps, approx. 100-200 ft range

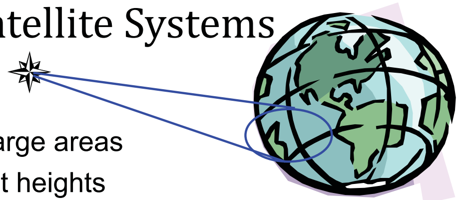
Many WLAN cards have all 3 (a/b/g)

- 802.11n (Hot stuff, standard close to finalization)
  - Standard in 2.4 GHz and 5 GHz band
  - Adaptive OFDM /MIMO in 20/40 MHz (2-4 antennas)
  - Speeds up to 600Mbps, approx. 200 ft range
  - Other advances in packetization, antenna use, etc.

WiMAX (802.16)

- Wide area wireless network standard
  - System architecture similar to cellular
  - Hopes to compete with cellular
- OFDM/MIMO is core link technology
- Operates in 2.5 and 3.5 MHz bands
  - Different for different countries, 5.8 also used.
  - Bandwidth is 3.5-10 MHz
- Fixed (802.16d) vs. Mobile (802.16e) Wimax
  - Fixed: 75 Mbps max, up to 50 mile cell radius
  - Mobile: 15 Mbps max, up to 1-2 mile cell radius

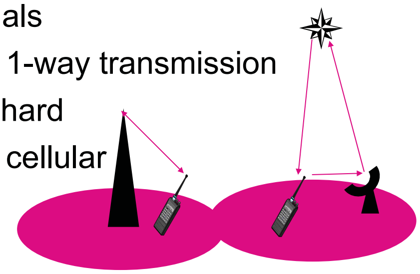
Satellite Systems



- Cover very large areas
- Different orbit heights
  - GEOs (39000 Km) versus LEOs (2000 Km)
- Optimized for one-way transmission
  - Radio and movie (SatTV, DVB/S) broadcasts
  - Most two-way systems struggling or bankrupt
- Global Positioning System (GPS) use growing
  - Satellite signals used to pinpoint location
  - Popular in cell phones, PDAs, and navigation devices

Paging Systems

- Broad coverage for short messaging
- Message broadcast from all base stations
- Simple terminals
- Optimized for 1-way transmission
- Answer-back hard
- Overtaken by cellular



Bluetooth

- Cable replacement RF technology (low cost)
- Short range (10m, extendable to 100m)
- 2.4 GHz band (crowded)
- 1 Data (700 Kbps) and 3 voice channels, up to 3 Mbps
- Widely supported by telecommunications, PC, and consumer electronics companies
- Few applications beyond cable replacement

Ultra-Wide Band Radio (UWB)

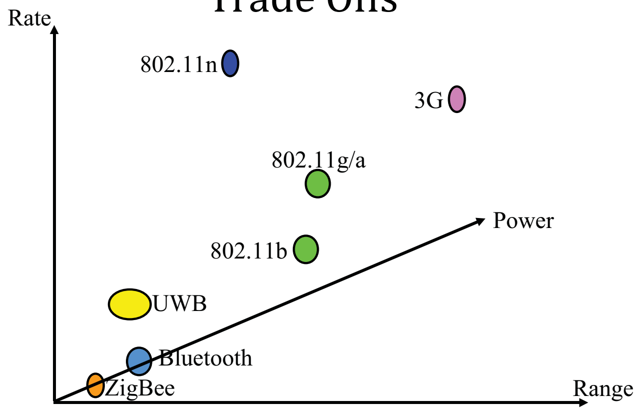
- UWB is an impulse radio: sends pulses of tens of picoseconds(10^{-12}) to nanoseconds (10^{-9})
  - Duty cycle of only a fraction of a percent
- A carrier is not necessarily needed
- Uses a lot of bandwidth (GHz)
- High data rates, up to 500 Mbps
- 7.5 GHz of “free spectrum” in the U.S.
- Multi-path highly resolvable: good and bad
- Limited commercial success to date

IEEE 802.15.4 /Zigbee

- Low-Rate WPAN
- Data rates of 20, 40, 250 Kbps
- Support for large mesh networking or star clusters
- Support for low latency devices
- CSMA-CA channel access
- Very low power consumption
- Frequency of operation in ISM bands

Focus is primarily on low power sensor networks

Trade Offs

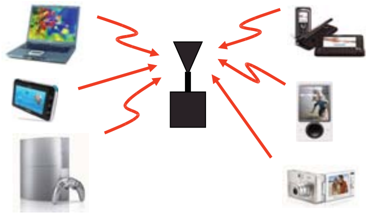


Spectrum Regulation

- Spectral Allocation in US controlled by FCC (commercial) or OSM (defence)
- FCC auctions spectral blocks for set applications.
- Some spectrum set aside for universal use
- Worldwide spectrum controlled by ITU-R
- Regulation is a necessary evil.

Coexistence Challenge:

Many devices use the same radio band



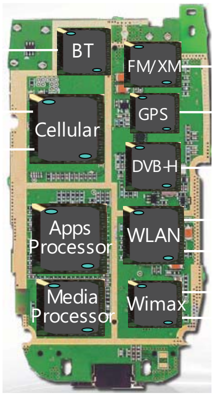
- Technical Solutions:
  - Interference Cancellation
  - Smart/Cognitive Radios

Next-Generation Devices

Everything Wireless in One Device

Multi Radio Integration Challenge

- RF Interference
- Where to put antennas
- Size
- Power Consumption



Standards

- Interacting systems require standardization
- Companies want their systems adopted as standard
  - Alternatively try for de-facto standards
- Standards determined by TIA/CTIA in US
  - IEEE standards often adopted
  - Process fraught with inefficiencies and conflicts
- Worldwide standards determined by ITU-T
  - In Europe, ETSI is equivalent of IEEE

Emerging Systems

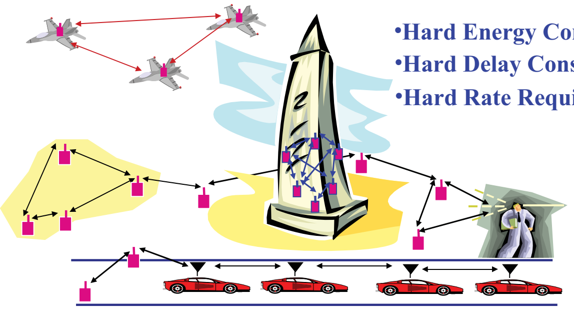
- 4th generation cellular (4G)
  - OFDMA will be PHY layer (like Wimax)
  - Other new features and bandwidth still in flux
- Ad hoc/mesh wireless networks
- Cognitive radios
- Sensor networks
- Distributed control networks

Ad Hoc/ Mesh Networks

Design Issues

- Ad-hoc networks provide a flexible network infrastructure for many emerging applications.
- The capacity of such networks is generally unknown.
- Transmission, access, and routing strategies for ad-hoc networks are generally ad-hoc.
- Cross-layer design critical and very challenging.
- Energy constraints impose interesting design trade-offs for communication and networking.

Wireless Sensor Networks
Data Collection and Distributed Control



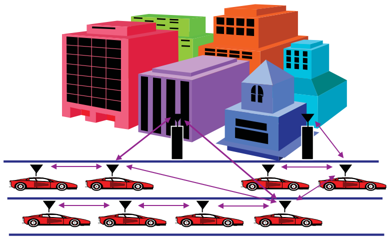
•Hard Energy Constraints
•Hard Delay Constraints
•Hard Rate Requirements

Nodes can cooperate in transmission, reception, compression, and signal processing.

Energy- Constrained Nodes

- Each node can only send a finite number of bits.
  - Transmit energy minimized by maximizing bit time
  - Circuit energy consumption increases with bit time
  - Introduces a delay versus energy trade-off for each bit
- Short-range networks must consider transmit, circuit, and processing energy.
  - Sophisticated techniques not necessarily energy-efficient.
  - Sleep modes save energy but complicate networking.
- Changes everything about the network design:
  - Bit allocation must be optimized across all protocols.
  - Delay vs. throughput vs. node/network lifetime trade-offs.
  - Optimization of node cooperation.

Distributed Control Over Wireless Links



Automated Vehicles
- **Cars**
- **UAVs**
- **Insect flyers**

- Packet loss and/or delays impacts controller performance.
- Controller design should be robust to network faults.
- Joint application and communication network design.

Main Points

- The wireless vision encompasses many exciting systems and applications
- Technical challenges transcend across all layers of the system design.
- Cross-layer design emerging as a key theme in wireless.
- Existing and emerging systems provide excellent quality for certain applications but poor interoperability.
- Standards and spectral allocation heavily impact the evolution of wireless technology

Wireless Communications

- Transmission Basics
- Analogue and Digital Data Transmission

Contents

- Transmission terminologies
- Analogue & Digital Data
- Analogue & Digital Signals
- Analogue & Digital Transmission

Terminology

Transmission system components:

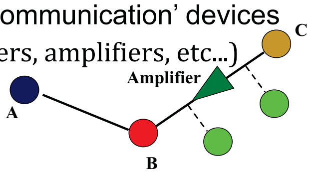
- Transmitter
- Receiver
- Medium
  - Guided media
    - e.g. twisted pair, coaxial cable, optical fibre
  - Unguided media (radiated)
    - e.g. air, water, vacuum

Terminology

Link Configurations:

- Direct link
  - No **intermediate** ‘communication’ devices

(these exclude repeaters, amplifiers, etc...)



Two types:

- Point-to-point (A-B)
  - Only 2 devices share link
- Multi-point (C-B)
  - More than two devices share the same link, e.g. Ethernet bus segment

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Analogue & Digital Data Transmission

- Data
  - Entities that convey meaning
- Signals
  - Electric or electromagnetic representations of **data**
- Transmission
  - Processing and propagation of **signals** that ultimately represent data

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Analogue & Digital Data in nature

- Analogue Data
  - Continuous values within some interval
  - Examples: audio, video
  - Typical bandwidths:
    - Human Voice: 100 Hz to 8 kHz
    - Speech over telephone: 300 Hz to 3400 Hz
    - Video: 0 to 4 MHz
- Digital Data
  - Discrete values (not necessarily binary)
  - Examples: integers, text characters,

Analogue and Digital Signals

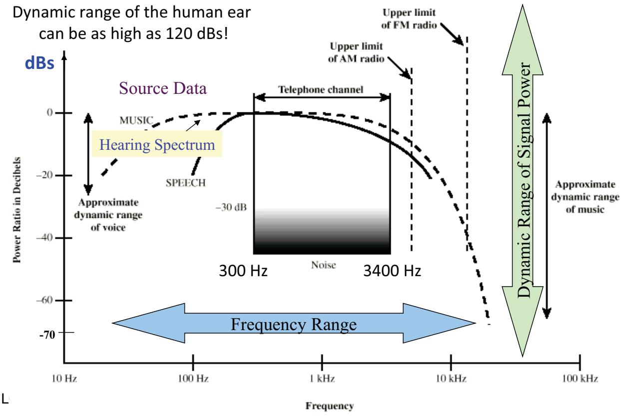
- Means by which data get represented for transmission over various media, e.g. wire, fibre optic, space, etc...
  - Analog signal:
    - Continuously variable in time and amplitude
  - Digital signal:
    - Uses a few (two or more) DC levels

Analogue data constructed as an analogue signals e.g. Speech Data

- Frequency range for human hearing: 20Hz - 20kHz
  - Almost fully utilized by music
  - Human voice: 100Hz - 7kHz
  - Telephone voice channel: Spectrum is further limited to 300 - 3400Hz (why?)
- Mechanical sound waves (data) are easily converted into electromagnetic signal for processing and transmission:
  - Mechanical waves (sound) of varying pitch and loudness (data)

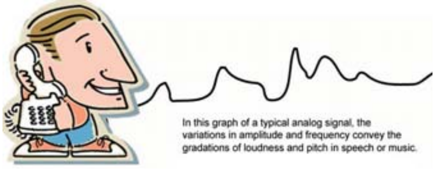
  is represented as electromagnetic signals of different frequencies and amplitudes (signal)

Example: The Acoustic Spectrum


Dynamic range of the human ear can be as high as 120 dBs!

Conventional Telephony Analogue data –> Analogue signal

- Telephone mouthpiece converts mechanical voice analogue data into electromagnetic analogue electrical signal
- Signal is transmitted over telephone lines in an analogue or digital manner.
- At receiver, speaker re-converts received electrical signal to voice

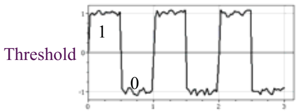

In this graph of a typical analog signal, the variations in amplitude and frequency convey the gradations of loudness and pitch in speech or music.

Digital data represented as digital (baseband) signals

Advantages:

- Cheaper and easier to generate: No extra processing needed
- Less susceptible to noise

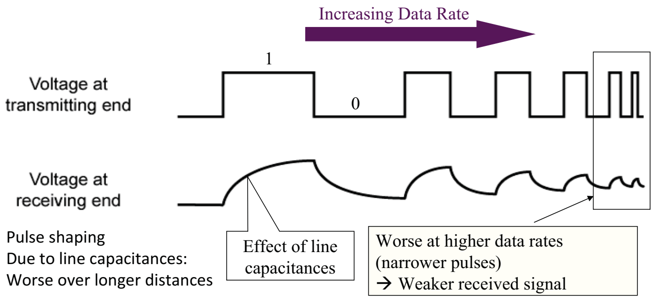
("Threshold effect")



Disadvantages:

- When noise is above threshold → Total data reversal (Bit error) (1→ 0, 0 →1)
- Greater attenuation
  - Line capacitances make pulses rounded and smaller in amplitude, leading to loss of information
  - More so at higher data rates and longer distances
  - So, use at low data rates over short distances

Attenuation of Digital Signals (later)



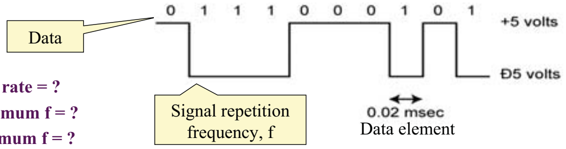
Pulse shaping
Due to line capacitances:
Worse over longer distances

Effect of line capacitances

Worse at higher data rates (narrower pulses)
→ Weaker received signal

Digital Binary (Base band) Signal

- Example: Between keyboard and computer
- Two bipolar DC levels (+ and – : Why?)
- Bandwidth required depends on the signal frequency, which depends on:
  - The data rate (bps) and
  - The actual data sequence transmitted

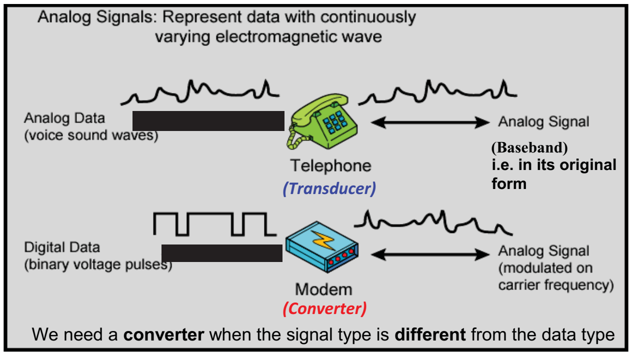


- Data rate = ?
- Maximum f = ?
- Minimum f = ?

Data and Signal combinations

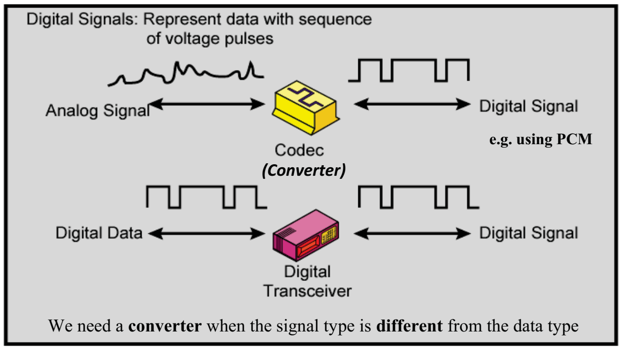
- We have seen above: (data and signal of same type)
  - Simple- one only needs a **transducer** or **transceiver**
  - Analogue signals carrying analogue data: Telephony, Video
  - Digital signals carrying digital data: Keyboard to PC
- But we may also have: (data and signal of different types)
  - More complex- Need a **converter**
  - Analogue signal representing digital data: e.g. digital data over telephone wires (use a modem)
  - Digital signal representing analogue data: CD Audio, PCM (pulse code modulation) (use a codec)
- So, all the four data-signal combinations are possible!

Analogue Signals can carry either Analogue Data or Digital Data


Analog Signals: Represent data with continuously varying electromagnetic wave

We need a **converter** when the signal type is **different** from the data type

Digital Signals can carry either Analogue Data or Digital Data


Digital Signals: Represent data with sequence of voltage pulses

We need a **converter** when the signal type is **different** from the data type

Four Data/Signal Combinations

		Signal	
		Analogue	**Digital**
Data	**Analogue**	*Two ways:* *Signal has same spectrum as data (baseband): e.g. Telephone to Exchange.* *Signal has different spectrum (through modulation): e.g. AM Radio.*	*Use a converter like a codec, e.g. for PCM (pulse code modulation)*
	Digital	*Use a converter like a modem e.g. V.90 standard*	*Simple unipolar signal e.g. NRZ-L or special encoding: e.g. Manchester*

Analogue Transmission of Analogue/Digital Signals

- Treats the signal as “analogue” regardless of what it represents i.e. not interested in the data content of signal.
- Following **attenuation** over distance, signal level is boosted using **amplifiers** or **boosters**.
- **Unfortunately, this also amplifies in-band noise**.
- With cascaded amplifiers (i.e. one after the other at locations along the link), effect on **noise** and **distortion** are **cumulative**, i.e. they get amplified again and again.
- **Effect of noise and distortion on analogue systems may be tolerated**, e.g. with telephony you can still manage to get the message! (Humans are good at filling-in the gaps!)
- But digital systems are more sensitive to the effects of excessive noise and distortion →unacceptable error rates!!!
- **So… Do not transmit a digital signal the analogue way!**

Digital Transmission of Analogue/Digital Signals

- Concerned with the **data content** of the signal.
- It assumes that the signal carries the digital data.
- Uses a **repeater** (NOT **amplifier**), which:
  - Receives the signal.
  - **Extracts** the data bit stream from it.
  - Retransmits a new **fresh (clean)**, **strong** signal representing the extracted bit stream.
- This way:
  - We overcome the effect of **attenuation.**
  - **Noise** and **distortion** are **NOT cumulative.**

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4 Signal/Transmission Combinations

		Transmission mode	
		Analogue	**Digital**
		- Uses amplifiers - Not concerned with what data the signal represents - Noise and distortion are cumulative - Associated with FDM	- Uses repeaters - Assumes signal represents digital data, extract this data and presents it as a new outbound signal - This way, noise and distortion are not cumulative - Associated with TDM
Signal	**Analogue**	OK if the analogue signal represents analogue data	Makes sense only if the analogue signal represents digital data! What data is the repeater going to extract?
	Digital	**Avoid**	**OK**

Advantages of Digital Transmission

- Use of digital technology
  - Lower cost, smaller size, and high speed VLSI technology (now!!!)
- Higher data integrity (reliability) as noise effects are not cumulative
  - Cover longer distances, at higher data rate, at low error rates, over lower quality lines, etc.
- Easier to implement 'muxing' for improved utilization of link capacity
  - High bandwidth links are now economical (Fibre, Satellite…)
  - To utilize them efficiently we need to do a lot of multiplexing
  - Done more efficiently using digital TDM rather than analogue FDM (later)
- Other related functions are also done digitally:
  - Compression to reduce bandwidth requirement
  - Encryption for data security and confidentiality
- Easier to integrate different data types
  - Convert analogue data to digital signals…and use one system to handle all voice, video, and data, e.g. one network for all types of traffic

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Wireless Communications

- Bandwidth, Channel Capacity
- Transmission Impairments

Contents

- Bandwidth and data rate
- The decibels notation for signal strength
- Transmission Impairments
- Channel Capacity

Maximum Data Rate (Channel capacity)

Considerations

- Bandwidth of transmission system
- Signal to noise ratio (SNR)
- Receiver type
- Specified acceptable error performance

