A Notebook on Wireless Communication Systems

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A Notebook on
Wireless Communication System

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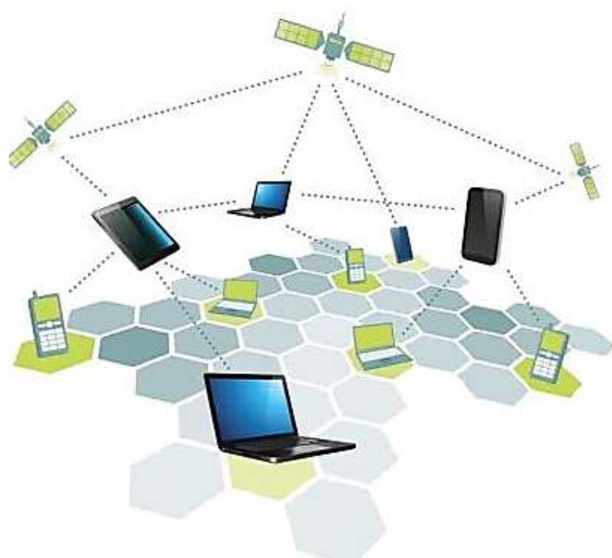

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Chapter 1: Introduction (Wireless Communication)

1.1 Introduction/Background Concept

Literally, any kind of communication without wired connection can be referred as the wireless communication e.g. speech, hand/light signals etc. Communication between two or more objects/entities through electromagnetic waves is called wireless communication. Wireless communication can be fixed, point to point and point to multi-point. In wireless mobile communication or simply mobile communication, a user can communicate wirelessly anytime and anywhere using a lightweight portable personalized communicator.

Issues & Challenges of Wireless Communication System

Once the customer is liberated from the confines of the wire and made free to move, the following issues arise:

- **Demand on the scarce radio resources:** The bandwidth of the RF spectrum available for the mobile communication is limited. However, service has to be provided for large number of subscribers notwithstanding the limitations in the availability of the frequency spectrum. This issue is resolved through a two-pronged attack. In the first place, no permanent RF channel is assigned for any customer and second, the same RF channels are reused at geographical locations that are considerably separated so as not to interfere with each other's communication.

- **Authentication of the customer:** Another important issue in mobile communication is the need to authenticate the genuineness of the customer whenever he receives or attempts to make communication. Since the media being open space, it is necessary to verify whether the customer is the one whom he claims to be before resource allocation. Security of the mobile account is to be ensured to prevent unauthorized use and also misuse of one's subscription. Privacy of the communication over radio is to be ensured though the radio signals are available everywhere for interception.

- **Security and Privacy on the radio:** One of the important issues for the customer is the availability of seamless service profile irrespective of his location. This is an essential feature of the mobile communication, particularly when the customer visits a service area served by an operator different from his own. Different dialing codes for accessing the same service in different networks can play havoc in realizing the service by the customer.

- **Keep track of the user as they move:** Yet another challenging issue in mobile communication is the need to keep track of the customer's location so that an incoming call can be connected to him. Equally important is the need to main established calls as the customer moves across areas covered by different radio transmitter / receiver.

- **Providing service across networks:** Since no service provider's network can provide coverage throughout the world, there is a need to offer service to one's customers through different other networks across the world to enable him to move freely. This necessitates inter network compatibility and connectivity with strict access safeguards.

- **Billing the customer whenever and wherever he makes and receives calls from:** Finally, since the customer is on the move and can avail service from any permissible network across the globe, there should be a foolproof billing and accounting mechanism that not only records the customer's liability for his consumption, but also that between the service providers.

Application of Wireless Communication System

Remote Sensing, Remote Controls, Radio Broadcast, Pager System, Wireless Headphone, Infrared and Bluetooth Devices, Wireless Mouse and Keyboard, Ham Radio, Mobile Phones, Cordless Phones, Wireless Networks, GPS Equipment, TV Broadcast, Military Communication, Satellite Communication, Deep Space Communication, etc.

1.2 History of Wireless Communication System

Important Milestones

- 1831: Faraday demonstrated electromagnetic induction.
- 1864: J. Maxwell (1831 – 1879), theory of electromagnetic fields, wave equations
- 1888: H. Hertz (1857 – 1894), demonstrated the existence of electromagnetic wave.
- 1895: G. Marconi first demonstrated wireless telegraphy.
- 1901: First transatlantic transmission
- 1915: First wireless voice transmission (New York and San Francisco)\
- 1946: First public mobile phone system (PMPS)/ 80 pound heavy equipment on truck.
- 1979: First cellular system deployed in Japan by Nippon Telephone and Telegraph Company (NTT)
- 1981: Nordic Mobile Telephone System (NMT450) was introduced in Scandinavia (Europe)
- 1982: Start of GSM specification
- 1983: Advanced Mobile Phone System (AMPS) is deployed in USA.
- 1985: European Total Access Cellular System (ETACS) was developed
- 1990: First phase of GSM specification published and deployed in Europe
- 1991: Digital AMPS (DAMPS/USDC) is standardized in USA
- 1993: Pacific Digital Cellular (PDC) used in Japan.
- 1993: IS-95 CDMA is introduced by Qualcomm Inc. in USA
- 1995: First commercial launch of CDMA
- 1997: 200GSM networks, 109 countries, 44 million users
- 1999: ITU-R releases IMT-2000 radio specification (UMTS, CDMA2000, TD-SCDMA)
- 2000: Introduction of GPRS
- 2001: Start of 3G services in Japan (FOMA: Freedom of Mobile Multimedia Access)
- 2002: WCDMA (UMTS) introduced and first deployed by NTT DoCoMo, Japan
- 2003: 3G services launched in UK (Hutchison 3/3/2003)
- 2005: Mobile TV first launched South Korea
- 2006: HSDPA (3.5G) launched worldwide.

Generation of Mobile Communication: a Comparative View

Description	***1G***	***2G***	***3G***
Era	1980's	1990's	2000's
Communication	Analog	Digital	Digital
Modulation	FM	QPSK, GMSK	QPSK, 8PSK, QAM
Duplexing	FDD	FDD	FDD, TDD
Multiple Access	FDMA	TDMA, CDMA	TDMA, CDMA
Services	Voice	Voice, Data	Voice, Video & High Speed Data
Roaming Services	NA	Available	Available
SMS Services	NA	Available	Available
Concept of SIM Card	NA	Available	Available
Technologies	AMPS, NMT	GSM, IS-95	WCDMA, CDMA2000 1X, CDMA 2000 3X

1.3 Fourth Generation Mobile Communication (4G)

- Successor to 3G and 2G families of standards.
- Refers to a change in the fundamental nature of the service, non-backwards compatible transmission technology and new frequency bands.
- All IP-Packet-Switched Networks, Mobile Ultra band (Giga bit speed) and multi-career transmission.
- WiMAX and first-release 3G LTE (Long Term Evolution) have been deployed as a pre-4G technologies.
- 4G system is expected to provide a comprehensive and secure all IP based solution facilitating IP telephony, ultra-broad band internet access, gaming services and streamed multimedia.
- Multi carrier transmission (OFDMA) combined with MIMO and dynamic channel allocation.
- 4G refers to IMT-Advanced (International Mobile Telecommunications Advanced) as defined by ITU-R (International Telecommunication Union – Radio Services), which has peak data rates up to ~ 100 Mbps for high mobility and up to 1Gbps for low mobility.

4G Technologies

1. LTE (Long Term Evolution)

- It has theoretical net bit rate capacity of up to 100 Mbps in the downlink and 50 Mbps in the uplink if a 20 MHz channel is used and more if MIMO (antenna array) is used.
- First publicly available in two Scandinavian capitals, Stockholm (Ericsson System) and Oslo (Huawei System) on 14^{th} December, 2009 and branded 4G.

2. WiMAX

- Offers peak data rate of 128 Mbps in downlink and 56 Mbps in uplink over 20 MHz wide channel.
- IEEE 802.16e-2005 standard, with the objective to fulfill the IMT-Advanced criteria of 1Gbps stationary reception and 100 Mbps for mobile reception.
- The world's first commercial mobile WiMAX was opened by KT in Seoul, South Korea on 30^{th} June, 2006 and later branded as 4G network.

3. UMB: Ultra Mobile Broadband [Formerly EV-DO (Evolution Data Optimized) Rev. C]

- Discontinued 4G project within 3GPP2 (3^{rd} Generation Partnership Project – 2) to improve CDMA 2000 mobile phone standard for next generation applications and requirements.
- Launched by Qualcomm in 2008 with and objective to achieve data speed over 275 Mbps downstream and over 75 Mbps upstream.

1.4 Basic Definition Related to Mobile Communication

Subscriber: A user who pays subscription charges for using a mobile communication system.

Mobile Station (MS): A station in the cellular radio service intended for use while in motion at unspecified locations. MS may be hand held personal units (portable) or installed in vehicles (mobiles).

Base Station (BS): A fixed station in a mobile radio system used for radio communication with mobile stations. BSs are located at the center or on the edge of a coverage region and consist of radio channels and transmitter and receiver antennas mounted on a tower.

Forward Channel (Downlink/Downstream): Radio channel used for transmission of information from the base station to the mobile.

Reverse Channel (Uplink/Upstream): Radio channel used for transmission of information for the mobile to base station.

Transceiver: A device capable of simultaneously transmitting and receiving radio signals.

Handoff: The process of transferring a mobile station from one channel or base station to another.

Roamer: A mobile station which operates in a service area (market) other than that from which service has been subscribed.

Mobile Switching Center (MSC): Coordinates the routing of calls in a large service area. In a cellular radio system, the MSC connects the cellular base stations and the mobiles to the PSTN. A MSC is called a mobile telephone switching office (MTSO).

Control Channel (CC): Radio channel used for transmission of call setup, call request, call initiation and other control purposes.

Simplex Systems: One-way communication.

Half Duplex Systems: Two-way alternate communication using the same radio channel.

Full Duplex Systems: Two-way simultaneous communication.

Page: A brief message which is broadcast over the entire service area, usually in simulcast fashion by many base stations at the same time.

FDD (Frequency Division Duplexing): Simultaneous radio transmission and reception between a subscriber and a base station, by providing two simultaneous but separate channels (frequency band).

TDD (Time Division Duplexing): Simultaneous radio transmission and reception between a subscriber and a base station, by providing two simultaneous but separate time slots on a single radio channel.

1.5 Examples of Wireless Communication Systems

Paging System

- Communication system that sends a brief message to a subscriber.
- The message may be either a numeric, an alphanumeric or a voice type.
- Typically used to notify a subscriber of the need to call a particular telephone number or travel to a known location to receive further instructions.
- Varies from simple paging system (limited range of 2 to 5 Kilo Meters) to wide area paging system (worldwide network).

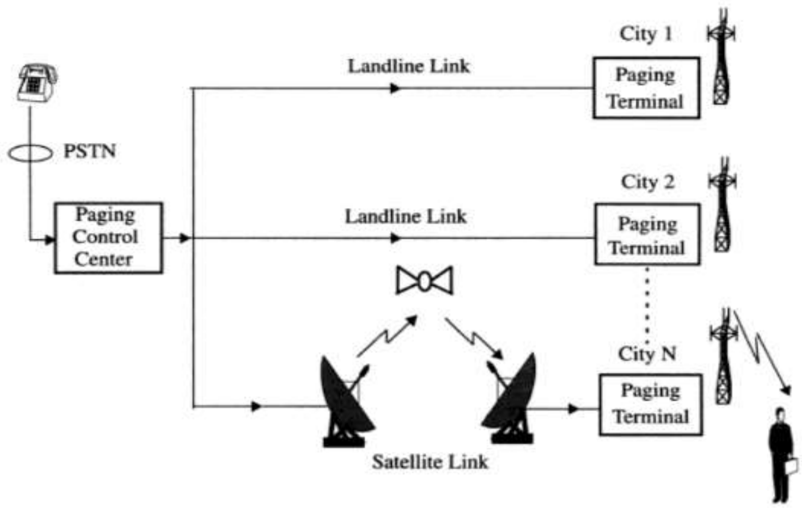


Figure: Wide Area Paging System

- The paging control center dispatches pages received from the PSTN throughout several cities at the same time, i.e. many base stations transmits/broadcast simultaneously known as simulcasting.
- The issued message is called a page. The paging system then transmits the page throughout the service area using base stations which broadcast the page on a radio carrier.

Cordless System

- Full duplex communication system.
- Use radio to connect a portable handset to a dedicated base station which then connected to a dedicated telephone line with a specific telephone number on PSTN (Public Switched Telephone Number).

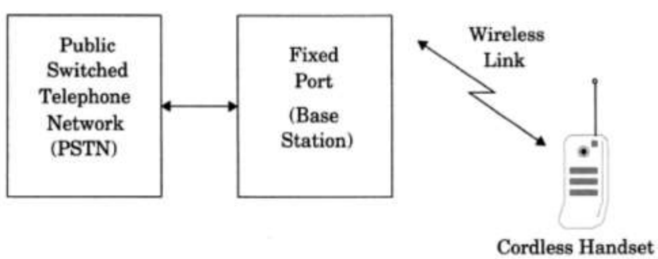


Figure: A Cordless Telephone System

- 1G Cordless (1980): communicates only over the distances of a few tens of meters, primarily used for in home system.
- 2G Cordless (1985): communicates over larger outdoor locations and also combined with paging system.

Chapter 2 - Speech Coding in Wireless System Applications

2.1 Introduction

- Speech coding techniques have assumed a considerable importance in communication systems as their performance, to a large extent, determines the quality of the recovered speech and the capacity of the system.
- In wireless communication systems, bandwidth is a precious commodity and service providers are continuously facing with the challenge of accommodating more users within a limited allocated bandwidth.
- Therefore, the goal of all speech coding systems is to transmit speech with the highest possible quality using the least possible channel capacity.

2.2 Classification of Speech Coding Techniques

- Based on the approach of achieving compression, speech-coders are classified into waveform coders and source coders.

Waveform Coders

- Waveform coders strive to reproduce time waveform of speech signal as closely as possible.
- They are source independent and hence can code equally well a variety of signals.
- It has minimal design complexity.
- They have moderate economy in transmission bit rate.
- Robust for wide range of speech characteristics and for noisy environments.
- Examples: PCM, DPCM, ADPCM, DM etc.

Source Coders

- Based on priory knowledge about the signal to be encoded and therefore, they are, in general, signal specific.
- Analyze the voice signal at the transmitter, transmit parameters derived from the analysis, and synthesize the voice at the receiver using those parameters.
- Much more complex than waveform coders and achieve very high economy in transmission rate.
- But they are less robust, and their performance tend to be talker dependent.
- Examples: Vocoders, LPC etc.

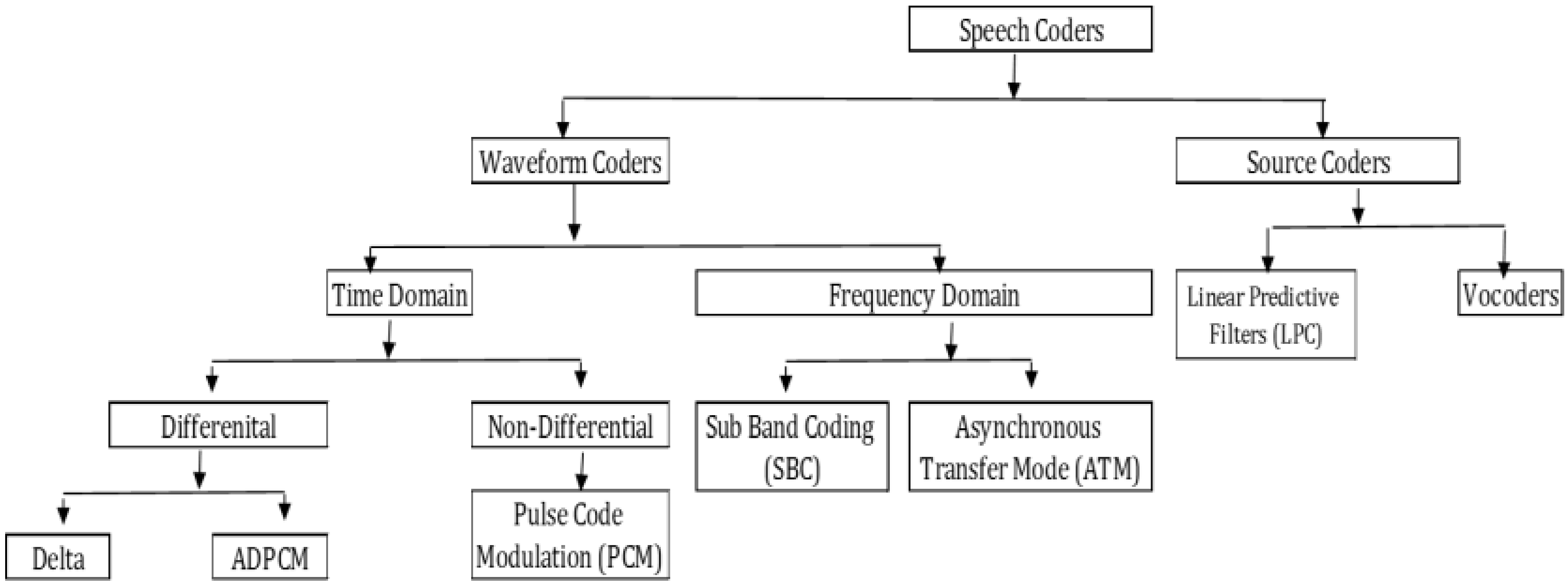


Figure: Major Classification of Speech Coding Techniques

2.3 Source Coders Examples

Vocoders

- Vocoders perform voice signal analysis by breaking down speech signal into small time segments.
- Derive parameters (amplitude, pitch, poles etc.) from analysis.
- Transmit parameters any synthesize at the receiver.
- Vocoders work by modeling speech generation process.

Vocoders Types

1. **Channel Vocoders**
   - Determine envelope of speech signal for number of frequency bands.
   - Sample, encode and multiplex voice, unvoiced and pitch frequency.

2. **Formant Vocoders**
   - Do not send sample of whole power spectrum envelope.
   - Transmit positions of peak (formants) of the spectral envelope.
   - Need at least 3 formants to represent speech.
   - Can operate below 1.2 Kbps.

3. **Ceptrum Vocoders**
   - Separate excitations from vocal tract spectrum.
   - High frequency: excitation coefficient and low frequency: spectral envelope
   - Periodic pulse train is formed at multiples of sampling period.

4. **Voice Excited Vocoders**
   - Eliminate need for pitch extraction and voicing decision.
   - PCM transmission for low frequency band.
   - Channel vocoding for high frequency band.
   - Pitch is generated at the synthesizer.
   - Range: 7.2 Kbps to 9.6 Kbps
   - Quality: superior.

Linear Predictive Coders (LPCs)

- LPCs belong to the time domain class of Vocoders, which attempts to extract the significant features of speech from the time waveform.
- Possible to transmit a good quality voice at 4.8 Kbps.
- Each channel's bandwidth is 0.2MHz making 124 channels per direction.
- Instead of transmitting quantized values of the error signals (as in ADPCM), LPC system transmits only selected characteristics of the error signal.
- The parameters include the gain factor, pitch information and voiced/unvoiced decision information, which allow approximation of the correct error signal.

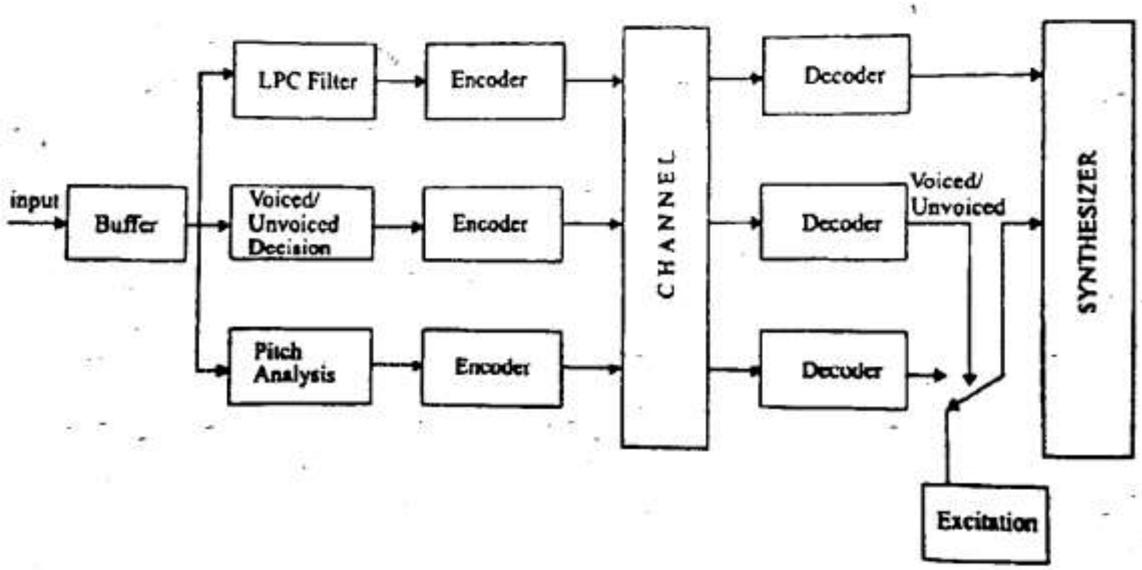


Figure: Linear Predictive Coder

Types of LPC

1. Multi Pulse Excited LPC (MPELP)

- Excitation through single pulse per pitch period produces audible distortions at the receiver. Thus, typically 8 pulses per period are used with adjustable amplitude and pulse position.
- Results in better speech quality because of several pulses per pitch period and multi-pulse algorithm that does not require pitch detection.

2. Code Excited LPC (CELP)

- Coder and decoder have a predetermined code book of stochastic (zero-mean white Gaussian) excitation signals. For each speech signal the transmitter searches through its code book for the one that gives the best perpetual match to the sound when used as an excitation to the LPC filter.
- The index of the code book where the best match was found is transmitted.
- CDMA (IS-95) uses a variable rate CELP codec at 1.2 to 14.4 Kbps.

3. Residual Excited LPC (RELP)

- Estimates model parameters (LP coefficients) and excitation parameters (voiced/unvoiced, pitch and gain) from a speech frame.
- The speech is synthesized at the transmitter and subtracted from the original speech signal to form a residual signal.
- Residual signal is quantized, coded and transmitted to the receiver along with LPC model parameters.
- At the receiver, the residual error signal is added to the signal generated using the model parameters to synthesize an approximation of the original speech signal.

2.4 Waveform Coders Examples

Frequency Domain Coding

- Speech signal is divided into a set of frequency components which are quantized and encoded separately.
- Different frequency bands are encoded preferentially according to some perceptual criteria for each band.
- The quantization noise is limited to that particular band only.
- Examples: SBC, ATC etc.

Types

1. SUB BAND CODING (SBC)

- Speech is typically divided into 4 or 8 sub bands.
- Bank of filters are used to divide speech.
- Each sub band is sampled at Nyquist rate.
- Sampled speech are encoded with different accuracy in accordance to perpetual criteria.
- Used for bit rate ranging from 9.6 Kbps to 32 Kbps.

2. ADAPTIVE TRANSFORM CODING (ATC)

- ATC is frequency domain coding.
- Encoding rate ranges from 9.6 Kbps to 20 Kbps.
- Input signal segmented into blocks of window size.
- Segments are represented by transform coefficients.
- Each segment is quantized and transmitted separately.
- At receiver, quantized coefficients are inverse transformed to reconstruct speech.
- Bits are allocated dynamically among frames while keeping bit rate constant.

Time Domain Coding

1. Pulse Code Modulation (PCM)

- Message signal is represented by a sequence of coded pulse, and is accomplished by representing the signal in discrete form in both time & amplitude.
- The basic operations performed in the transmitter of a PCM system are sampling, quantization and encoding. Low-pass filtering (anti-aliasing filter) prior to sampling is included only to prevent aliasing of the message signal.
- The quantizing and encoding operations are performed in an A/D converter.
- The basic operations in the receiver are regeneration of impaired signals, decoding and reconstruction of the train of quantized samples.

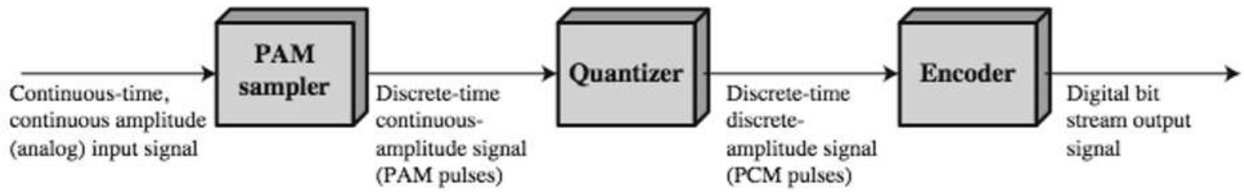


Figure: Block Diagram of PCM Transmitter

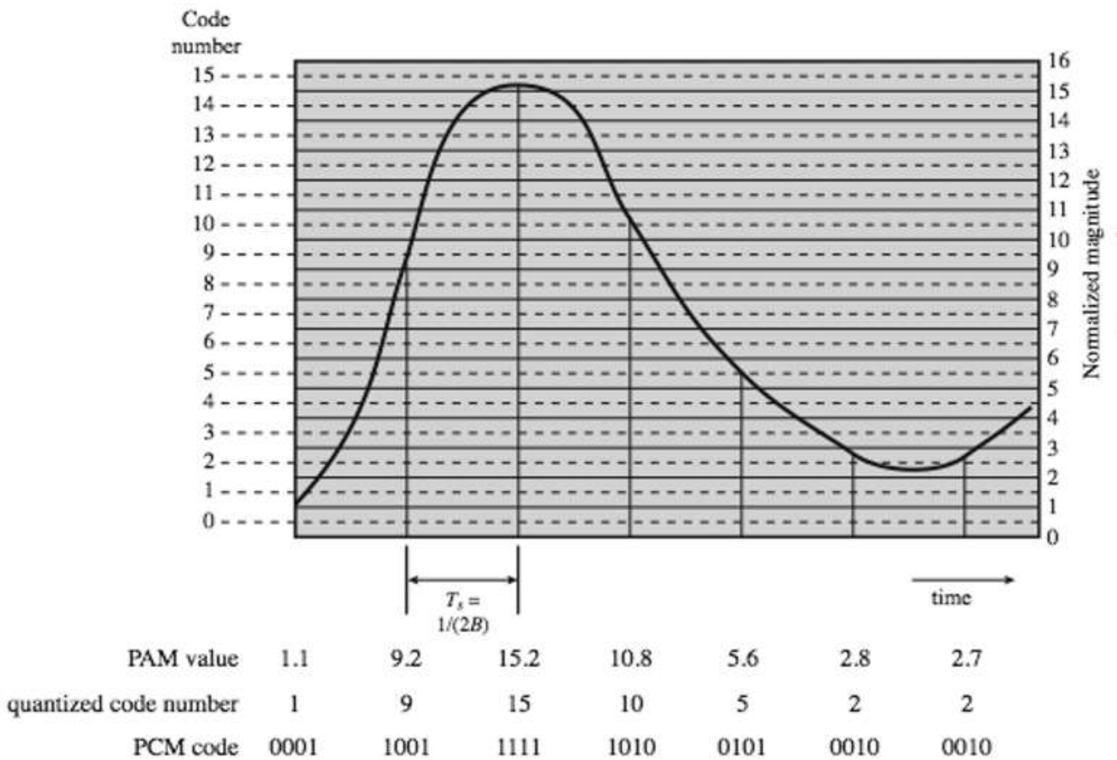


Figure: PCM Bit Stream

Operations

1. **Sampling**: The incoming message signal is sampled with a train of rectangular pulses, narrow enough to closely approximate the instantaneous sampling process. The sampling rate must be greater than twice the highest frequency component of the message signal. An anti-alias (low-pass) filter is used at the front end of the sampler in order to exclude frequencies higher than *W* prior to sampling.

2. **Non-uniform Quantization:** The use of a non-uniform quanitzer is equivalent to passing the message signal through a compressor and then applying the compressed signal to a uniform quanitzer. A particular form of compression law is called μ-**law, A-law.**

3. **Encoding**: An encoding process is to translate the discrete set of sample values to a more appropriate form of signal. If *R* denotes the number of bits per sample, we can represent a total of *2R* distinct numbers.
   For example, a sample quantized into one of 256 levels may be represented by an 8-bit code word.

2. Delta Modulation (DM)

- A variety of techniques have been used to improve the performance of PCM or to reduce its complexity. One of the most popular alternatives to PCM is delta modulation (DM).
- With delta modulation, an analog input is approximated by a staircase function that moves up or down by one quantization level (δ) at each sampling interval (T_s).
- The important characteristic of this staircase function is that its behavior is binary: At each sampling time, the function moves up or down a constant amount 'δ'. Thus, the output of the delta modulation process can be represented as a single binary digit for each sample.
- In essence, a bit stream is produced by approximating the derivative of an analog signal rather than its amplitude: A 1 is generated if the staircase function is to go up during the next interval; a 0 is generated otherwise.

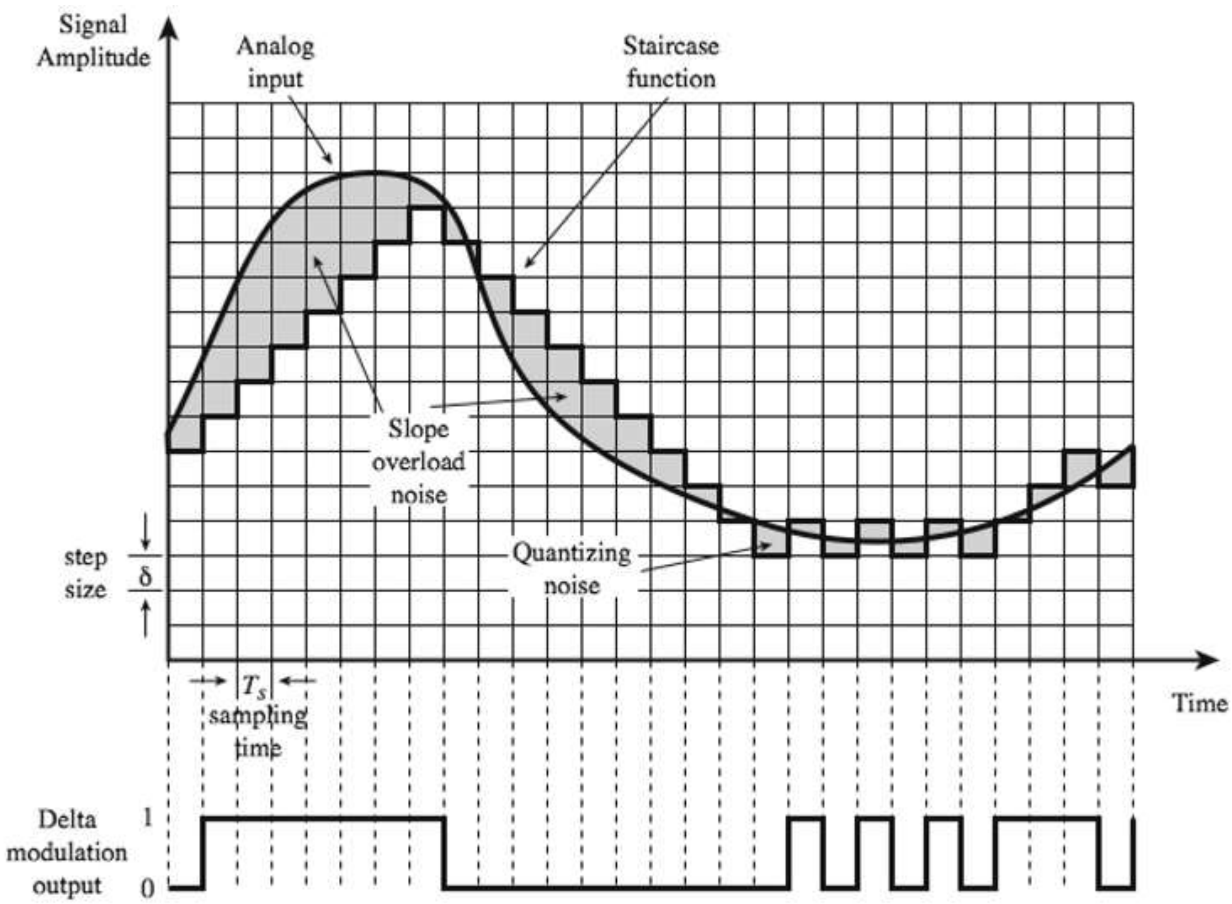


Figure: Example of Delta Modulation

2.5 Channel Coding and Error Control

In digital communication system, coding has specially two purposes: (i) for binary representation of discrete sources and (ii) for error detection and correction. Some of the codes used for digital representation are:

(a) Baudot Code: Used in Telegraph. Here 5-bit code is used to represent 32-characters.
(b) BCD Code: Binary coded decimal is used for numeric processing.
(c) ASCII Code: Seven bit code that represent alphabets and numbers. Eighth bit is used for parity.
(d) EBCDIC: Extended binary coded decimal interchange code is eight bit code similar to ASCII with no parity bit.

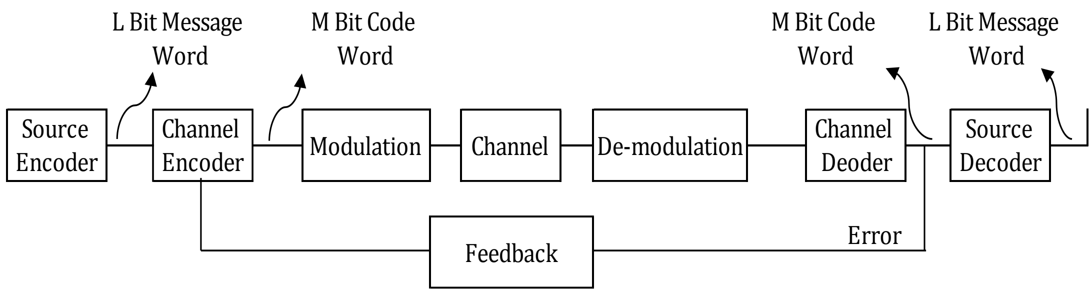


Figure: Channel Coding and Error Control in Digital Communication

In another part, coding is mainly concentrated to error detection and correction, because: (i) error probability in digital communication is a function of input SNR and data rate, (ii) maximum channel power and channel bandwidth is restricted to some level and (ii) channel noise is fixed by the distribution. Hence, error coding is performed by channel encoder at transmitting end and decoding is performed by channel decoder at the receiving end. It is possible to detect and correct by adding extra bit. But it is not possible to correct all errors unless redundancy is very high. (Extra redundant bit reduces efficiency).

Error Control Methods

Error control refers to mechanisms to detect and correct errors that occur in the transmission of frames. Data are sent as a sequence of frames; frames arrive in the same order in which they are sent; and each transmitted frame suffers an arbitrary and potentially variable amount of delay before reception. In addition, we admit the possibility of two types of errors:

- ***Lost Frame***: Frame fails to arrive at the other side, e.g. because a noise burst damaged a frame so badly it is not recognized
- ***Damaged Frame***: A recognizable frame does arrive, but some of the bits are in error (have been altered during transmission).

The most common techniques for error control are based on some or all of the following ingredients:

- ***Error Detection***: Detecting whether the received bits are altered as of sent by the transmitter.
- ***Positive Acknowledgment***: The destination returns a positive acknowledgment to successfully received, error-free frames.
- ***Retransmission after Timeout***: The source retransmits a frame that has not been acknowledged after a predetermined amount of time.
- ***Negative Acknowledgment and Retransmission***: The destination returns a negative acknowledgment to frames in which an error is detected. The source retransmits such frames.

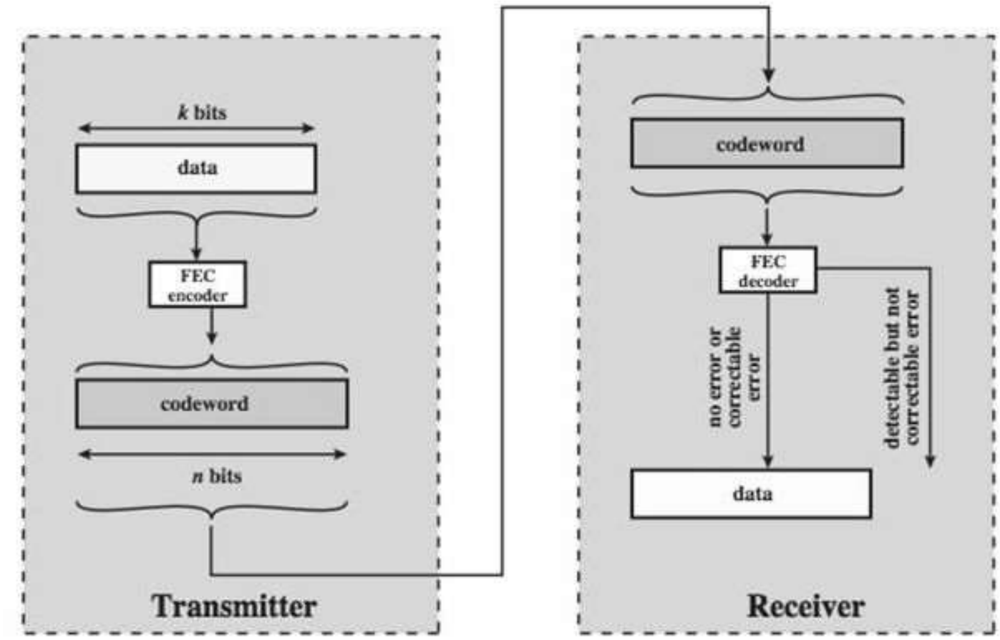


Figure: Error Detection Process

For a given frame of bits, additional bits that constitute an **error-detecting code** are added by the transmitter. This code is calculated as a function of the other transmitted bits. Typically, for a **data block of *k* bits**, the error-detecting algorithm yields an **error-detecting code of *n* – *k* bits**, where (n – k) < k. The error-detecting code, also referred to as the **check bits**, is appended to the data block to produce a **frame of *n* bits**, which is then transmitted. The receiver separates the incoming frame into the ***k* bits of data** and **(*n* – *k*) bits of the error-detecting code**. The receiver performs the same error-detecting calculation on the data bits and compares this value with the value of the incoming error-detecting code. A detected error occurs if and only if there is a mismatch.

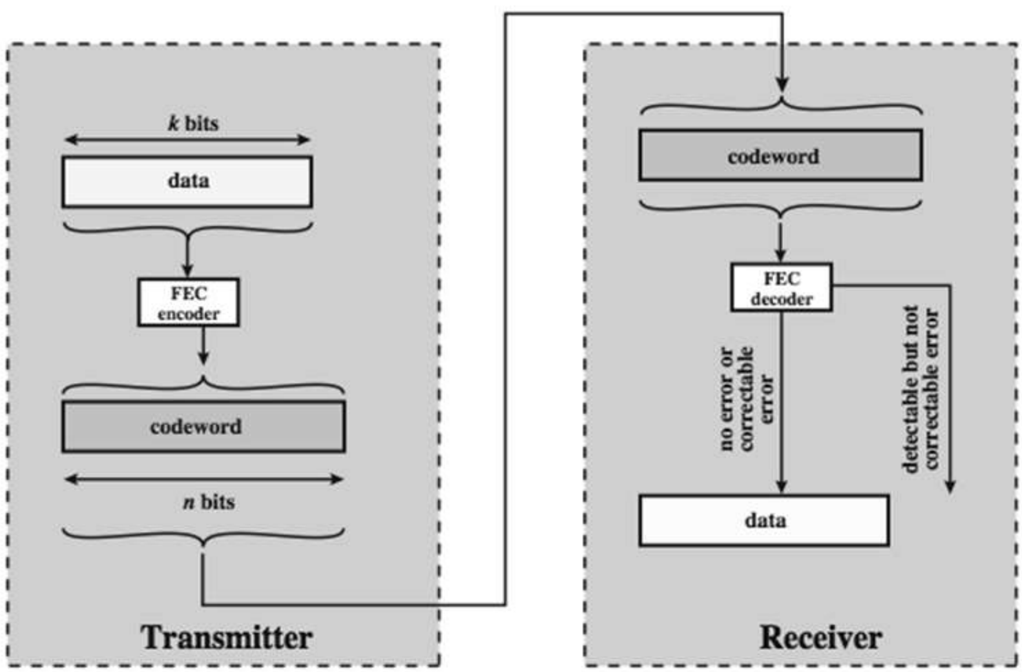


Figure: Error Correction Process

On the transmission end, each k-bit block of data is mapped into an n-bit block ($n > k$) called a **codeword**, using an FEC (forward error correction) encoder. The codeword is then transmitted. During transmission, the signal is subject to impairments, which may produce bit errors in the signal. At the receiver, the incoming signal is demodulated to produce a bit string that is similar to the original codeword but may contain errors. This block is passed through an FEC decoder, with one of four possible outcomes:

1. If there are no bit errors, the input to the FEC decoder is identical to the original codeword, and the decoder produces the original data block as output.
2. For certain error patterns, it is possible for the decoder to detect and correct those errors, the FEC decoder is able to map this block into the original data block.
3. For certain error patterns, the decoder can detect but not correct the errors, the decoder simply reports an uncorrectable error.
4. For certain, typically rare, error patterns, the decoder does not detect that any errors have occurred and maps the incoming data block into a block different from the original.

Chapter 3 – GSM & Cellular System Fundamentals

3.1 GSM – Global System for Mobile Communication

- GSM was the world's first digital cellular system (2G – Cellular System) in Europe to specify the digital modulation and network level architectures and services.
- GSM was originally developed to serve as pan-European cellular service in 900MHz (now also available in 1800MHz frequency band).
- GSM standard is set by ETSI – European Telecommunication Standards Institute.

Objectives

The design objectives of the GSM system can be briefly states as below:

- Excellent speech quality and High security and privacy
- Low module terminal cost and Low service and facilities cost
- Design of sleek and handled mobile terminals
- International roaming and Wide range of services and facilities
- Ability to adopt to new and innovative features
- Narrowband ISDN compatibility and High Spectral Efficiency
- Digital Radio

The digital radio uses the 900/1800 MHz band. The mobile terminal vary in power class 20 watts to as low as 0.8 watts. A GSM cell can cover a maximum distance of up to 30 kilo meters. The system can provide service to mobile customers traveling up to a maximum speed of 250 Km/hr.

GSM Network Elements and Architecture

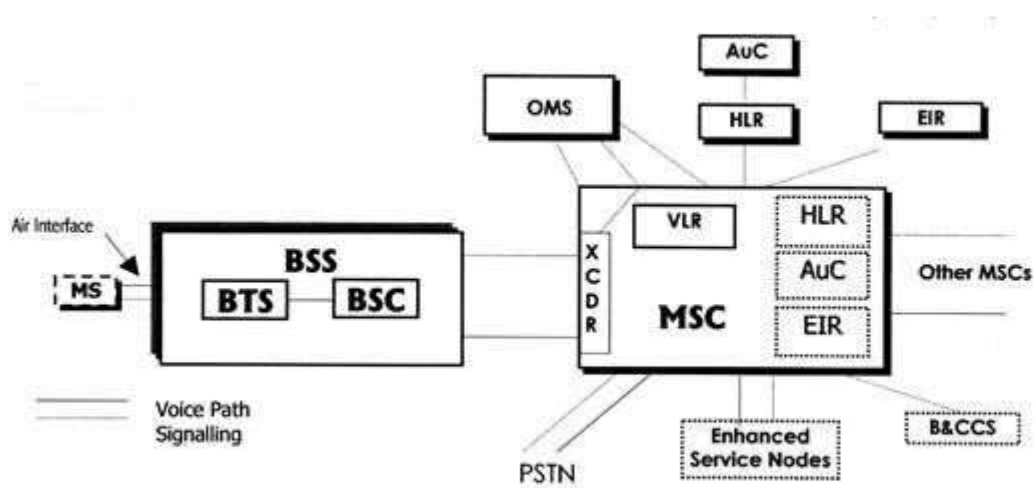


Figure: GSM System Architecture

Mobile station distinctly include two entities: Mobile Terminal Equipment (MT) and Subscriber Identity Module (SIM). SIM is one of the most remarkable features of GSM, which is a memory device that stores information such as subscriber's identification number, the networks and countries where the subscriber is entitled to service, privacy keys and other user specific information. Another remarkable features of GSM are 'Global Roaming' and 'On-the-Air-Privacy' which is provided by the system and therefore it is virtually impossible to eavesdrop (listen without the speaker's knowledge) on GSM radio transmission.

GSM system architecture consists of three major interconnected subsystems. These subsystems interact between themselves and with the users through certain network interfaces.

1. **Base Station Subsystem (BSS)**
   BSS is also known as Radio Subsystem. It provides and manages radio transmission paths between the mobile station (MS) and the Mobile Switching Center (MSC). It consists of many Base Station Controllers (BSCs) which connect the MS to the NSS via MSCs. Mobile handoffs (handover) between tow BTSs under the control of the same BSC are handled by the BSC itself and not by the MSC which reduce the switching burden to MSC.

2. **Network and Switching Subsystem (NSS**) and
   NSS manages the switching functions of the system and allows the MSCs to communicate with other networks such as the PSTN and ISDN. MSC is the central unit in NSS and controls the traffic among all the BSCs. NSS includes three different databases:
   a) *Home Location Register* (HLR): Contains subscriber information and location information for each user;
   b) *Visiting Location Register* (VLR) – which temporarily stores the IMSI (International Mobile Subscriber Identify) and customer information for each roaming subscriber;
   c) *Authentication Center* (AUC) – a strongly protected database which handles the authentication and encryption keys for every single subscriber in HLR and VLR. It contains a register called Equipment Identity Register (EIR) which identifies stolen or fraudulently altered phones.

3. **Operation Support Subsystem (OSS)**
   OSS supports the operation and maintenance of GSM and allows system engineers to monitor, diagnose and troubleshoot all aspects of GSM system.

Besides, GSM includes ***Enhanced Service Subsystem*** (ESS) which includes:

Wireless Application Protocol (WAP) System – allows wireless access to Internet sites from customer mobile handsets and manages telephony events.

Intelligent Network (IN) System – provides value added features such as Pre-Paid Service (PPS), Free Phone Service (FPH), Premium Rate Service (PRM), Mobile Virtual Private Network (MVPN), Universal Access Number (UAN) etc.

Content and Location Based Services (C&LBS) System and Unified Messaging System (UMS) e.g. Voice Mail, Fax Mail, E-Mail, Visual Mail, Short Message Service (SMS) etc.

While, ***Billing and Customer Care System*** (B&CCS) is responsible for obtaining the call details of each of the customers from the HPLMN as well as from all VPLMNs for raising the invoice. It incorporates a powerful and flexible rating engine that would enable the service provider to offer innovative and competitive tariff packages. This subsystem also incorporates a sophisticated printing subsystem for distributed printing of the customer invoices.

GSM Interfaces

- Each BSC typically controls up to several hundred Base Transceiver Stations (BTSs). The MS communicate with the BSS over the Radio-Air-Interface called ***Um-Interface***.

- The interface that connects a BTS to a BSC is called ***Abis-Interface***, which carriers traffic and maintenance data. In practice, Abis for each GSM BS manufacturer has a subtle differences, and therefore, the service providers has to use the same manufacturer for the BTS and BSC equipment.

- The BSCs are physically connected via dedicated/leased lines or microwave link to the MSC and the interface between a BSC and a MS is called the ***A-Interface***. This allows a service provider to use the BSs and switching equipment made by different manufacturers.

GSM Radio System

- GSM utilizes two bands (890-915 and 935 – 960) of 25 MHz band.
- The 890 – 915 MHz band is used for ***Subscriber-to-Base Transmissions (Reverse Link/Uplink)*** and the 935 – 960 MHz band is used for ***Base-to-Subscriber (Forward/Downlink)***
- GSM use FDD (Frequency Division Duplexing) and combination of TDMA (Time Division Multiple Access) and FHMA (Frequency Hopping Multiple Access) schemes to provide base stations with simultaneous access to multiple users.
- The available forward and reverse frequency bands are divided into 200 KHz wide channels called ARFCN (Absolute Radio Frequency Numbers).
- ARFCN denotes a forward and reverse channel pair which is separated in frequency by 45 MHz
- Each of eight subscribers uses the same ARFCN and occupies a unique time slots per frame.
- Radio transmission on both forward and reverse link are made at a channel data rate of 270.833 kbps using GMSK Modulation.

3.2 Cellular System Fundamentals (System Design Fundamentals)

Cellular Concept

- A concept to solve the problem of spectral congestion and user capacity.
- Cellular system offers very high capacity in a limited spectrum allocation without any major technological changes.
- This concepts replaces a single, high power transmitter (large cell) with many low power transmitters (small cells), and each providing coverage to only a small portion of the service area.
- Each BS is allocated a portion of the total numbers of channels available to the entire system and nearby BSs are assigned different groups of channels so that all the available channels are assigned to a relatively small number of neighboring BSs.
- Neighboring BSs are assigned different groups of channels so that the interference between BSs is minimized.
- The available channels are distributed throughout the geographic region and may be used as many times as necessary so long as the interference between co-channel stations is kept below acceptable level.
- As the demand for the service increases (i.e. as more channels are needed within a particular market or serving area), the number of BSs may be increase (along with the corresponding decrease in transmitter power to avoid added interference), thereby providing additional radio capacity with no additional increase in radio spectrum.

Concept of Frequency Reuse

- Each cellular BS is allocated a group of radio channels to be used within a small geographic area called 'Cell'.
- The BS antennas are designed to achieve the desired coverage within the particular cell.
- By limiting the coverage area to within the boundaries of a cell, the same group of channels may be used to cover different cells that are separated from one another by distances large enough to keep interference levels within tolerable limits.

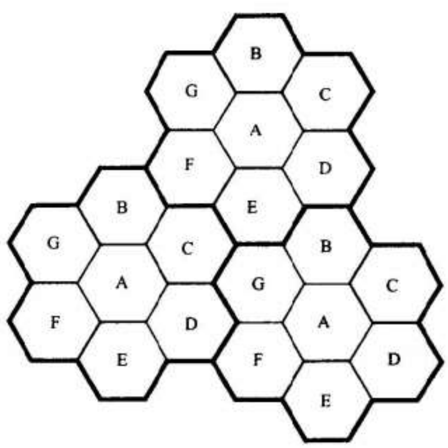


Figure: Hexagonal Cell Geometry Illustrating Frequency Reuse

- The design process of selecting and allocation channel groups for all the cellular BSs within a system is called frequency reuse or frequency planning.
- Figure above illustrates the concept of cellular frequency reuse, where cells labeled with the same letter use the same group of channels.
- The hexagonal cell shape shown in figure is conceptual and is simplistic model of the radio coverage for each BS.
- Hexagon has been universally adopted since it permits easy and manageable analysis of a cellular system.

Why to use hexagonal Cell Geometry?

- It might seem natural to choose a circle to represent the coverage area of a BS, adjacent circles cannot be overlaid upon a map without leaving gaps or creating overlapping regions.
- When considering geometric shapes, which cover an entire region without overlap and with equal area, there are three sensible choices – a square, and equilateral triangle and a hexagon.
- For a give distance between the center of polygon and its farthest perimeter points, the hexagon has the largest area of the three.

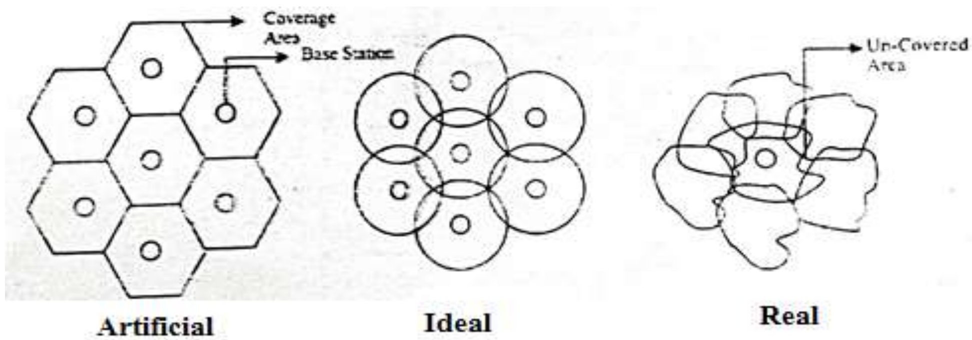


Figure: Hexagonal Approximation of Cell Coverage

- Thus, by using the hexagonal geometry, the fewest number of cells can cover a geographic region, and the hexagon closely approximates a circular radiation pattern, which would occur for an Omni-directional BS antenna and free space propagation.

Cluster and Frequency Reuse Factor

- A group of cells forms cluster when the entire available spectrum is divided equally among the cells. Cells in this group have a disjoint set of frequencies. The number of cells in a cluster must be determined so that the cluster can be repeated continuously within the coverage area of a service provider.

- The frequency reuse factor is given by 1/N, since each cell within a cluster is only assigned 1/N of the total available channels in the system.

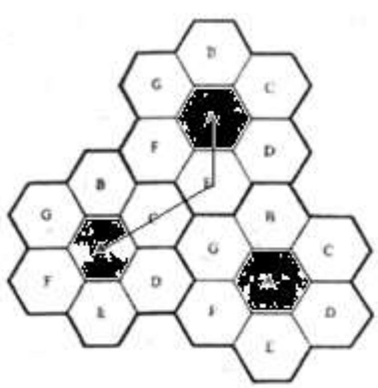


Figure: Seven Cluster Cell Geometry

- The typical clusters contain 4, 7, 12, or 21 cells. Only certain cluster sizes and cell layout are possible. The number of cells/cluster can only has values which satisfy - $\boldsymbol{N = i^2 + ij + j^2}$, where, *i* and *j* are non – negative integers.

- To find the nearest co-channel neighbors of a particular cell, one must do the following:

  i) Move '***i***' cells along any chain of hexagon and then
  ii) Turn 60º CW/CCW and move '***j***' cells as illustrated in alongside figure (N = 7).