Bandwidth & Data Rates: Trade-offs…

- Increasing the data rate (bps) while keeping BW the same (to economize) means working with inferior (poorer) waveforms at the receiver, which may require:
  - ☐ Ensuring higher signal to noise ratio (SNR) at RX
    - Larger transmitted power (may cause interference to others!)
    - Shorter link distances
    - Use of more en-route repeaters/amplifiers
    - Better shielding of cables to reduce noise, etc.
  - ☐ Using a more sensitive (& costly!) receiver
  - ☐ Suffering from higher bit error rates
    - Tolerate them?
    - Add more efficient means for error detection and correction- **this also increases overhead!.**

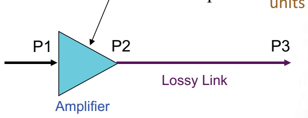
Decibels & Signal Strength

- The decibel notation (dB) is a logarithmic measure of the ratio between two signal power levels
  - N_{dB} = number of decibels
  - P_1 = input power level (Watts)
  - P_2 = output power level (Watts)

$$N_{dB} = 10\log_{10}\frac{P_2}{P_1}$$

P1, P2 Should have the same units

- e.g. → Amplifier gain
  → Signal loss
  (attenuation) over a link



Example:

- A signal with power level of 10mW is inserted into a transmission line
- Measured power some distance away is 5 mW
- Power "gain" in dBs is expressed as 10 log (out/in)

N_{dB}=10 log (5/10)=10(-0.3)= -3 dB (negative gain is loss)

- negative dBs: $P_3 < P_2$ (Loss), positive dBs: $P_2 > P_1$ (Gain)

Relationship between dB Values and Power ratio (P_2/P_1)

Log (Compressed)

Power Ratio	dB	Power Ratio	dB
1	0		
10^1	10	10^{-1}	-10
10^2	20	10^{-2}	-20
10^3	30	10^{-3}	-30
10^4	40	10^{-4}	-40
10^5	50	10^{-5}	-50
10^6	60	10^{-6}	-60
2	3	1/2	-3

1

1,000,000

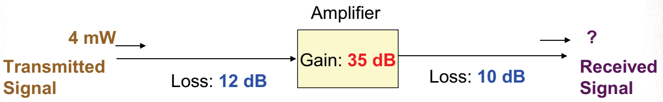
Linear

Decibels and Signal Strength

- Decibel notation is a relative (not absolute) measure:
  - A loss of 3 dB halves the power (e.g. 100 to 50, 16 to 8, …)
  - A gain of 3 dB doubles the power (e.g. 5 to 10, 7.5 to 15, …)
- Will see shortly how we can handle absolute levels
- Advantages of using dBs:
  - The "log" allows replacing:
    - Multiplication with Addition
      C = A * B
      Log C = Log A + Log B
    - and Division with Subtraction
      A = C / B
      Log A = Log C - Log B

Decibels and Signal Strength

- Example: Transmission line with an intermediate amplifier



- Net power gain over transmission path:

+ 35 – 12 – 10 = 13 dB (positive means there is net gain)

$$\therefore \quad 13 = 10\log_{10}\left[\frac{\text{Received Signal Power}}{4mW}\right]$$

$$\left[\frac{\text{Received Signal Power}}{4mW}\right] = \log^{-1}{}_{10}\left[\frac{13}{10}\right]$$

- Received signal power = (4 mW) $\log_{10}^{-1}(1.3)$ = 4 x $10^{1.3}$
  = 4 x $10^{1.3}$ mW = 79.8 mW

Transmission Impairments

- Signal received is often a degraded form of the signal transmitted
- Why? What happens en-route?... Impairments:
  - Attenuation:
    - Limits the bandwidth of the received signal
    - In-band signals arrive weaker
    - Attenuation distortion (Attenuation is not uniform over bandwidth)
  - Delay
  - Delay distortion
  - Noise and interference (including crosstalk)
- Effect:
  - On analogue data - Some degradation in signal quality
  - On digital data – Fatal bit errors (total bit reversals)

Attenuation

- Signal strength falls off with distance traveled
- Nature of loss in signal power depends on medium:
  - Guided (Wires, etc.):
    Signal power after travelling distance d
    - Exponential drop is signal power with distance: $P_d = P_0 e^{-\alpha d}$

      $10 \ln (P_d/P_0) = -\alpha\ d$

      $10 \log (P_d/P_0) = -\alpha' d$

    →Loss: α' dBs per km (α' depends on medium type e.g. fiber, twisted pair, cable)
  - Unguided (Open-space):
    - Inverse square law spread with distance: $P\ \alpha\ P_0/d^2$

    →Loss: 6 dBs for each distance doubling
    - Absorption and scattering by objects
    - May also depend on weather, e.g. rain, sunspots, etc

Effects of Attenuation

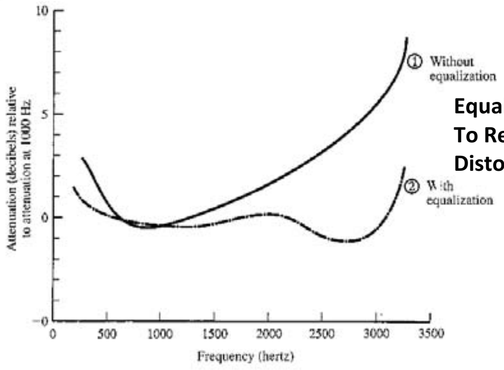
- Received signal strength must be:
  - Large enough to be detected
  - Sufficiently higher than noise to be interpreted correctly (without error)
- To overcome these problems:
  - Use amplifiers (analogue transmission mode)

  or repeaters (digital transmission mode) en-route
  - Amplifier gains should not be too large as this may cause signal distortion due to saturation (non-linearities)
  - Problem with networks: distance actually travelled (hence attenuation) will depend on actual route taken through the network!

Attenuation Distortion

- Attenuation usually increases with frequency
- This causes bandwidth limitation (understood)
- Moreover, over the transmitted bandwidth itself:
  - Different frequency components of the signal get attenuated differently →Signal distortion
  - Affects analogue signals more
- To overcome this problem:
  - Use **equalizers** that reverse the effect of frequency-dependent attenuation distortion:
    - **Passive**: e.g. loading coils in telephone circuits
    - **Active**: Amplifier gain designed specifically for this purpose

Attenuation Distortion

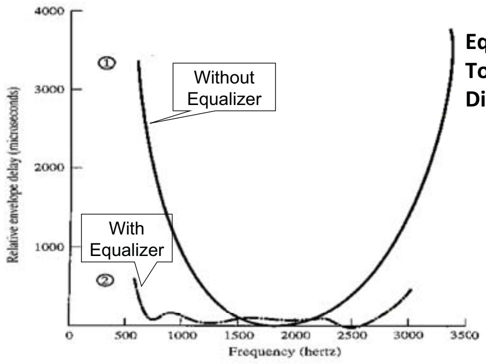


Equalization To Reduce Attenuation Distortion

Delay Distortion

- Happens only on guided media
- Wave propagation velocity varies with frequency:
  - Highest at the centre frequency (minimum delay)
  - Lower at both ends of the bandwidth (larger delay)
- Effect: Different frequency components of the signal arrive at slightly different times! (Dispersion in time)
- Affects digital data more: due to bit spill-over (timing is more critical here than for analogue data)
- Again, equalization can help overcome the problem

Delay Distortion

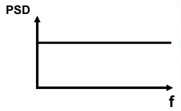


Equalization To Reduce Delay Distortion

Noise

- Definition: Any additional unwanted signal inserted between transmitter and receiver
- The most limiting factor in communication systems
- Noise Types:
  - Thermal Noise
  - Inter-modulation Noise
  - Crosstalk Noise
  - Impulse Noise

Noise



- Thermal (White) Noise
  - Due to thermal agitation of electrons

  (Increases with temperature)
  - **Uniformly distributed over frequency** (White noise)

  →Difficult to eliminate

  (exists even in the same bandwidth as your signal!, gets amplified!)
  - Effect is more significant on weak received signals, e.g. from satellites

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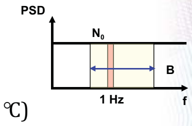
Thermal Noise, Contd.



- Thermal noise power density in 1 Hz of bandwidth, N_0 (Constant, Independent of frequency):

  $$N_0 = kT \quad (W / Hz)$$
  - k Boltzmann's constant = 1.38×10^{-23} J/K
  - T temperature in degrees Kelvin (= 273 + t ℃)
- Thermal noise power in a bandwidth of *B* Hz:

$$N = N_0 B = kTB \quad \text{(watts)}$$

$$= -228.6 + 10 \log T + 10 \log B \quad \text{(dBw)}$$

10 log k

Can you see some disadvantage now in having a larger BW?

Example: at t = 21 ℃ (T = 294 ºK) and for a bandwidth of 10 MHz:

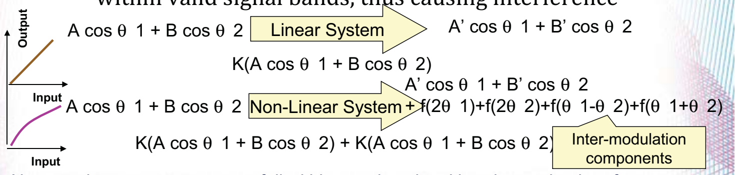
$N = -228.6 + 10 \log 294 + 10 \log 10^7$

$= -133.9$ dBW

Noise

- Inter-modulation Noise
  - Signals having the sum and difference (frequency mixing) of original frequencies sharing a transmission system

    f1, f2 →(f1+f2) and (f1-f2)
  - Caused by non-linearities in the medium and equipment, e.g. due to overdrive and saturation of amplifiers
  - Danger: Resulting new frequency components may fall within valid signal bands, thus causing interference



A cos θ 1 + B cos θ 2 Linear System A' cos θ 1 + B' cos θ 2

K(A cos θ 1 + B cos θ 2)

A cos θ 1 + B cos θ 2 Non-Linear System A' cos θ 1 + B' cos θ 2 + f(2θ 1)+f(2θ 2)+f(θ 1-θ 2)+f(θ 1+θ 2)

K(A cos θ 1 + B cos θ 2) + K(A cos θ 1 + B cos θ 2)

Inter-modulation components

New spurious components can fall within genuine signal bands causing interference

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Noise

- Crosstalk Noise
  - A signal from one channel picked up by another channel in close *proximity*
  - Examples:
    - Physical proximity: coupling between adjacent twisted pair channels

    → Shield cables properly
    - Directional proximity: antenna pick up from other directions

      → Use directional antennas
    - Spectral proximity: leakage between adjacent channels in frequency division multiplexing (FDM) systems

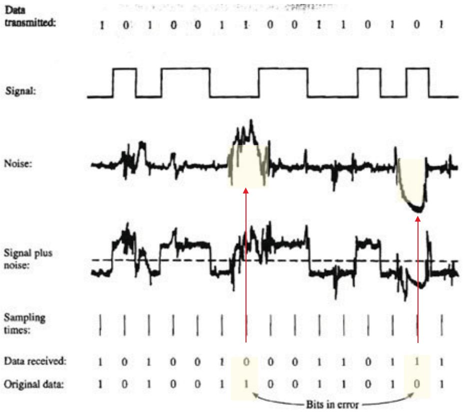
    → Use guard bands between adjacent channels

Noise

- Impulse Noise
  - Pulses (spikes) of irregular shape and high amplitude lasting short durations
  - Causes: External electromagnetic interference due to switching large currents, car ignition, lightning, …
  - Minor effect on analog signals (e.g. crackling noise in voice channels)
  - Major effect on digital signals- Bit reversal error!
  - More damage at higher data rates

  (a noise pulse of a given width can destroy a larger block of bits)

Effect of Impulse Noise on a Digital Signal



Q: What is the effect of the same noise at 10 times the data rate?

Channel Capacity

- Channel capacity: Maximum data rate usable under a given set of communication conditions
- How channel BW (B), signal level, noise and impairments, and the amount of data error that can be tolerated limit the channel capacity?
- In general, maximum possible data rate, C, on a given channel

= Function (B, Signal wrt noise, Bit error rate allowed)

  - Maximum data rate: Maximum rate at which data can be communicated on the channel, bits per second (bps)
    - Bandwidth: BW of the transmitted signal as constrained by the transmission system, cycles per second (Hz)
    - Signal relative to Noise, SNR = signal power/noise power ratio (Higher SNR → better communication conditions → higher C)
    - Bit error rate (BER) allowed: in (bits received in error)/(total bits transmitted). Equal to the bit error probability. e.g. Higher allowed → higher usable data rates → higher C

Channel Capacity, C

- So, in general: C bps = F(B, SNR, BER)
- Three formulations under different assumptions:

Idealistic

Assumptions	Formulation
Ideal: Noise-free, Error-free: C = F(B)	Nyquist
Noisy: Error-free: C = F(B, SNR)	Shannon
Practical: Noisy, Error: C = F(B, SNR, BER)	E_b/N_0 vs Error Rate

Realistic

Bandwidth or Spectral Efficiency (BE)

$$BE = \frac{Channel\ Capacity\ C}{Bandwidth\ B}, \qquad bps/Hz$$

- Measures how well we are utilizing a given bandwidth to send data at a high rate….
- Can be greater than 1 (not like engineering efficiencies)
- The larger the better

Nyquist Capacity (Noise-free, Error-free)

- Idealized, theoretical, assumes a noise-free → error-free channel
- Nyquist showed that (without noise, without errors): If rate of signal transmission is 2B then a signal with frequency components up to B Hz is sufficient to carry that signalling rate
- In other words: Given bandwidth B, highest signalling rate possible is 2B signal elements per second
- How much data rate does this represent?

(depends on how many bits are represented by each signal element!)

  - Given a binary signal (1,0), data rate is same as signal rate → Data rate supported by a BW of B Hz is 2B bps → C = 2B
  - For the same B, data rate can be increased by sending one of M different signals (symbols): as each signal level now represents $\log_2 M$ bits
- Generalized Nyquist Channel Capacity, $C = 2B \log_2 M$ bits/s (bps)
- Bandwidth efficiency = $C/B = 2 \log_2 M$, dimensionless quantity

---

Nyquist Bandwidth: Example

- $C = 2B \log_2 M$ bits/s
  - C = Nyquist Channel Capacity
  - B = Bandwidth
  - M = Number of discrete signal levels (symbols) used
- Data on telephone channel:
- B = 3400-300 = 3100 Hz
- With a binary signal (M = 2 levels)

$C = 2B \log_2 2 = 2B \times 1 = 6200$ bps

- With a quadnary signal (M = 4 levels)

$C = 2B \log_2 4 = 2B \times 2 = 4B = 12{,}400$ bps

- Channel capacity increased, but larger number of signal levels (M) makes it more difficult for the receiver to determine data correctly in the presence of noise

Shannon Capacity (Noisy, Error-Free)

- Highest error-free data rate in the presence of noise
- Signal power to noise power S/N = signal/noise levels

  $SNR_{dB} = 10 \log_{10} (S/N)$
- Errors are less likely with lower noise (larger SNR). This allows higher error-free data rates i.e. larger Shannon channel capacities
- Shannon Capacity $C = B \log_2(1+S/N)$

  Caution! $\log_2$ Not $\log_{10}$
- For a given BW, the larger the SNR the higher the data rate I can use without introducing errors
- C/B: Spectral (bandwidth) efficiency, BE, (bps/Hz) (>1)
- Larger BEs mean better utilization of a given bandwidth B for transmitting data fast.

Shannon Capacity: Comments

- Formula says: for data rates ≤ calculated C, it is theoretically possible to find an encoding scheme that achieves error-free transmission at the given SNR… But it does not say how!

It is a theoretical approach based on thermal (white) noise only. But in practice, we also have impulse noise, attenuation and delay distortions, etc…

  - So, maximum error-free data rates measured in practice are expected to be lower than the C predicted by the Shannon formula due to the greater noise
- However, maximum error-free data rates can be used to compare practical systems: The higher that rate the better the system…

Shannon Capacity: Comments

- Formula suggests that changes in B and SNR can be done arbitrarily and independently… but

→ In practice, this may not be the case!

  - Higher SNR obtained through excessive amplification may also introduce non-linearities → increased distortion and inter-modulation noise … which reduces SNR!
  - High Bandwidth B opens the system up for more thermal noise (kTB), and therefore reduces SNR!

Shannon Capacity: Example

Spectrum of communication channel extends from 3 MHz to 4 MHz

SNR = 24dB

Then B = 4MHz – 3MHz = 1MHz, $SNR_{dB} = 24dB = 10 \log_{10} (S/N)$

$S/N = \log^{-1}_{10} (24/10) = 10^{24/10} = 251$

- Using Shannon's formula: $C = B \log_2 (1+ S/N)$

  $C = 10^6 * \log_2(1+251) \sim 10^6 * 8 = 8$ Mbps
- Based on Nyquist's formula, determine M that gives the above channel capacity: $C = 2B \log_2 M$

  $8 * 10^6 = 2 * (10^6) * \log_2 M$

  $4 = \log_2 M$, therefore M = 16

E_b/N_0 vs Error Rate Formula

- Handling both noise and a quantified error rate simultaneously
- We introduce E_b/N_0: A standard quality measure of three channel parameters (B, SNR, R) and can also be independently related to the error rate

  R is the data rate. Max value of R is the channel capacity C
- It expresses SNR in a manner related to the data rate, R
  - E_b = Signal energy in one bit interval (Joules)

    = Signal power (Watts) x bit interval T_b (second)

    = S x (1/R) = S/R
  - N_0 = Noise power (watts) in 1 Hz = kT.

$T_b = 1/R$

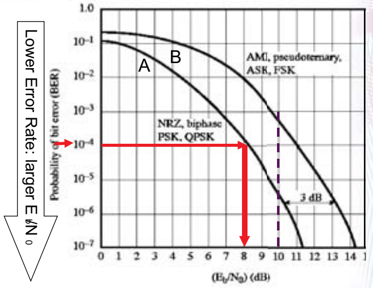
$$\frac{E_b}{N_0} = \frac{ST_b}{N_0} = \frac{S/R}{kT} = \frac{S}{kTR} \qquad \frac{E_b}{N_0} = \frac{S/R}{N_0} = \frac{S}{N}\frac{B_T}{R} = SNR\left(\frac{B_T}{R}\right)$$

= SNR/BE

E_b/N_0

- Bit error rate for digital data is a decreasing function of E_b/N_0 for a given signal encoding scheme
- **Analysis**: ⇒For a given system (SNR, B, R) → (E_b/N_0), determine error rate BER
- **Design**: ⇒Given a desired error rate BER, get E_b/N_0 to achieve it, then determine other parameters from formula, e.g. S, SNR, R, etc.
- Effect of S, R, T on error performance
- Which encoding scheme is better: A or I?

BER vs E_b/N_0 curve for a given encoding scheme



$$\left(\frac{E_b}{N_0}\right)_{dB} = S_{dBW} - 10\log R - 10\log k - 10\log T$$

$$= S_{dBW} - 10\log R + 228.6\ dBW - 10\log T$$

$$\frac{E_b}{N_0} = \frac{S/R}{N_0} = \frac{S}{N}\frac{B_T}{R} = SNR\left(\frac{B_T}{R}\right) = \frac{\mathbf{SNR}}{\mathbf{BE}}$$

Max R = C, BE = C/B

Example:

$$\left(\frac{E_b}{N_0}\right)_{dB} = S_{dBW} - 10\log R - 10\log k - 10\log T$$

$$= S_{dBW} - 10\log R + 228.6\ dBW - 10\log T$$

- Given:
  - The effective noise temperature, T, is 290°K
  - The data rate, R, is 2400 bps
  - Would like to operate with a bit error rate of 10^{-4} (e.g. 1 error in 10^4 bits)

What is the minimum signal level required for the received signal?