System Capacity

If a cluster is replicated 'M-times' within the system, the total number of duplex channels 'C' can be used as a measure of capacity and is given by:

$$C = M * S = M * K * N = k * (M * N) = k * N_{BS}$$

Where, the total number of duplex channel available and distributed among 'N' cells; is denoted by 'S' and each cell is allocated a group of 'k' channels ($k < S$ and $S = k*N$).

Hence, the cellular system is directly proportional to the number of times a cluster is replicated in a fixed service area. $N_{BS} = M*N$ is the total number of BSs (cells) installed.

The smaller the number of cells per cluster is, the bigger the number of channels per cell will be. The capacity of each cell will therefore be increased. However, a balance must be found in order to avoid the interference that could occur between neighboring clusters.

Channel Assignment Strategies

Channel assignment deals with the allocation of channels to cells in a cellular network. Once the channels are allocated, cells may then allow users within the cell to communicate via the available channels. Channels in a wireless communication system typically consist of time slots, frequency bands and/or CDMA pseudo noise sequences. Basically there are two types of channel assignment strategies.

Fixed Channel Assignment (FCA)

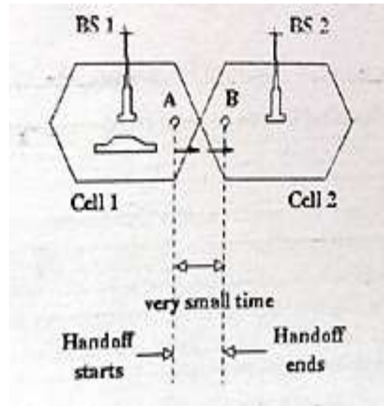
- Each cell is allocated a fixed number of voice channels. Any communication within the cell can only be made with the designated unused channels of that particular cell.
- If all the channels are occupied, then the call is blocked and subscriber has to wait.
- This is the simplest of the channel assignment strategies as it requires very simple circuitry but provides worst channel utilization. Later there was another approach in which the channels were borrowed from adjacent cell if all of its own designated channels were occupied. This was named as borrowing strategy. In such cases the MSC supervises the borrowing process and ensures that none of the calls in progress are interrupted.

Dynamic Channel Assignment (DCA)

- Channels are temporarily assigned for use in cells for the duration of the call.
- Each time a call attempt is made from a cell the corresponding BS requests a channel from MSC. The MSC then allocates a channel to the requesting the BS. After the call is over the channel is returned and kept in a central pool.
- To avoid co-channel interference any channel that is in use in one cell can only be reassigned simultaneously to another cell in the system if the distance between the two cells is larger than minimum reuse distance.
- When compared to the FCA, DCA has reduced the likelihood of blocking and even increased the trunking capacity of the network as all of the channels are available to all cells, i.e. good quality of service. But this type of assignment strategy results in heavy load on switching center at heavy traffic condition.

Handover Process/Handoff Strategies

- When a mobile moves into a different cell while a conversation is in progress, the MSC automatically transfers the call to a new channel belonging to the new BS and this is referred to as '*handoff operation'*.
- Handoff operation not only involves identifying a new BS, but also requires that the voice and control signals be allocated to channels associated with the new BS.
- Many handoff strategies prioritize handoff requests over call initiation requests when allocating unused channels in a cell site.
- Handoffs must be performed successfully and as infrequently as possible. System designers must specify an optimum signal level at which to initiate a handoff.



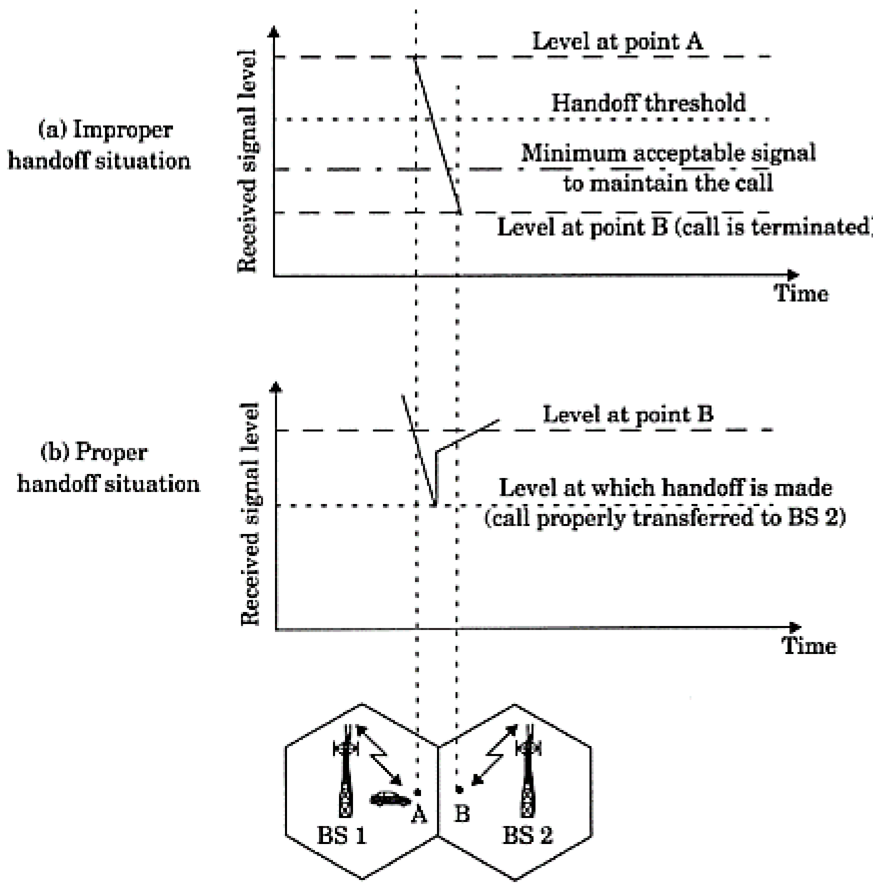


Figure: Handoff Strategies

- Once a particular signal level is specified as the minimum usable signal for acceptable voice quality at the BS receiver (normally taken as between -90dbm to -100dbm), a slightly stronger signal level is used as a threshold at which a handoff is made. This margin is given by:

$$\Delta = \boldsymbol{P_{r,\ handoff}} - \boldsymbol{P_{r,\ minimum\ usable}}$$

- This margin should not be too large (unnecessary handoffs which burden the MSC) or too small (there may be insufficient time to complete a handoff before a call is lost due to weak signal conditions).
- In deciding when to handoff, it is important to ensure that the drop in the measured signal level is not due to momentary fading and that the mobile is actually moving away from the serving BS. In order to ensure this, BS monitors the signal level for a certain period of time before a handoff is initiated. This running average measurement of signal strength should be optimized so that unnecessary handoffs are avoided.
- The time over which a call may be maintained within a cell, without handoff, is called ***'dwell time'.*** Which is governed by number of factors, including propagation, interference, distance between MS and BS and other time varying effects.

- By using different antenna heights (often on the same building or tower) and different power levels, it is possible to provide "large" and "small" cells, which are co-located at a single location. This technique is called '***Umbrella Cell Approach'*** and is used to provide large area coverage to high speed users while providing small area coverage to users travelling at low speeds. This techniques ensures a minimized handoffs for high speed users.

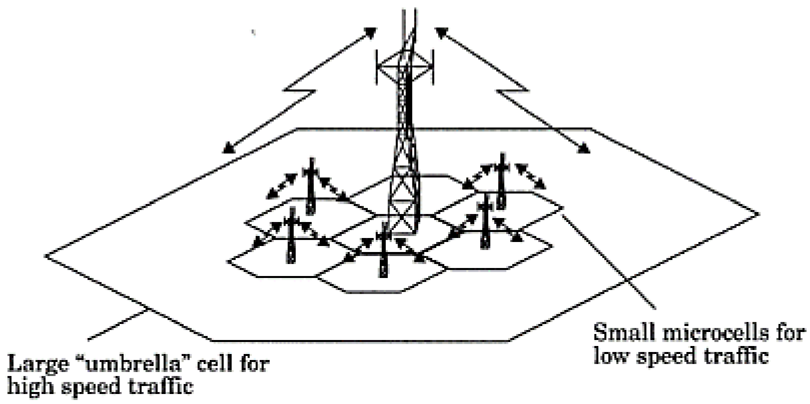


Figure: Umbrella Cell Approach

- In 1G analog cellular systems, signal strength measurements are made by the BS and supervised by the MSC. In 2G (TDMA) system, handoff decisions are mobile assisted called ***Mobile Assisted Handoff*** (MAHO), while if a mobile moves from one cellular system toe a different cellular system, handoff is controlled by different MSC known as intersystem handoff.

Types of Handoff

1. **Hard Handoff/Handover:** The channel in the source cell is released and only then the channel in the target cell is engaged. Thus the connection to the source is broken before the connection to the target is made – for this reason such handoffs are also known as ***break-before-make***. This phenomenon is most common in non CDMA networks.

2. **Soft Handoff/Handover:** The channel in the source cell is retained and used for a while in parallel with the channel in the target cell. The connection to the target is established before the connection to the source is broken, hence this handoff is called ***make-before-break.*** The interval, during which the two connections are used in parallel, may be brief or substantial. This phenomenon is used in CDMA networks.

Interference and System Capacity

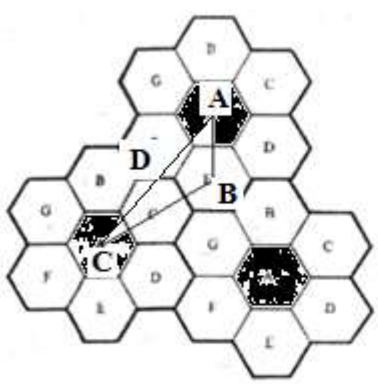
Interference is a limiting factor in the performance of cellular systems. It can occurs from clash with another mobile in the same cell or because of a call in the adjacent cell. There can be interference between the BSs operating at same frequency band or any other non-cellular system's energy leaking inadvertently into the frequency band of the cellular system. If there is an interference in the voice channels, cross talk is heard which appear as noise between the users.

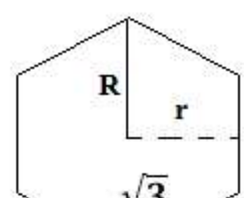
Adjacent Channel Interference (ACI)

- Interference resulting from signals which are adjacent in frequency to the desired signal is called Adjacent Channel Interference (ACI). It results from imperfect receiver filters, which allow nearby frequencies to leak into the pass band.
- ACI can be minimized through careful filtering and channel assignments. By keeping the frequency separation between each channel in a given cell as large as possible, ACI may be reduced considerably.

Co-Channel Interference (CCI)

- Frequency reuse implies that in a given coverage area, there are several cells that use the same set of frequencies.
- The cells that use the same set of frequencies are called co-channel cells, and the interference between signals from these cells is called co-channel interference (CCI).
- CCI cannot be combated by simply increasing the carrier power of a transmitter because CCI is independent of the transmitted power and becomes a function of the radius of the cell ® and the distance between centers of the nearest co-channel cells (D).
- By increasing the ratio of D/R, the spatial separation between co-channel cells relative to the coverage distance of a cell is increased, and interference is reduced from improved isolation of RF energy from the co-channel cell.





$$r = \frac{\sqrt{3}}{2}R$$

- In triangle ABC

$$D^2 = [i.2r]^2 + [j.2r]^2 - 2.[i.2r].[j.2r].cos120^o$$

$$D = \sqrt{[(2.r)^2(i^2 + i.j + j^2)]} = \sqrt{3R^2N} = R\sqrt{3N}$$

$$\frac{D}{R} = \sqrt{3N}$$

Where, the ratio, the spatial separation between co-channel cells relative to the coverage distance of a cell, D/R is known as co-channel reuse ration (Q). For a hexagonal geometry, the co-channel reuse ratio for different cluster size is given below.

i, j	*Cluster Size (N)*	*Co-Channel Reuse Ratio, Q = D/R*
i = 1, j = 1	3	3
i =1, j = 2	7	4.58
i =2, j = 2	12	6
i =1, j = 3	13	6.24

- If i_0 be the number of co-channel interfering cells; then, the signal-to-interference ratio, S/I or SIR for a mobile receiver which monitors a forward channel can be expressed as:

$$\frac{S}{I} = \frac{S}{\sum_{i=1}^{i_0} I_i}$$

Where S is the desired signal power form the desired BS and I_i is the interference power caused by the i^{th} interfering co-channel cell BS. Thus, if the signal level of co-channel cells are known, then the S/I ratio for the forward link can be found using above equation.

- When transit power of each BS is equal and the path loss exponent is the same throughout the coverage area, S/I for a mobile can be approximated as:

$$\frac{S}{I} = \frac{\left(\frac{D}{R}\right)^n}{i_0} = \frac{\left(\sqrt{3N}\right)^n}{i_0}$$

Which relates S/I to the cluster size N, which in turn determines the overall capacity of the system.

Trunking and Grade of Service

- The concept of trunking allows a large number of users to share the relatively small number of channels in a cell by providing access to each user, on demand, from a pool of available channels.
- In a trunked radio system, each user is allocated a channel on a per call basis, and upon termination of the call, the previously occupied channel is immediately returned to the pool of available channels
- In a trunked mobile radio system, when a particular user requests service and all of the radio channels are already in use, the user is blocked, or denied access to the system. In some system, a queue may be used to hold the requesting users until a channel becomes available.

- The grade of service (GOS) is a measure of the ability of a user to access a trunked system during the busiest hour. It is a benchmark used to define the desired performance of a particular trunked system.
- GOS is typically given as the likelihood that a cell is blocked.
- A GOS of 2% blocking implies that the channel allocations for cell sites are designed so that 2 out of 100 calls will be blocked due to channel occupancy during the busiest hour.

- Erlang (Erl) is used in mobile traffic engineering to represent the amount of traffic intensity carried by a channel that is completely occupied.
- There are two types of trunked systems which are commonly used, a) Blocked Calls Cleared – Erlang B Formula and b) Delayed Call Cleared – Erlang C Formula.
- The Erlang B Formula determines the probability that a call is blocked and is a measure of the GOS for a trunked system which provides no queuing for blocked calls.

The traffic intensity offered by each user (A_u) is equal to the call request rate (λ) multiplied by the holding time (t_m). That is, each user generates a traffic intensity of A_u Erlang given by: $A_u = \lambda * t_m$. For a system contacting U users and an unspecified number of channels, the total offered traffic intensity A is given as, $A = U * A_u$.

Trunking Efficiency

- This is the measure of the number of users which can be offered a particular GOS with a particular configuration of fixed channels.
- The way in which the channels are grouped can substantially alter the number of users handled by a trunked system.
- For example, 10 trunked channels at GOS of 0.01 can support 4.46 Erlang of traffic, whereas two group of 5 trunked channels can support 2 x 1.36 = 2.72 Erlang of traffic.
- Clearly 10 channel trunked together support 60% more traffic at specific GOS than do two 5 channel trunks
- Thus the allocation of channels in a trunked radio system has a major impact on overall system capacity.

Cellular System Capacity Improvement

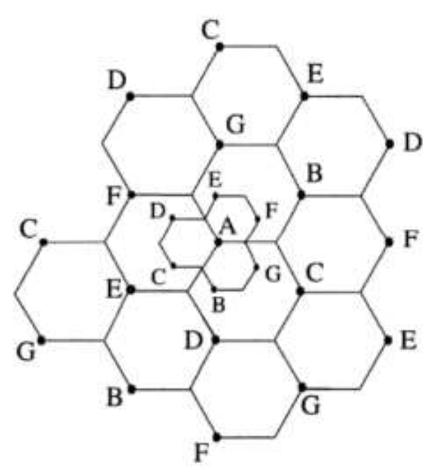
- As the demand for wireless service increases, the number of channels assigned to a cell eventually becomes insufficient to support the required number of users.
- Therefore, cellular design techniques are needed to provide more channels per unit coverage area. Techniques used to expand the capacity of cellular systems are, ***Cell Splitting, Cell Sectoring & Coverage Zone Approaches.***

Cell Splitting

- Cell splitting is the process of subdividing a congested cell into smaller cells, each with its own BS and a corresponding reduction in antenna height and transmitter power.

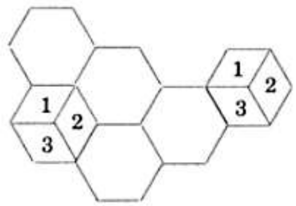
- It increases the capacity of cellular system since it increases the number of times that channels are reused. By installing smaller cells between the existing cells, capacity increases due to additional number of channels per unit area.

- At the same time cell radius is decreased but the co-channel reuse ratio, Q=D/R remain unchanged. So, it is necessary to ensure that the frequency reuse plan for the new micro and pico-cells behave exactly as for original cells.



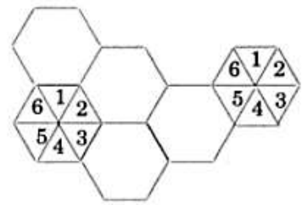
Cell Sectoring

- Sectoring increases the channel capacity with cell radius remaining unchanged but by decreasing the ratio of D/R, thus by reducing the cluster size, N using directional antennas.
- The co-channel interference in a cellular system is reduced by replacing a single omnidirectional antenna at the BS by several antenna radiating within a specified sector. The factor by which the co-channel interference gets reduced depends on the amount of sectoring used. Sectoring improves S/I.



(a)

120° sectoring;



(b)

60° sectoring.

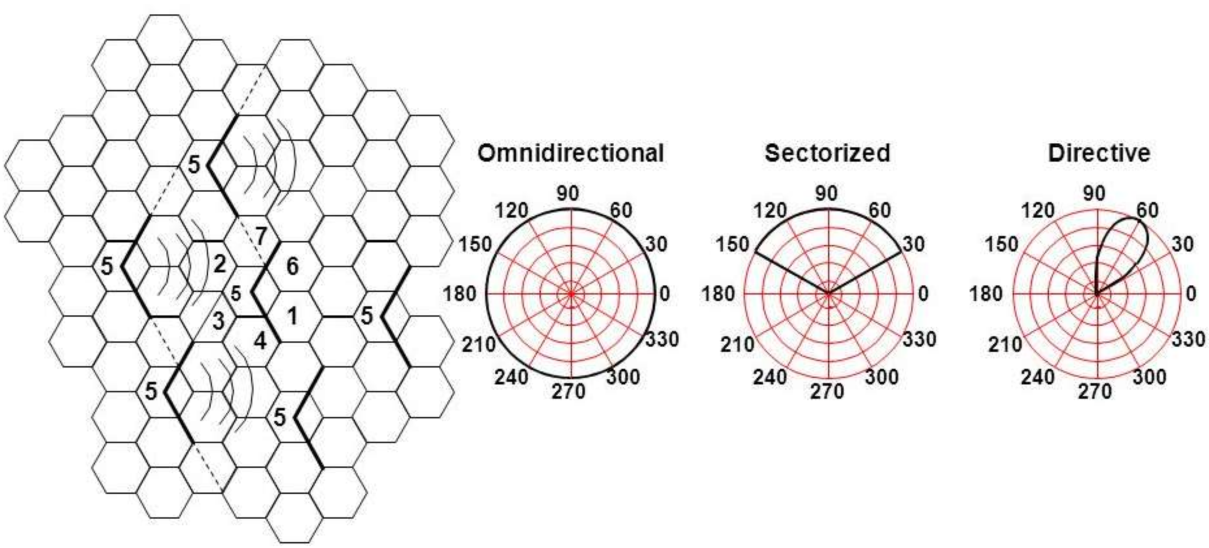


Figure: Sectorization and Antenna Radiation Pattern

- For a specified requirement of S/I, if i_0 (number of co-channel interference) is decreased, the value of N (cluster size) may be made lower; which increases the capacity with the same quality. However, sectoring degrades trunking efficiency.

Cluster Size (N)	*Number of Co-Channel Interfering Cells*		
	No Sectoring	120° Sectoring	60° Sectoring
3	6	3	2
4	6	2	1
7	6	2	1

Table: Sectoring and Number of Co-Channel Interfering Cells

 # Note: *The corresponding numerical problems had been exercised on lecture class, besides: a separate numerical assignment sheet and solution (probably) will be provided later or before the board exam*.

Chapter 4 – Mobile Radio Propagation

4.1 Introduction/Background Concept

- The Mobile Radio Propagation occurs in the form of Electro-Magnetic Wave (EMW) propagation. The mechanisms behind the EMW propagation are diverse; however, can generally be attributed to reflection, diffraction and scattering.
- Most cellular radio systems operate in urban areas where there is no direct Line-of-Sight (LoS) path between Transmitter (TxR) and the Receiver (RxR) and where the presence of high-rise buildings causes severe diffraction loss.
- Even though there exist a direct LoS between TxR and RxR, due to the multiple reflections from various objects, eth EMW travel along different paths of varying lengths, which cause multipath fading at a specific location, and the strength of waves decreases as the TxR/RxR separation increases.

Problems Unique to Wireless Systems

- Radio channels are ***extremely random*** and ***difficult to analyze.***
- Path can vary from ***simple LOS to ones that are severely obstructed*** by buildings, mountains and foliage.
- Interference may occur from other service providers ***– Out of band nonlinear transmitter emissions***.
- Interference from other user (same network) – ***CCI*** (Co-channel Interference) due to frequency reuse and ***ACI*** (Adjacent Channel Interference) due to TxR/RxR design limitation and large users sharing finite bandwidth (BW).
- ***Shadowing*** – Obstructions to LOS paths cause areas of weak received signal strength
- ***Fadin***g: when no clear LOS path exists, signals are received that are reflections off obstructions and diffractions around obstructions and due to multipath signals received at specific point that interfere with each other.
- ***Fixed Wireless Channel*** (FWC) – random and unpredictable, must be characterize in a statistical fashion and field measurements often needed to characterize radio channel performance.
- The ***Mobile Radio Channel*** (MRC) has unique problems that limit performance – A mobile receiver in motion influences rate of fading, the faster the mobile moves, more quickly characteristics change

Goals of Propagation Modeling

1. ***Predict average received signal strength for a given TxR/RxR separation***

   a. **Characterization of Received Signal Strength** – Over the distances from 20m to 20km.
   b. **Large Scale Fluctuation/Fading -** In general, large scale path loss decays gradually with distance from the transmitter and is affected by geographical features like hills and buildings. Hence, large scale radio wave propagation models are needed to estimate coverage area of BS.

2. ***Predict magnitude and Rate of received signal strength fluctuations over short distances/time duration***

   a. **Short** – Typically in a few wavelengths or seconds at 1 GHz ($\lambda = C/f = 3*10^8/1*10^9 = 0.3$ meter), received signal strength can vary drastically by 30 to 40 dB.
   b. **Small Scale Fluctuation/Fading** – Caused by multiple signals due to reflections and scattering received together which can be destructively added together by being out-of-phase.

4.2 Large Scale Fading

Propagation models that predict the mean signal strength for an arbitrary transmitter-receiver (T-R) separation distance are useful in estimating the radio coverage area of a transmitter and are called ***large-scale-propagation-model.***

Free Space Propagation Model (Free Space Path Loss Model)

- Used to predict the received signal strength when the transmitter and receiver have a clear, unobstructed LoS path between them.
- Satellite communication systems and microwave LoS radio links typically undergo free space propagation.
- The free space power received by a receiver antenna which is separated from a radiation transmitter antenna by a distance -'***d***', is given by the ***'Friis Free Space Equation'.***

$$P_r(d) = \frac{P_t G_t G_r \lambda^2}{(4\pi)^2 d^2 L} \dots (i)$$

Where;

P_t = Transmitted Power | $P_r(d)$ = Received power | d = T-R Separation

G_t = Transmitting Antenna Gain | G_r = Receiving Antenna Gain | λ = Wavelength in meters

L = System loss factor no related to propagation ($L \geq 1$)

- The Friis free space model is only for values of '***d***' which are in the far-field of the transmitting antenna. The far-field or Fraunhofer region, of a transmitting is defined as the region beyond the far-field distance 'd_f', which is related to the largest linear dimension of the transmitter antenna aperture and the carrier wavelength. The Fraunhofer distance is given by:

$$d_f = \frac{2D^2}{\lambda} \dots (ii) \; (\therefore d_f \gg D, \lambda)$$

- The system losses (L) are usually due to transmission line attenuation, filter losses and antenna losses in the communication system. The gains of antennas are dimensionless quantity and is related to its effective aperture (A_e), by:

$$G = \frac{4\pi A_e}{\lambda^2} \dots (iii) \; (\therefore \lambda = \frac{c}{f} = \frac{c}{\omega/2\pi} = \frac{2\pi c}{\omega})$$

- The path loss, which represents signal attenuation as a positive quantity measured in dB, is defined as the difference (in dB) between the effective transmitted power and the received power, and may or may not include the effect of the antenna gains. The path loss for the free space model when antenna gains are included is given by:

$$Path\ Loss\ (PL) = P_t - P_r$$

$$i.e.\ PL(dB) = 10.\log(PL) = 10.\log(P_t - P_r) = 10.\log\left(\frac{P_t}{P_r}\right) = 10.\log\left(\frac{P_t}{\frac{P_t G_t G_r \lambda^2}{(4\pi)^2 d^2 L}}\right)$$

$$= -10.\log\left(\frac{G_t G_r \lambda^2}{(4\pi)^2 d^2}\right) = -10.\log\left(\frac{\lambda^2}{(4\pi)^2 d^2}\right) \dots (iv) \; (\therefore For\ L = 1\ and\ G_t = G_r = 1)$$

- For a close in reference point 'd_0' *(typically 100m to 1km for outdoor system and 1m for indoor system),* the received power '$P_r(d)$' is given by:

$$P_r(d) = P_r(d_0)\left(\frac{d_0}{d}\right)^2 \dots (v) \; (\therefore For,\ d > d_0 > d_f)$$

Basic Radio Wave Propagation Mechanisms

1. Reflection

- When a radio wave propagating in one medium impinges upon another medium having different electrical properties (permittivity, permeability, conductance, specular reflexivity etc.), the wave is partially reflected and partially transmitted.
- If a plane wave is incident on a perfect dielectric, part of the energy is transmitted into the second medium and part of energy is reflected back into the first medium, and there is no loss of energy in absorption.
- If the second medium is perfect conductor, then all incident energy is reflected back into the first medium without loss of energy.
- The reflection coefficient is a function of the material properties, and generally depends on the wave polarization, angle of incidence, and the frequency of the propagating wave.
- Reflection occurs when a propagating EMW impinges upon an object which has very large dimensions when compared to the wavelength of the propagating wave.

2. Scattering

- When a radio wave impinges on a rough surface, the reflected energy is spread out (diffused) in all directions known as scattering.
- Scattering occurs when the medium through which the wave travels consists of objects with dimensions that are small compared to the wavelength of propagating wave and where the number of obstacles per unit volume is large.
- Objects such as lamp posts and trees tend to scatter energy in all directions, thereby providing additional radio energy at a receiver.

3. Diffraction

- Diffraction occurs when the radio path between the transmitter and receiver is obstructed by a surface that has sharp irregularities (edges).
- The phenomenon of diffraction can be explained by Huygens's principle, which states that "all points on a wavefront can be considered as point sources for the production of secondary wavelets, and that these wavelets combine to produce a new wavefront in the direction of propagation".
- The secondary waves resulting from the obstructing surface are present throughout the space and even behind the obstacle, giving rise to a bending of waves around the obstacle, even when a LoS path does not exist between TxR and RxR.