- From curve, a minimum E_b/N_o needed to achieve a bit error rate of 10^{-4} = 8.4 dB
- 8.4 = S(dBW) – 10 log 2400 + 228.6 dBW – 10 log290
  = S(dBW) – (10)(3.38) + 228.6 – (10)(2.46)

S = -161.8 dBW

E_b/N_0 in terms of BE, assuming Shannon channel capacity

- From Shannon's formula:

$$C = B \log_2(1+SNR)$$

We have:

$$SNR = (2^{C/B} - 1) = (2^{BE} - 1)$$

- From the E_b/N_0 formula:

$$\frac{E_b}{N_0} = \frac{SNR}{BE} = \frac{1}{BE}(2^{BE} - 1)$$

C/B (bps/Hz) is the spectral (bandwidth) efficiency BE based on Shannon channel capacity

---

Example

- Find the minimum E_b/N_0 required to achieve a Shannon bandwidth efficiency ($BE=C_{Shannon}/B$)of 6 bps/Hz:

$$\frac{E_b}{N_0} = \frac{1}{BE}(2^{BE} - 1)$$

- Substituting in the equation above:

$E_b/N_0 = (1/6)\ (2^6 - 1) = 10.5 = 10.21$ dB

Wireless Communications

- Transmission Media

Contents

- Overview
- Radiated Transmission
  - Antennas
  - Terrestrial Microwaves
  - Satellite Microwaves
  - Broadcast Radio
  - Infrared

Overview

- Media:
  - Guided – wire or fibre
  - Unguided - wireless
- Transmission characteristics and quality determined by:
  - Signal
  - Medium
- For guided, the medium is more important
- For unguided, the bandwidth provided by the antenna is more important

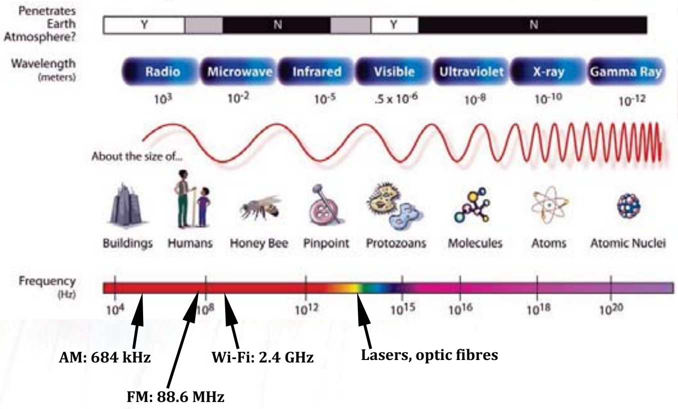
Design Issues

- Key communication objectives are:
  - High data rate
  - Low error rate
  - Long distance
  - Bandwidth economy: Trade-off
    - Want larger BW for higher data rates: C α B
    - But limited by economy: Larger BW is costly e.g. Coaxial vs TP
- Transmission impairments
  - Attenuation: Twisted Pair > Cable > Fibre (best)
  - Interference:

    Worse with unguided… (the medium is shared!)
- Number of receivers
  - In multi-point links of guided media:

    More connected receivers introduce more attenuation

The Electromagnetic Spectrum



AM: 684 kHz

Wi-Fi: 2.4 GHz

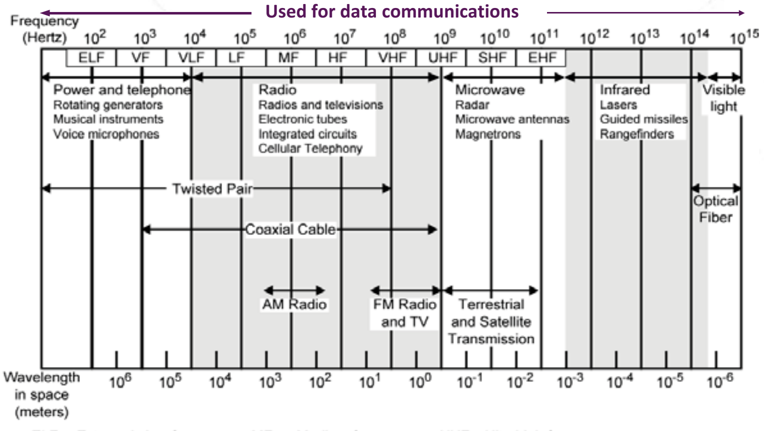
Lasers, optic fibres

FM: 88.6 MHz

SI Units Prefixes 10^{-24} to 10^{+24}

yotta: Y 10^{24} = 1,000,000,000,000,000,000,000,000
zetta: Z 10^{21} = 1,000,000,000,000,000,000,000
exa: E 10^{18} = 1,000,000,000,000,000,000
peta: P 10^{15} = 1,000,000,000,000,000
tera: T 10^{12} = 1,000,000,000,000
giga: G 10^{9} = 1,000,000,000
mega: M 10^{6} = 1,000,000
kilo: k 10^{3} = 1,000
hecto: h 10^{2} = 100
deka: da 10^{1} = 10
deci: d 10^{-1}= 0.1
centi: c 10^{-2}= 0.01
milli: m 10^{-3} = 0.001
micro: µ 10^{-6} = 0.000001
nano: n 10^{-9} = 0.000000001
pico: p 10^{-12} = 0.000000000001
femto: f 10^{-15} = 0.000000000000001
atto: a 10^{-18} = 0.000000000000000001
zepto: z 10^{-21} = 0.000000000000000000001
yocto: y 10^{-24} = 0.000000000000000000000001

Electromagnetic Spectrum



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

Wireless Transmission

- Free-space is the transmission medium
- Need efficient radiators, called **antennas**
  - Signal fed from transmission line (wireline) and radiated it into free-space (wireless)
- Popular applications
  - Radio & TV broadcast
  - Cellular Communications
  - Microwave Links
  - Wireless Networks

Wireless Transmission Frequency Ranges

- Radio: 30 MHz to 1 GHz
  - Omni-directional
    - Broadcast radio e.g. FM
- Microwaves: 1 GHz to 40 GHz
  - Highly directional beams
    - Point to Point (Terrestrial)
    - Satellite
- Infrared Light: 0.3 THz to 20 THz (below light)
  - Localized communications (confined spaces)

Antennas

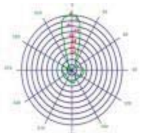
- Electrical conductor (or system of conductors) used to radiate / collect electromagnetic energy into/from surrounding space
- Transmission
  - Radio frequency electrical energy from
  - Converted into electromagnetic energy
  - Radiated into surrounding space
- Reception
  - Electromagnetic energy impinging on antenna
  - Converted to radio frequency electrical energy
  - Fed to receiver
- Same antenna often used for both Tx and Rx in 2-way communication systems

Radiation Pattern

- Power radiated in all directions, but usually not with the same efficiency
- **Isotropic antenna**
  - A hypothetical point source in space (Small dimensions relative to λ)
  - Radiates equally in all directions giving a spherical radiation pattern
  - Used as a reference for other antennae
- **Directional Antenna**
  - Concentrates radiation in a given desired direction hence point-to-point, line of sight communications
  - Gives antenna **‘gain’** in that direction relative to isotropic for both Tx and Rx
  - **Larger dimensions relative to λ → Greater directivity**

Radiation Patterns

Isotropic

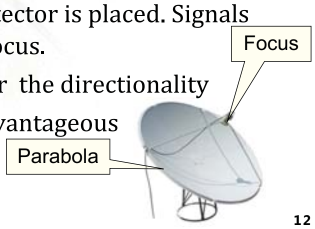


Directional

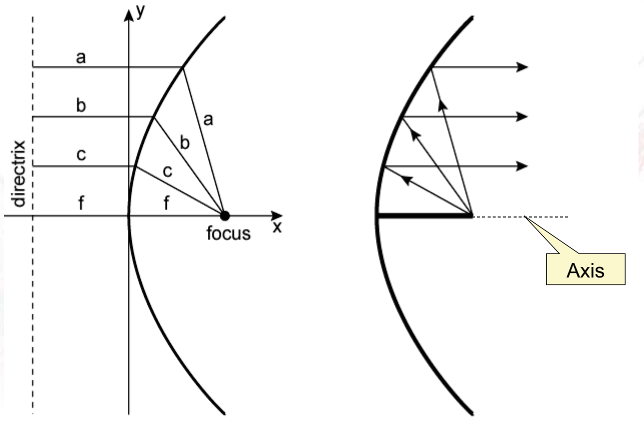
Parabolic Reflective Antenna

- Used for terrestrial and satellite microwave
- Source placed at the focal point will produce waves that get reflected from parabola *parallel* to the parabola axis
  - Creates a (theoretically) parallel beam of light/sound/radio that does not spread (disperse) in space
  - In practice, some divergence (dispersion) occurs, because source at focus has a finite size (not exactly a point!)
- On reception, only signal from the axis direction is concentrated at focus, where detector is placed. Signals from other directions miss the focus.
- The larger the antenna the better the directionality

→so, using Higher frequency is advantageous



Parabolic Reflective Antenna



(a) Parabola

(b) Cross-section of parabolic antenna showing reflective property

Parabolic Antenna Gain, G

- A measure of antenna directionality
- Power output of the antenna in a particular direction compared to that produced by a perfect isotropic antenna
- Can be expressed in decibels (dB, dBi) (i = relative to isotropic)
- Increased power radiated in one direction causes less power radiated in another direction (Total power is fixed)
- Effective area A_e:
  - Related to size and shape of antenna
  - Determines the antenna gain G

$$G = \frac{4\Pi A_e}{\lambda^2} = \frac{4\Pi f^2 A_e}{c^2}$$

Antenna Gain, G

$$G = \frac{4\Pi A_e}{\lambda^2} = \frac{4\Pi f^2 A_e}{c^2}$$

- An isotropic antenna has a gain G = 1 (0 dBi)
- i.e. $A_e = \frac{\lambda^2}{4\pi}$ $(\approx 0.1\,\text{cm}^2 \text{ at } 30\,\text{GHz - a 'Point Source''})$
- A parabolic antenna has:

$$A_e = 0.56A$$

A = Actual Area = Πr^2

-
- Substituting we get: $G = \frac{4\pi\,(0.56A)}{\lambda^2} \approx \frac{7A}{\lambda^2}$
- Gain in dBi = 10 log G
- Important: Gains apply to *both* Tx and Rx antennas

Propagation Attenuation

- As signal propagates in space, its power drops with distance according to the inverse square law

$$P_d \; \alpha \; \frac{1}{d'^2}$$

d' = distance in λ 's

While with a guided medium, signal drops exponentially with distance… giving larger attenuation and lower repeater spacing

i.e. loss in signal power over distance travelled, d

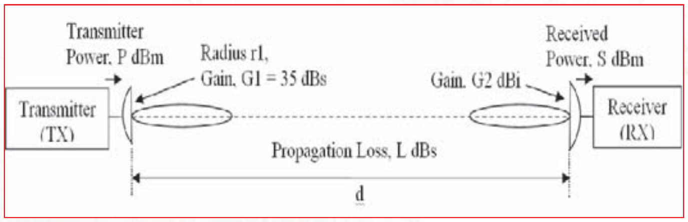
$$L = \frac{P_t}{P_r} = \frac{(4\Pi d)^2}{\lambda^2} = \frac{(4\Pi f d)^2}{c^2}$$

$$L_{dB} = 10 log L$$

- **Show that L increases by 6 dBs for every doubling of distance d.**
- **For guided medium, corresponding attenuation = α d dBs, α in dBs/km**

A disadvantage for operating at higher frequency?

Microwave Link Calculations



TX-RX Net attenuation A dBs = L – G1 – G2

S dBm = P dBm – A dBs

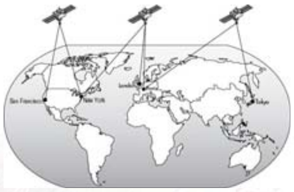
Terrestrial Microwave

- Parabolic dish
- Focused beam (with antenna gain)
- Line of sight requirement:
  - Beam should not be obstructed
  - Curvature of earth limits maximum range
  - Use relays to increase range (multi-hop link)
  - Link performance sensitive to antenna alignment
- Applications:
  - Long haul telecommunications
    Many voice/data channels over long distances between large cities, through intermediate relays. Competes with coaxial cable and fibre
  - Short wireless links between buildings:
    - CCTV links
    - Wireless links between LANs in close-by buildings
  - Cellular Telephony

Terrestrial Microwave: Transmission Properties

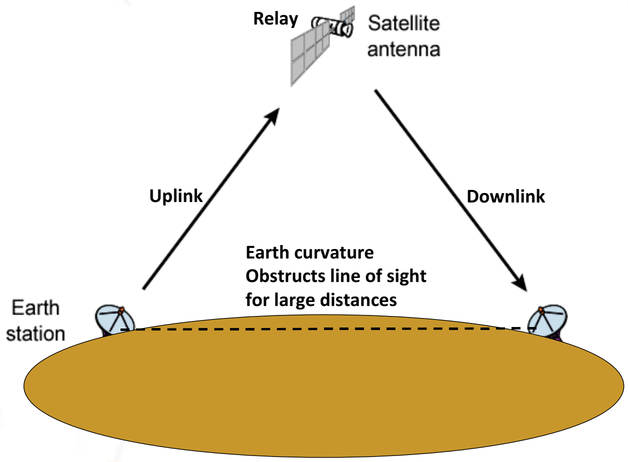
- Operating Frequency range: 1 - 40 GHz
- Higher frequency advantages:
  - Larger bandwidth, B implies higher data rate
  - Smaller λ implies smaller (lighter hence cheaper) antenna for a required antenna gain.
- But higher frequency implies larger attenuation due propagation and absorption by rain. So,
  - Long-haul links (long distances) operate at lower frequencies (4-6 GHz,11 GHz) to avoid large attenuation
  - Short links between close-by buildings operate at higher frequencies e.g. 22 GHz. (Attenuation is not a big problem for the short distances, smaller antenna size)

Satellite Microwave

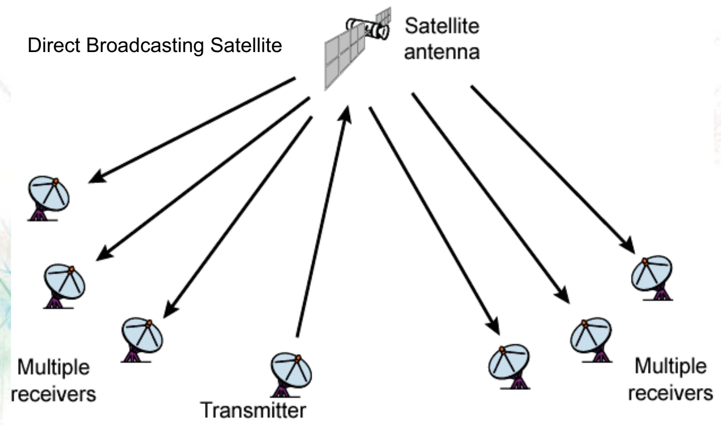


- Satellite acts as a relay station for the link
- Satellite receives on one frequency (uplink), amplifies or repeats signal and re-transmits it on another frequency (downlink)
- Spatial angular separation (e.g. 3°) to avoid interference from neighbouring TXs
- Require a geo-stationary orbit (satellite rotates at the same speed of earth rotation, so appears stationary):
  - Height: 35,784 km (long link, large transmission delays)
- Applications:
  - Television direct broadcasting
  - Long distance telephony
  - Private business networks linking multiple company sites worldwide

a. Satellite Point to Point Link



b. Satellite Broadcast Link



Satellite Microwave: Transmission Properties

- 1-10 GHz
- Frequency Trade offs:
    - Lower frequencies: More noise and interference
    - Higher frequencies: Larger rain attenuation, but smaller antennas
- Downlink/Uplink frequencies recently going higher: 4/6 GHz →12/14 →20/30 (better receivers becoming available)
- Delay = 0.25 s →noticeable for telephony
- Inherently a broadcasting facility

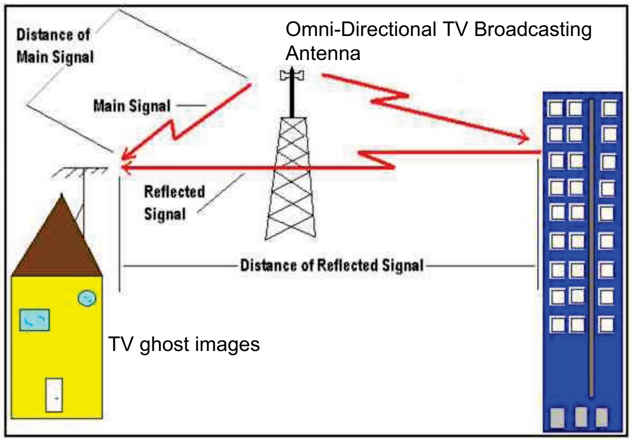
Broadcast Radio: 30 MHz – 1 GHz

- Omni directional (no need for antenna directionality horizontally)
    - No dishes
    - No line of sight requirement
    - No antenna alignment requirement/problems
- Applications:
    - FM radio
    - UHF and VHF television
- Choice of frequency range:

  Reflections from ionosphere < 30 MHz -1 GHz < Rain
- Propagation attenuation:

  Lower than for Microwaves (as λ is larger)
- Problems caused by omni directionality: Interference due to multi-path reflections
    - e.g. TV ghost images

Multi-Path effects of omni-directionality



Infra-red

- Data modulates a non-coherent infra-red light
- Relies on line of sight (or reflections through walls or ceiling)
- Blocked by walls (unlike microwaves)
- No licensing required for frequency allocation
- Applications:
    - TV remote control
    - Transfer between mobile devices e.g. notebooks, cellular phones, digital camera, etc.

Wireless Communications

- Encoding & Modulation (Part 1)

Contents

- Encoding and Modulation (Part I)
  - Digital data into digital signals
  - Digital data into analogue signals

Four Data/Signal Combinations

		Signal	
		Analogue	**Digital**
Data	**Analogue**	- Same spectrum as data (baseband): e.g. Telephony - Different spectrum (carrier modulation): e.g. AM, FM, PM ④	Use a (converter): codec, e.g. PCM (pulse code modulation) ③
	Digital	Use a (converter): modem e.g. ASK, FSK, PSK ②	-Two signal levels: e.g. NRZ -More complex encoding: e.g. Manchester ①

Encoding Techniques

Digital data as digital signal

Digital data as analogue signal: Converter (modem)

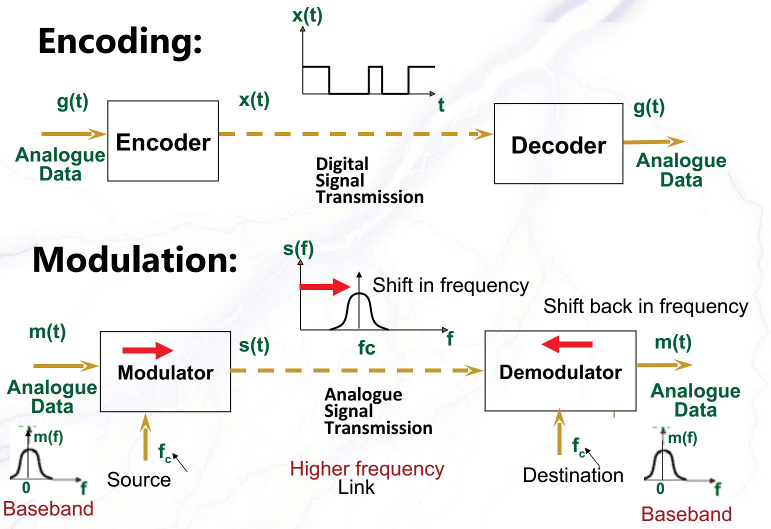
Analogue data as digital signal: Converter (codec)

Analogue data as analogue signal

In general:

- When the outcome is a <u>digital signal</u> we use an **Encoding** process
- When the outcome is an <u>analogue signal</u> we use a **Modulation** process
- The modulation of an <u>analogue signal</u> from digital data is called **shift-keying**

Encoding:



Modulation:

Encoding and Modulation: Remarks

- Encoding is simpler and less expensive than modulation
- Encoding into digital signals allows use of modern digital transmission and switching equipment
  - Basis for Time Division Multiplexing (TDM)
- Modulation shifts baseband signals to a higher region in the frequency spectrum (needs same f_cs at **both** ends)
  - Basis for Frequency Division Multiplexing (FDM)
- Optical fibers and unguided media can carry only analogue signals

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Terminology

- **Unipolar Signals**
  - Binary data represented by signals of the *same* polarity, e.g. 0 -> +5 V, 1 -> +10 V ⇒DC content
- **Bipolar (Polar) Signals**
  - Binary data represented by signals of *opposite* polarity, e.g. 0 -> +5 V, 1 -> -5 V ⇒ideally Zero DC content
- **Mark and Space**
  - Binary 1 and Binary 0 respectively
- **Duration of a bit (T_b)**
  - Time taken for transmitter to emit a <u>data bit</u>
- **Data rate, R (= 1/T_b)**
  - Rate of data transmission measured in **bits per second (bps)**
- **Duration of a Signal Element (T_s)**
  - <u>Minimum</u> duration of a signal pulse
- **Modulation (signaling) rate, D (1/T_s)**
  - Rate at which the signal level changes with time measured in **bauds = signal elements per second**

Not always Tb = Ts !!!
e.g. Multi-symbol transmission
(M = 4, 8, …): Tb < Ts

Example: Two different coding methods

Data rate = $1/1\mu s$

= 1 Mbps

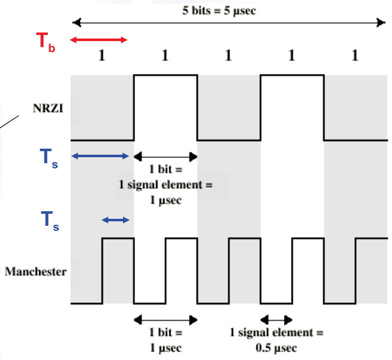
(in both cases)

Signaling Rate for NRZ-I: = $1/1\mu s$

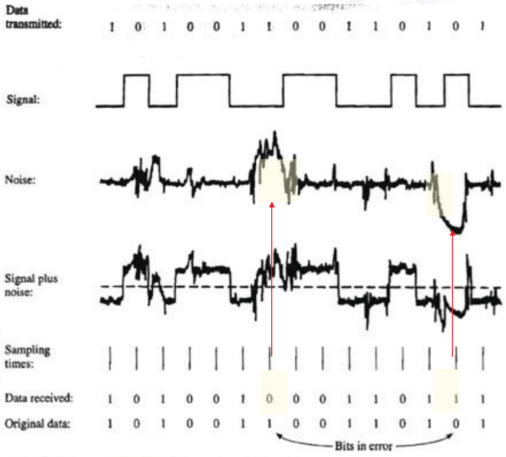
= 1 Mbauds

Signaling Rate for Manchester: = $1/0.5\mu s$

= 2 Mbauds



Interpretation of the Received Signal



Interpreting Received Signals

- Requirements at RX:
  - Determine timing of bits – Bit start and end (When to look)
    ⇒Need Synchronization (later)
  - Detect signal levels at mid-bit points
  - Compare signal level with a threshold level to decide on data
- Factors affecting successful signal interpretation
  (Affect bit error rate)
  - Bandwidth
  - Signal to noise ratio
  - Data rate
  - **Also Encoding/Modulation scheme, e.g. binary or multi-level**

---

1. Digital Data, Digital Signal

- Digital signal
  - Voltage/current pulses having a few discrete levels (2 levels for binary)
  - Each pulse is a signal element
  - Binary data is encoded into those signal elements
  - Encoding is the mapping of data to signal elements

Encoding Schemes

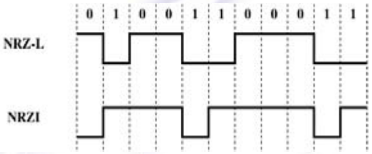
Schemes for encoding digital data as digital signals

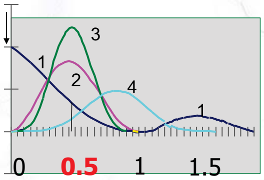
- **Non-return to Zero** (NRZ) Group:
  - Non-return to Zero-Level (NRZ-L)
  - Non-return to Zero Inverted (NRZ-I)
- **Multi-level Binary** Group:
  - Bipolar-AMI (Alternate Mark Invert)
  - Pseudoternary
- **Bi-Phase** (RZ) Group:
  - Manchester
  - Differential Manchester
- **Scrambling** Group:
  - B8ZS (Bipolar with 8-Zeros Substitution)
  - HDB3 (High Density Bipolar 3-Zeros)

Aspects of comparison between schemes

- **Clocking:** Synchronizing RX to TX can be achieved using:
  - An external clock, or better:
  - A built-in synchronizing mechanism in the **signal** itself! (so, a code with many signal transitions is better)
- **Error detection**
  - Mostly handled by higher layers, e.g. data-link control
  - But error detection capabilities built into the signal encoding scheme would help!
    Advantage: Implemented much faster (in hardware)
- **Performance with interference and noise**
  - Some encoding schemes perform better than others: e.g. with differential encoding: data is encoded as signal transition/no signal transition, and data detection at RX is **less affected by noise.**
- **Cost and complexity**
  - Some codes require signaling at a rate greater than the data rate (e.g. RZ) At higher signaling rates this requires higher bandwidth, faster circuits, etc. (larger costs)

NRZ Group
Pros and Cons:

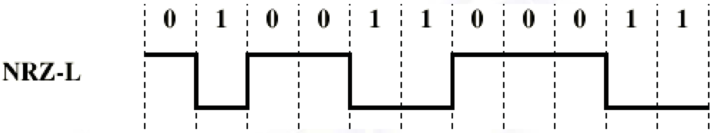




- Pros
  - Easy to implement
  - Modest bandwidth requirements
- Cons
  - Large DC component
  - Poor TX-RX synchronization:

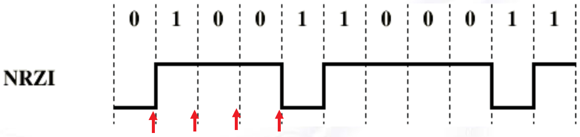
  e.g. **No signal transitions for long strings of 0's or 1's** (so few edges are available for synchronization)
- Used for magnetic recording
- Not used much for signal transmission

NRZ-L: Non-return to Zero Level



- Two different signal voltages for the 0 and 1 data bits
- Voltage level is constant (no return to zero, so no signal transition) for the full duration of the data bit interval
- e.g. positive voltage for space and 0V for mark
- More often, negative voltage for one data value and positive for the other (bipolar signal) (Why?)
- An example of absolute encoding: Mapping data **directly** to signal **levels**

NRZ-I: Non-return to Zero Invert

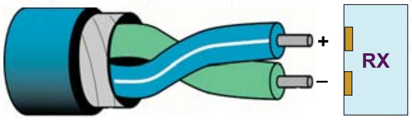


- Still constant voltage level for bit duration of (hence NRZ)
- But data is encoded as presence or absence of signal transition at the beginning of bit time:
  - Transition (low to high or high to low): Denotes binary 1
  - No transition: Denotes binary 0
- This is an example of differential encoding:Encoding data as a change/no change in signal level.

Differential Encoding

- Data is represented by signal transitions rather than signal levels
- Advantages;
  - With noise, signal transitions (or lack of them) are detected more easily than signal levels ⇒Better noise immunity
  - In complex transmission layouts, it is easy to accidentally lose sense of polarity

Effect of swapping terminals on:
- **NRZ-L**
- **NRZI**

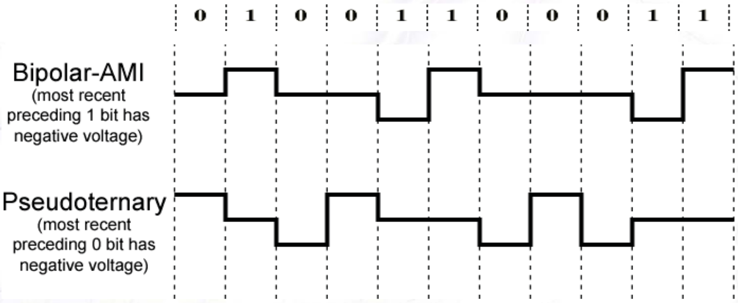


The Multilevel Binary Group

- Uses more than two signal levels (3 in this case)
- Signal is multi-level but data is still binary!
- Bipolar-AMI (Alternate Mark (1) Inversion)
  - 0 data is represented by no line signal
  - 1 data represented by positive or negative pulse
  - The "1" pulses alternate in polarity (why? 2 reasons!)
  - Advantages:
    - No net DC component (for any data sequence!)
    - Lower bandwidth than NRZ
    - No loss of sync with a long string of 1'z (but zeros still a problem- Will try to solve it later)
    - Alteration of pulse polarity also useful for error detection

---

Bipolar-AMI and Pseudoternary



Pseudoternary

- Opposite of Bipolar-AMI:
  - 1 represented by no line signal
  - 0 represented by alternating positive and negative pulses
- Could be called Bipolar-ASI: (Why?)
- No advantage or disadvantage over bipolar-AMI

The Multilevel Binary Group: Advantages

- No net DC component
- Spectrum centered at the middle of the BW
- Lower bandwidth than NRZ
- No loss of sync with a long string of 1’z (but zeros still a problem- Will try to solve it later)
- Alteration of pulse polarity also useful for error detection: Next slide

---

Disadvantages of Multilevel Binary

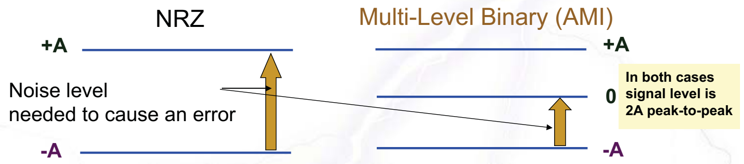
$N = \text{Log}_2\,(M)$

No. of bits sent during each signal element

No. of signal levels used

- Coding scheme not as efficient as NRZ:
  - We send only one bit at a time (1 or 0 data) ⇒Only $M = 2^1 = 2$ signal levels should be enough, but we are sending 3 levels > 2 !
  - We use 3 levels ⇒Enough to represent $\log_2 3 = 1.58$ bits > 1 bit !
- Receiver Design and Noise Performance
  - Now receiver must distinguish between three signal levels (+A, -A, 0) ⇒Need better receiver design
  - Requires approximately 3dB higher SNR for the same probability of bit error (bit error rate)

Performance with noise: NRZ Vs AMI



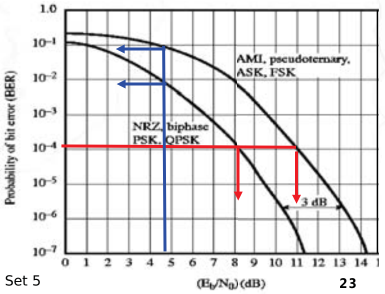
In both cases signal level is 2A peak-to-peak

- For the same error rate: AMI requires higher SNR noise (lower noise) ⇒i.e. higher E_b/N_0 (for same B and R)

(hence the 3 dBs difference between the two curves)

- For the same SNR (same E_b/N_0) AMI has higher error rate
- i.e. AMI has poorer performance with noise

$$\frac{E_b}{N_0} = SNR\left(\frac{B}{R}\right)$$



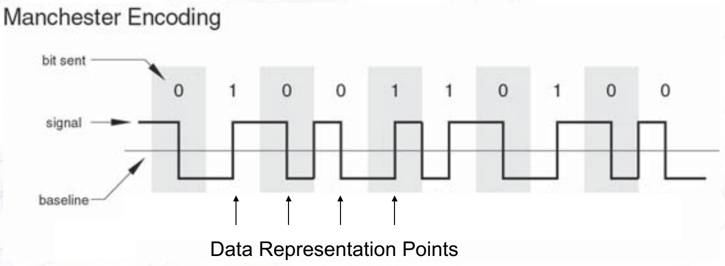
The Biphase Group (2 signal phases per bit)

- Manchester
  - Transition in middle of each bit period
  - Transition serves both as a clock edge and data representation
    - Low to high represents 1
    - High to low represents 0
  - Used by the IEEE 802.3 specification for Ethernet LAN (short distances)
- Differential Manchester
  - Dedicated mid-bit transition used **only** for clocking
  - Data representation is at start of bit:
    - No transition at start of a bit period represents 1
    - Transition at start of a bit period represents 0 (Inverts on 0’s – opposite of NRZ-I)
  - An example of differential encoding
  - Used by IEEE 802.5 specification for Token Ring LAN

Manchester Encoding

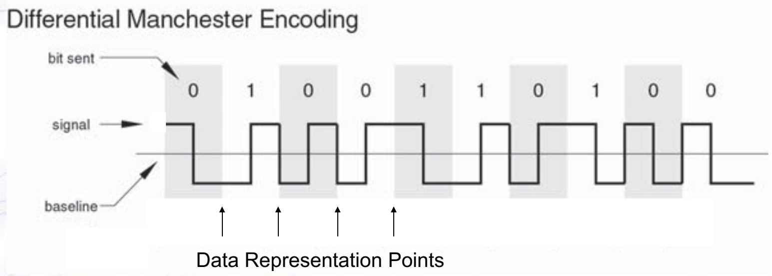
- **Mandatory transition in middle of each bit period**
  → Low to high represents 1
  → High to low represents 0
- **Transitions at start of bit only where required**

Note: This is not differential



Differential Manchester Encoding

- **Mandatory mid-bit transition for clocking**
- **Data represented by transition or no transition at bit start:**
  - **→ Transition (either direction) represents 0 (Invert on zeros)**
  - **→ No transition represents 1**



Biphase Pros and Cons

- Pros
  - Guaranteed mid bit transitions
    - Synchronization facility (self clocking codes)
  - Ideally no dc component (using bipolar signals)
  - Error detection
    - Detecting absence of expected (mandatory) transitions
- Cons
  - At least one transition per bit time and possibly two
    - Modulation (signaling) rate as high as twice that of NRZ
    - So, requires more bandwidth
    - Therefore, used over shorter distances (in LANs)

Data rate & Modulation (signaling) rate

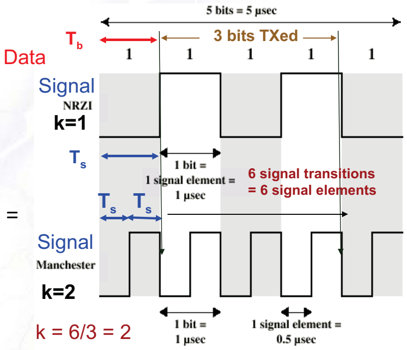
- Data rate, R = 1/T_b bps
- Signaling Rate, D = 1/T_s bauds

If we use k signal elements per bit, then:

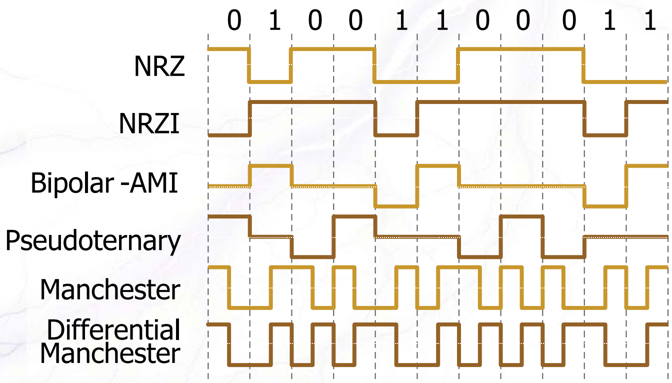
- Signaling (modulation) rate, D = Data rate, R (bit/s)
  x k (signal elements/bit)

Signal elements/s (bauds)

- k = No. of signal elements/bit
  = No. of signal transitions (both ways)÷ No. of bits transmitted, n (over a given period of n T_bs)



1. Digital data, Digital signal Encoding



Scrambling Group: B8ZS, HDB3

- Use bit scrambling to replace data bit sequences that would otherwise produce a <u>constant</u> signal voltage, with a more appropriate bit sequence producing signal changes
- Helps overcome constant DC problems with Multilevel Binary codes (poor synchronization)
- So, a “filling” (replacement) bit sequence is inserted where necessary
- Criteria for a “Filling sequence”
  - Should produce enough transitions for synchronization
  - Must be recognized by receiver for replacement with original data
  - Not likely to be generated by noise (difficult for noise/interference to produce it)
  - Should occupy the same bit length as original data (so no extra overhead in the data rate)

Scrambling Group: B8ZS, HDB3

- Advantages:
  - No long sequences of zero level line signal
  - No DC component
  - No reduction in useful data rate (No extra data sent)
  - Built-in error detection capability

B8ZS

- Bipolar With 8 Zeros Substitution
- Improvement on bipolar-AMI
- If an octet of 8 zeros and the last pulse preceding was positive (+):Transmitter encodes the 8 zeros as 000+-0-+

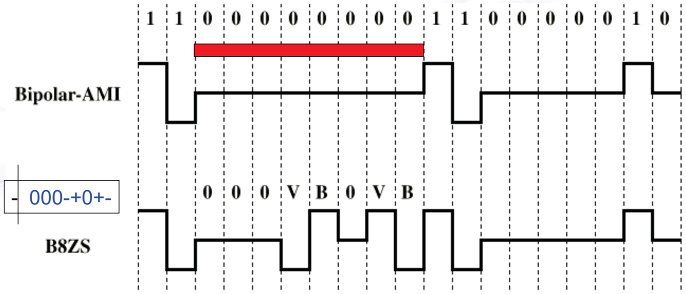
  (how many level changes does this introduce?)
- If an octet of 8 zeros and last voltage pulse preceding was negative (-): Transmitter encodes as 000-+0+-

  (shown in Fig. 5.6)
- Each insertion has *two* intentional violations of the basic AMI code rule: (violations alternate in polarity- no net DC added)

  +000+-0-+

  -000-+0+-
- A strange event ⇒unlikely to be caused by noise
- Receiver should detect it and interpret as an octet of 8 zeros (original data)
- No additional data sent ⇒No penalty on genuine data rate

B8ZS



V: Violation

B: Bipolar (Valid)

See how the insertion satisfies the 5 requirements:

-Detectable at RX

-Difficult for noise to generate

-Introduces transitions

-Does not introduce DC (alternate violations)

-Error detection capability

HDB3

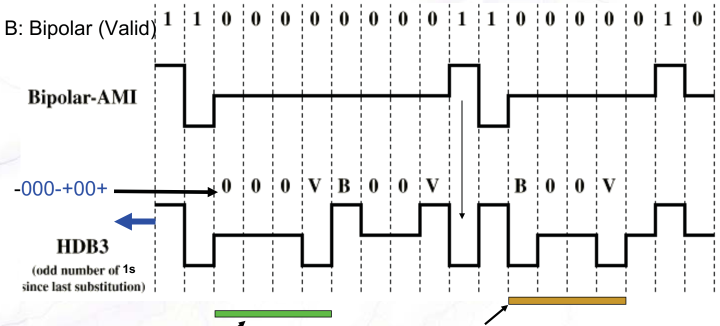
- **H**igh **D**ensity **B**ipolar **3** Zeros
- Also based on bipolar-AMI
- 4th zero always replaced with an intentional code violation
- String of four zeros replaced with either:
  - 1 pulse - 000- or + 000+ (violation with preceding pulse)
  - or 2 pulses - +00+ or + -00- (internal violation **within** the insertion)
- What determines whether 1 or 2 pulses?
  - Successive insertion violations must *alternate* in polarity:

    - 00000000 ⇒- 000-+00+ or + 00000000 ⇒+ 000+-00-
  - With insertions separated by n ‘1’ pulses: The new insertion is determined by the following rules:
    - If n is even, with last pulse p (+ or -) ⇒**p00p**
    - If n is odd, with last pulse p (+ or -) ⇒**000p**

HDB3

V: Violation

B: Bipolar (Valid)