Ground Reflection (2-Ray) Model

- Free space propagation model is inaccurate when used alone, therefore 2-ray model could be a useful propagation model for large-scale strength model over distances of several kilometers for mobile radio systems that use tall towers (heights which exceeds 50m).

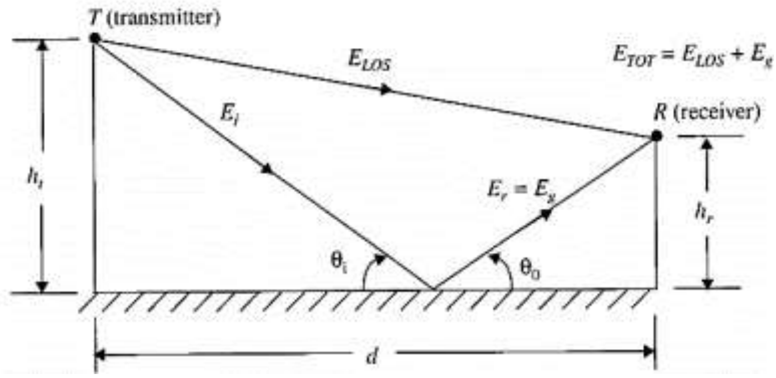


Figure: Two-Ray Ground Reflection Model

- In figure two TxR and RxR antenna with height 'h_t' and 'h_r' respectively are separated by a distance -'*d*'. E_{TOT} is the electric field that results from a combination of a direct LoS path and a ground reflected path.

$$\overrightarrow{E_{TOT}} = \overrightarrow{E_{LOS}} + \overrightarrow{E_g} \ \dots (i)$$

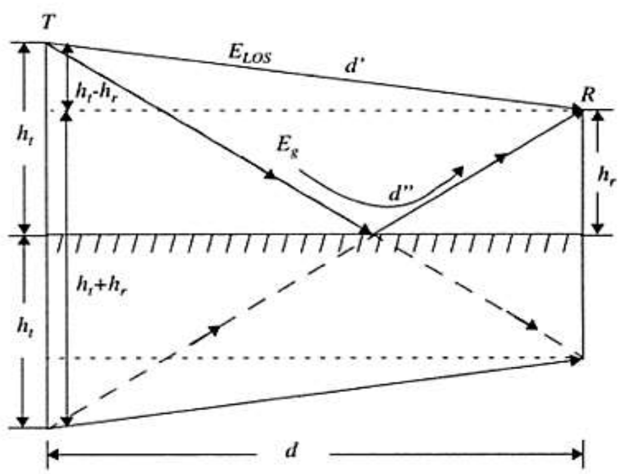


Figure: The method of images used to find the path difference between LoS and Ground Reflected Path

- Now using the method of images, which is demonstrated by the geometry of above figure, the path difference (Δ), between the LoS and the ground reflected paths cane be expressed as:

$$\Delta = d'' - d' = \sqrt{(h_t + h_r)^2 + d^2} - \sqrt{(h_t - h_r)^2 + d^2} \ = d\sqrt{\left(\left(\frac{h_t+h_r}{d}\right)^2 + 1\right)} - d\sqrt{\left(\left(\frac{h_t-h_r}{d}\right)^2 + 1\right)}$$

$$= d\left(1 + \frac{1}{2}\left(\frac{h_t+h_r}{d}\right)^2\right) - d\left(1 + \frac{1}{2}\left(\frac{h_t-h_r}{d}\right)^2\right) \text{ (Applying Taylor's Series Expansion)}$$

$$= \frac{1}{2d}\left((h_t + h_r)^2 - (h_t - h_r)^2\right)$$

$$= \frac{1}{2d}\left(\left({h_t}^2 + 2h_t h_r + {h_r}^2\right) - \left({h_t}^2 - 2h_t h_r + {h_r}^2\right)\right);$$

i.e. $\Delta = \frac{2h_t h_r}{d}$ … *(ii)*, Which works well for, d >> (h_t + h_r)

- Once the path difference is known, the phase difference, θ_Δ between the two E-field components and the time delay $\tau_d$between the arrivals of the two components can be easily computed using the following relations.

$$\theta_\Delta = \frac{2\pi}{\lambda}\,\Delta = \frac{\Delta\omega_c}{c} = \frac{2h_t h_r}{d}.\frac{\omega_c}{c} \text{ and } \tau_d = \frac{\Delta}{c} = \frac{2h_t h_r}{d.c} \;\ldots (iii)$$

- In another relating power to electric field, we will get:

$$P_r(d) = \frac{|E_{TOT}|^2}{120\pi}.A_e = \frac{P_t G_t G_r \lambda^2}{(4\pi)^2 d^2} \ldots (iv)$$

- Using trigonometric identities, |ETOT| is found to be: $2.\frac{E_0 d_0}{d}.\sin\left(\frac{\theta_\Delta}{2}\right)$, for small angle, $\sin(\theta) = \theta$, Then

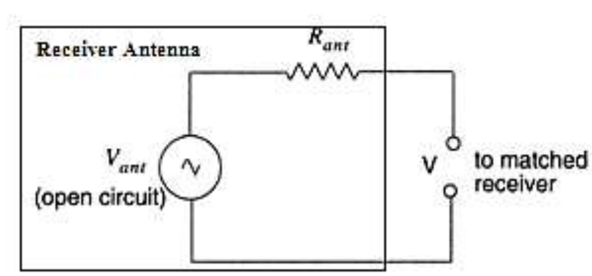
$$|E_{TOT}| = 2.\frac{E_0 d_0}{d}.\frac{\theta_\Delta}{2} = 2.\frac{E_0 d_0}{d}.\frac{\frac{2\pi}{\lambda}\Delta}{2} = 2.\frac{E_0 d_0}{d}.\frac{2\pi h_t h_r}{\lambda d} = k/d^2 \;\ldots (v)$$

$$\therefore P_r(d) = \frac{|E_{TOT}|^2}{120\pi}.A_e = \frac{\left(2.\frac{E_0 d_0}{d}.\frac{2\pi h_t h_r}{\lambda d}\right)^2}{120\pi}.A_e = \frac{16\pi^2 h_t^2 h_r^2 d_0^2}{\lambda^2 d^4}\left(\frac{|E_0|^2}{120\pi}.A_e\right) = \frac{16\pi^2 h_t^2 h_r^2 d_0^2}{\lambda^2 d^4}\left(\frac{P_t G_t G_r \lambda^2}{(4\pi)^2 {d_0}^2}\right)$$

$$i.e. P_r(d) = P_t G_t G_r.\frac{h_t^2 h_r^2}{d^4}$$

Circuit Model for Antenna System

- It is useful to relate the received power level to a receiver input voltage, as well as to an induced E-field at the receiver antenna. If the receiver antenna is modeled as a matched resistive load to the receiver, then the receiver antenna will induce an rms voltage into the receiver which is half of the open circuit voltage at the antenna. Thus, if V is the rms voltage at the input of a receiver (measured by a high impedance voltmeter), and sR$_{ant}$ is the resistance of the matched receiver, the received power is given by:



$$P_r(d)\mathrm{W} = \frac{V^2}{R_{ant}} = \frac{\left(\frac{V_{ant}}{2}\right)^2}{R_{ant}} = \frac{V_{ant}^2}{4R_{ant}}$$

Knife-Edge Diffraction Geometry/Model

- In mobile communication, the great advantage is: by diffraction (and scattering, reflection), the receiver is able to receive the signal even when not in LoS of the transmitter. Diffraction is the phenomena that explains the digression of a wave from a straight line path, under the influence of an obstacle, so as to propagate behind the obstacle.
- Though the intensity received gets smaller as receiver is moved into the shadowed region (region behind the obstacle), the radiation from a point source radiating in all directions can be received at any point. This is explained by ***Huygens-Fresnel Principle***.

Huygens-Fresnel Principle

It states that "all points on a wavefront can be considered as the point source for secondary wavelets which form the secondary wavefront in the direction of the propagation". That is in the case of an obstacle, the effect of wave source behind the obstacle cannot be felt and the sources around the obstacle contribute to the secondary wavelets in the shadowed region, leading to bending of wave.

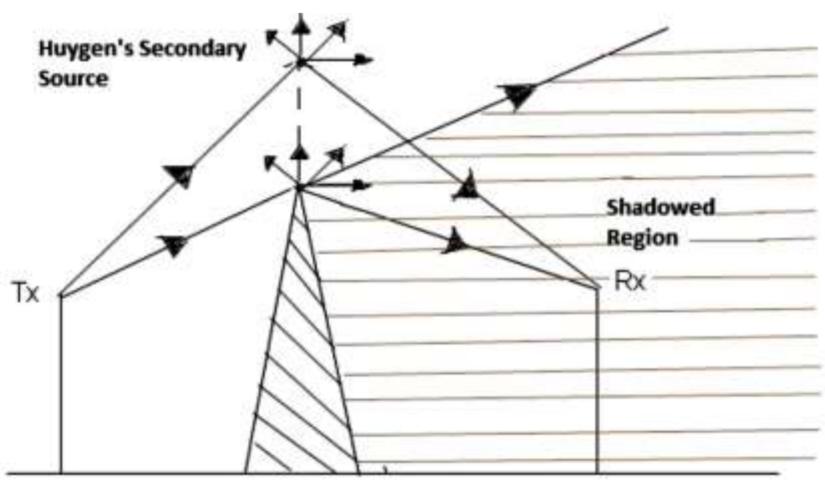


Figure: Huygens's Secondary Wavelets

- Let us consider that there is an impenetrable obstruction of height *'h'* at a distance of *'d_1'* from the transmitter and *'d_2'* from the receiver. The path difference between direct path and the diffracted path is calculated as:

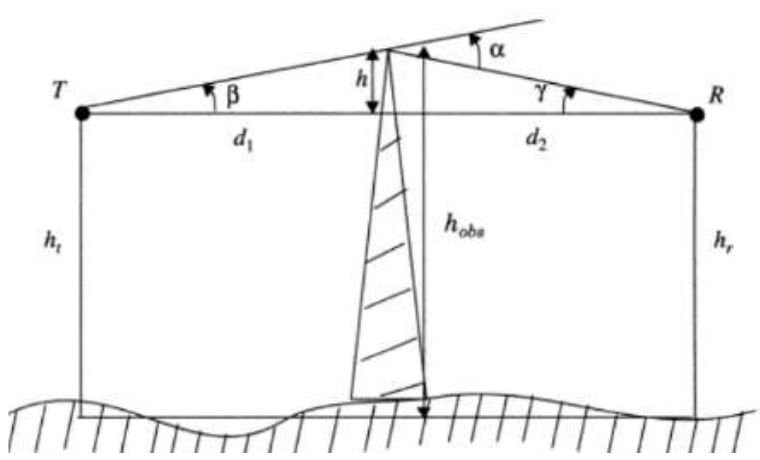


Figure: Diffraction through Sharp Edge

$$\Delta = \sqrt{d_1^2 + h^2} + \sqrt{d_2^2 + h^2} - (d_1 + d_2) \ = d_1\sqrt{\left(\left(\frac{h}{d_1}\right)^2 + 1\right)} + d_2\sqrt{\left(\left(\frac{h}{d_2}\right)^2 + 1\right)} - (d_1 + d_2)$$

$$= d_1\left(1 + \frac{1}{2}\left(\frac{h}{d_1}\right)^2\right) + d_2\left(1 + \frac{1}{2}\left(\frac{h}{d_2}\right)^2\right) - (d_1 + d_2) \text{ (Applying Taylor's Series Expansion)}$$

$$= \frac{h^2}{2d_1} + \frac{h^2}{2d_2} = \frac{h^2}{2}\frac{(d_1 + d_2)}{d_1 d_2}$$

Then, phase difference, $\theta_\Delta = \frac{2\pi}{\lambda}\ \Delta = \frac{2\pi}{\lambda}\ \frac{h^2}{2}\frac{(d_1+d_2)}{d_1 d_2}$

The diffraction gain due to presence of Knife Edge can be given as: $G_d(dB) = 20.\log|F(\vartheta)|$

i.e. $\quad G_d(dB) = 0; for\ \vartheta \leq -1$

$G_d(dB) = 20.\log(0.5 - 0.62\vartheta)\ ; for\ -1 \leq \vartheta \leq 0$

$G_d(dB) = 20.\log(0.5\exp(0.95\vartheta))\ ; for\ 0 \leq \vartheta \leq 1$

$G_d(dB) = 20.\log\big(0.4\,\text{sqrt}\big(0.1184 - (0.38 - 0.1\vartheta^2)\big)\big)\ ; for\ 1 \leq \vartheta \leq 2.4$

$G_d(dB) = 20.\log(0.225/\vartheta)\ ; for\ 2.4 \leq \vartheta$

Where, ϑ is a ***Fresnel-Kirchhoff Diffraction Parameter,*** given by: $\vartheta = h\sqrt{\frac{2(d_1+d_2)}{\lambda d_1 d_2}}$

From figure: $\alpha = \beta + \gamma \quad \rightarrow \quad tan\alpha = tan\beta + tan\gamma$

$\alpha = tan\beta + tan\gamma$ (For small angle $\alpha,\ tan\alpha = \alpha$)

$$\alpha = \left(\frac{h}{d_1}\right) + \left(\frac{h}{d_2}\right) = \frac{h(d_1+d_2)}{d_1 d_2}$$

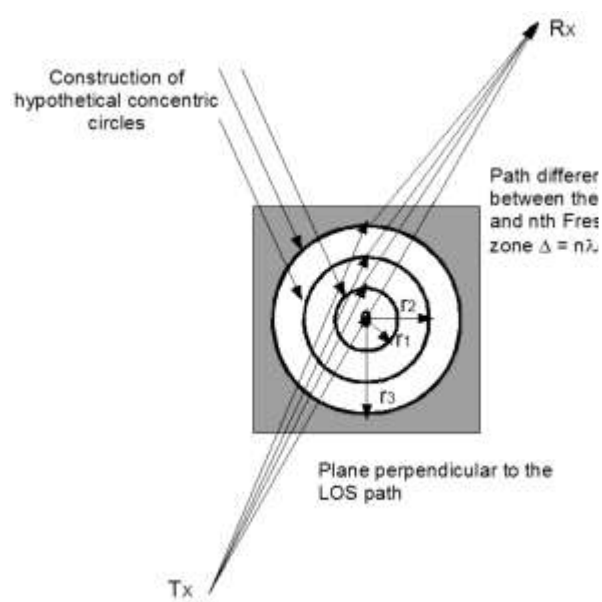
Then $\quad \vartheta = h\sqrt{\frac{2(d_1+d_2)}{\lambda d_1 d_2}} = \alpha\sqrt{\frac{2d_1 d_2}{\lambda(d_1+d_2)}}$

Fresnel Zones: The Concept of Diffraction Loss

- The more is the object in the shadowed region, greater is the diffraction loss of the signal. The effect of the diffraction loss is explained by Fresnel zones as a function of the path difference.

- The successive Fresnel zones are limited by the circular periphery through which the path difference of the secondary waves is (nλ/2) greater than total length of the LoS path, as shown in figure. Thus the successive Fresnel zones have phase difference of π which means they alternatively provide constructive and destructive interference to the received signal.

- The radius of each Fresnel zone is maximum at middle of transmitter and receiver (i.e. $d_1 = d_2$) and decreases as moved to either side.



Scattering

- It is often stronger that the radio signal impinges on a rough surface causes a reflected energy spread out (diffused) in all directions due to scattering. Objects such as lamp posts and trees tend to scatter energy in all directions, thereby providing additional radio energy at the receiver.

- Flat surfaces that have much larger dimension than a wavelength may be modeled as reflective surfaces. However, the roughness of such surfaces often induces propagation effects.

- ***Roughness Test:*** Surface roughness is often tested using Rayleigh criterion, which defines a critical height (h_c) of surface protuberances for a given angle of incidence θ_i, given by: $h_c = \lambda/8.\sin(\theta_i)$; If '$h < h_c$' the surface is called smooth surface otherwise called rough surface

- ***Radar Cross Section (RCS) Model:*** It is defined as the ratio of the power density of the signal scattered in the direction of the receiver to the power density of the radio wave incident upon the scattering object, and has units of square meters.

- For urban mobile radio system, the received power due to scattering in the far field is given by:

$$P_R(dBm) = P_T(dBm) + G_T(dBi) + 20.\log(\lambda) + RCS[dBm^2] - 30.\log(4\pi) - 20.\log(d_R) - 20.\log(d_T)$$

Where, d_T and d_R are the distance from the scattering object to the transmitter and receiver respectively. The scattering object is assumed to be in the far field (Fraunhofer Region). The variable *RCS* can be approximated by the surface area of the scattering objects.

Practical Link Budget Design Using Path Loss Models

Path loss models are used to predict large scale coverage for mobile communication system design. By using path loss models to estimate the received signal level as a function of distance, it becomes possible to predict the SNR for a mobile communication system.

a. Log - Distance Path Loss Model

The average received signal power decreases logarithmically with distance. The average large-scale path loss for an arbitrary T-R separation is expressed as a function of distance by using path loss exponent (n).

i.e. $\overline{PL(d)} \propto \left(\frac{d_0}{d}\right)^n$ as $P_r(d) = P_r(d_0)\left(\frac{d_0}{d}\right)^n$

$\overline{PL(d)}|_{dB} = \overline{PL(d_0)} + 10.n.\log(\frac{d_0}{d})$

Where d_0 is the close-in reference distance in the far field of the antenna.

Environment	*Path Loss Exponent, n*
Free Space	2
Urban Area Cellular Radio	2.7 to 3.5
Shadowed Urban Cellular Radio	3 to 5
In Building Line of Sight	1.6 to 1.8
Obstructed in Building	4 to 6
Obstructed in Factories	2 to 3

b. Log - Normal Shadowing

Log-Distance model does not consider the fact that surrounding environmental clutter may be vastly different at two different locations having the same T-R separation. This leads to measured signals which are vastly different than average value predicted by Log-Distance-Model. Measurements have shown that any value of d, the path loss at a particular location is random and distributed log-normally about the mean distance-dependent value.

i.e. $PL(d)|_{dB} = \overline{PL(d_0)} + 10.n.\log(\frac{d_0}{d}) + X_\sigma$

Where, X_σ is a zero-mean Gaussian distributed random variable with standard deviation, σ.

4.3 Small Scale Fading

Propagation models that characterize the rapid fluctuations of the received signal strength over very short travel distances (a few wavelengths) or short time durations (on the order of seconds are called ***small-scale-propagation-model or small-scale-fading-model.*** Generally, caused by the interference between two or more versions of the transmitted signal which arrive at the receiver at slightly different times; called multipath waves which combine at the receiving antenna to give a resultant signal that vary widely in amplitude and phase.

Factors Influencing Small Scale Fading

Multipath Propagation: The presence of reflecting objects and scatters in the channel creates constantly changing environment, resulting in multipath waves.
Speed of the Mobile: Relative motion between BS and MS results in shift in frequency (random frequency modulation), called Doppler Shift; which is positive or negative depending on whether the MS is moving towards or away from the BS.
Speed of the Surrounding Objects: The motion of objects induce time varying Doppler Shift on multipath components; if the surrounding objects move at a greater rate than the mobile, then this effects dominates small scale fading, otherwise it is ignored
The Transmission Bandwidth of the Signal: If the transmitted signal has a narrow bandwidth as compared to the channel, the amplitude of the signal will change rapidly, but signal will not be distorted in time.

Doppler Shift

- Consider a mobile moving at a constant velocity '*v*' along a path segment having a length '*d*' between points '*X*' and '*Y*', while it receives signals from a remote source '*S*' as shown in the figure. The difference in path lengths travelled by the wave from source '*S*' to the mobile at points '*X*' and '*Y*' is:

  $$\Delta l = d.cos(\theta) = \vartheta.\Delta t.cos(\theta)$$

  Where, Δt is the time required for the mobile to travel from X to Y and θ is assumed to be the same at points X and Y since source is assumed to be very far away. The phase change in the received signal due to the difference in path length is: $\Delta\emptyset = \frac{2\pi}{\lambda}\Delta l = \frac{2\pi}{\lambda}\vartheta.\Delta t.cos(\theta)$
- The apparent change in frequency or Doppler shift is given by: $f_d = \frac{1}{2\pi}\left(\frac{\Delta\emptyset}{\Delta t}\right) = \frac{\vartheta}{\lambda}\cos(\theta)$

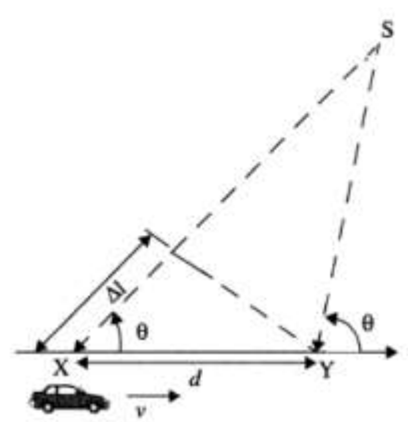


Figure: Illustration of Doppler Shift

Parameters of Mobile Multipath Channels

Multipath channel parameters are derived from the power delay profile. Power delay profile is a plot of relative received power as a function of excess delay with respect to a fixed time delay reference. Power delay profiles are found by averaging the instantaneous power delay profile measurements over a local area.

a. **Coherence Bandwidth (B_c) and Delay Spread (σ_τ)**

- Coherence bandwidth is a statistical measure of range of frequencies over which the channel can be considered "flat" (i.e. a channel which passes all spectral components with approximately equal gain and linear phase).
- Two sinusoids with frequency separation greater than B_c are affected quite differently by the channel.
- If the coherence bandwidth is defined as the bandwidth over which the frequency correlation function is above 0.9, then the coherence bandwidth is approximately $B_c = 1/50\sigma_\tau$, where, σ_τ is RMS delay spread (spectral broadening caused by multipath delay).
- If the frequency correlation function is above 0.5, then the coherence bandwidth, $B_c = 1/5\sigma_\tau$.

b. **Doppler Spread (B_D) and Coherence Time (T_c)**

- Doppler spread (B_D) is a measure of the spectral broadening caused by the time rate of change of the mobile radio channel and defined as the range of frequencies over which the Doppler spectrum is essentially non-zero.
- When a pure sinusoidal tone of frequency, f_c is transmitted, the received signal spectrum, called the Doppler spectrum, will have components in the range $(f_c - f_d)$ to $(f_c + f_d)$, where, f_d is the ***Doppler Shift.***
- The amount of spectral broadening depends on 'f_d' which is function of relative velocity of the mobile and the angle between the direction of motion of the mobile and direction of the scattered waves.
- Doppler Spread and Coherence Time (at which all spectral components are about to appear in order and in consistent way) are inversely proportional to one another, i.e. a ***Thumb Rule***: $T_C \propto 1/f_m \rightarrow 0.423/f_m$, where, '$f_m$' is the maximum Doppler shift given as: $f_m = v/\lambda$.

Types of Small Scale Fading

- Types of fading experienced by a signal propagating through a mobile radio channel depends on the nature of the transmitted signal with respect to characteristics of the channel.
- The time dispersion and frequency dispersion mechanisms in a mobile radio channel lead to ***four possible distinct effects.***
- Multipath delay spread leads to time dispersion and frequency selective fading, whereas Doppler spread leads to frequency dispersion and time selective fading.
- The two propagation mechanisms are independent of one another.

Small Scale Fading
(Based on Multipath Time Delay Spread)

Flat Fading	Frequency Selective Fading
1. BW of Signal < BW of the Channel	1. BW of Signal > BW of Channel
2. Delay Spread < Symbol Period	2. Delay Spread > Symbol Period

Small Scale Fading
(Based on Doppler Spread)

Fast Fading	Slow Fading
1. High Doppler Spread	1. Low Doppler Spread
2. Coherence Time < Symbol Period	2. Coherence Time > Symbol Period
3. Channel variations faster than baseband signal variation	3. Channel variations slower than baseband signal variation

Figure: Types of Small-Scale-Fading

a. Flat Fading (Amplitude Varying/Narrowband Channel)

If the mobile radio channel has a constant gain and linear phase response over a bandwidth which is greater than the bandwidth of the transmitted signal, then the received signal will undergo flat fading. In flat fading, spectral characteristics of the transmitted signal preserved at the receiver; however, the strength of received signal changes with time due to fluctuations in the gain of the channel caused by multipath.

b. Frequency Selective Fading (Wideband Channels)

If the channel has a constant gain and linear phase response over a bandwidth which is smaller than the bandwidth of the transmitted signal, then the channel creates frequency selective fading on received part. Received signal include multiple versions of transmitted signal which are attenuated and delayed in time and hence distorted. Time dispersion of the transmitted symbols induces Inter Symbol Interference.

c. Fast Fading (Time Selective Fading)

Channel impulse response changes rapidly within the symbol duration and this cause frequency dispersion and leads to signal distortion. Signal distortion increases with increasing Doppler spread. Fast fading occurs at only low data rates.

d. Slow Fading

Channel impulse response changes at a rate much slower than the transmitted baseband signal. Doppler spread is much less than the bandwidth of the baseband signal and channel can be assumed to be static. The velocity of the mobile of objects in the channel and baseband signaling determines whether the signal goes fast fading or slow fading.

4.4 Outdoor Propagation Models

Radio transmission in a mobile communication system often takes place over irregular terrain. The terrain profile of a particular area needs to be taken into account for estimating the path loss. These models vary widely in their approach, complexity and accuracy. Most of these models are based on a systematic interpretation of measurement data obtained in the service area.

a. Longley-Rice Model (1967-1978)

Applicable to point to point communication systems in frequency range from 40 MHz to 100 GHz over different kind of terrain. It does not consider the effects of buildings and foliage and the multipath.

b. Durkin's Model (1969)

Adopted by the Joint Radio Committee (JRC) in the U.K. for the estimation of effective mobile radio coverage areas. It models simply LoS and diffraction from obstacles but excludes the reflections from other surrounding objects and local scatters.

c. Okumura Model

Most widely used models for signal prediction in urban areas in the frequency range from 150 to 1920 MHz over the distance from 1 to 100km with antenna height 30m to 1000m. It is totally based on measured data but does not provide any analytic explanation. This method is not so good for rural areas. It was a standard for system planning in modern land mobile radio system in Japan. Common standard deviations between predicted and measured path loss is found to be 10dB to 14dB. The model is expressed as:

$$\boldsymbol{L_{50}(dB) = L_F + A_{mu}(f,d) - G(h_{re}) - G(h_{te}) - G_{Area}}$$

Where, L_{50} is the 50^{th} percentile value of propagation path loss, L_F is the free space propagation loss, A_{mu} is the median attenuation relative to free space, $G(h_{te})$ is the BS antenna height gain factor, $G(h_{re})$ is the mobile antenna height gain factor and G_{area} is the gain due to the type of environment.

d. Hata Model (1990)

Empirical formulation of the graphical path loss data provided by Okumura. This model is applied for the frequency range from 150 to 1500 MHz, the model is expressed for different areas:

$$\boldsymbol{L_{50}Urban(dB) = 69.55 + 26.16\log(f_c) - 13.82\log(h_{te}) - a(h_{re}) + (44.9 - 6.55log(h_{te}))logd}$$

Where '$\boldsymbol{d}$' is the T-R separation in kilometers and $\boldsymbol{a(h_{re})}$ is the correction factor for effective mobile antenna height.

e. PCS Extension to Hata Model

Extended Hata model for 2GHz PCS by European Co-operative for Science and Technical Research (EUROCOST). The model is extended as:

$$\boldsymbol{L_{50}Urban(dB) = 46.3 + 33.9\log(f_c) - 13.82\log(h_{te}) - a(h_{re}) + (44.9 - 6.55log(h_{te}))logd + C_M}$$

Where, $C_M = \begin{matrix} 0\ dB & for\ medium\ sized\ city\ and\ suburban\ areas \\ 3\ dB & for\ metropolitan\ centers \end{matrix}$

4.4 Indoor Propagation Models

The indoor radio channels differs from the traditional mobile radio channel in two aspects – the distances covered are much smaller, and the variability of the environment is much greater for a much smaller range of T-R separation. Propagation within the buildings is strongly influenced by specific features such as the layout of the building, the construction materials, and the building type.

a. **Partition Losses (Same Floor)**

   Partition that are formed as part of the building structure are called *hard partitions*, and partitions that may be moved and which don not span to the ceiling are called *soft partition.* Partitions vary widely in their physical and electrical characteristics, making it difficult to apply general models to specific indoor installation. But the radio path is always obstruction by common building materials which causes a signal to be lost.

b. **Partition Losses (Between Floors)**

   The loss between floors of a building are determined by the external dimensions and materials of the building, as well as the type of construction used to create the floors and the external surroundings. Even the number of windows in a building and the presence of tinting (which attenuates radio energy) can impact the loss between the floors. The attenuation increases with each additional floor.