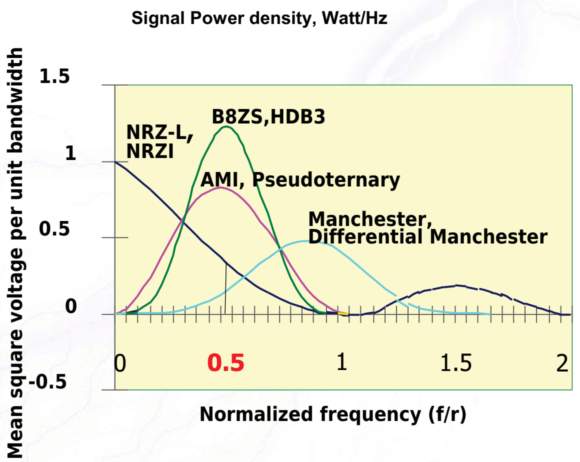
ODD number of 1s after last substitution, with the last pulse (-) ⇒ **000p** ⇒000-

p

EVEN number of 1s after last substitution, with the last pulse (+) ⇒**p00p** ⇒-00-

p

B8ZS, HDB3 Spectrum



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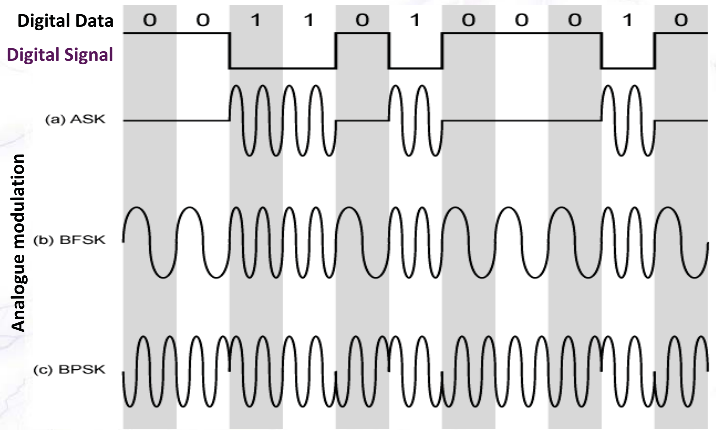
2. Digital Data, Analog Signal Encoding

- e.g. over public telephone system
  - 300Hz to 3400Hz
  - Use modem (modulator-demodulator)
- Modulation (here called shift keying) manipulates one or more property of a carrier sine wave:
  - Amplitude shift keying (ASK)
  - Frequency shift keying (FSK)
  - Phase shift keying (PSK)

Modulation Techniques



Amplitude Shift Keying (ASK)

- Values represented by different amplitudes of the carrier sine wave
- Usually, one amplitude is zero
  - i.e. presence and absence of carrier

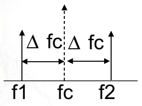
$$s(t)=\begin{cases}A\cos(2\pi f_c t) & \text{binary } 1\\ 0 & \text{binary } 0\end{cases}$$

  - e.g. switching the light sent through a fiber on and off
- Susceptible to noise and sudden changes in gain
- Up to 1200bps on voice grade lines
- Used over optical fiber

Frequency Shift Keying (FSK)

- Most common form is binary FSK (BFSK)
- The two binary data values represented by two different frequencies (near and on both sides of a central carrier frequency f_c)

$$s(t)=\begin{cases}A\cos(2\pi f_1 t) & \text{binary } 1\\ A\cos(2\pi f_2 t) & \text{binary } 0\end{cases}$$



- Less susceptible to noise than ASK

(Same as with FM Radio: Frequency can be detected correctly in the presence of noise better than amplitude)

Applications:

- Up to 1200bps on voice grade lines
- Also used at High frequency radio (3-30 MHz)
- And at even higher frequencies on LANs using coaxial cables

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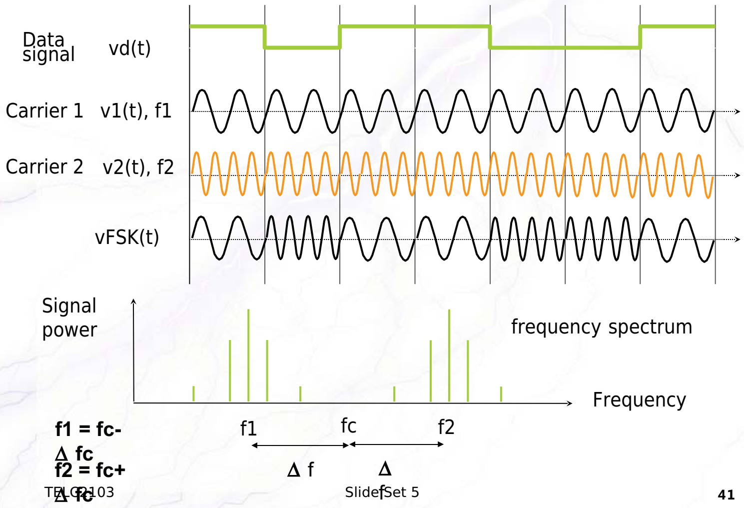
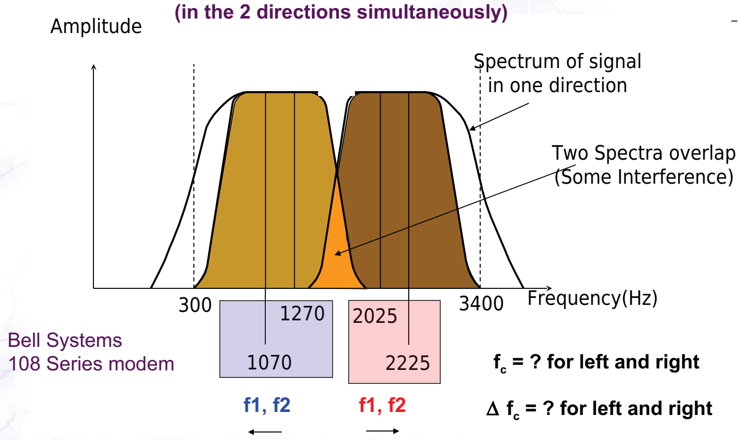
FSK



FSK for digital data on Voice Grade Lines

Full Duplex Communication
(in the 2 directions simultaneously)



Bell Systems
108 Series modem

f_c = ? for left and right

Δ f_c = ? for left and right

Multiple FSK (MFSK)

To improve BW utilization (efficiency) we send one of multiple signal symbols (frequencies) every signal element ⇒ More than 1 bit at a time

- More than two frequencies used
- An example of multi-level coding (M levels)
- Each signalling element conveys more than one bit (L bits, $L = \log_2 M$)
- This increases bandwidth efficiency

(high BE = C/B values) (Higher data rates for the same signalling rate)

- But in general, multi-level coding is more prone to error due to noise

(Unless you do something about it, e.g. orthogonally)

Multiple FSK (MFSK)

$$s_i(t) = A\cos 2\pi f_i t, \qquad 1 \le i \le M \qquad (5.4)$$

$f_i = f_c + (2i - 1 - M)f_d$

f_c = the carrier frequency

f_d = the difference frequency (Half the frequency separation Δ fc as before)

M = number of different signal elements = 2^L

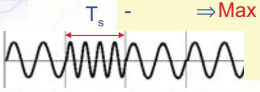
L = number of bits per signal element

i.e. different frequencies

Important Parameters

- Frequency separation = 2 f_d
- Bandwidth Required = M ($2f_d$)
-
- Minimum T_s (signal element duration) = $1/(2f_d)$
- ⇒Max signaling rate D = $1/T_s = 2f_d$

⇒Max data rate R = D L = $2f_d$ L



The closer the two frequencies are, the larger Ts needed to discriminate between them

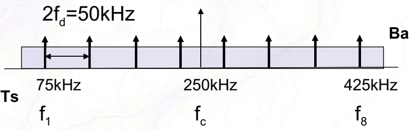
Multiple FSK (MFSK)

$$f_i = f_c + (2i - 1 - M)f_d$$

Example 5.1 With f_c = 250 kHz, f_d = 25 kHz, and M = 8 (L = 3 bits), we have the following frequency assignments for each of the eight possible 3-bit data combinations:

f_1 = 75 kHz 000 f_2 = 125 kHz 001 f_3 = 175 kHz 010 f_4 = 225 kHz 011

f_5 = 275 kHz 100 f_6 = 325 kHz 101 f_7 = 375 kHz 110 f_8 = 425 kHz 111

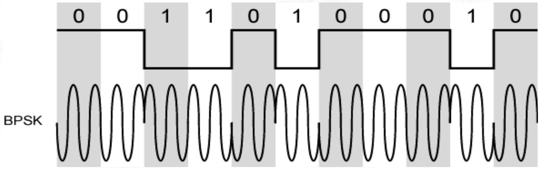


Min T_s = 1/ ($2f_d$)
= 1/50 KHz
= 20 μ s
Max signaling rate = 1/Ts
= $2f_d$
= 50 kBauds
Max Data rate = Max Signaling rate x L = 50 KHz x 3 = 150 Kbps

Bandwidth = M ($2f_d$)
= 8 x 50
= 400 kHz
(< 2 f_c, so OK)

Phase Shift Keying (PSK)

- Phase of carrier signal is shifted to represent data
- Binary PSK: Absolute
  - Two phases (spaced at 180°) represent the two binary digits



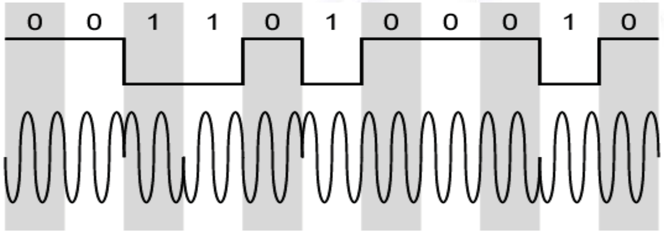
$$s(t) = \begin{cases} A\cos(2\pi f_c t) \\ A\cos(2\pi f_c t + \pi) \end{cases} = \begin{cases} A\cos(2\pi f_c t) & \text{binary 1} \\ -A\cos(2\pi f_c t) & \text{binary 0} \end{cases} \qquad (5.5)$$

$$s_d(t) = A\,d(t)\cos(2\pi f_c t) \qquad (5.6)$$

Where d(t) = +1 for '1' data and -1 for '0' data

Differential PSK (DPSK)

Phase shifted relative to the previous signal element, rather than some reference signal:



- **0**: Do not reverse phase **1**: Reverse phase (as with NRZI, invert on 1))
  (A form of differential encoding)
- Advantage:
  - No need for a reference oscillator at RX to determine **absolute** phase

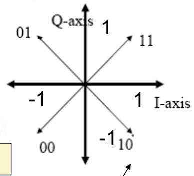
Quadrature PSK (QPSK)

- 4 different phases spaced at π /2 (90°)
- Multilevel signaling, so:
  - More efficient use of bandwidth (i.e higher data rate for the same signaling rate)
- Each signal element represents $\log_2 4 = 2$ bits

$$s(t) = \begin{cases} A\cos\left(2\pi f_c t + \dfrac{\pi}{4}\right) & \text{binary } 11 \\ A\cos\left(2\pi f_c t + \dfrac{3\pi}{4}\right) & \text{binary } 01 \\ A\cos\left(2\pi f_c t + \dfrac{5\pi}{4}\right) & \text{binary } 00 \\ A\cos\left(2\pi f_c t + \dfrac{7\pi}{4}\right) & \text{binary } 10 \end{cases}$$

-3π /4

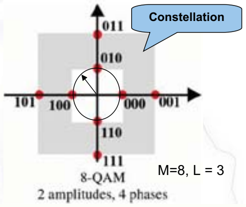
- π / 4

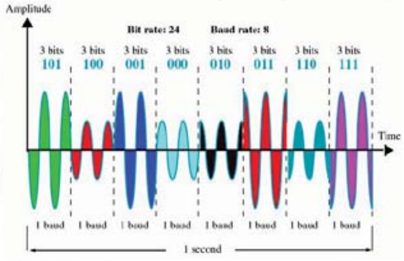


Bit pair transmitted

Quadrature Amplitude Modulation (QAM)

- An extension of the QPSK just described
- Combines both ASK and PSK
- For example, ASK with 2 levels and PSK with 4 levels give 4 x 2 i.e. 8-QAM
- M = 8, L = 3
- Up to M=256 is possible
- Large bandwidth savings
- But some susceptibility to noise
- QAM used on asymmetric digital subscriber line (ADSL) and some wireless systems

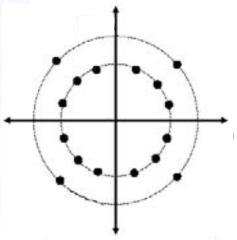




True Multilevel PSK (MPSK)

- Can use more phase angles and more than one amplitude
- For example, 9600 bps modems use 12 phase angles, four of which have 2 amplitudes
- Gives 16 different signal elements ⇒M = 16 and L = $\log_2(16)$ = 4 bits
- Every signal element carries 4 bits (Data sent 4 bits at a time)
- Baud rate D required is only 9600/4 = 2400 bauds (required BW is low … i.e. can use on a voice grade lines!)

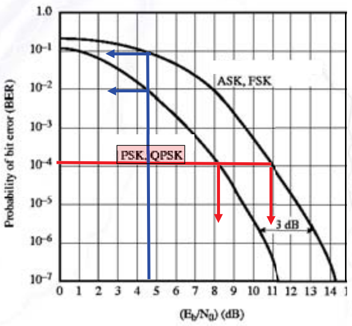
- Complex signal encoding allows high data rates to be sent on voice grade lines having a limited bandwidth



Performance of D-A Modulation Schemes

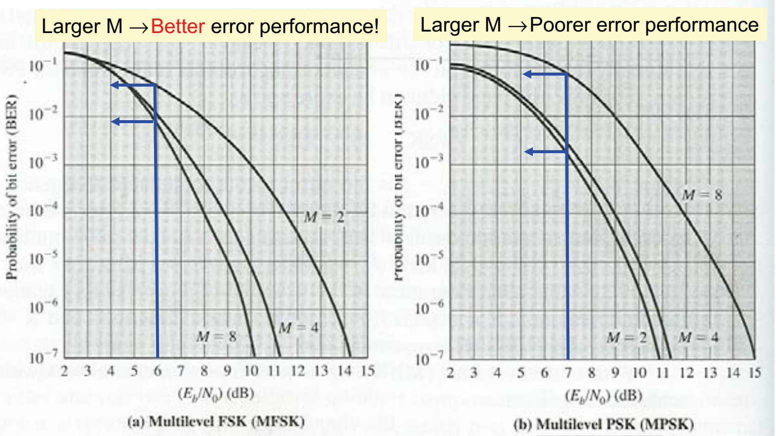
b. Performance with noise: ASK, FSK, PSK, QPSK

- Bit error rate (BER) Plotted Vs E_b/N_0 (dBs)
- Curves to the left give better performance:
  - Lower S/N required for same Error rate
  - Lower Error rate obtained for same SNR
- Why QPSK and PSK give the same performance?
  - 2 phase levels (+1,-1) in *both* cases
  - Remember QPSK gave 4 phase levels for the price of 2!



Performance of D-A Modulation Schemes

b. Performance with noise: MFSK, MPSK



Larger M →Better error performance!

Larger M →Poorer error performance

(a) Multilevel FSK (MFSK)

(b) Multilevel PSK (MPSK)

Orthogonal FSK

As expected

E_b/N_0 in terms of the bandwidth efficiency (BE) (for binary transmission)

B_T is the Transmission Bandwidth

$$BE_{dB} = \left|\frac{R}{B_T}\right|_{dB}$$

$$\frac{E_b}{N_0} = \frac{\frac{ST_b}{N}}{B_T} = \frac{\frac{S}{R}}{\frac{N}{B_T}} = \frac{\frac{S}{N}}{\frac{R}{B_T}}$$

$$\left|\frac{E_b}{N_0}\right|_{dB} = \left|\frac{S}{N}\right|_{dB} - \left|\frac{R}{B_T}\right|_{dB}$$

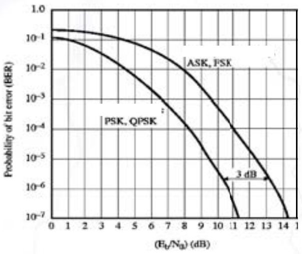
$$= \left|\frac{S}{N}\right|_{dB} - BE_{dB}$$

Example

- What is the bandwidth efficiency (BE) for ASK and PSK, for a bit error rate (BER) of 10^{-7} on a channel with a SNR of 12dB ?

- For ASK (binary): At BER = 10^{-7}, E_b/N_0 = 14.3 dBs
- Substituting in:

$$\frac{E_b}{N_0} = SNR_{dB} - [BE]_{dB}$$



- $BE_{ASK,FSK}$ = -14.3 + 12 = -2.3 dBs → $R/B_T = 10^{-.23}$ = 0.6
- However, for PSK → E_b/N_0 = 11.3 dBs):

$BE_{PSK} = R/B_T$ = 1.2 (doubled: 3dB higher- improvement)

Wireless Communications

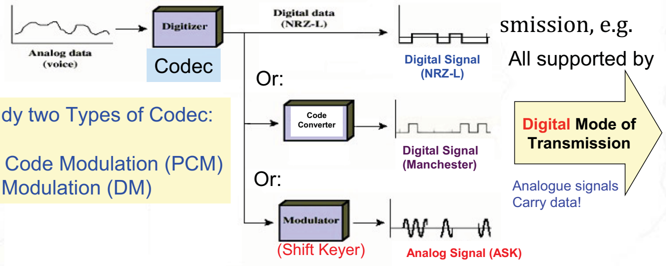
- Encoding & Modulation (Part 2)

Contents

- Encoding and Modulation (Part II)
  - Analogue data into Digital signals
  - Analogue data into Analogue signals

Analogue Data, Digital Signal

- Digitization
  - Conversion of analogue data into signals suitable for the digital mode of transmission/storage
    - The digital data can be transmitted digitally as is (e.g. NRZ-L)
    - Or converted to a more appropriate digital code, e.g. Manchester



smission, e.g.

All supported by

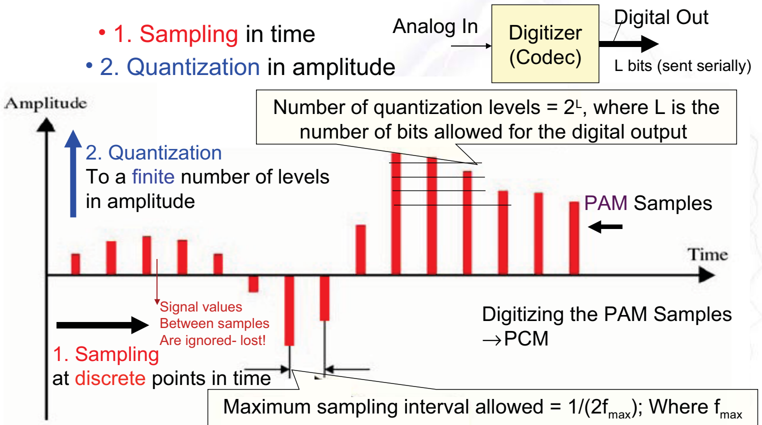
Will study two Types of Codec:

- Pulse Code Modulation (PCM)
- Delta Modulation (DM)

Two basic tasks to be performed by a digitizer:

"Analogue" is continuous in **both** time and amplitude… Must discretize it in both!