Chapter 5 – Modulation Techniques

5.1 Introduction/Background Concept

- Modulation is the process of combining or encoding baseband message signal with carrier wave or the process of converting analog or digital information to a waveform suitable for transmission over a given medium.

- The modulation process translates a baseband message signal to a bandpass signal. It involves varying some characteristics or parameters of a carrier wave (amplitude, frequency or phase) in accordance with the amplitude of the message signal.

- On its counterpart, demodulation is the process of extracting the baseband message signal from the carrier wave.

Analog Vs Digital Modulation

- Analog modulation schemes were employed in the 1G mobile radio systems such as AMPS, ETACS etc. Since, digital modulation offers numerous benefits over the analog modulation, the conventional analog modulation systems have been replaced with the digital modulation techniques.

- Advancement in VLSI (Very Large Scale Integration) and DSP (Digital Signal Processing) technology have made digital modulation more cost effective than analog transmission systems.

- Digital modulation offers many advantages over analog modulation, e.g. ***greater noise immunity*** and ***robustness to channel impairments***, ***easier multiplexing of various forms of information*** (voice, data, image and video) and ***greater security***. Furthermore, digital transmissions accommodate ***digital error-control codes*** which detect and/or correct transmission errors, and ***support complex signal conditioning and processing techniques*** e.g. source coding, encryption and equalization to improve the performance of the overall communication link.

Factors that Influence the Choice of Digital Modulation

A modulation scheme of low bit error rates at low received signal-to-noise ratio (SNR) is desirable to perform well in multipath and fading conditions. The performance of digital modulation scheme is measured in terms of its power efficiency and bandwidth efficiency.

Power efficiency (PE)

- Describes the ability to preserve the fidelity of the digital message at low power levels.
- In digital transmission, in order to increase noise immunity, it is necessary to increase the signal power (with acceptable bit error probability).
- Expressed as the ratio of signal energy per bit to noise power spectral density (E_b/N_o) required at the receiver input for certain probability of error.

Bandwidth Efficiency (BE)

- Describes the ability of a modulation scheme to accommodate data within a limited bandwidth.
- In general, increasing the data rate implies decreasing the pulse width of a digital symbol, which increases the bandwidth of the signal. So, BE reflects how efficiently the allocated bandwidth is utilized.
- If ***R*** be the data rate in bps and ***B*** be the bandwidth occupied by the modulated ***RF*** signal, then ***BE = R/B*** (bps/Hz).
- There is a fundamental upper bound on achievable BE. ***Shannon's Channel Coding Theorem*** states that for an arbitrarily small probability of bit error rate (BER), the maximum possible BE is limited by the noise in the channel, and is given by: $\boldsymbol{(BE)_{max} = C/B = \log_2 (1+SNR)}$. Where, C - channel capacity (in bps), B - RF bandwidth (in Hz).
- Adding error control coding to a message increases the bandwidth occupancy (and this, in turn, reduces the bandwidth efficiency), but at the same time reduces the required received power for a particular BER, and hence trades bandwidth efficiency for power efficiency.
- On the other hand, higher level of modulation schemes (M-ary Keying) decrease bandwidth occupancy but increase the required received power, and hence trades the power efficiency for bandwidth efficiency.
- Hence, power and bandwidth efficiency considerations are very important for the choice of digital modulation schemes.

5.2 Digital Modulation Techniques

- In digital communication system, there is finite set of waveforms to be transmitted whereas in an analog communication, the set of waveforms is virtually infinite. It is not necessary to detect and reproduce the exact replica of the transmitted signal. It is sufficient only to decide whether the signal is present or absent.

- In demodulation of digitally modulated signal, a decision making device decides whether the signal is present or absent in the presence of additive noise and interferences with minimum error probability. A demodulator with least probability of error in decision making is called optimum detector or matched filter. A matched filter is an impulse response $h(t) = x(t-t_m)$, where t_m is a decision making instance and $x(t)$ is received signal. The output signal of the matched filter has maximum SNR at decision making instance, $t = t_m$. This effect is equivalent to amplifying signal and attenuating noise at that instant.

Review of Line Coding (Signaling Formats) and Binary Modulation Techniques

- Digital data (a sequence of binary digits) can be transmitted by various pulse waveforms. These pulse waveforms are also called line codes. To illustrate the number of signal formats for the transmission of binary data 01001100011 is given below (Figure: Encoding Schemes).

- On the other hand, in digital signals, at the modulator input take on only one of the two possible values, the communication is referred to as binary communication. Because baseband binary (digital) signals have sizeable power at low frequencies, they are suitable for transmission over a pair of wires or coaxial cables. So baseband binary signals cannot be transmitted over a radio link. For such purpose, we use analog modulation techniques in which the binary messages are used to modulate a high frequency continuous wave carrier. This corresponds to the process of keying (or switching) the amplitude, frequency or phase of the continuous wave carrier between either of two values corresponding to the binary symbols 0 and 1 (Figure: Binary Modulation Schemes).

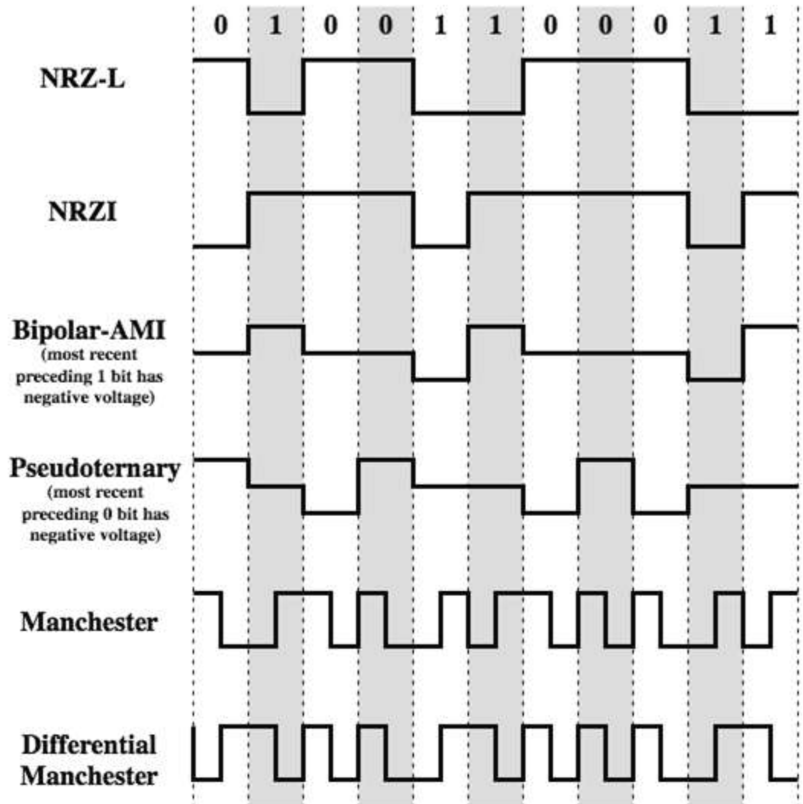


Figure: Encoding Schemes

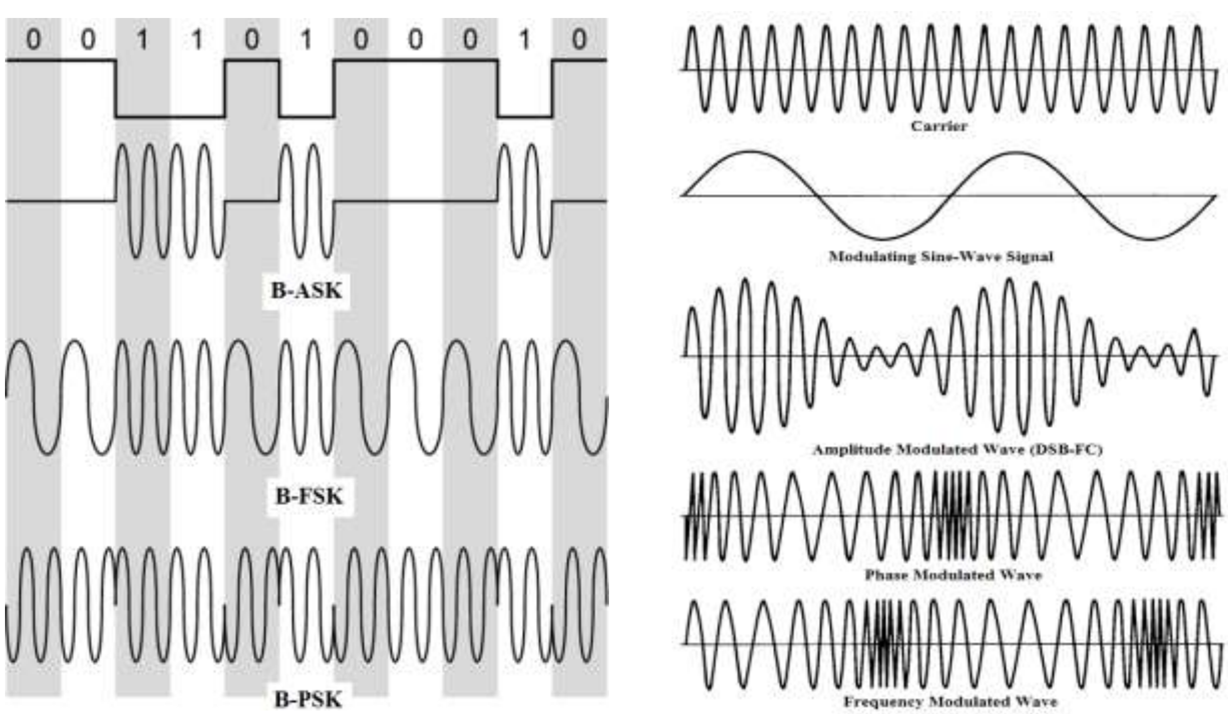


Figure: Binary Modulation Schemes and Analog Modulation Review

Binary Amplitude Shift Keying (BASK)

- In binary ASK, binary symbol 1 is represented by transmitting high frequency carrier wave of fixed amplitude A_c and fixed frequency f_c for a bit duration T_b.
- The binary symbol 0 is represented by switching off the carrier for T_b seconds. Thus the amplitude of the carrier varied according to the modulating binary signal. In ASK, the modulated signal can be expressed as:

$$s(t)|_{BASK} = \begin{cases} A_c Cos 2\pi f_c t & for\ b(t) = 1 \\ 0; & for\ b(t) = 0 \end{cases}$$

Binary Frequency Shift Keying (BFSK)

- In binary FSK, two sinusoidal waves of same amplitude A_c but different frequencies f_1 and f_2 are used.
- Binary symbol 1 is represented by carrier wave of frequency f_1 and binary symbol 0 is represented by carrier wave of frequency f_2. The modulated signal can be expressed as:

$$s(t)|_{BFSK} = \begin{cases} A_c Cos 2\pi f_1 t & for\ b(t) = 1 \\ A_c Cos 2\pi f_2 t; & for\ b(t) = 0 \end{cases}$$

Binary Phase Shift Keying (BPSK)

- In binary PSK, a sinusoid of fixed amplitude A_c and fixed frequency f_c is used to represent both the symbols 1 and 0.
- It has one fixed phase when data is at one level (i.e. for symbol 1) and when data is at another level (i.e. symbol 0), the phase is different by 180^0. Thus the transmitted signal is:

$$s(t)|_{BPSK} = \begin{cases} A_c Cos 2\pi f_c t & for\ b(t) = 1 \\ A_c Cos(2\pi f_c t + \pi) = -A_c Cos(2\pi f_c t); & for\ b(t) = 0 \end{cases}$$

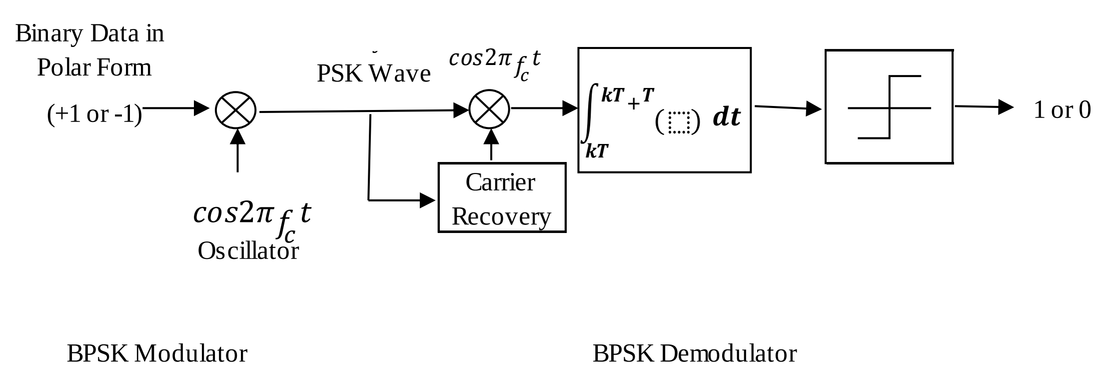


Figure: BPSK Modulator and Demodulator

* M-ary Signaling

- A baseband M-ary signal consists of M-possible amplitude levels; which are transmitted.
- In this system, the information source emits a sequence of symbols from an alphabet that consists of M-symbols.
- Each amplitude level at the transmitter output corresponds to a distinct symbol so that there are M distinct amplitude levels to be transmitted.
- Under similar conditions i.e. $T_b = T_s$, where T_b is the bit duration for binary system and T_s is the symbol duration for M-ary system, the data bit for M-ary system is $\log_2 M$ times the data rate for binary system, i.e. $\boldsymbol{R_s = R_b . \log_2 M}$

Disadvantage of M-ary Signaling:

- It requires ***(M-1)*** threshold levels, hence the receiver for M-ary system becomes more complex as number of comparators required is also ***(M-1).***
- The transmitted power must be increased by the factor $\boldsymbol{M^2/\log_2 M}$ compared to binary system.

M-ary FSK / Multiple FSK (MFSK)

- In M-ary FSK modulation, the binary data stream is divided into n-tuples of bits, where $\boldsymbol{n = \log_2 M}$.
- There are M signals with different frequencies to represent this M messages. The expression of MFSK is:

$$s(t)|_{MFSK} = A_c Cos(2\pi f_i t + \emptyset_i)\ ;\ \ for\ kT \le t \le (k+1)T; for\ M_i$$

- Where, T is the symbol period which is ***n*** times the bit period. It is a non-linear, constant envelope modulation technique.

$$s(t)|_{4FSK} = \begin{Bmatrix} A_c Cos(2\pi f_1 t) & for\ dibit\ 00 \\ A_c Cos(2\pi f_2 t\ ) & for\ dibit\ 01 \\ A_c Cos(2\pi f_3 t\ ) & for\ dibit\ 10 \\ A_c Cos(2\pi f_4 t\ ) & for\ dibit\ 11 \end{Bmatrix}$$

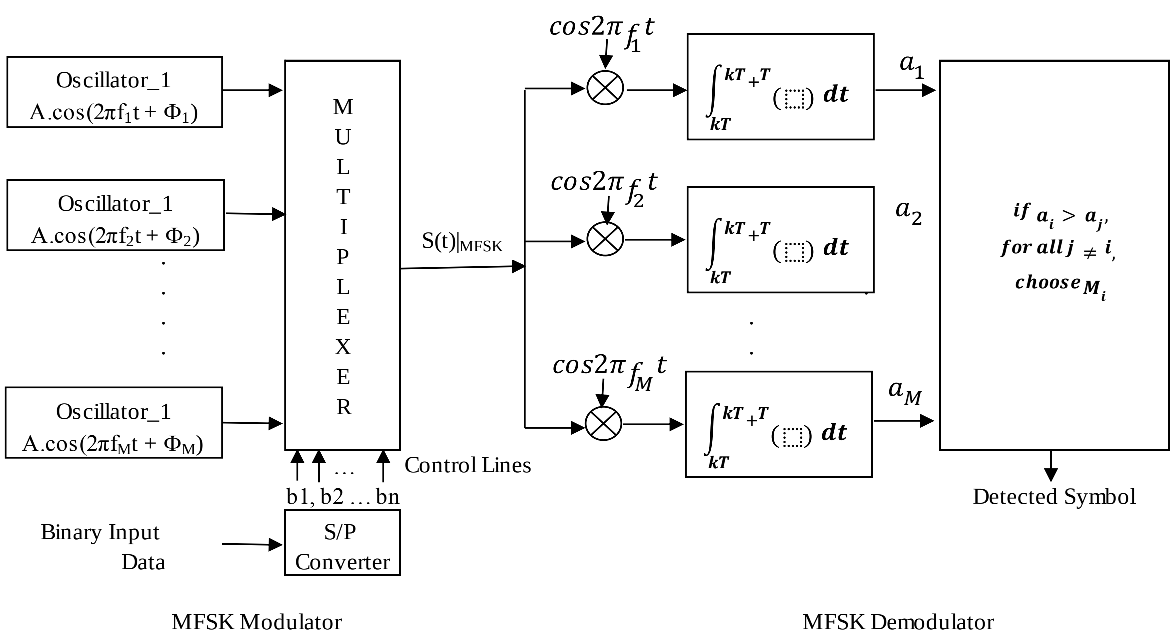


Figure: MFSK Modulator and Demodulator

M-ary PSK / Multiple PSK (MPSK)

- In M-ary FSK modulation, the binary data stream is divided into n-tuples of bits, where $n = log_2M$.
- There are M signals with different phase to represent this M messages. The expression of MPSK is:

$$s(t)|_{MPSK} = A_c Cos(2\pi f_c t + 2\pi i/M)\ ;\ \ for\ kT \le t \le (k+1)T; for\ M_i$$

Where, T is the symbol period which is ***n*** times the bit period.

- All symbol waveforms have the same amplitude, hence each MPSK signals have constant amplitude/envelope.

Special Cases:

BPSK: M = 2, n = $log_2M = log_2 2 = 1$ (i.e. n = either 1 or 0)

$$s(t)|_{BPSK} = A_c Cos(2\pi f_c t + 2\pi i/M) = A_c Cos(2\pi f_c t + 2\pi i/2) = A_c Cos(2\pi f_c t + \pi i);\ \text{for i = 0 and 1}$$

QPSK: M = 4, n = $log_2M = log_2 4 = 2$ (i.e. n = 00, 01, 10, 11/dibit)

$$s(t)|_{QPSK} = A_c Cos(2\pi f_c t + 2\pi i/M) = A_c Cos(2\pi f_c t + 2\pi i/4) = A_c Cos(2\pi f_c t + \pi i/2\ );\ \text{for i = 0 to 3}$$

$$s(t)|_{QPSK} = s(t)|_{4PSK} = \begin{Bmatrix} A_c Cos(2\pi f_c t + 0) & for\ dibit\ 00 \\ A_c Cos(2\pi f_c t + \pi/2\ ) & for\ dibit\ 01 \\ A_c Cos(2\pi f_c t + \pi\ ) & for\ dibit\ 10 \\ A_c Cos(2\pi f_c t + 3\pi/2\ ) & for\ dibit\ 11 \end{Bmatrix}$$

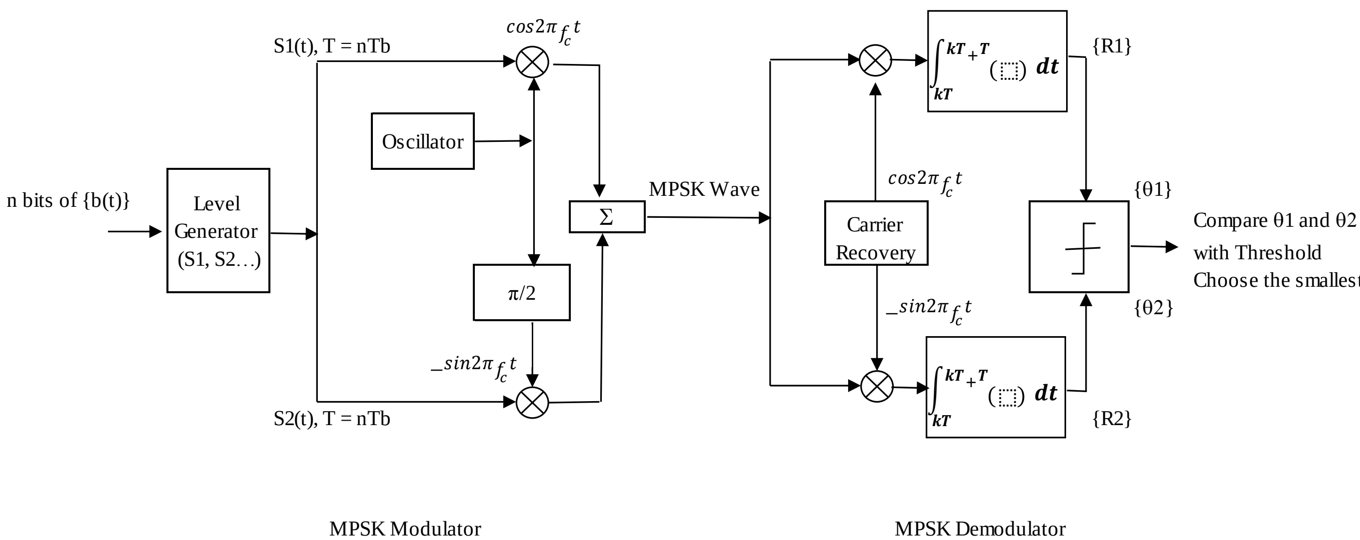


Figure: MPSK Modulator and Demodulator

Similarly for 8PSK: M = 8, n = $log_2M = log_2 8 = 3$ (i.e. n = 000, 001, 010, 011, 100, 101, 110, 111/tribit)

$$s(t)|_{BPSK} = A_c Cos(2\pi f_c t + 2\pi i/M) = A_c Cos(2\pi f_c t + 2\pi i/8) = A_c Cos(2\pi f_c t + \pi i/4)\ ;\ \text{for i = 0 to 7}$$

* Constant Envelope/Non-Linear Modulation Techniques

- Many practical mobile radio communication systems use these modulation methods, where the amplitude of the carrier signal is constant, regardless of the variation in the modulating signal.
- The constant envelope family of modulation techniques offers the following advantages
    - Power efficient ***Class-C*** amplifiers can be used
    - Low out-of band radiation of the order of -60dB to -70dB can be achieved
    - Limiter-Discriminator Detection can be used which simplifies receiver design and provides high immunity against random FM noise and signal fluctuations.
- However, its main disadvantage is that it occupies larger bandwidth than linear modulation techniques. So, it may not suitable in those situations where bandwidth efficiency is more important than power efficiency.

Minimum Shift Keying (MSK)

- ✓ **MSK** signifies a minimum frequency separation with continuous phase but changing frequency.
- ✓ In MSK modulation scheme instead of rectangular pulse wave; half cycle sine wave is used. Frequency of logical zero and logical one differs by half of data rate. For this, baseband modulation (which starts with a bit stream of 0's and 1's with a bit clock) is converted first into baseband signal using NRZ filter. Then FM is applied to produce MSK (ISI may be introduced).
- ✓ The expression for MSK modulation is outlined below.

$$\boldsymbol{S(t) = A.I(t)\cos\left(\frac{\pi t}{2T}\right)\cos(2\pi f_c t) + A.Q(t)\sin\left(\frac{\pi t}{2T}\right)\sin(2\pi f_c t)}$$

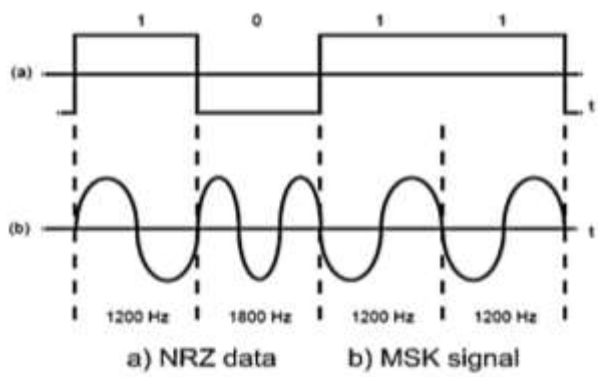


Figure: MSK Modulation in Time Domain

- ✓ It is a special type of ***Continuous Phase Frequency Shift Keying*** (CPFSK) wherein the peak frequency deviation is equal to ¼ th the bit rate; i.e. MSK is continuous phase FSK with a modulation index of 0.5.

- ✓ The modulation index of FSK is given by: ***(h) = 2Δf/R$_b$,*** where, *Δf* is the peak frequency deviation ($f_H - f_C$) and R_b is the bit rate *(=1/T_b)*. Hence for h = 0.5, $\Delta f = 0.25R_b \rightarrow T_b/4$

- ✓ Modulation index of 0.5 corresponds to the minimum frequency spacing that allows two FSK signals to be coherently orthogonal and the name MSK implies minimum frequency separation that allows orthogonal detection.

Gaussian Minimum Shift Keying (GMSK)

- GMSK is a form of continuous phase FSK in which the phase is changed between symbols to provide a constant envelope. Consequently, it is a popular alternative to QPSK. It is used in GSM/CDPD (Cellular Digital Packet Data) technologies.
- GMSK is advanced version of MSK. It is developed to improve the spectrum property of MSK by using a pre-modulation Gaussian filter. The transfer function of Gaussian filter is:

$$H(f) = \exp\{-\alpha^2 f^2\} = \exp\left\{-\left(\frac{f}{B}\right)^2 \frac{ln2}{2}\right\}, where\ \alpha = \frac{\sqrt{ln2}}{\sqrt{2}B} = \frac{0.5887}{B}$$

Where, B is the 3 dB bandwidth. The GMSK filter is completely defined from B and baseband symbol duration T (= 1/f). Therefore, GMSK is defined by it BT product.