- 1. Sampling in time
- 2. Quantization in amplitude



Number of quantization levels = 2^L, where L is the number of bits allowed for the digital output

Maximum sampling interval allowed = $1/(2f_{max})$; Where f_{max} is the maximum frequency in the analogue signal

$2f_{max}$ is the minimum acceptable sampling rate

Sampling

- **Nyquist Sampling Theorem:**

If a signal is sampled at regular intervals at a rate higher than *twice* the **highest** signal frequency f_{max}, the samples contain all the information in the original signal

- Original signal may be reconstructed from these samples using an **ideal** low-pass filter
- Example: Voice data limited to 4000Hz
  - Requires sampling at a rate of at least 8000 sample per second

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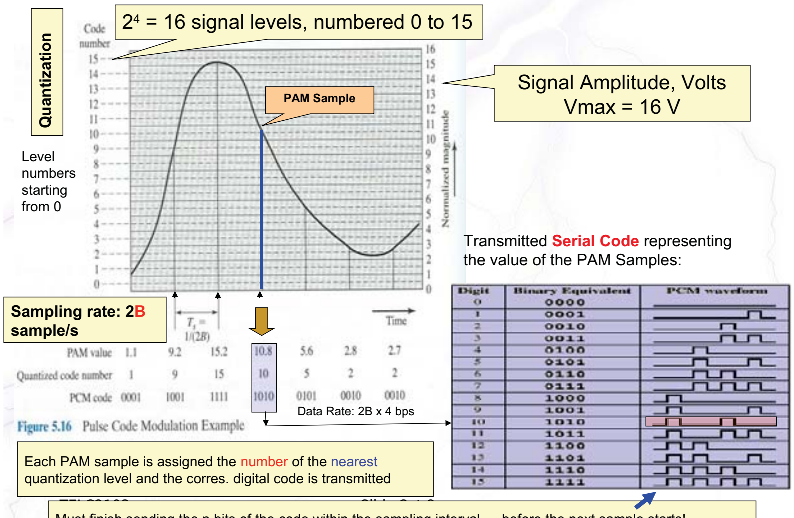
Quantization using 4 bits



2^4 = 16 signal levels, numbered 0 to 15

Signal Amplitude, Volts Vmax = 16 V

Level numbers starting from 0

Sampling rate: 2B sample/s

PAM value	1.1	9.2	15.2	10.8	5.6	2.8	2.7
Quantized code number	1	9	15	10	5	2	2
PCM code	0001	1001	1111	1010	0101	0010	0010

Figure 5.16 Pulse Code Modulation Example

Each PAM sample is assigned the number of the nearest quantization level and the corres. digital code is transmitted

Transmitted **Serial Code** representing the value of the PAM Samples:

Digit	Binary Equivalent	PCM waveform
0	0000	
1	0001	
2	0010	
3	0011	
4	0100	
5	0101	
6	0110	
7	0111	
8	1000	
9	1001	
10	1010	
11	1011	
12	1100	
13	1101	
14	1110	
15	1111	

Must finish sending the n bits of the code within the sampling interval ….before the next sample starts!

Pulse Code Modulation (PCM)

- Start with the analogue sampled pulses (Pulse Amplitude Modulation, PAM)
- Assign each sample a digital value (= number of the closest quantization level)
- n = 4 bit system gives M = 16 levels ($M = 2^n$)
- Quantization error or noise
  - Larger for small M (number of levels)
  - Approximations mean it is impossible to recover the original signal exactly
  - SNR for quantization error using n bits is

$$SNR = 20\log_{10} 2^n + 1.76\,dB = 6.02\,n + 1.76\,dB$$

  - Each additional bit used for quantization increases SNR by about 6 dB (a power factor 4)
- 256 quantization levels: n = 8 bits, SNR ≈ 50 dB

→ Quality comparable with analogue transmission

- Voice: **2** x 4000 = 8000 samples per second, with of 8 bits per sample, this is a data rate of 8000*8 = 64 kbps

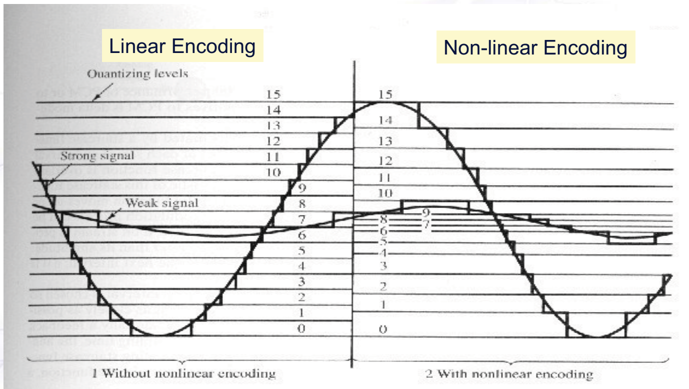
PCM Example

- Suppose we want to encode an analogue signal that has voltage levels 0 – 5 V using 2-bit PCM (n = 2 bits) (M = 2^2 = 4 levels)
- We divide the maximum voltage level into four intervals, so the size of each interval is 5/4=1.25 V
  - Level intervals: 0-1.25, 1.25-2.5, 2.5-3.75, 3.75-5
- We select the quantization levels at the middle of each level interval
  - i.e. selected levels are: 0.625, 1.875, 3.125, 4.375
  - This guarantees a maximum quantization error of ½ (5V /4) = 0.625 (=1/2 LSB)
  - and quantization SNR = 6 x 2 + 1.76 = 13.76 dB

Problem with Linear (Uniform) Encoding

- Absolute quantization error for each sample is the same regardless of signal level
  - Signals with lower amplitudes are relatively more distorted
- One Solution: make quantization levels not evenly spaced (denser for low amplitudes)
- i.e. higher number of quantization steps for lower amplitudes and smaller number for larger ones
- Reduces overall signal distortion
- This is non-linear encoding

Effect of Nonlinear Coding



Quantization error is fixed- same for both weaker and stronger signals

Weaker signals have smaller quantization errors

Example

- Consider an audio signal with spectral components in the range of 300 to 3000 Hz. Assuming a sampling rate of 7000 samples per second will be used to generate the PCM signal
  - 7000 > 2 x 3000 → OK
  - To obtain a quantization SNR of 30 dB, what is the number of uniform quantization levels needed?
    - $(\mathrm{SNR})_{dB} = 6.02\, n + 1.76 = 30$ dB

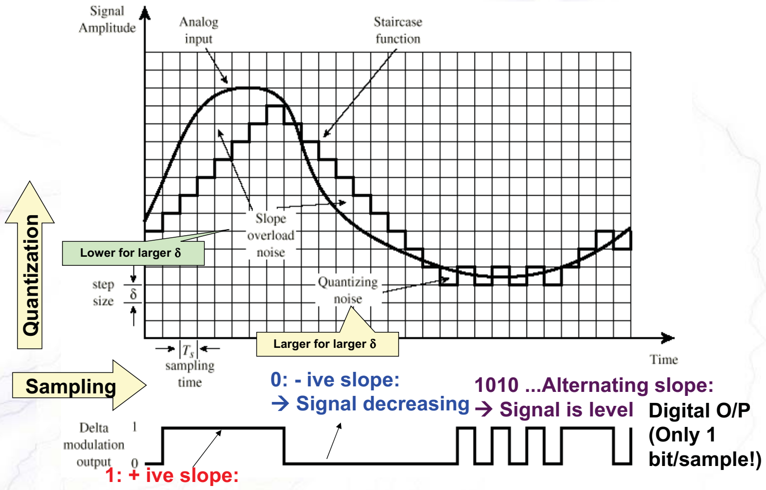
    $n = (30 - 1.76)/6.02 = 4.69$

    Always round off to the next higher integer ⇒ $n = 5$ bits ⇒ $2^5 = 32$ quantization levels
  - What is the **data rate** required?
    - R = 7000 samples/sec × 5 bits/sample = 35 Kbps

Delta Modulation: An alternative to PCM

- An attempt to reduce complexity (and large R) for PCM
- Analog input is approximated by a staircase function
  - Move up or down one fixed amplitude increment (δ) at each sample interval to track changes in the analogue waveform
- A single bit stream is produced to approximate the derivative of the analogue signal rather than its amplitude
  - Generate a 1 if staircase goes up (slope +ve)
  - Generate a 0 if staircase goes down (slope -ve)
- Transmit this sequence of 1,0 data (1-bit per sample)
- Receiver uses this bit stream to reconstruct the staircase waveform and approximate the original analogue waveform

Delta Modulation - example



0: - ive slope: → Signal decreasing

1010 ...Alternating slope: → Signal is level

Digital O/P (Only 1 bit/sample!)

1: + ive slope: → Signal increasing

Delta Modulation - Implementation

- At mid-sampling interval, compare the analogue input to current value of the approximating staircase function
  - If input *exceeds* staircase function, transmit a 1 and increment staircase by δ for the next sample
  - Otherwise generate a 0 and decrement staircase by δ for the next sample
- Output of the DM is a binary bit sequence to be used for generating the staircase function at RX
  - Reconstruct staircase function at receiving end and smooth by a low pass filter to reconstruct an approximation of the analogue signal

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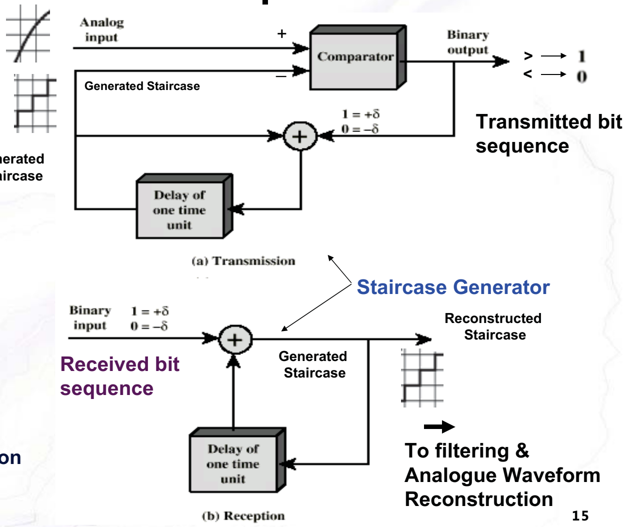
Delta Modulation - Implementation

At Source

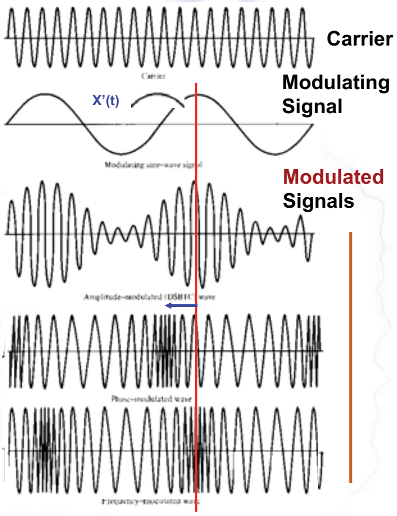


At Destination

Analogue Data, Analogue Signals

- Modulation
  - Combining an input signal *m(t)* and a carrier at frequency f_c to produce signal *s(t)* with bandwidth centered at f_c
- We had to use a form of modulation (shift keying) to represent digital data as analogue signals.
- But why modulate signals that are *already* analogue?
  - Higher frequency may be needed for effective transmission
    - For unguided transmission: impossible to send low frequency baseband signals, e.g. speech, as required antennas would have dimensions in kilometers!
  - Allows implementing frequency division multiplexing (FDM)

Types of Analogue Modulation



Signal to be Transmitted, x(t)

Amplitude Modulation (AM)

A α x(t)

Angle Modulation:

1. Phase, PM

ϕ α x(t)

A sin (ω t)
ϕ (t) = ω t

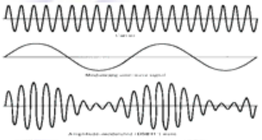
$$\omega = \frac{d\phi(t)}{dt} = \phi'(t)$$

2. Frequency, FM

f α x(t)

Effect of modulation on signal power? **Effect of modulation on signal BW?**

Amplitude Modulation (AM)



- Simplest form of modulation
- $A_c \cos 2\pi f_c t$ is the carrier,
- and x(t)= $A_m \cos 2\pi f_m t$ is the input modulating signal
- Modulated signal expressed as:

Amplitude of modulated wave

$$s(t) = \boxed{[1 + n_a \cos 2\pi f_m t]A_c} \cos 2\pi f_c t$$

Portion of the modulating signal

- **n_a is the modulation index ($0 < n_a \leq 1$):**

$$n_a = \frac{A_m}{A_c}$$

Units of n_a?

- **Added '1' is a DC component to prevent loss of information - there will always be a carrier**
- **Scheme is known as double sideband transmitted carrier (DSBTC)**

Angle Modulation

What parameters can I change to change the angle of the modulated signal?

- Includes:
  - Frequency modulation (FM) and
  - Phase modulation (PM)
- Modulated signal is given by

Total Angle

$$s(t) = A_c \cos[\boxed{2\pi f_c t + \phi(t)}] \qquad \phi(t) = n_p x(t)$$

- Phase modulation (PM): (the direct way)
  - Instantaneous phase is proportional to the modulating signal:
  - n_p is the phase modulation index
- Frequency modulation (FM): (the indirect way)
  - Instantaneous angular frequency deviations from ω_c is proportional to the modulating signal, $\phi'(t) = n_f x(t) = \delta\omega(t) = 2\pi\,\delta f(t)$
  - and we have:
  - So make the *derivative* of ϕ proportional to modulating signal
  - n_f is the frequency modulation index

Angle Modulation

- The total phase angle of s(t) at any instant is $[2\pi f_c t+\phi(t)]$
- Instantaneous phase deviation from that of the carrier is $\phi(t)$
- Phase Modulation (PM):
  - $\phi(t) = n_p x(t)$, instantaneous phase variations are directly proportional to m(t)
- Frequency Modulation (FM):
  - Instantaneous angular frequency, $\omega_i(t)$, can be defined as the rate of change of total phase
  - So, for the modulated signal, s(t)

$$\omega_i(t) = 2\pi f_i(t)$$

$$= \frac{d}{dt}[2\pi f_c t + \phi(t)] = 2\pi f_c + \phi'(t)$$

$$\therefore f_i(t) = f_c + \frac{1}{2\pi}\phi'(t)$$

Phase Modulation (PM)- Example

- Derive an expression for a phase-modulated signal s(t) and its instantaneous frequency given: A_c= 5V, and the modulating signal

  $x(t) = 3 \sin 2\pi f_m t$
- We know that s(t):

$$s(t) = A_c \cos[2\pi f_c t + \phi(t)]$$

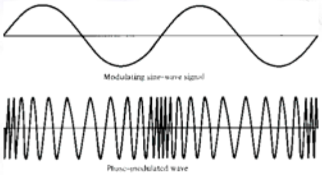
- For PM, $\phi(t)$ is given by:

$$\phi(t) = n_p x(t)$$

n_p is Radians/Volt

- Then s(t) is:

$$s(t) = 5\cos[2\pi f_c t + n_p 3\sin 2\pi f_m t]$$



- Instantaneous frequency of s(t) is:

Peak frequency deviation for the PM signal

$$f_i(t) = f_c + \frac{3n_p(2\pi f_m)}{2\pi}\boxed{\cos} 2\pi f_m t = f_c + \boxed{3n_p f_m}\cos 2\pi f_m t \quad f_i(t) = \frac{1}{2\pi}\frac{d}{dt}[\text{total phase}]$$

Note: Frequency variations in s(t) *phase-lead* x(t) amplitude variations by 90°

Frequency Modulation: FM

$$f_i = f_c + \frac{1}{2\pi}\phi'(t)$$

$$and\ \phi'(t) = n_f x(t)$$

$$and\ x(t) = A_m \sin(2\pi f_m t)$$

- From equations opposite,

  Peak frequency deviation Δ F is given by:

$$\Delta F = \frac{1}{2\pi} n_f A_m \ \text{Hz}$$

  - Where A_m is the peak value of the modulating signal x(t)
- An increase in the amplitude A_m of x(t):

  increases Δ F → increases bandwidth requirement B_T
- But average power level of the FM modulated signal is fixed at $A_C^2/2$, (does not increase with A_m)
- i.e. in Frequency Modulation (angle modulation in general), A_m affects the BW but not the power budget
- While in Amplitude Modulation, A_m affects the power budget but not the bandwidth

---

Frequency Modulation - Example

- Derive an expression for a frequency-modulated signal s(t) with A_c= 5V, given the modulating signal

  $x(t) = 3 \sin 2\pi f_m t$
- The FM modulated signal s(t) is:

$$s(t) = A_c \cos[2\pi f_c t + \boxed{\phi(t)}]$$

- For FM, $\phi'(t)$ is given by:

$$\phi'(t) = n_f x(t)$$

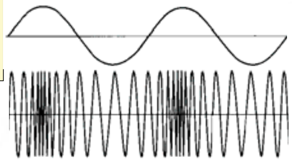
n_f is (Radians/s)/Volt

- Then $\phi(t)$ is:

$$\phi(t) = \int \phi'(t)dt = \int n_f 3\sin 2\pi f_m t \ dt = \frac{-3n_f}{2\pi f_m}\cos 2\pi f_m t$$

- We have: $\Delta F = \frac{3}{2\pi} n_f$ Hz
- Substituting for Δ F we get:

But frequency varies as ϕ ’, i.e. as sin not as – cos !!



$$s(t) = 5\cos[2\pi f_c t - \frac{\Delta F}{f_m}\cos 2\pi f_m t]$$

Bandwidth Requirement

- All AM, FM, and PM result in a modulated signal whose bandwidth is centered around f_c
- Let B be the bandwidth of the modulating signal (0-B Hz)
- AM gives only sums & differences of frequencies with f_c, and we have: B_T = 2B for DSB systems
- Angle modulation includes a term of the form cos(…+cos()) which is a nonlinear term producing a wide range of frequencies f_c+f_m, f_c+2f_m, … (the Bessel function)
- i.e. Theoretically, an infinite bandwidth is required to transmit an FM or PM signal

---

Wireless Communications

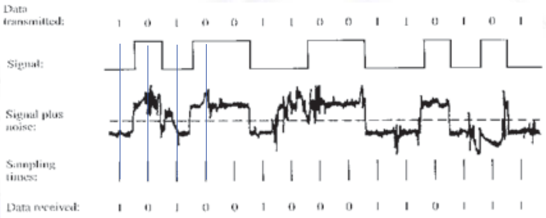
- Digital Data Communication Techniques

Contents

- Digital Data Communications Techniques
  - Asynchronous and Synchronous Transmission
  - Types of Errors
  - Error Detection
    - Parity Checks
    - Cyclic Redundancy Checks

Asynchronous and Synchronous Transmission:

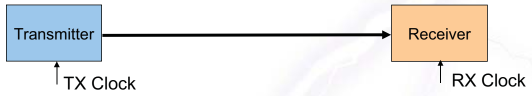
- To communicate meaningful data serially between TX and RX, signal timing should be the same at both
- Timing considerations include:
  - Rate,
  - Duration,
  - Spacing,
  - Etc.

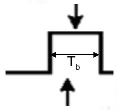


- We need to achieve some synchronism between RX & TX
- Two ways to achieve this:
  - Asynchronous Transmission
  - Synchronous Transmission

- RX needs to sample the received data at mid-points
- So it needs to establish:
  - Bit arrival time
  - Bit duration

Need for RX and TX Synchronization:





- Clock drift (example):
  - If the receiver clock drifts by 1% every bit sample time,
  - Total drift after 50 bit intervals = 50 X 0.01 = 0.5 T_b
  - i.e. instead of sampling at the middle of the bit, the receiver will sample bit # 50 at the edge of the bit – Bit 51 will be <u>wrongly</u> sampled
  - RX and TX clocks become out-of-synch ⇒Communication Error!
- In general, No. of correctly sampled bits = 0.5 $T_b/(n/100)T_b$ = 50/n, where n is the % timing error between TX and RX clocks
- Two approaches for correct reception:
  - Send only a few bits (e.g. a character) at a time (that RX can sample correctly before losing sync) →Asynchronous Transmission
  - Keep receiver clock properly synchronized with the transmitter clock all the time →send as many bits as you want…Synchronous Transmission

---

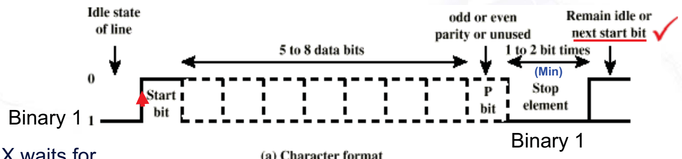
Asynchronous Transmission: Character-Level Synchronization

- Avoids the timing problem by NOT sending long, uninterrupted streams of bits
- So data is transmitted only one short character at a time: (so drift will not be a serious problem). Characters consists of:
  - A distinct start bit,
  - Say 5 to 8 data/parity bits
  - A distinct stop bit
- Character is *delimited* (at start & end) by known signal elements: start bit – stop element
- Sync needs to be maintained only for the short duration of the character (easier to achieve, allows some clock drift)
- The receiver has a new opportunity to resynchronize at the beginning of next character (Start bit)

⇒ i.e. Timing errors do not accumulate from character to character

---

Asynchronous Transmission

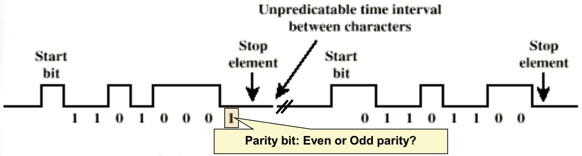


(a) Character format

RX waits for A character following the end of the previous character

RX “knows” how many bits To expect in a character, and keeps counting them following the ‘start’ bit

- The ‘stop’ bits confirm end of character →otherwise: Framing Error
- Stop bits continue (idling) till next character



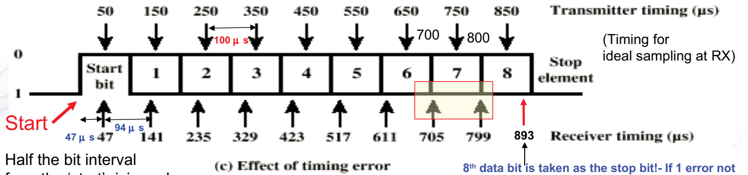
(b) 8-bit asynchronous character stream

Asynchronous Transmission

Errors due to lack of sync for an 8-bit system

- Let data rate = baud rate = 10 kbps
- Bit interval = signal element width = 1/10k = 100 μ s
- Clock Drift: Let RX’s clock is *faster* than TX by 6% (10.6 KHz)
  (So RX *thinks* that the bit interval is 1/10.6 KHz = 94 μ s)

- RX checks mid-bit data: after 47 μ s and then at 94 μ s intervals
- Data bit 8 is wrongly sampled within bit 7 (bit 7 is read twice!)
- Actual data bit 8 is missed and is seen by RX as a stop bit!



(c) Effect of timing error

Asynchronous Transmission: Efficiency

What are we paying to compensate for lack of proper TX to RX synchronization?

- Each Char uses 1 start bit & 2 stop bits: (3 non-data bits)

with 8-bit data and no parity:

⇒Efficiency = Useful Data / Total Data = 8/(8+3) = 72%

⇒Overhead = Non Data bits / Total Data = 3/(8+3) = 28%

Asynchronous Transmission – Pros & Cons

Pros:

- Simple
- Cheap
- Good for data with large gaps in between (e.g. terminal to a computer like keyboard or mouse)

Cons:

- Overhead of 2 or 3 bits per short character (~20%)
- Limit on character size
- Timing errors accumulate within large characters

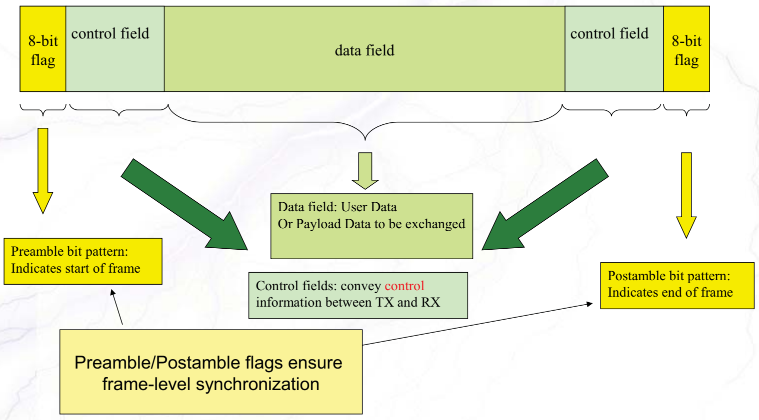
Synchronous Transmission: Bit- Synchronization

- Allows transmission of large blocks of data (frames)
- Need both bit-level and frame-level synchronization
- Bit-level synchronization (to prevent timing drift):
  - Use a separate clock line between TX and RX
    - OK over short distances
    - Subject to transmission impairments over long distances
  - Or: Embed clock signal in data using:
    - Self-clocking codes, e.g. Manchester or Differential Manchester encoding
    - Or carrier frequency for analog signals (shift keying)
- Frame-level synchronization:

Preamble & Postamble flags

Synchronous Frame Format

Typical Frame Structure



8-bit flag	control field	data field	control field	8-bit flag

Preamble bit pattern: Indicates start of frame

Data field: User Data Or Payload Data to be exchanged

Control fields: convey control information between TX and RX

Postamble bit pattern: Indicates end of frame

Preamble/Postamble flags ensure frame-level synchronization

Synchronous Transmission: Efficiency

- Example: HDLC data link protocol uses a total of 48 bits for control, preamble, and postamble fields per frame:

With a data block consisting of 1000 characters (8-bits each),

⇒ Efficiency = 8000/(8000+48) = 99.4%

⇒ Overhead = 48/8048 = 0.6% (compare 20% for async)

- Note higher efficiency and lower overhead compared to asynchronous transmission

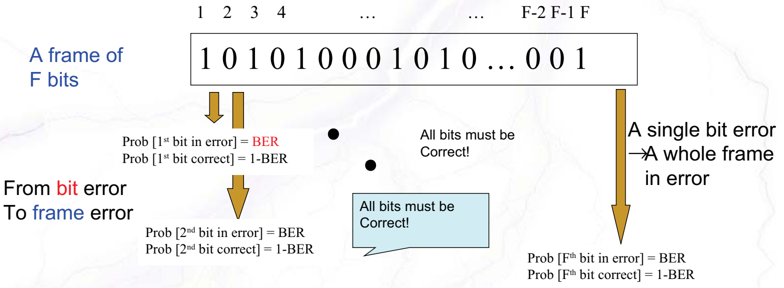
Errors in Digital Transmission

- Error occurs when a bit is altered between transmission and reception (0 ⇒1 or 1 ⇒0)
- Two types of errors:
  - Single bit errors
    - One bit altered
    - Isolated incidence, adjacent bits not affected
    - Typically caused by white noise
  - Burst errors
    - Contiguous sequence of bits in which first, last, and any number of intermediate bits are in error
    - Caused by impulse noise or fading (in wireless communication)
    - More common, and more difficult to handle
    - Effect is greater at higher data rates
- What to do about these errors?:
  - Do nothing? (Is this acceptable?)
  - Detect them (at least, so we can ask TX to retransmit!)
    - and Correct them (if we can)

▪ Will show that - Without error detection/correction - rate of erroneous frames received would be unacceptably large!

Frame Error Rate

- We know about the bit error rate (BER)
- But we send data as large frames → We are more interested in frame error rate (FER)
- How does BER affect FER?



1 2 3 4 … … F-2 F-1 F

A frame of F bits

1 0 1 0 1 0 0 0 1 0 1 0 … 0 0 1

Prob [1^{st} bit in error] = BER
Prob [1^{st} bit correct] = 1-BER

All bits must be Correct!

A single bit error →A whole frame in error

From bit error To frame error

Prob [2^{nd} bit in error] = BER
Prob [2^{nd} bit correct] = 1-BER

All bits must be Correct!

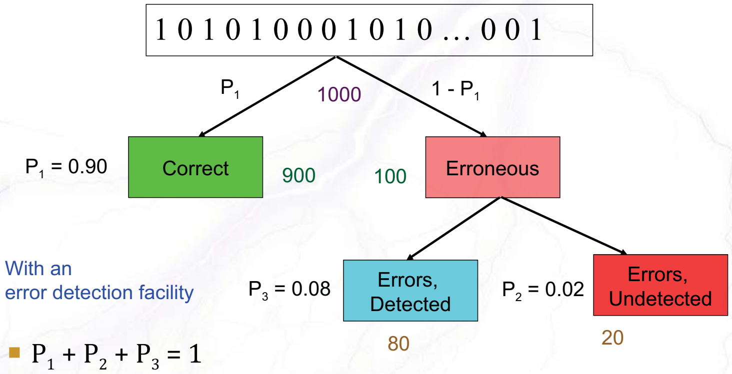
Prob [F^{th} bit in error] = BER
Prob [F^{th} bit correct] = 1-BER

- Hence, for a frame of F bits,
  Prob [frame is correct] = $(1\text{-BER})^F$: Decreases with increasing BER & F (bad)
- Prob [frame is erroneous] = 1 - $(1\text{-BER})^F$ = Frame Error Rate (FER) Increases with increasing BER & F (bad)

Motivation for Error Detection & Correction: Example

- ISDN specifies a BER = 10^{-6} for a 64kbps channel
- Let frame size F = 1000 bits
- What is the FER?
  - FER = $1 - (1 - \text{BER})^F = 1 - (0.999999)^{1000} = 10^{-3}$
- Assume a continuously used channel…
  - How many erroneous frames in one day ?
- Number of frames sent/day = (64,000/1000 frames/s) × (24 × 3600 s/day)
  = 5.5296×10^6 Frames/day
- Number of erroneous frames/day:
  - = $5.5296 \times 10^6 \times 10^{-3} = 5.5296 \times 10^3$
- Typical requirement: Maximum of 1 erroneous frame /day!
- i.e. frame error rate is too high to be tolerated!
  ⇒We definitely need error detection & correction!

Frame Error Probabilities:

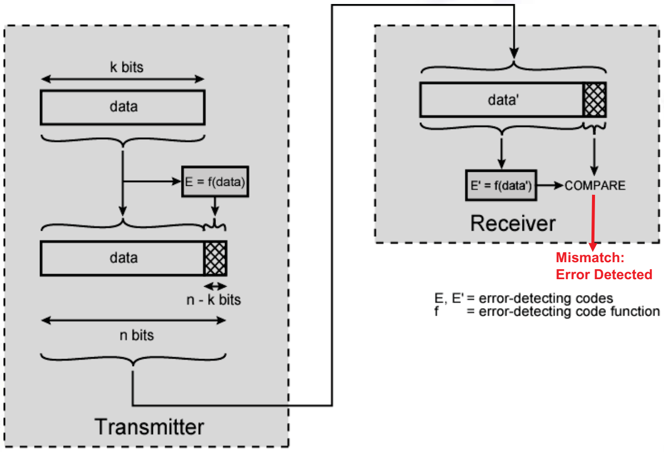


- $P_1 + P_2 + P_3 = 1$
- Without error detection facility: $P_3 = 0$, and:
  $P_2 = 1 - P_1$ (all errors are undetected)

Error Detection Techniques

- Two main error detection techniques:
  - Parity Check
  - Cyclic Redundancy Check (CRC)
- Both techniques use *additional bits* that are appended to the "payload data" by the transmitter for the purpose of error detection at the receiver

Error Detection: Implementation



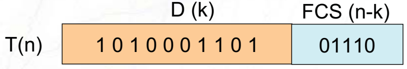
E, E' = error-detecting codes
f = error-detecting code function

Parity Check

- Simplest error-detection scheme
- Appending one extra bit:
  - Even Parity: Will append "1" such that the total number of 1's is even
  - Odd Parity: Will append "1" such that the total number of 1's is odd
- Example: If an even-parity is used, RX will check if the total number of 1's is even
  - If it is not ⇒error occurred
- Problem: even number of bit errors go undetected!

Cyclic Redundancy Check (CRC)

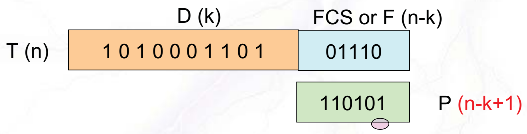
- Burst errors will most likely go undetected by a simple parity check scheme
- Instead, we use a more elaborate technique: Cyclic Redundancy Check (CRC)
- CRC appends **redundant** bits to the frame trailer called Frame Check Sequence (FCS)
  - The FCS bits are used at RX for error detection
- In a given frame containing a total of n bits, we define:
  - k = the number of original data bits
  - (n – k) = the number of added bits in the FCS field
  - So, that the total frame length is k + (n – k) = n bits



CRC Generation

- CRC generation at TX is all about finding the FCS, given the data (D) and a divisor (P)

that makes T exactly divisible by P (i.e. with 0 remainder)



- There are three equivalent ways to see how the CRC code is generated:
  - Modulo-2 Arithmetic Method
  - Polynomial Method (not covered)
  - Digital Logic Method (not covered)

What is F that makes T divide P exactly? i.e. with no remainder

---

Modulo-2 Arithmetic

- Binary arithmetic *without* carry
- Equivalent to XOR operation
- i.e.:

0 ± 0 = 0; 1 ± 0 = 1; 0 ± 1 = 1; 1 ± 1 = 0

- Examples:

$$\begin{array}{r}1111\\+1010\\\hline 0101\end{array}\qquad\begin{array}{r}1111\\-1010\\\hline 0101\end{array}\qquad\begin{array}{r}11001\\\times\ 11\\\hline 11001\\11001\\\hline 101011\end{array}\qquad\begin{array}{r}1010\\+1010\\\hline 0000\end{array}$$

Subtraction is the same as addition!

A+A = A-A = 0!

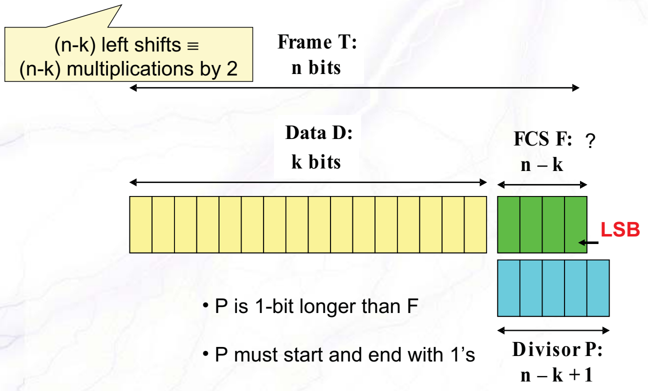
CRC Error Detection Process

- Given k-bit data (D), the TX generates an (n – k)-bit FCS field (F) such that the *total* n-bit frame (T) is exactly divisible by a predefined (n-k+1) bit divisor (P) (i.e. gives a zero remainder)
- In general, the received frame may or may not be identical to the sent frame
  - Let the received frame be (T')
  - Only in error-free transmissions that we have T' = T
- RX divides (T') by the same known divisor (P) and checks if there is any remainder
  - If division yields a remainder then the frame is erroneous
  - If the division yields zero remainder then the frame is error-free unless many erroneous bits in T' resulted in a new exact division by P (we now know what **cyclic** means!)
    - This is unlikely but possible, causing an **undetected** error!

CRC Generation

$T = 2^{(n-k)} \times D + F$

(n-k) left shifts ≡ (n-k) multiplications by 2



- P is 1-bit longer than F
- P must start and end with 1's

CRC Generation

- $T = 2^{(n-k)} \times D + F$, What is F that makes T divide P exactly ?
- Claim: F is the **remainder** obtained from dividing $\{2^{(n-k)} \times D\}$ by divisor P

$$\frac{2^{(n-k)}D}{P} = Q + \frac{F}{P} \qquad (1)$$

where Q is the quotient and F is the remainder

- If this is the correct F, T should now divide P with Zero remainder

$$\frac{T}{P} = \frac{2^{(n-k)}D}{P} + \frac{F}{P}, \text{Substituting for the first term from (1)}$$

$$= Q + \frac{F}{P} + \frac{F}{P} = Q + \frac{F+F}{P}$$

$$= Q + \frac{0}{P}; \qquad (F + F = 0 \text{ in modulo 2 Arithmetic, XOR Operation})$$

It does!... T divides P exactly!

- Note: For F to be a remainder when dividing by P (in step 1), P should be 1-bit longer, that is why P is (n-k)+1 bits....

CRC Generation: 1. Modulo-2 Arithmetic Method

- At TX: CRC Generation (using the rules):
  1. Multiply: $2^{(n-k)} \times D$ (left shift by (n-k) bits)
  2. Divide: $\{2^{(n-k)} \times D\} / P$
  3. Use the resulting (n – k)-bit remainder as the FCS
- At RX: CRC Checking: RX divides the received T (i.e. T’) by the known divisor (P) and checks if there is any remainder:
  - Non-zero remainder ⇒ Error (for sure)
  - Zero Remainder ⇒Assume no error. You could be wrong- (undetected error) but with a small probability… see slide 41)

Example – Modulo-2 Arithmetic Method

- Given
  - D = 1 0 1 0 0 0 1 1 0 1 At the Transmitter (source) side
  - P = 1 1 0 1 0 1
- Find the FCS field
- Solution:
  - First we note that:
    - The size of the data block D is k = 10 bits
    - The size of P is (n – k + 1) = 6 bits

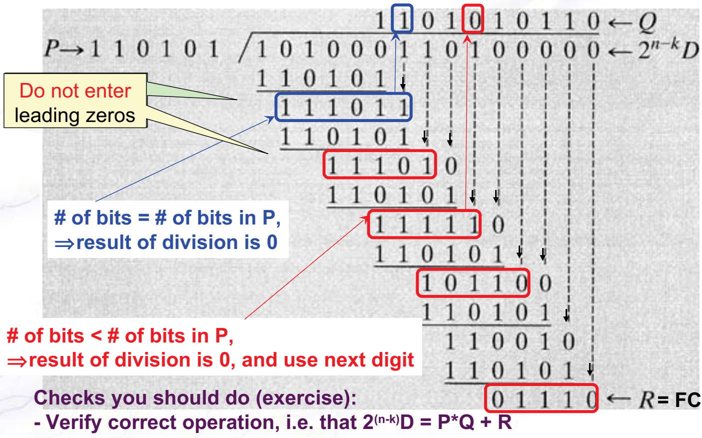
      ⇒Hence the FCS length is n – k = 5

      ⇒Total size of the frame T is n = 15 bits

Example – Modulo-2 Arith. Method

- Solution (continued):
  - Multiply $2^{(n-k)} \times D$
    - $2^{(5)} \times$ 1 0 1 0 0 0 1 1 0 1 = 1 0 1 0 0 0 1 1 0 1 0 0 0 0 0
    - This is a simple shift to the left by five positions
  - Divide $2^{(n-k)} \times D / P$ (see next slide for details)
    - 1 0 1 0 0 0 1 1 0 1 0 0 0 0 0 ÷ 1 1 0 1 0 1 yields:
      - **Quotient Q = 1 1 0 1 0 1 0 1 1 0**
      - **Remainder R = 0 1 1 1 0**
  - So, FCS = R = 0 1 1 1 0: Append it to D to get the full frame T to be transmitted
  - $T = \underbrace{1\,0\,1\,0\,0\,0\,1\,1\,0\,1}_{D}\,\underbrace{0\,1\,1\,1\,0}_{FCS}$

Example – Modulo-2 Arith. Method



Do not enter leading zeros

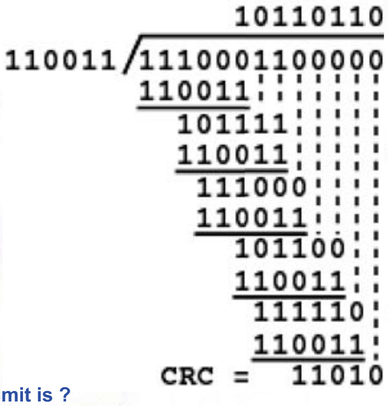
# of bits = # of bits in P,
⇒result of division is 0

# of bits < # of bits in P,
⇒result of division is 0, and use next digit

Checks you should do (exercise):
- Verify correct operation, i.e. that $2^{(n-k)}D = P*Q + R$
- Verify that the obtained T (=101000110101110) divides P (110101) exactly (i.e. with zero remainder)

Example

- For P = 110011 & D = 11100011, find the CRC



T to transmit is ?