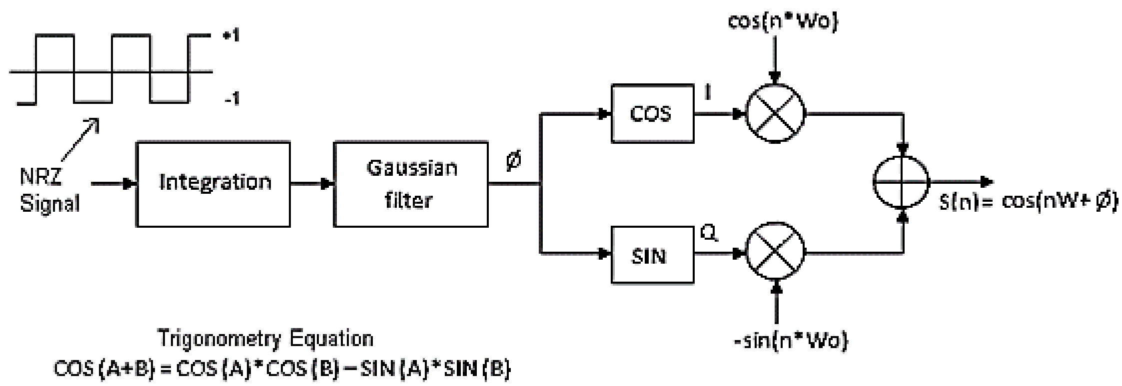


Figure: GMSK Modulation

- As shown in the GMSK modulator Gaussian filter is applied to NRZ signal after it is passed through integrator block. This gives Φ.
- By applying ***cos( )*** and ***sin( )*** function to this Φ gives out I and Q components which bits with ***cos( )*** and ***sin( )*** function respectively using mixing function.
- Both the chains are summed up and will give out ***S(n)***. The same process is understood well by trigonometric equation mentioned.

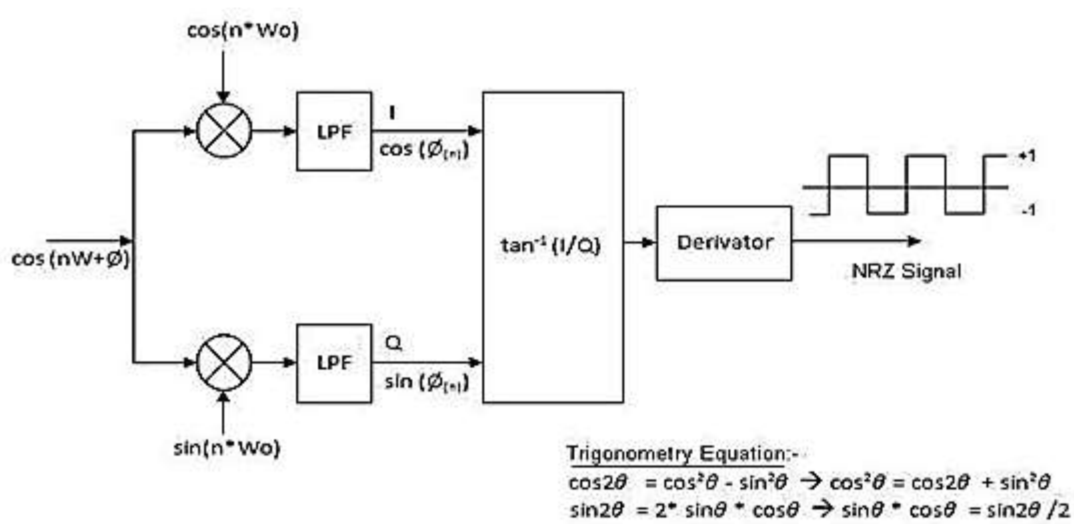


Figure: GMSK Demodulator

- GMSK demodulator basically derives back Φ using arc-tan function, which is applied to derivator block to obtain NRZ signal back.
- Before doing this mixing a low pass filtering is done to obtain I and Q components from two chains. The same is understood well by trigonometric equations mentioned.
- Bandwidth efficiency for MSK and GMSK is 1 bit/Hz also known as spectral efficiency. Figure below depicts MSK, GMSK waveforms.
- GMSK's power spectrum drops much quicker than MSK's. Furthermore, as BT product is decreased, the roll-off is much quicker.

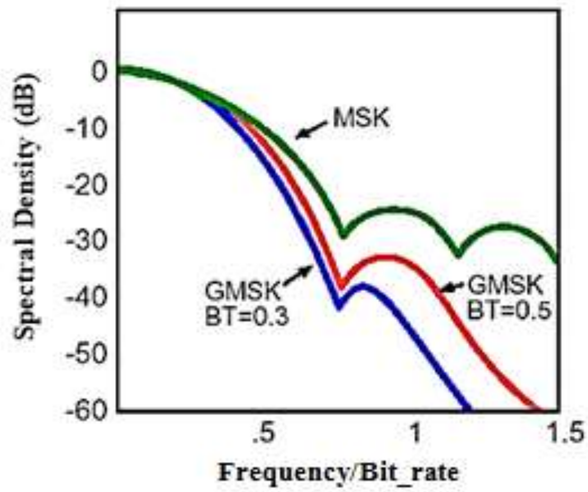


Figure: MSK/GMSK Spectral Density

- With lower time-bandwidth products, the pulse is spread over a longer time, which can cause Inter-Symbol-Interference.

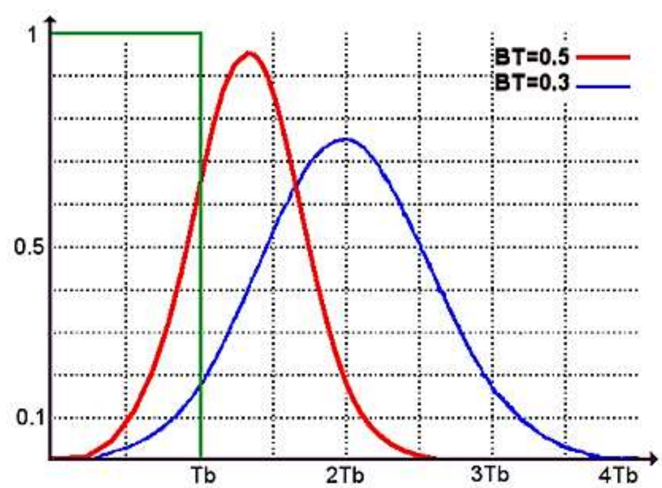


Figure: Spectrum Vs BT Product

- Therefore, as a compromise between spectral efficiency and time-domain performance, and intermediate time-bandwidth (BT) product must be choosen.

Chapter 6 – Equalization & Diversity Techniques

6.1 Introduction/Background Concept

- Equalization and diversity techniques are used independently or in a tandem to improve received signal quality and radio-link performance (i.e. to minimize the instantaneous bit error rate).

- ISI (Inter-Symbol-Interference) problem in digital data transmission.

    - When rectangular pulse (digital messages) are passed through a band limited channel, dispersion in the channel causes an overlap in time between successive symbols, which is referred to as ISI.
    - In another words: when modulation bandwidth exceeds the coherence bandwidth of the radio channel, ISI occurs and modulation pulses are spread in time into adjacent symbols causing bit error at the receiver.
    - ISI causes an increased probability of the receiver making error while detecting a symbol
    - Unless corrective measures are taken, ISI can pose a serious problem to the quality of reception and may impose a limit on the attainable data rate, which is far below the physical capability of the channel.

6.2 Fundamental of Equalization

- Equalization is a signal processing or filtering operation within the receiver.
- Equalization is used for compensating ISI, which compensate for the average range of expected channel amplitude and delay characteristics.
- Equalizer must be adaptive as the channel is generally unknown and time varying.

Adaptive Equalizer

- An adaptive equalizer is a time varying filter which is constantly tuned i.e. the weights are updated continuously by the adaptive algorithm either by sample basis or block by block basis.

- The adaptive algorithm is controlled by error signal, *e(t)*, derived by comparing the output of equalizer, *d(t)*, with some signal *d^(t)*, which is either exact scaled replica of the transmitted signal *x(t)* or which represents a known property of transmitted signal as shown in figure below:

  Working Principle:

  $y(t) - x(t) * f(t) + n_b(t)$
  $d(t) = x(t) * f(t) * h_{eq}(t) + n_b(t) * h_{eq}(t)$
  $d(t) = x(t) * g(t) + n_b(t) * h_{eq}(t)$

  If, $n_b(t)=0$; $d(t) = x(t)$, such that $g(t) = f(t) * h_{eq}(t)=\delta(t)$

  Which implies that the equalizer is an inverse filter of the channel.

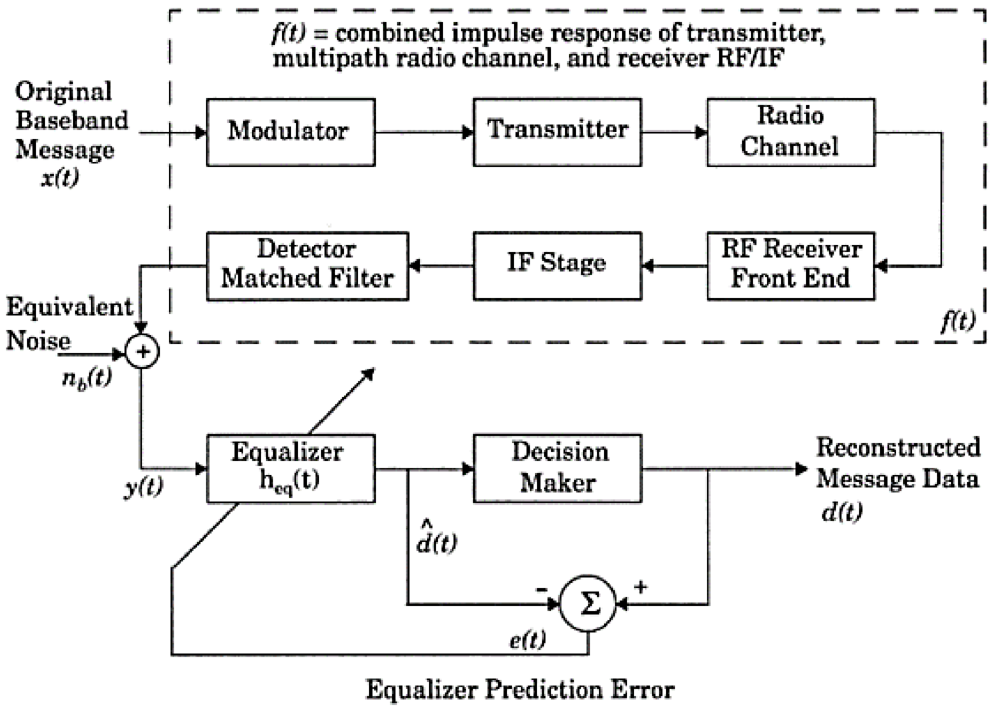


Figure: Block Diagram of Adaptive Equalizer

Operating Modes of Adaptive Equalizer

1. **Training Mode**

- Initially a known fixed length training sequence is sent by the transmitter, so that the receiver's equalizer adapt to a proper setting for minimum bit error rate (BER) detection.
- Training sequence is pseudo-random sequence or a fixed prescribed bit pattern.
- User data is sent immediately following the training sequence.
- The adaptive equalizer utilizes algorithm to estimate the filer coefficients (maximum delay, deepest fades, maximum ISI etc.) near the optimum value to compensate for the distortion.

2. **Tracking Mode**

- When user data is received, the adaptive algorithm tracks the changes in the channel.
- As a result, adaptive equalizer continuously changes the filter characteristics over time i.e., weight of the filter changes over time.
- When an equalizer is properly trained, it is said to be converged, equalizers are widely used in TDMA system.

Classification of Equalization Techniques

- Equalizers are classified based on how the output of an adaptive equalizer is used for subsequent control of the equalizer.
- In general, the decision maker determines the value of the digital data bit being received and applies thresholding operation.

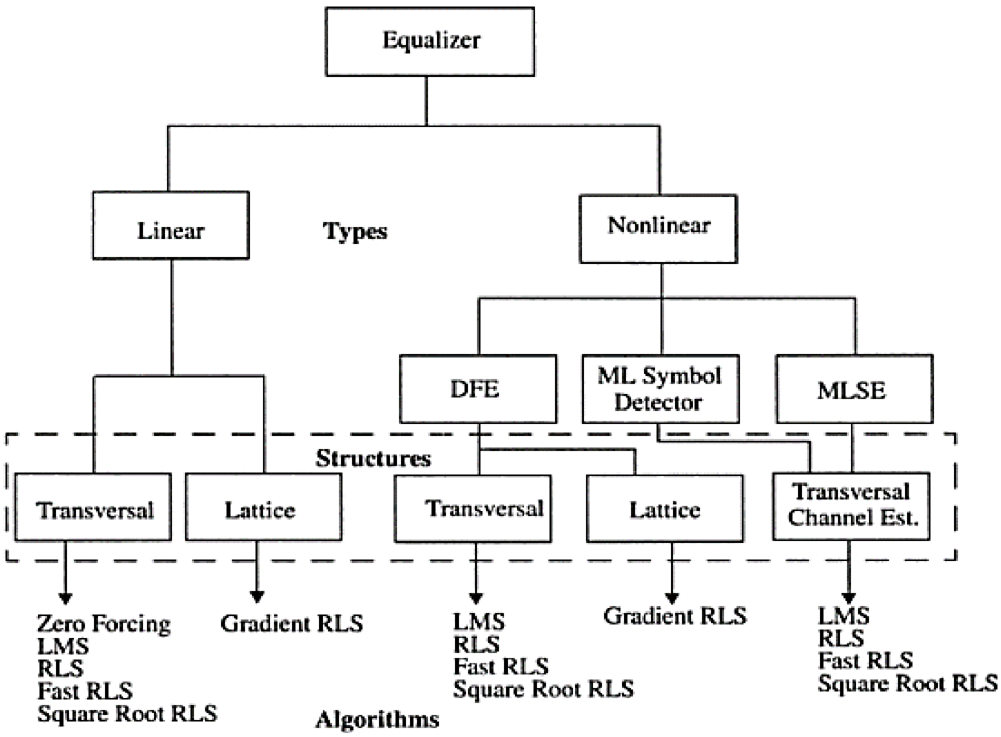


Figure: Classification of Equalizers

1. Linear Equalizers

- In linear equalization, the output signal *d(t)* is not used in feedback path to adapt the equalizer.
- Linear equalizer can be implemented as FIR filter.

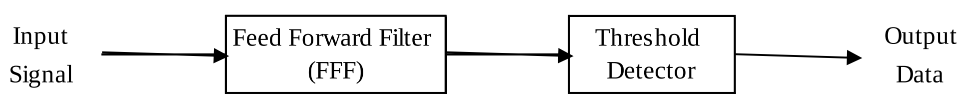


Figure: Linear Equalizer Structure

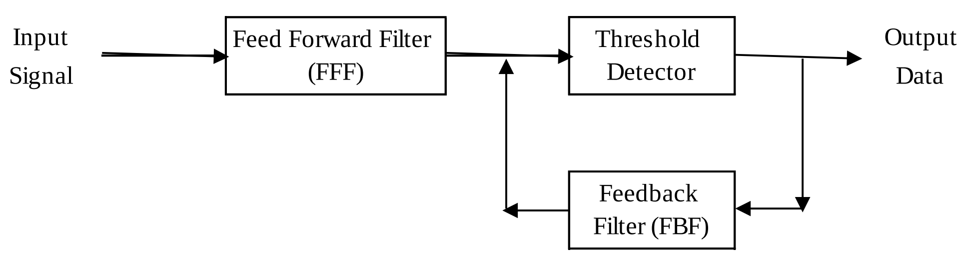


Figure: Non-Linear Equalizer Structure

2. Non-Linear Equalizers

- In non-linear equalization, the output signal *d(t)* is used as feedback to change the subsequent output of the equalizer.
- Non-linear equalizers are used in applications, where channel distortion is too severe for linear equalizer to handle.

6.3 Diversity Techniques

- A diversity scheme is a method that is used to develop information from several signals transmitted over independent paths.
- It is used to compensate for fading channel impairments and is usually implemented by using two or more receiving antennas.
- Diversity exploits the random nature of radio propagation by finding independent (uncorrelated) signal paths for communication.
- Diversity is employed to reduce depth and duration of the fades experienced by the receiver in a flat fading (narrow band) channel.
- No training and tracking is required.
- Diversity provides significant link improvement with little added cost. Diversity decisions are made by the receiver and are unknown to the transmitter.
- If one radio path undergoes an deep fade another independent path may have strong signal and by having more than one path to select from, both instantaneous and average SNR, at the receiver may be improved often by as much as 20 dB to 30 dB.

General Types of Diversity

1. **Macroscopic Diversity**

   - Prevents large scale fading.
   - Large scale fading is caused by shadowing due to variation in both the terrain profile and the nature of surroundings.
   - Large scale fading is normally distributed.
   - This fading is prevented by an antenna which is not shadowed when others are, this allows increase in SNR.

2. **Microscopic Diversity**

   - Prevents small scale fading.
   - Small scale fading is caused by multiple reflections from the surroundings. It is characterized by deep and rapid amplitude fluctuation which occur as the mobile moves over distances of a few wavelengths.
   - This fading is prevented by selecting an antenna which gives a strong signal that mitigates small scale fading.

Commonly used Diversity Techniques

1. **Space Diversity**

   - Also called Antenna-Diversity
   - Signals received from spatially separated antennas on the mobile would have essentially uncorrelated envelopes for antenna separations of one-half wavelengths or more.

   "*A method of transmission or reception or both, in which the effects of fading are minimized by the simultaneous use of two or more physically separated antennas, ideally separated by one half or more wavelength*".

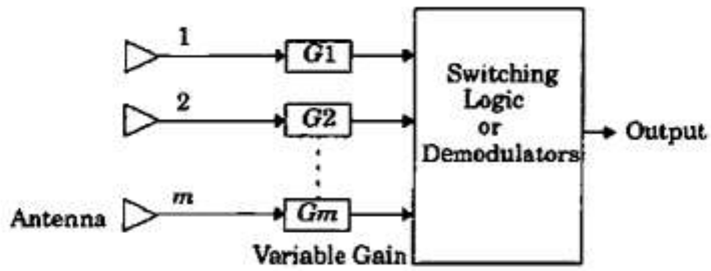


Figure: Block diagram of Space Diversity

2. **Polarization Diversity**

- At the base station, space diversity is considerably less practical than the mobile because the narrow angle of incident field require large antenna spacing.
- In such situation, polarization diversity is preferred.
- Horizontal and vertical polarization or circular and liner polarization provides uncorrelated signals at the receiver.
- It does not require antenna spacing.
- Dramatically reduce multipath delay spread.
- Require antenna orientation to achieve orthogonal polarization to be used.

3. **Frequency Diversity**

- Transmit information on more than one carrier frequency.
- Separation between the carriers should be at least the coherent bandwidth (BW = df = B_c).
- Uncorrelated frequency suffer different level of fading in frequency selective channels i.e. different copies undergo independent fading.
- Only one antenna is needed.

4. **Time Diversity**

- Transmit the desired signal repeatedly in M-different periods of time (each symbols is transmitted M-times), which exceeds the coherence time of the channel.
- The interval between transmission of same symbol should be at least the coherence time (dt = T_C)
- Different copies undergo independent fading.
- Reduction in efficiency (effective data rat < real data rate).
- RAKE receiver for spread spectrum CDMA is based on time diversity.

Space Diversity Reception Methods

1. **Selection Diversity**

- It offers an average improvements in the link margin without requiring additional transmitter power or sophisticated receiver circuitry.
- M-demodulators are used to provide M-diversity branches.
- The receiver branches having the highest instantaneous SNR is connected to the demodulator.

2. Feedback/Scanning Diversity

- Scanning all the signals in a fixed sequence until one with SNR more than a predetermined threshold is identified.

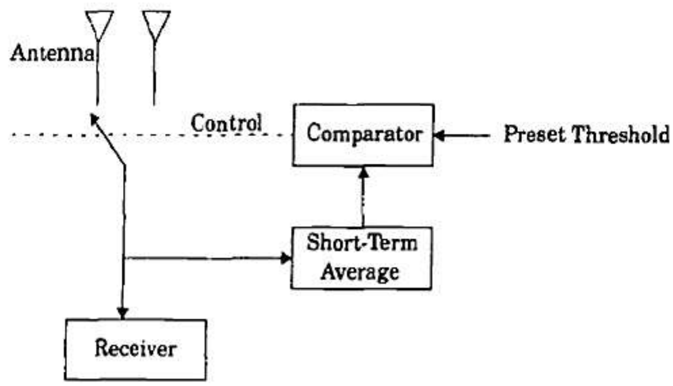


Figure: Feedback/Scanning Diversity

- Signal received continuously until it falls below threshold from the branch having signal above predetermined threshold.
- Require only one receiver.

3. Maximal Ratio Combining

- Signals from all of the M-branches are weighted according to the signal voltage to noise power ratios and then summed.
- Produces output – SNR equal to the sum of the individual SNRs.
- Gives best statistical reduction of fading of any known linear diversity combiner.

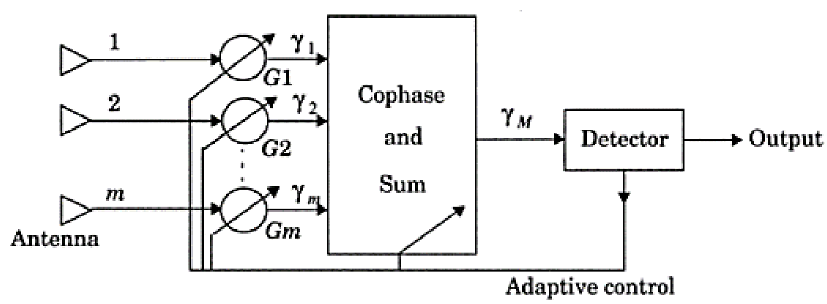


Figure: Maximal Ratio Combiner

4. Equal Gain Combining

- Instead of variable weighting, each branch weight is set to unity.
- So combining all the signals in a co-phased manner with unity weights for all signal levels so as to have the highest achievable SNR at the receiver at all times.
- The performance is marginally inferior to maximal ratio combining and superior to selection diversity.

RAKE Receiver

- Since, multipath components of the signal carry important information. They are time delayed versions of the original signal.
- RAKE receiver collects the time shifted versions of the original signal by providing a separate correlation receiver for each of the multipath signals.

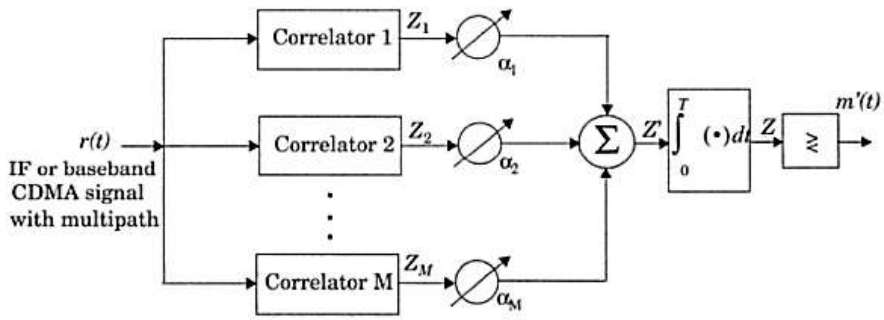


Figure: RAKE Receiver

Where, $Z' = \sum_{m=1}^{M} \alpha_m Z_m$ and $\alpha_m = \frac{{Z_m}^2}{\sum_{m=1}^{M} {Z_m}^2}$

Definition

- *A receiver technique which uses several baseband correlators to individually process several multipath components. The correlators outputs are combined to achieve improve communication reliability and performance.*
-
- RAKE receivers are mostly/widely used in CDMA, spread spectrum system. In CDMA, the chip-rate is typically much greater that the flat-fading bandwidth of channel. CDAM spreading codes are designed to provide very low correlation between successive chips and thus if the multipath versions of the transmitted signals are delayed in time by more than one chip duration, they appear like uncorrelated noise at receiver, an equalization is not required.
- RAKE receiver in CDMA, combine delayed version of the signals to improve the received SNR –collect the time shifted version of the original signal by providing a separate correlation receiver for each signals.
- RAKE receiver uses multiple correlators to separately detect M strongest multipath components. The output of each correlators are then weighted to provide a better estimate of a transmitted signal than is provided by the signal component
- Demodulation and bit decisions are then based on the weighted outputs of the M – correlators.
- Weighting coefficients are based on the power or the SNR from each correlators output.

Interleaving

- Used to obtain time diversity in a digital communication system without adding any overheads.
- Extremely useful in 2G and 3G wireless system due to rapid proliferation of digital speech coder, which transform analog voices into efficient digital messages that are transmitted over wireless links.
- The function of the interleaver is to spread the source bits out in time, so that if there is a deep fade or noise burst, the important bits from the block of source data are not corrupted at the same time.
- By spreading the information bit over time, it becomes possible to make use of error control coding (channel coding) which protects the source date against the channel error. Interleaver are of two types:

a) Block Interleaver

- Which formats the encoded data into a rectangular array of size 'm x m', and interleave 'nm' bits at a time.
- Usually each row contains a word of source data having 'n' bits.

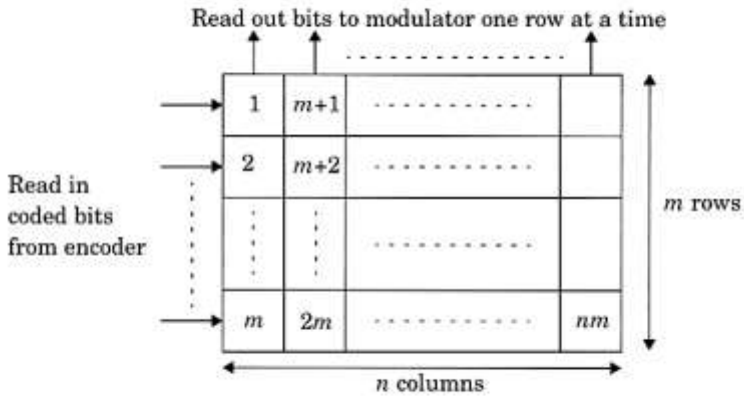


Figure: Block Interleaver

- Source bit are placed into the interleaver by sequentially increasing the row number for each successive bits and filling the columns.
- The source data is then read out row-wise and transmitted over the channel .This has the effect of the spreading the original source bit by 'm' bit period. The effect is that in any case of burst error even if a block of data is lost, this has an effect of loss of single bit in source data.

b) Convolution Coder

- Convolution codes are fundamentally different from block codes in that the information sequence are not grouped into distinct blocks and encoded, instead a continuous sequence of information bits are mapped onto a continuous sequence of encoder output bits.
- A convolution code is generated by passing the information sequence through a finite state shift registers.
- In general, the shift registers contain 'N' k-bit stages and 'm' linear algebraic function generator based on the generator polynomials as shown below.