Chances of missing an error by CRC error detection

- Let E be an n-bit number with a bit = 1 at the position of each error bit error occurring in T
- Error occurring in T causes bit reversal
- Bit reversal is obtained by XORing the bit with 1
- So, received $T_r = T \oplus E$
- Error is missed (not detected) if T_r is divisible by P
- Since T is made divisible by P, this requires E also to be divisible by P ! (can be proven)
- That is a ‘bit’ unlikely! But it can happen- causing a missed error that we could not detect…

Choice of P(X)

- How should we choose the polynomial P(X) (or equivalently the divisor P)?
- The answer depends on the types of errors that are likely to occur in our communication link
- As seen before, an error pattern E(X) will be undetectable only if it is divisible by P(X)
- It can be shown that the following error types are detectable :
  - All single-bit errors, if P(X) has two terms or more
  - All double-bit errors, if P(X) has three terms or more
  - Any odd number of errors, if P(X) contains the factor (X+1)
  - Any burst error whose length is less than the FCS length (n – k)
  - A fraction ($=1-2^{-(n-k-1)}$) of error bursts of length (n-k+1)
  - A fraction ($=1-2^{-(n-k)}$) of error bursts of length > (n-k+1)

Choice of P(X): Probability of undetected errors

- If all error-patterns are equally likely, and n - k = length of the FCS, then:
  - For a burst error of length (n-k+1), the probability of undetected error is $1/2^{(n-k-1)}$
  - For a longer burst error i.e. length > (n-k+1), the probability of undetected error is $1/2^{(n-k)}$

To improve error delectability use long divisors → (n-k+1) is large
…. but this increases FCS overhead, (n-k) large, and processing time…

FCS is 1-bit shorter than P:

P(X) in practical systems

FCS is 1-bit shorter than P:

- There are four widely-used versions of P(X)
  - CRC-12: $P(X) = X^{12} + X^{11} + X^3 + X^2 + X + 1$ (13 bits) (r = 13 -1 = 12)
  - CRC-16: $P(X) = X^{16} + X^{15} + X^2 + 1$ (r = 17 -1 = 16)
  - CRC-CCITT: $P(X) = X^{16} + X^{12} + X^5 + 1$
  - CRC-32: $P(X) = X^{32} + X^{26} + X^{23} + X^{22} + X^{16} + X^{12} + X^{11} + X^{10} + X^8 + X^7 + X^5 + X^4 + X^2 + X + 1$ (r = 33 -1 = 32)

FCS Size, not P size

- CRC-32 is used for the IEEE 802.3 (Ethernet) LAN standard

Note: P(X) always starts (& ends) with 1

Error Correction

- Once an error is detected, what action can the RX take?:
- Two alternatives:
  - RX asks for a retransmission of the erroneous frame
    - Adopted by data-link protocols such as HDLC (later) and transport protocols such as TCP
    - i.e. A **Backward Error Correction** (BEC) strategy
  - RX attempts to correct the errors if enough redundancy exists in the received data
    - TX uses **Block Coding** to allow RX to correct potential errors
    - i.e. A **Forward Error Correction** (FEC) strategy
    - Used in applications that do not tolerate the extra time required for retransmission, e.g. VoIP.

UNIVERSITY of TECHNOLOGY, MAURITIUS

BEng (Hons) Telecommunications

BTEL/10A/FT

Examinations for 2011 / Semester 2

MODULE: WIRELESS COMMUNICATIONS

MODULE CODE: TELC2103

DURATION: 3¼ HOURS

Instructions to Candidates

1. Answer **ALL FIVE QUESTIONS**
2. Always start a new question on a fresh page.
3. Questions do not carry equal marks.
4. Total marks: 100
5. Appendix is attached.
6. Use of silent calculators is allowed in the Examination Room.

This exam paper contains 5 questions and consists of 7 pages.

Question 1 (25 marks)

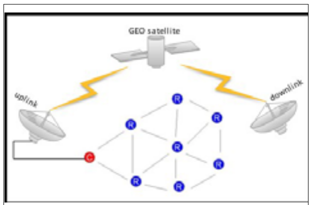


Figure 1: Wireless Mesh Sensor topology

The ground temperature of a volcanic region can be monitored by *SB-PT100 "Hockey Puck"* wireless transmitters arranged in a mesh topology as shown in *Figure 1*. Each transmitter is fitted with a Platinum Thermometer sensor (*PT100*) whose resistance changes with temperature. The transmitters act as routers **R** and communicate with each other and with the coordinator **C** using the *IEEE 802.15.4* standard *via ZigBee®*. The *SB-PT100* has an in-built ADC in the form of a 24-bit PCM encoder. The transmitters operate at 2.4 GHz using Offset QPSK among themselves and the coordinator. The coordinator periodically send the collected digital data as a stream which is then transmitted by a satellite uplink using a powerful, 14 GHz carrier frequency microwave signal to a geostationary satellite. Refer to the specifications of the *SB-PT100* in the *Appendix* in answering some of the following:

(a) Calculate the maximum operating bandwidth of the *PT100* sensors.

(2 marks)

(b) Calculate the quantization error in dB of the PCM encoder.

(3 marks)

(c) The *ZigBee®* transmitters uses Direct Sequence Spread Spectrum (DSSS). Give a DSSS transmitter block diagram and state two benefits of this technique.

(4+2 marks)

(d) *ZigBee®* radios operate at three frequency bands. State the maximum data rate achievable per channel in the 2.4 GHz band.

(2 marks)

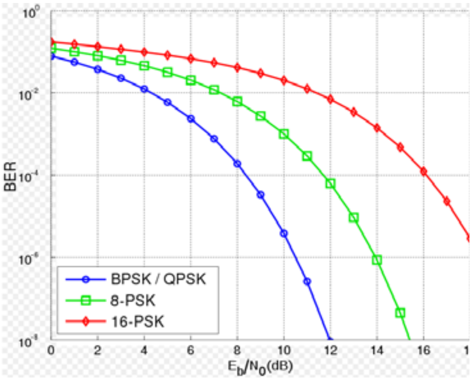
(P.T.0)

(e) Using the data rate above, calculate the minimum bandwidth needed for transmitter communication with a SNR of 5 dB assuming error free transmission?

(3 marks)

(f) Now suppose that the channel becomes error-prone as well, what is the minimum signal power in dBW that is needed to sustain a data rate of 100 kbps with a BER of 10^{-3} at 300 degrees Celsius?

(5 marks)



. **Figure 2: BER v/s Eb/N0**

(g) Calculate the free space loss in dB of the signal centred at the carrier frequency on the satellite uplink.

(4 marks)

Question 2 (15 marks)

(a) *Bluetooth™* technology uses Multiple Frequency Shift Keying (M-FSK) and Frequency Hopping Spread Spectrum techniques. Assuming that M = 2 and that the difference frequency (f_D) is 500 kHz at a carrier frequency of 2402 MHz.

i. Assuming no guard bands, calculate the maximum bit rate achievable.
ii. Using the bit rate above, and assuming a **1101010** user bit stream, sketch possible frequency v/s time graphs for slow-hopping and fast-hopping modes and calculate the dwell time assuming that each hop occurs instantaneously.
iii. Give two disadvantages of Frequency Hopping Spread Spectrum.

(2+4+2 marks)

(b) Consider a communication system that uses 64-QAM modulation. It is desired that the system communicates 36 megabits of information each second.

i. What should the symbol rate be?
ii. With the symbol rate above, how many bits per second can we transmit with 256-QAM?
iii. It seems that in a square QAM constellation, you can keep adding points to the constellation by increasing the size of the square that contains all constellation points, e.g. 64-QAM, 256-QAM, 1024-QAM. Mention two issues that prevents you from getting higher and higher bit rates?

(1+2+2 marks)

(c) *WiMAX* (*IEEE 802.16-2004*) has a spectral efficiency of 3.7 bps/Hertz, what is the minimum E_b/N_0 required to achieve this?

(2 marks)

Question 3 (20 marks)

(a) Assume that the following digital bit stream **10100110** is to be encoded in:

  i. Pseudoternary
  ii. Manchester encoding

Sketch the waveforms for each of the code above showing clearly the bit durations, signal levels and transitions wherever necessary.

(Assumption: The signal level for the last space was negative for pseudoternary).

(3+4 marks)

(b) A wireless *IEEE 802.11g* frame transmitter is using the generator polynomial x^4+x+1 and that the header contains the above bit pattern: **10100110**. Derive the transmitted CRC header checksum.

(4 marks)

(c) Given the following AM signal expression: $s(t)=[1+0.7\sin 400t]7\cos 250t$

  i. evaluate the carrier frequency f_c.
  ii. evaluate the amplitude of the modulating signal A_m.

(3+2 marks)

(d) Derive an expression for a FM signal $s(t)$ given then carrier amplitude A_C of 3V and given the modulating signal is $5\sin(2\pi f_m)t$.

(4 marks)

Question 4 (15 marks)

(a) Code Division Multiplexing (CDM) is popular used in cellular networks.

  i. What is the main reason behind multiplexing, in general?
  ii. Explain what is meant by the term spreading factor?
  iii. Give **two** benefits and **two** disadvantages of this multiplexing technique.

(1+2+4 marks)

(P.T.0)

(b) Cellular communication systems are becoming commonplace nowadays.

  i. Describe the three main components of a basic cellular system.
  ii. Give three limiting factors to the size and weight of a mobile handset.

(3+3 marks)

(c) Contrast two factors which renders Personal Communication Services (PCS) different from traditional cellular telephony.

(2 marks)

Question 5 (25 marks)

(a) VSAT technology is becoming more widespread by providing low-cost connectivity in very remote areas.

  i. Explain which type of satellite and frequency band are used in VSAT
  ii. Illustrate three common configurations used for satellite communication.

(2+3 marks)

(b) Consider a hypothetical network layout at UTM as shown in *Figure 3*.

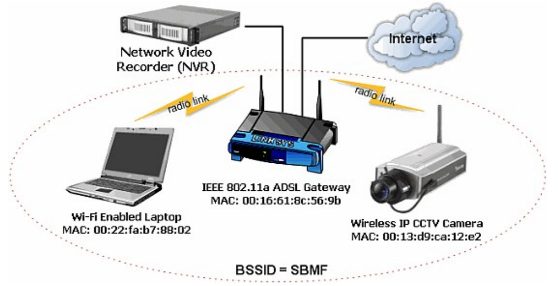


Figure 3

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  i. What do you understand by term BSSID?

(2 marks)

  ii. Give the generic structure of an *IEEE 802.11x* data frame.

(5 marks)

  iii. The wireless gateway uses *IEEE 802.11a* technology. Describe this technology in terms of modulation type, maximum data rate, range and operating frequency.

(4 marks)

  iv. How many non-overlapping channels are available for the *IEEE 802.11a* technology?
  Describe a situation where non-overlapping channels are useful?

(1+3 marks)

  v. Give the values of the relevant fields in the wireless frame if the wireless IP CCTV Camera sends a wireless frame destined to Laptop *via* the wireless gateway.

(5 marks)

******END OF EXAMINATION PAPER******

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wireless sensors

SB-PT100 "Hockey Puck"

Wireless Sensor with PT100

The SB-PT100 family connects PT100 temperature sensors without the expense and hassle of cabling. Its integrated low power IEEE 802.15.4 transceiver provides long battery life and seamless wireless connectivity in hard to reach areas, when portability is essential, or in locations where running cables is inconvenient or not cost-effective. Enabled by our industrial-grade mesh networking protocol, the SB-PT100 provides highly reliable performance even in harsh environments to ensure delivery of critical temperature measurements.

The integrated PT100 interface incorporates a 24-bit ADC for accurate conversion of the PT100 ohm resistance value into temperature data, in accordance with the DIN EN60751 standard. When activated, the SB-PT100 adapter will automatically sample, log and transmit these readings in fully user-configurable intervals. Bi-directional communication is also supported, allowing the adapter to be wirelessly controlled and re-configured.

SB-PT100 electronics are sized to fit in standard sensor heads designed for wired "hockey puck" style transmitters providing a familiar form factor and can be supplied as transmitter only, transmitter mounted in sensor head or complete with PT100 sensor for single source simplicity

Features and Benefits

- Integrated PT100 measurement transducer for direct connection to PT100 probes and sensors
- Nodes run on batteries, with a battery life of multiple years
- Self-forming, self-healing mesh network topology for maximum resilience and ease of deployment
- Up to 90% installation and commissioning cost savings over traditional cable-based solutions
- Time synchronization of all nodes assures accurate time stamping of individual measurements
- Embedded software provides autonomous data logging and reporting, triggers/alarms, battery condition monitoring, over-the-air configuration, firmware upgrades, and many other advanced capabilities

wireless sensors
12 Old Powerhouse Rd.
Falmouth, ME 04105
T 888.928.4362
info@WirelessSensors.com

Specifications

Wireless		General	
Radio type	IEEE 802.15.4 compliant	Sample rate (max.)	1 kHz
Frequency band	2.4 GHz	Scan cycle (typical)	10 s - 1 day
Standby current	20 µA	Scan cycle (min.)	100 ms
Active measurement current	2 mA	Sensor interface	1 x PT100, 2-wire
Transmit current	55 mA	Sensor temperature range	-100ºC - +400ºC
Receive current	50 mA	Resolution	0.1ºC / 24-bit
Node-to-node hops (max.)	3[1]	Battery lifetime[2] @ 3000 mAh	▪ 15 min heartbeat - up to 7 years (Leaf), 3 years (Router) ▪ 5 min heartbeat - up to 5 years (Leaf), 18 months (Router)
Line of sight range (max.)	250 m (820') node-to-node[3]	Accuracy	±0.25% of maximum range
In-building range (typical)	70 m (230') node-to-node	Power source	2 x AA batteries (3,000 mAh) or external 12 - 24 VDC
Receiver sensitivity	-92 dBm	Data log buffer	98 readings
Output power (max.)	2 dBm	Terminal cross-section	0.5 mm² - 1.5 mm²
Output power (typical)	0 dBm	EMC noise immunity	According to DIN-EN300328, DIN-EN50371
		EMC compatibility	According to DIN-EN60950
		Certifications	R/TTE, DIN-EN301489-1, DIN-EN301489-1, CE authorized for

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[1] Extendable to up to 5 hops
[2] Longer battery life on request
[3] Extendable to up to 2 km (6,500')

Complementary Products and Accessories

A Gateway device is required to establish and manage the wireless mesh network and to provide connectivity to backend systems and is specified separately

Accessory	Order Code	Accessory	Order Code
PT100 sensor, 2-wire	ACC-DS-PT100	Battery pack, 3,000 mAh	ACC-BP-S

wireless sensors
12 Old Powerhouse Rd.
Falmouth, ME 04105
T 888.928.4362
info@WirelessSensors.com

UNIVERSITY of TECHNOLOGY, MAURITIUS

BEng (Hons) Telecommunications

BTEL/09/FT

Examinations for 2010 - 2011 / Semester 2

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MODULE CODE: TELC2103

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Question 1 (25 marks)

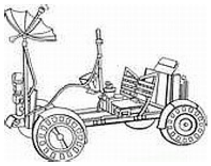


Fig. 1

Figure 1 shows a prototype vehicle to navigate on the Moon. The vehicle is fitted with an array of sensors which gathers lunar information and relays it down to Earth. The vehicle contains four Infra-red analogue distance sensors on each side of the vehicle to measure the distance from an obstacle. The sensors derive their power from photovoltaic cells mounted on the vehicle itself and the sensors have an operating bandwidth of 50 Hz. The sensors can measure distances between 180 cm to 20 cm by varying the output voltage signal from +3 to +5 Volts respectively. The four signals are then time-division multiplexed with each sensor being given a time slot of 0.25 second. The output of the multiplexer is then fed to a PCM encoder. The resultant digital output data stream is then transmitted as a powerful, 10 GHz carrier frequency, 64-QAM microwave signal back to Earth.

(a) Calculate the minimum sampling frequency needed to fully characterize the analogue sensor signals?

(2 marks)

(b) Calculate the number of quantization levels needed in the PCM if we wish the sensors to have a sensitivity of 5 cm.

(2 marks)

(c) Calculate the bit rate and the quantization error in dB of this PCM?

(2+3 marks)

(d) Assuming a bandwidth of 75 kHz, calculate the maximum Nyquist output data rate achievable using 64-QAM downlink signal?

(3 marks)

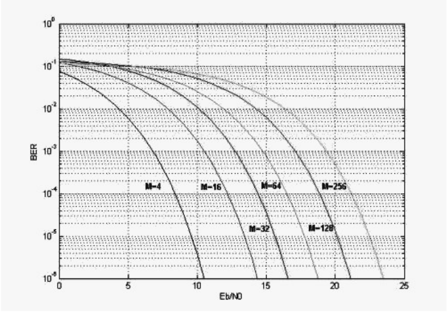
(P.T.0)

(e) Calculate the maximum allowable SNR_{dB} of the downlink if the data rate has to be a minimum of 150 kbps?

(4 marks)

(f) Now suppose that the downlink becomes error-prone as well, what is the minimum signal power in dBW that is needed to sustain a data rate of 300 kbps with a BER of 10^{-4} at 25 degrees Celsius?

(5 marks)



(g) Assuming the Moon is 375,000 km from Earth, calculate the free space loss in dB of the signal centred at the carrier frequency.

(4 marks)

Question 2 (15 marks)

(a) Figure 2 shows a cellular network topology made up of **regular** hexagonal cells with each cell **side** 2 km long.

i. Derive the area of one such hexagonal cell.
ii. Derive the distance between the centres of any two adjacent cells.

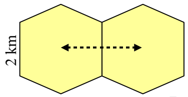


Fig. 2

(3+3 marks)

(b) Consider an area of 500 Km^2 covered by a cellular network. Each user requires 50 kHz for communication, and the total available bandwidth is 10 MHz:

i. If each cell has a radius of 3 kilometres. How many cell sites would be required, assuming hexagonal cells?
ii. How many users can be supported without frequency reuse?
iii. How many users can be supported with cluster size of 7?

(2+1+2 marks)

(c)

i. What do you understand by the Bandwidth Efficiency or Spectral Efficiency?
ii. What is the minimum E_b/N_0 required to achieve a Shannon Bandwidth Efficiency of 5 bps/Hz?

(2+2 marks)

Question 3 (20 marks)

(a) Assume that the following digital bit stream **10100110** is to be encoded in:

i. Alternate Mark Inversion
ii. Differential Manchester

Sketch the waveforms for each of the above-mentioned codes showing clearly the bit durations, signal levels and transitions where necessary.

(Assumption: The signal level for the most recent preceding mark was negative for AMI).

(3+4 marks)

(b) A wireless IEEE 802.11x frame transmitter is using a Generator Polynomial **x^4+x+1** and that the header contains the above bit pattern: **10100110**. Derive the transmitted CRC header checksum.

(4 marks)

(c) Describe the structure of a Bluetooth piconet.

(5 marks)

(d) What is the meant by the elevation of a communication satellite?
Why must satellites usually have an elevation greater than 10° ?

(2+2 marks)

Question 4 (15 marks)

(a) GSM uses a variant of Time Division Multiplexing (TDM) technique.

i. What is the name of this variant?
ii. Explain briefly how it works?
iii. Give **two** benefits of this variant multiplexing technique.

(1+2+4 marks)

(b) Direct Sequence Spread Spectrum (DSSS) is used in cellular networks.

i. Explain briefly how it works
ii. Give **two** advantages of this spread spectrum technique.

(4+4 marks)

Question 5 (25 marks)

(a) Give **two** advantages of Bluetooth transmission over infra-red transmission.

(2 marks)

(b) Consider the following WLAN layout in an office as shown in Figure 3.

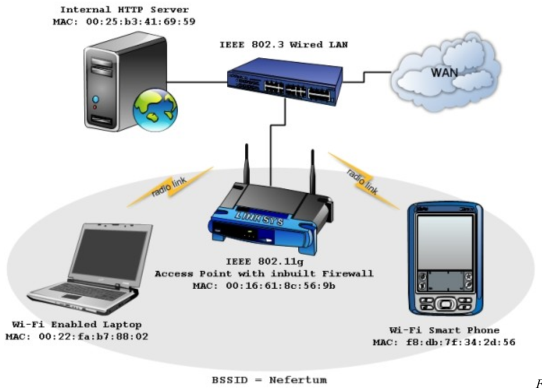


Fig. 3

i. What do you understand by term BSSID?

(2 marks)

ii. Give the generic structure of an IEEE 802.11x data frame.

(5 marks)

[P.T.O]

iii. The Access Point uses IEEE 802.11g technology. Describe this technology in terms of modulation type, maximum data rate, range and operating frequency.

(4 marks)

iv. How many non-overlapping channels are available for the IEEE 802.11g technology?
Give a situation where non-overlapping channels are useful?

(1+3 marks)

v. Give the values of the relevant fields in the wireless frames if the smart phone sends a wireless frame destined to the laptop **via** the Access Point.

(5 marks)

(c) Bluetooth can be used for audio transfer using SCO.

i. What is the meaning of SCO?
ii. Give **four** properties of this method of transfer.

(1+2 marks)

******END OF EXAMINATION PAPER******