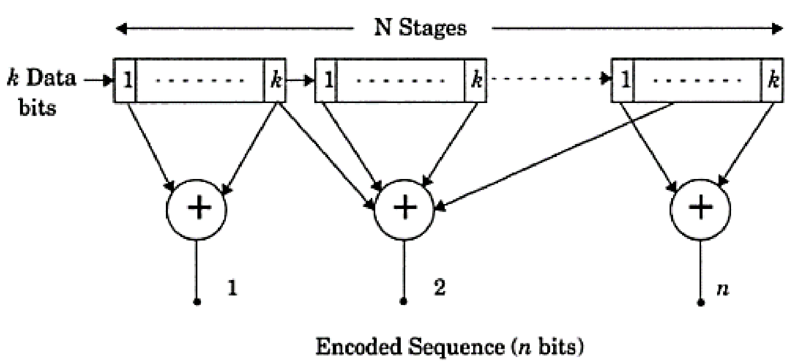
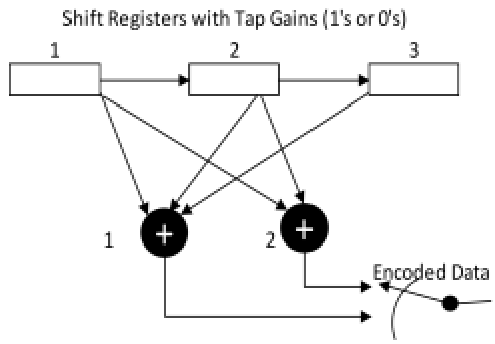


Figure: General Block Diagram of Convolution Encoder

- In convolution codes, the message bits and check bits are continuously interleaved and the code words for i^{th}-memory bit depends not only upon the i^{th}-input but also upon $(N-1)^{th}$ bit, where N is the number of shift registers.

Example: Let a convolution code with no. of shift register (N) = 3 and no. of modulo-2 adder (y) = 2 is taken as an example with following configuration.



For a message word: {11010}, the encoded output is produced as:

{11 00 01 01 11}.

Hence, for k-bits data word, it generates k times y number of bits in code word.

Chapter 7 – Multiple Access Techniques

7.1 Overview of Duplexing Techniques

Duplexing refers to the way downlink and uplink data is arranged in a two-way wireless transmission. The downlink carries information from a Base Station (BS) to Subscriber Stations (SSs, which could be a handset or a PC). Downlink is also known as forward link. The uplink carries information from a SS to a BS. It is also called reverse link. There are two types of duplexing schemes, FDD and TDD.

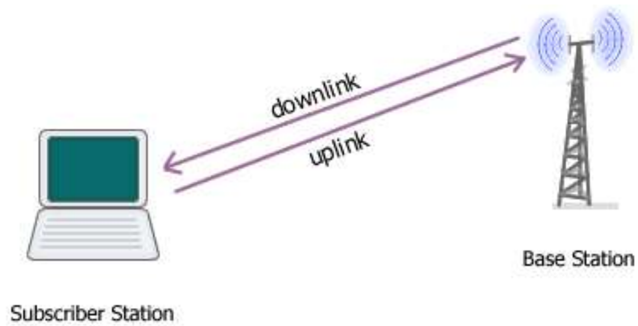


Figure: Duplexing Techniques

FDD (Frequency Division Duplex)

- It requires two distinct frequency channels for transmitting downlink sub-frame and uplink sub-frame at the same time slot (meaning simultaneously).
- FDD is suitable for bi-directional voice service since it occupies a symmetric downlink and uplink channel pair. Voice produces equal amount of data in both uplink and downlink directions constantly and hence referred to as symmetric service. FDD is commonly used in mobile networks (e.g. 2G and 3G).

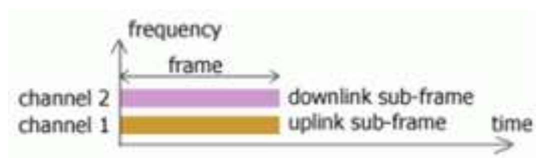


Figure: Frequency Division Duplexing (FDD)

- FDD is inefficient for handling asymmetric data services (browsing, downloads) because traffic is generated in one direction (up or downlink) and other link remains reserved but unused, and data traffic may only occupy a small portion of a channel bandwidth at any given time

TDD (Time Division Duplex)

- It requires only one channel for transmitting downlink and uplink sub-frames at two distinct time slots (at different times). TDD therefore has higher spectral (radio frequency) efficiency than FDD.
- Same single frequency is used to transmit & receive data in turns (doing only one thing at one time).
- Using TDD, downlink to uplink (DL/UL) ratio can be adjusted dynamically based on demand. TDD can flexibly handle both symmetric and asymmetric broadband traffic.

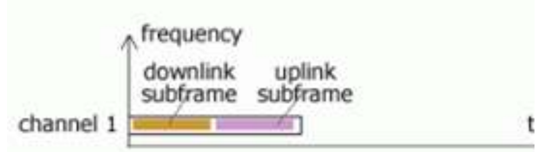


Figure: Time Division Duplexing (TDD)

7.2 Multiple Access Techniques

- Multiple access techniques are used to allow a large number of subscriber to share the allocated spectrum in the most efficient manner.
- As the spectrum is limited, sharing is required to increase the capacity by allowing the available bandwidth to be used at the same time by different users.
- Sharing must be done in a way such that the quality of service does not degrade within the existing users.

Multiple Access Protocols (MAC-Protocols)

- Single shared communication channel, i.e. simultaneous use of a communication system by more than one users at a time.
- Each user's signal must be kept uniquely distinguishable from other user's signals, to allow private communication on demand.
- Use distributed algorithm which determines how stations share channel
- Users can be separated in many ways: physically on separate wires or by arbitrarily defined 'channels' established in frequency, time or many other variable imaginable.
- Ideally, when one nodes wants to transmit - it can send at rate 'R' bps, when 'M' nodes want to transmit, each can be send at average rate of 'R/M' bps.

MAC Classification

Three broad classification of multiple access techniques.

a. **Channel Partitioning Techniques**
   - Divides channel into smaller "pieces" (time slots, frequency bands etc.)
   - Allocate each piece to nodes for exclusive use.
b. **Random Access Techniques**
   - Allow collisions and recover from collision simultaneously (ALOHA, CSMA).
c. **Taking Turns Techniques**
   - Tightly coordinate shared access to avoid collisions.

7.3 Frequency Division Multiple Access (FDMA)

Initial multiple-access technique, in which each individual user is assigned a pair of frequencies while making or receiving a call. One frequency is used for downlink and one pair for uplink. This is called frequency division duplexing (FDD).

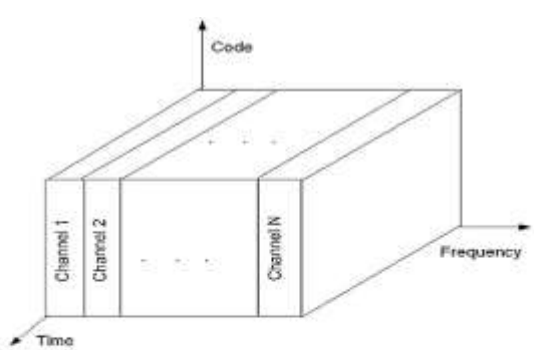


Figure: The basic concept of FDMA.

Features

- Allocated frequency pair is not used in the same cell or adjacent cells (in wireless communication) during the call so as to reduce the co channel interference.
- Different users can use the same frequency in the same cell except that they must transmit at different times. If an FDMA channel is not in use, then it sits idle and it cannot be used by other users to increase share capacity.
- FDMA requires tight filtering to minimize the adjacent channel interference.

Advantages	Disadvantages
- Easy to realize - Less Inter-Symbol-Interference, equalizer not needed. - Network synchronization not needed. - Easy bit time recovery and frame synchronization. - Voice coder not needed.	- Guard band is needed to reduce interference between frequencies. - Encryption is difficulty. - Low efficiency for non-voice data communication. - Restricted capacity due to the low spectrum utilization: frequency reusing factor is 7.

7.4 Time Division Multiple Access (TDMA)

Entire bandwidth is available to the user but only for a finite period of time. Available bandwidth is divided into fewer channels compared to FDMA and the users are allotted time slots during which they have the entire channel bandwidth at their disposal, as shown in figure below. TDMA uses different time slots for transmission and reception. This is referred as Time Division Duplexing (TDD).

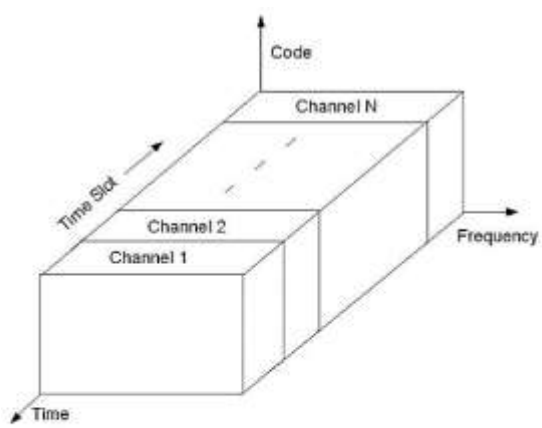


Figure: The basic concept of TDMA.

Features:

- TDMA requires careful time synchronization since users share the bandwidth in the frequency domain. Since, the number of channels are less, inter channel interference is almost negligible.
- TDMA shares a single carrier frequency with several users where each users makes use of non-overlapping time slots. Since it uses different time slots for transmission and reception thus duplexers are not required.
- It is possible to allocate different numbers of time slots per frame to different users. Thus bandwidth can be supplied on demand to different users by concatenating or reassigning time slot based on priority.

- Each user occupies a specific frequency but only during an assigned time slot. The frequency is used by other user during other time slots.
- The medium is accessed and information is transmitted at different time slots. Receiver can stay on same frequency just to listen on a different time slot.
- TDMA is more flexible and dynamic.

Advantages	Disadvantages
- Frequency sharing among N users by time scheduling. - Variable bit rate by changing slots. - Less stringent power control due to reduced inter-user interference: dedicated frequencies and slots. - Mobile assisted/controlled handoff enable by available measurement slots.	- Pulsating power envelop – Interference with devices link hearing aids have been detected. - Complexity inherent in slot/frequency allocation. - High data rate imply need for equalization to overcome Inter-Symbol-Interference. - Large overhead and complex hardware. - Restricted capacity by frequency band and time slot.

7.4 Code Division Multiple Access (CDMA)

Same bandwidth is occupied by all the users, however they are all assigned separate codes, which differentiates them from each other as shown below. Code set partitioning is used mostly in wireless broadcast channels (cellular, satellite etc.) CDMA utilize a spread spectrum technique in which a spreading signal is used to spread the narrow band message signal. All users share same frequency, but each user has own "chipping" sequence (i.e. code) to encode data. It allows multiple users to "coexist" and transmit simultaneously with minimal interference (if coders are "orthogonal").

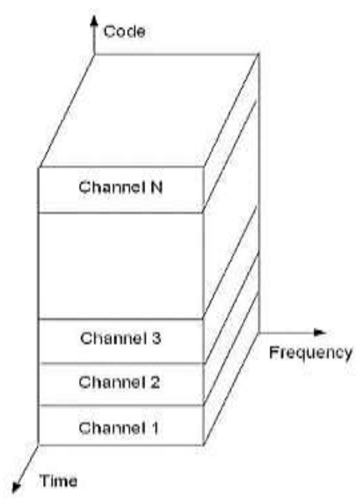


Figure: The basic concept of CDMA.

What is Spread Spectrum (SS) Techniques?

A transmission technique in which a pseudo-noise code, independent of the information data, is employed as a modulation waveform to "spread" the signal energy over a bandwidth much greater than the signal information bandwidth. At the receiver the signal is "de-spread" using a synchronized replica of the pseudo-noise code.

1. Direct Sequence Spread Spectrum

- A pseudo-noise sequence or PN code generated at the modulator is used in conjunction with m-ary PSK modulation to shift the phase of the PSK signal pseudo randomly, at the chipping rate ($R_C = 1/T_C$): a rate that is an integer multiple of the symbol rate ($R_S = 1/T_S$).
- The transmitted bandwidth is determined by the chip rate and by the baseband filtering.
- The bits (message) are chipped or sampled at higher frequency.

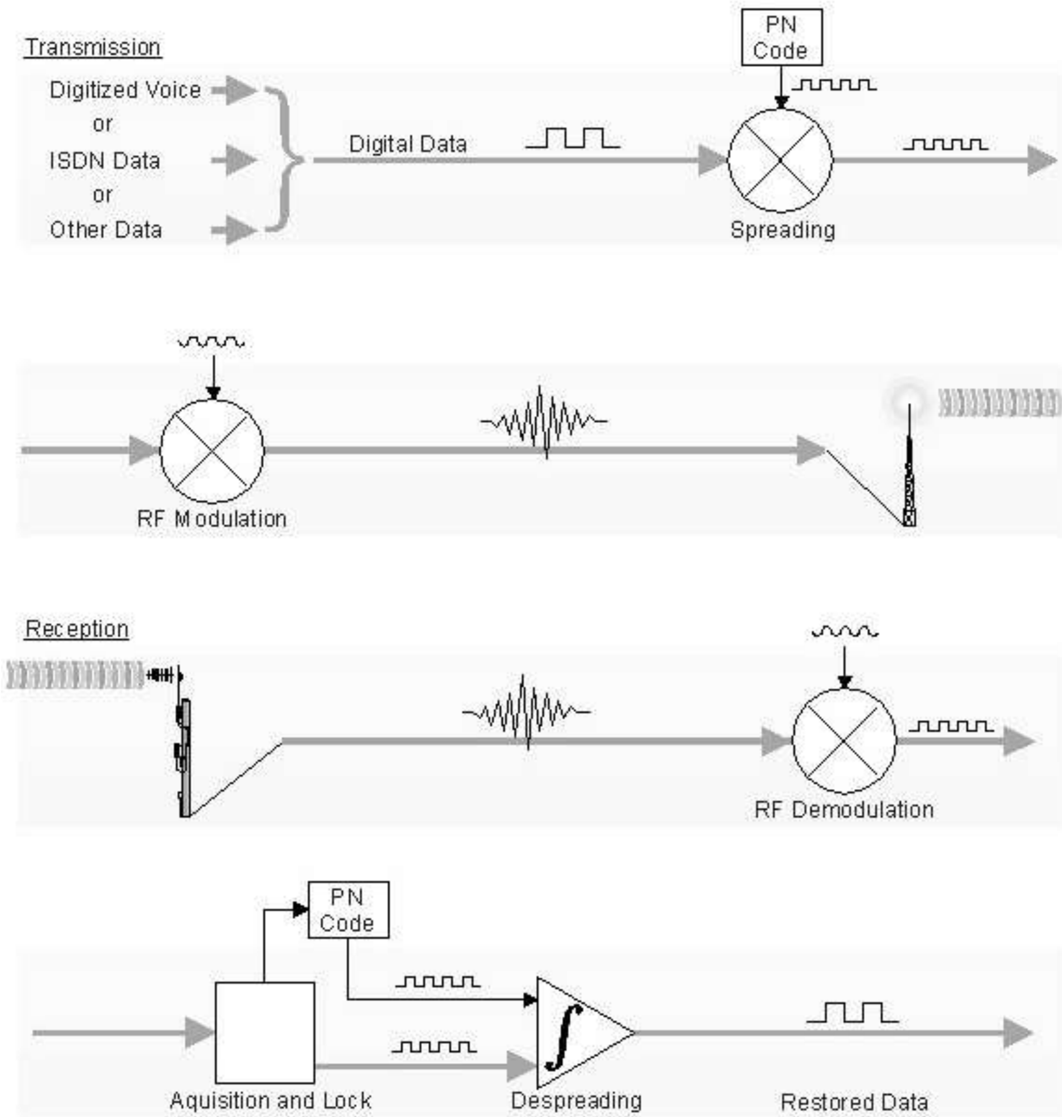


Figure: Direct Sequence Spread Spectrum Technique

Signal transmission consists of following steps:

1. A pseudo-random code is generated, different for each channel.
2. The Information data modulates the pseudo-random code.
3. The resulting signal modulates a carrier.
4. The modulated carrier is amplified and broadcast.

Signal reception consists of the following steps:

1. The carrier is received and amplified.
2. Received signal is mixed with a local carrier to recover the spread digital signal.
3. A pseudo-random code is generated, matching the anticipated signal.
4. The receiver acquires the received code and phase locks its own code to it.
5. The received signal is correlated with the generated code, extracting the Information data.

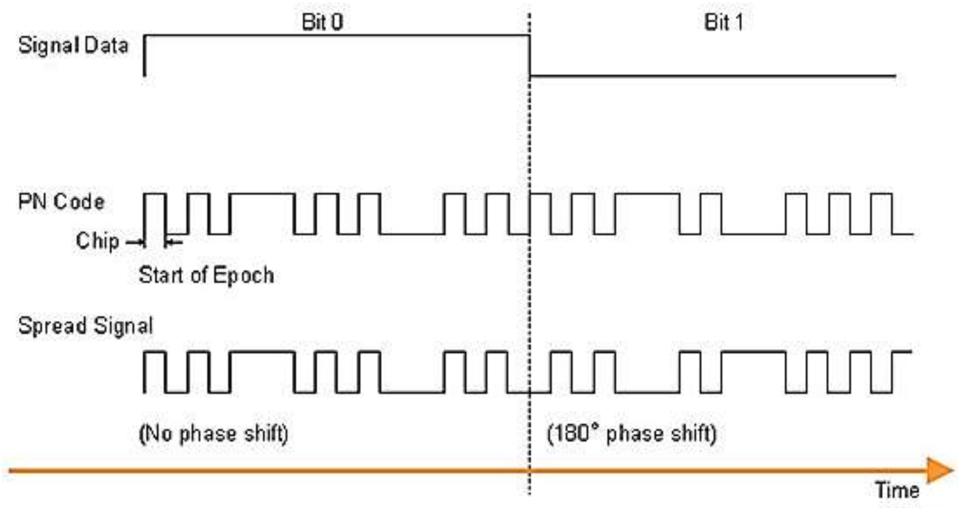


Figure (a): Pseudo Noise Spreading of Message Bits

The Information data modulates the PN code, as in Figure (a). Some terminology related to the PN Code are:

- Chipping Rate (R_c): the bit rate of the PN code.
- Information Rate (R_s): the bit rate of the digital data.
- Chip: One bit of the PN code.
- Epoch: The length of time before the code starts repeating itself (code period).

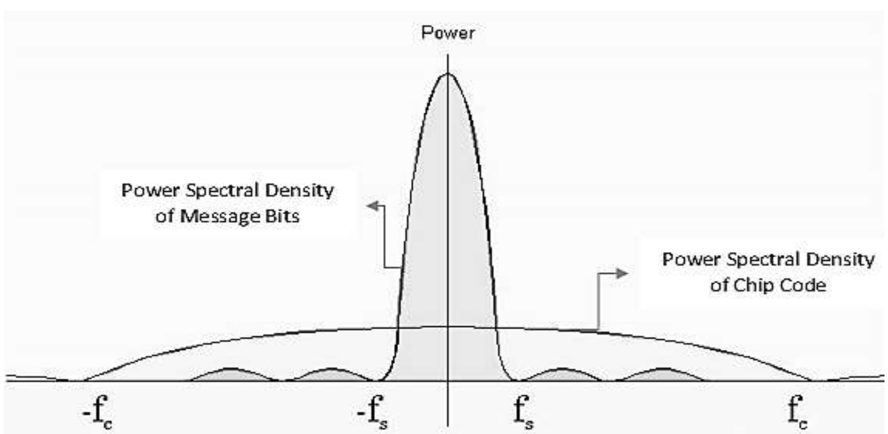


Figure (b): Frequency Spreading

Figure (b) shows the process of frequency spreading. In general, the bandwidth of a digital signal is twice its bit rate. The bandwidths of the information data (f_i) and the PN code are shown together. The bandwidth of the combination of the two, for $f_c > f_i$, can be approximated by the bandwidth of the PN code

- In case of BPSK modulation, the DS-SS signal is then: $S_\omega(t) = d(t)c(t)\cos(2\pi f_0 t)$
- At the receiver, the wideband signal is multiplied with c(t) once again, giving:

  $\bar{S_n}(t) = c(t)S_\omega(t)\cos(2\pi f_0 t) = d(t)\cos(2\pi f_0 t)$

  Which is the narrowband data-modulated carrier again. The time frequency occupancy of a DS-SS signal will be as shown alongside.

Figure: Time Frequency Occupancy of DS-SS Signal

2. Frequency Hopping SS Technique

- Another common method to spread the transmission spectrum of a data signal is to (pseudo) randomly hop the data signal over different carrier frequencies, called *Frequency Hopping Spread Spectrum (FH-SS).*
- Usually, the available band is divided into non-overlapping frequency called *bins.*
- The data signal occupies one and only one *bin* for a duration 'T_C' and hops to another *bin* afterward.

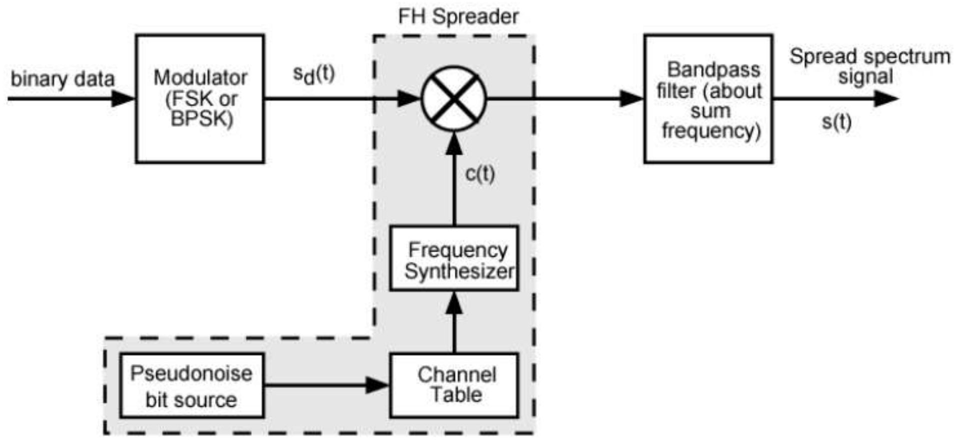


Figure (a): FH-SS Transmitter Section

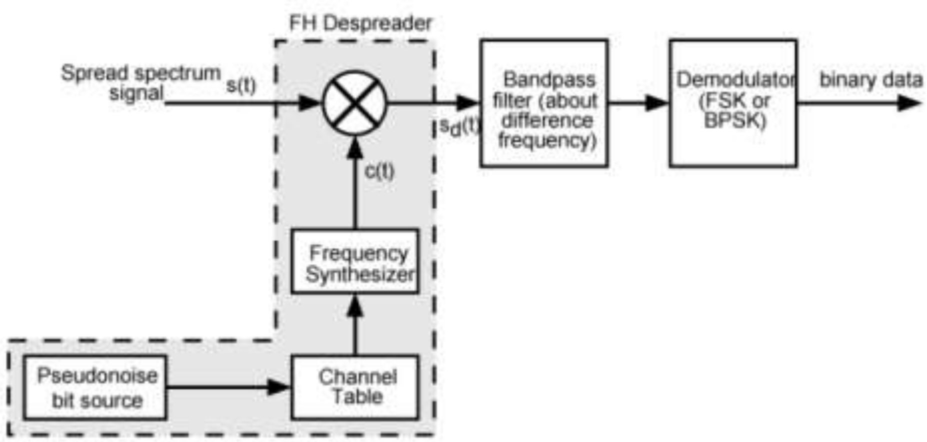


Figure (b): FH-SS Receiver Section

1. **Fast FH-SS**
   When the hopping rate is faster than the symbol rate, the FH scheme is referred to as fast hopping. Where, $\mathbf{R_C = nR_s}$ (i.e. $\mathbf{T_C < T_S}$) and Chip Rate, $R_{Ch} = \max(R_S, R_h)$. The carrier frequency changes (or hops) several times during the transmission of one symbol.

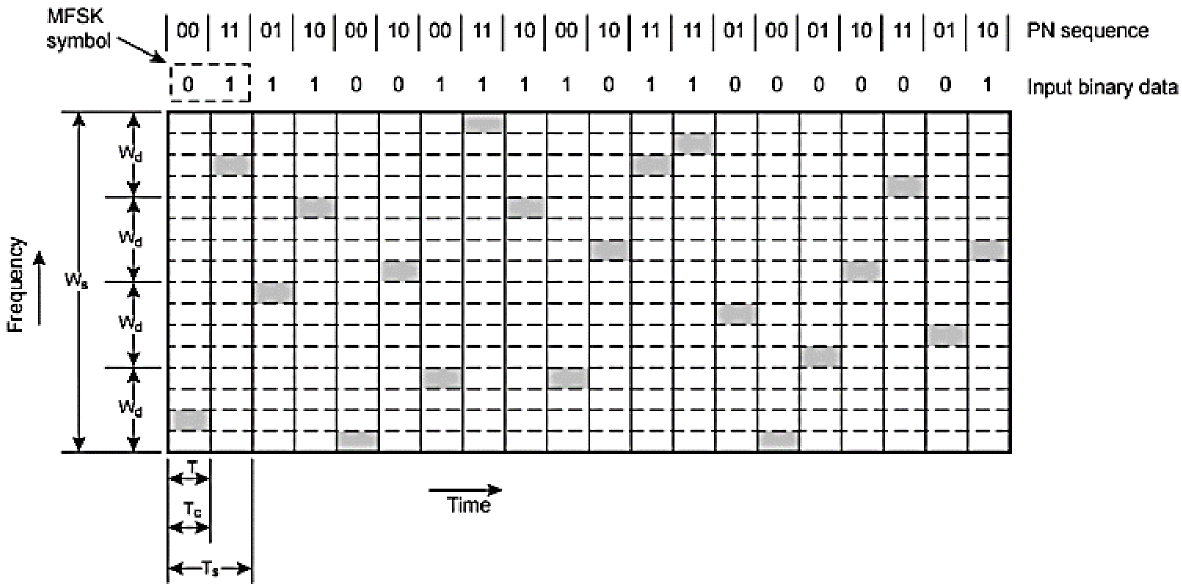
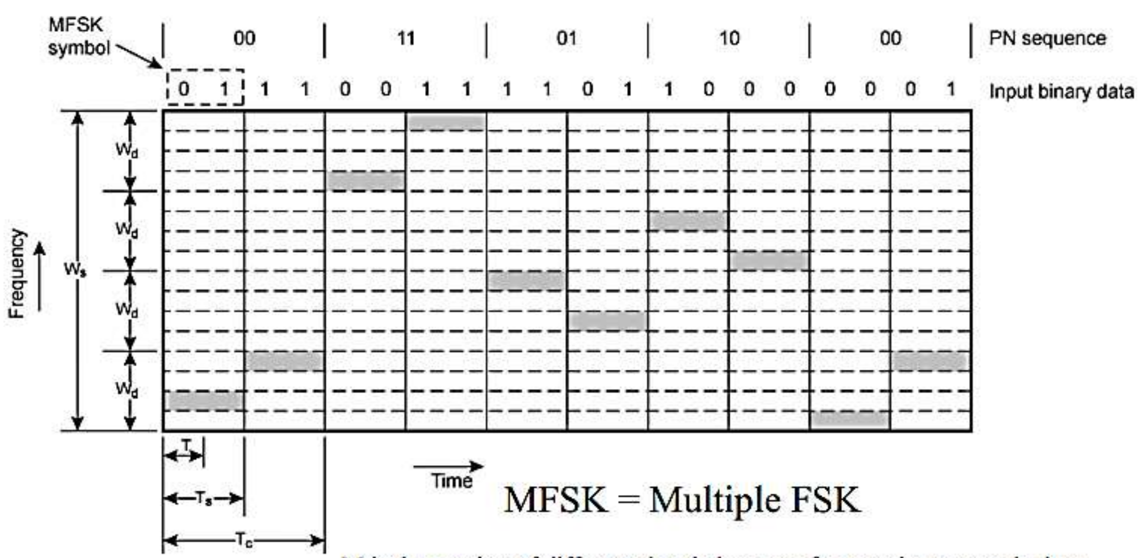


Figure: Fast Hopping Spread Spectrum

2. **Slow FH-SS**
   When the hopping rate is slower than the symbol rate, FH scheme is referred to as slow hopping. Where, $\mathbf{R_S = nR_C}$ (i.e. $\mathbf{T_C \geq T_S}$) and Chip Rate, $R_{Ch} = \max(R_S, R_h)$



MFSK = Multiple FSK

M is the number of different signal elements, frequencies to encode data,
2^k is the number of channels, each of width W_d

Figure: Slow Hopping Spread Spectrum

Code Division Multiple Access (CDMA): Elaboration

- CDMA is a multiplexing mechanism with spread spectrum techniques, which breaks each data bit (of data rate ‘D’ per second) into ‘k’ chips according to fixed pattern specific to each user known as user’s code producing the chip rate of ‘kD’ per second for new channel.
- In CDMA, each ‘n’ users use different orthogonal PN sequence, it modulates each user data stream using BPSK.

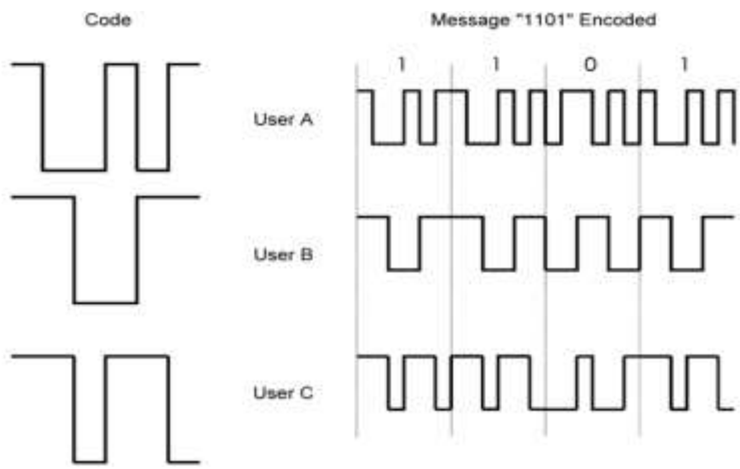


Figure: CDMA Example

- In above figure, three users (A, B & C) communication with receiver with following codes.
    - Code for A = {1, -1, -1, 1, -1, 1}
    - Code for B = {1, 1, -1, -1, 1, 1}
    - Code for C = {1, 1, -1, 1, 1, -1}
- If the communication is said to be already synchronized then, A wants to send a ‘1’ with chip pattern (A’s code): {1, -1, -1, 1, -1, 1} and {-1, 1, 1, -1, 1, -1} for a ‘0’ exactly the compliments of A’s code.
- Decoder ignores other sources when using A’s code to decode.

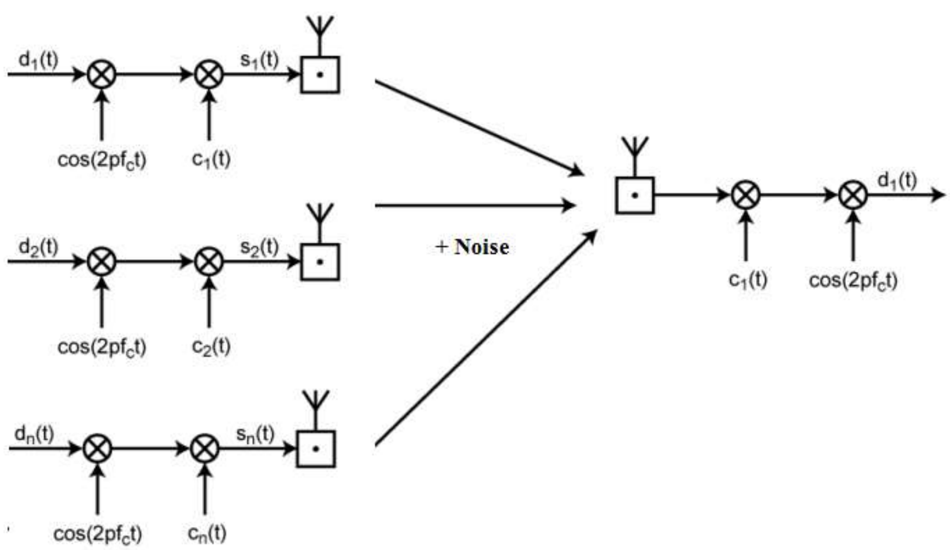


Figure: CDMA in DSSS Environment

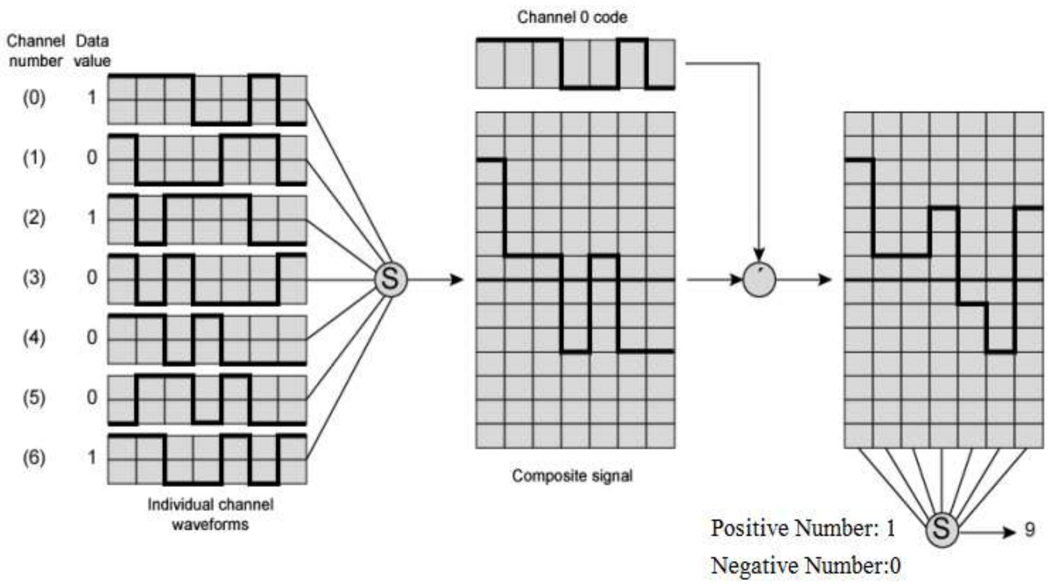


Figure: Seven Channel CDMA Coding and Decoding

Pseudo-Random Noise (PN) Code

- A PN code sequence acts as a noise like (but deterministic) carrier used for bandwidth spreading of the signal energy.
- The selection of a good code is thus important because the type and length of the code sets bounds on the system capability.
- The PN code sequence is Pseudo-Noise or Pseudo-Random-Sequence of 1's and 0's, but not a real random sequence (because it is periodic).

Classes of periodic PN sequence

- Maximal-length linear shift register sequence (or m-sequence): Length: $N = 2^m - 1$ (m = number of shift registers)
- Quadratic residue sequence: Length: N = 4k-1 = Prime Number (k is an integer)
- Hall sequence: Length: $N = 4k-1 = 4q^2+27$ = Prime Number (both k and q are integers)
- Twin primes: Length: N = p(p+2) (both p and p+2 are prime numbers)

Properties of the m-sequence

- Balance Property:
  Number of 1's = Number of 0's + 1 or
  Number of 0's = Number of 1's + 1
- Run Property:
  Run – A sequence of a single type binary digits.
  Length – The number of digits in the run.
  ½ the runs are of length 1.
  ¼th the runs are of length 2

  1/8th the runs are of length 3 … and so on

- Correlation Property:
  The correlation function of an m-sequence is periodic and binary valued (or two valued).

$$R_C(k) = \frac{1}{N}\sum_{n=1}^{N} c(n)c(n-k) = \begin{Bmatrix} 1 & if\ k = lN\ (l; Integer) \\ -\frac{1}{N} & if\ k \neq lN \end{Bmatrix}$$

 # PN Code Generation: An Example

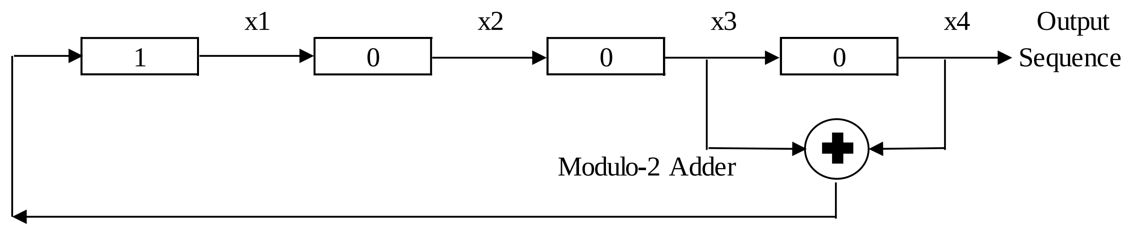


Figure: PN Sequence Generation (for m = 4)

Initial state and the succession of states:

- 1000→0100→0010→1001→1100→0110→1011→0101→1010→1101→1110→1111→0111→0011→0001→1000
- The output sequence, c(n): 000100110101111 (N = 2^4-1 = 15)
- Number of 1’s = 8 and number of 0’s = 7
- Eight runs: {000}, {1}, {00}, {11},{0},{1},{0},{1111}
- For c(n): -1-1-1+1-1-1+1+1-1+1-1+1+1+1+1
- $R_c(k) = \begin{cases} 1, if\ k = lN \\ -\frac{1}{15}, if\ k \neq lN\ (l: integer) \end{cases}$

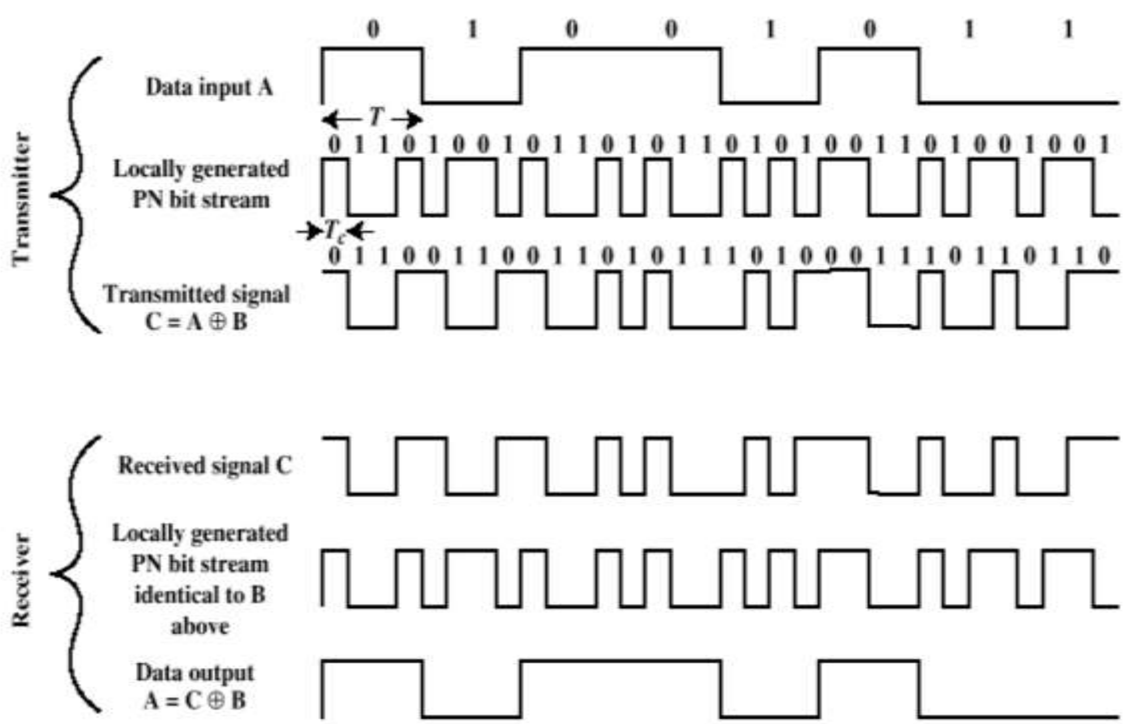


Figure: DSSS Transmitter and Receiver using PN Code

Processing Gain (PG)

- The main parameter in spread spectrum system is the 'processing gain' – the ration of transmission and information bandwidth.

- It is basically the "spreading factor", which determines the number of users that can be allowed in a system, the amount of multi-path effect reduction, the difficulty to jam or detect a signal etc.

- When the processing gain is large enough, the spread spectrum signal hides below the white noise level. Without knowledge of the signature sequences, an unintentional receiver cannot de-spread the received signal. Therefore, it is hard for the unintentional receiver to detect the presence of the spread spectrum.

- Given a set of 'D' orthogonal signals in an 'N' dimensional space (D << N), then Signal to Jammer Ratio (SJR) = $E_S N/E_W D$, where E_S is the average energy of each orthogonal signal and E_W is the total energy of the jammer waveform.

- Therefore, Processing Gain (PG) = N/D ~ $R_C/R_B = T_B/T_C$ (or 10 log PG in dB), where: R_B is the bit rate (bandwidth of the source), R_C is the chip rate (bandwidth of spread spectrum), T_B and T_C are bit duration and chip duration respectively.

Other Features of CDMA

- Each code is approximately orthogonal with all other codes.
- Share the full spectrum of resource asynchronously.
- CDMA does not require an external synchronization network, which is essential in TDMA.
- CDMA offers a gradual degradation in performance as the number of users are increased.
- CDMA offers an external interference rejection capacity, i.e. multipath rejection or resistance to deliberate jamming.

Chapter 8 – Wireless Communication Systems & Standards

8.1 GSM Radio System

- GSM utilizes two bands (890-915 and 935 – 960) of 25 MHz band.
- The 890 – 915 MHz band is used for ***Subscriber-to-Base Transmissions (Reverse Link/Uplink)*** and the 935 – 960 MHz band is used for ***Base-to-Subscriber (Forward/Downlink)***
- GSM use FDD (Frequency Division Duplexing) and combination of TDMA (Time Division Multiple Access) and FHMA (Frequency Hopping Multiple Access) schemes to provide base stations with simultaneous access to multiple users.
- The available forward and reverse frequency bands are divided into 200 KHz wide channels called ARFCN (Absolute Radio Frequency Numbers).
- ARFCN denotes a forward and reverse channel pair which is separated in frequency by 45 MHz
- Each of eight subscribers uses the same ARFCN and occupies a unique time slots per frame.
- Radio transmission on both forward and reverse link are made at a channel data rate of 270.833 kbps using GMSK Modulation.
- The signaling bit duration is 3.692 μs and effective channel transmission rate per user is 33.854 kbps.
- Each time slot has a time duration of 576.25 μs and a single GSM TDMA frame spans 4.615 ms.

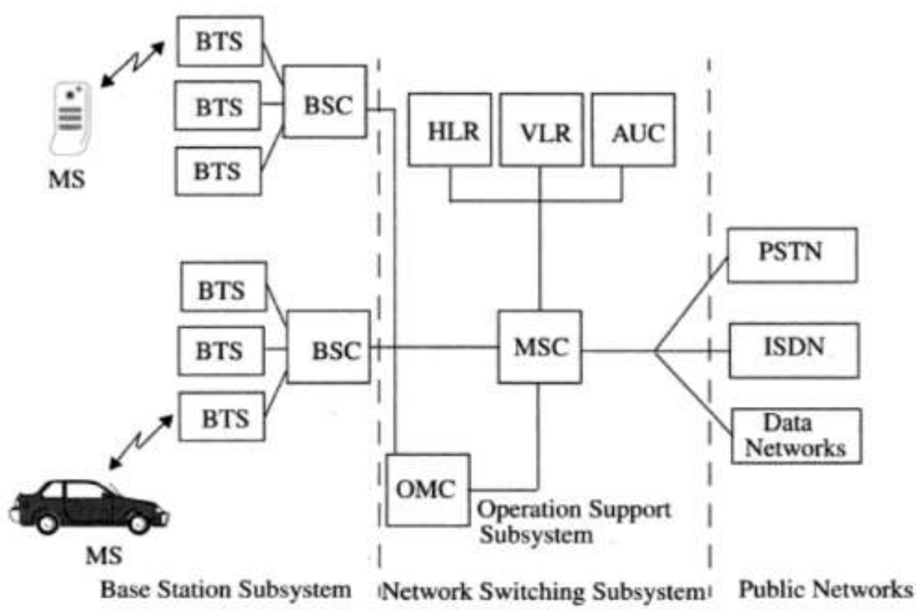


Figure: GSM System Architecture

GSM Air Interface Specifications Summary

Parameter	***Specifications***
Reverse Channel Frequency	890 – 915 MHz
Forward Channel Frequency	935 – 960 MHz
ARFCN Number	0 to 124 and 975 to 1023
Tx/Rx Frequency Spacing & Tx/Rx Time Slot Spacing	45 MHz & 3 Time Slots
Modulation Data Rate	270.83 kbps
Frame Period	4.615 ms
Users Per Frame (Full Rate)	8
Time Slot Period	576.9 μs
Bit Period	3.692 μs
Modulation	0.3 GMSK
ARFCN Channel Spacing	200 KHz
Interleaving (Maximum Delay)	40 ms
Voice Coder Bit Rate	13.4 kbps

8.2 GSM Channel System

GSM divides each ARFCN into 8 time slots. These 8 timeslots are further broken up into logical channels. Logical Channels can be thought of as just different types of data that is transmitted only on certain frames in a certain timeslot. Different time slots will carry different logical channels, depending on the structure the BSS uses. There are two main categories of logical channels in GSM, Control Channels (CCH) and Traffic Channels (TCH).

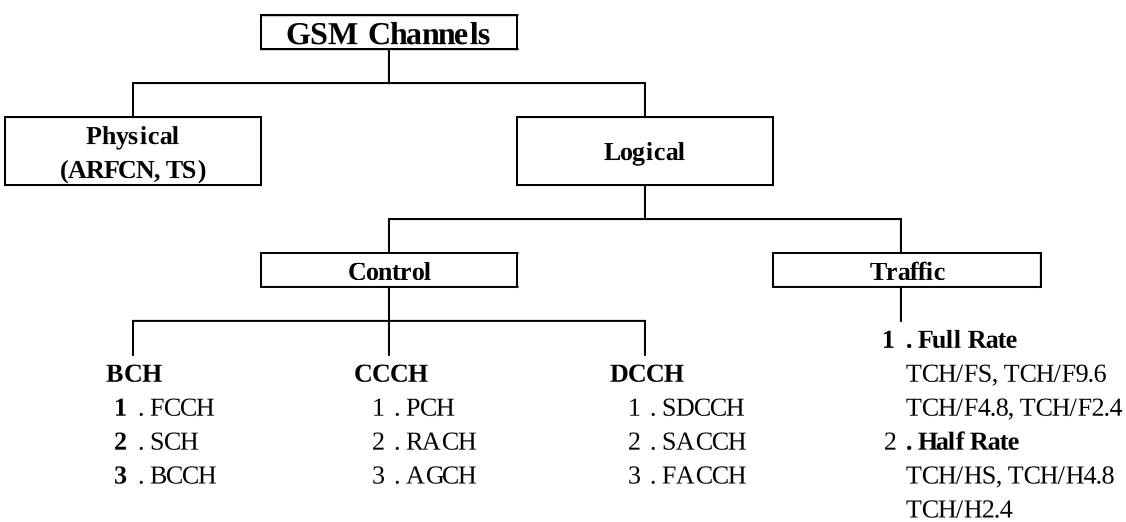


Figure: Channels Types in GSM

Traffic Channels (TCH)

Traffic channels carry digitally encoded user speech or user data and control channels carry signaling and synchronizing commands between the base station and the mobile station. Control channels are also called signaling channels.

GSM traffic channels may be either full-rate or half-rate and may carry either digitized speech or user data. When transmitted as full-rate, user data is contained within one TS per frame. When transmitted as half-rate, user data is mapped onto the same time slot, but is sent in alternate frames. That is, two half-rate channel users would share the same time slot, but would alternately transmit during every other frame.

Control Channels (CCH)

a. ***Broadcast Channels (BCH):*** Transmitted by the BTS to the MS. This channel carrier system parameters needed to identify the network, synchronize time and frequency with the network, and gain access to the network.

1. Broadcast Control Channel (BCCH) – Downlink: This channel contains system parameters needed to identify the network and gain access to the network. These parameters include the Location Area Code (LAC), the Mobile Network Code (MNC), the frequencies of neighboring cells and access parameters.
2. Frequency Correction Channel (FCCH) – Downlink: This channel is used by the MS as a frequency reference. This channel contains frequency correction bursts.
3. Synchronization Channel (CCH) – Downlink: This channel is used by the MS to learn the Base Station Information Code (BSIC) as well as the TDMA frame number (FN). This lets the MS know what TDMA frame they are on within the hyper-frame.

b. ***Common Control Channels (CCCH):*** Used for signaling between the BTS and the MS and to request and grant to the network.

1. Paging Channel (PCH) – Downlink: This channel is used to inform the MS that is has incoming traffic. The traffic could be a voice call, SMS, or some other form of traffic.
2. Random Access Channel (RACH) – Uplink: This channel is used by a MS to request an initial dedicated channel from the BTS. This would be the first transmission made by a MS to access the network and request radio resources. The MS sends an Access Burst on this channel in order to request access.
3. Access Grant Channel (AGCH) – Downlink: This channel is used by a BTS to notify the MS of the assignment of an initial SDCCH for initial signaling.

c. ***Dedicated Control Channels (DCCH):*** Used for call setup, measurement report and handover

1. Standalone Dedicated Control Channel (SDCCH) – Uplink/Downlink: This channel is used for signaling and call setup between the MS and the BTS unless a traffic channel (TCH) is assigned to the MS.
2. Fast Associated Control Channel (FACCH) – Uplink/Downlink: This channel is used for control requirements such as handoffs. There is no TS and frame allocation dedicated to a FACCH. The FACCH is a burst-stealing channel; it steals a timeslot from a traffic channel.
3. Slow Associated Control Channel (SACCH) – Uplink/Downlink: This channel is a continuous stream channel that is used for control and supervisory signals associated with the traffic channels. It is used by MS to send measurement report to BTS.

8.3 GSM Frame Structure

- In GSM, frequency band of 25 MHz is divided into 200 KHz of smaller bands, each carry one RF carrier, this gives 125 carriers. As one carrier is used as guard channel between GSM and other frequency bands, 124 carriers are useful RF channels. This division of frequency pool is called FDMA.

- Now each RF carrier will have 8 time slots. This time wise division is called TDMA. Here each RF carrier frequency is shared between 8 users hence in GSM system, the basic radio resources is a time slot with duration of about 577 μs. This time slot carriers 156.25 bits which leads to bit rate of 270.833 kbps.

- The GSM frame structure is designated as hyperframe, superframe, multiframe, and frame. The minimum unit being framed (or TDMA frame) is made of 8 time slots.

- One GSM hyperframe is composed of 2048 superframe. Each GSM superframe is composed of either 26 or 51 multiframe. Each GSM multiframe is composed of either 51 or 26 frames based on multiframe type and each frame is composed of 8 time slots. Hence, there will be total of 2715648 TDMA frames available in GSM and the same cycle continues.

- There are two variant of multiframe structure:

  a) 26 – Frame Multiframe: Called traffic multiframe, composed of 26 bursts in duration of 120 ms, out of these 24 are used for traffic, one for SACCH and one is not used.

  b) 51 – Frame Multiframe: Called control multiframe, composed of 51 bursts in duration of 235.365 ms. Last frame here is also idle.

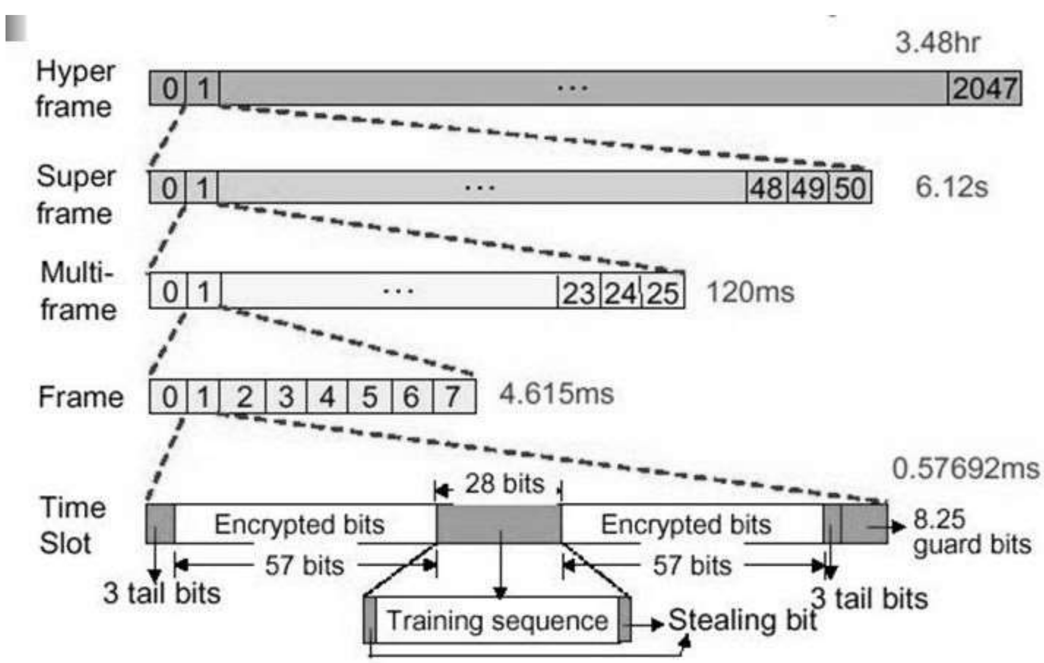


Figure: GSM – TDMA Structure

GSM Burst Types

The fields in each slot are described as:

- ✓ Trail Bits (TB) – at beginning and end of each time slot, used for synchronization.
- ✓ Coded Data (CD) – encrypted/coded data placed in each time slot.
- ✓ Stealing Bit (SB) – to indicate decoder at receiver whether the incoming burst is carrying signaling data or it is carrying user data.
- ✓ Training Sequences (TS) – used for multipath equalization, this is used to extract the desired signal from unwanted reflections. This training sequence also used to determine channel the burst has travelled, this helps in correcting rest of the frame and hence helps in decode the transmitted information properly.
- ✓ Guard Bits (GB) – used to avoid overlap of different bits.

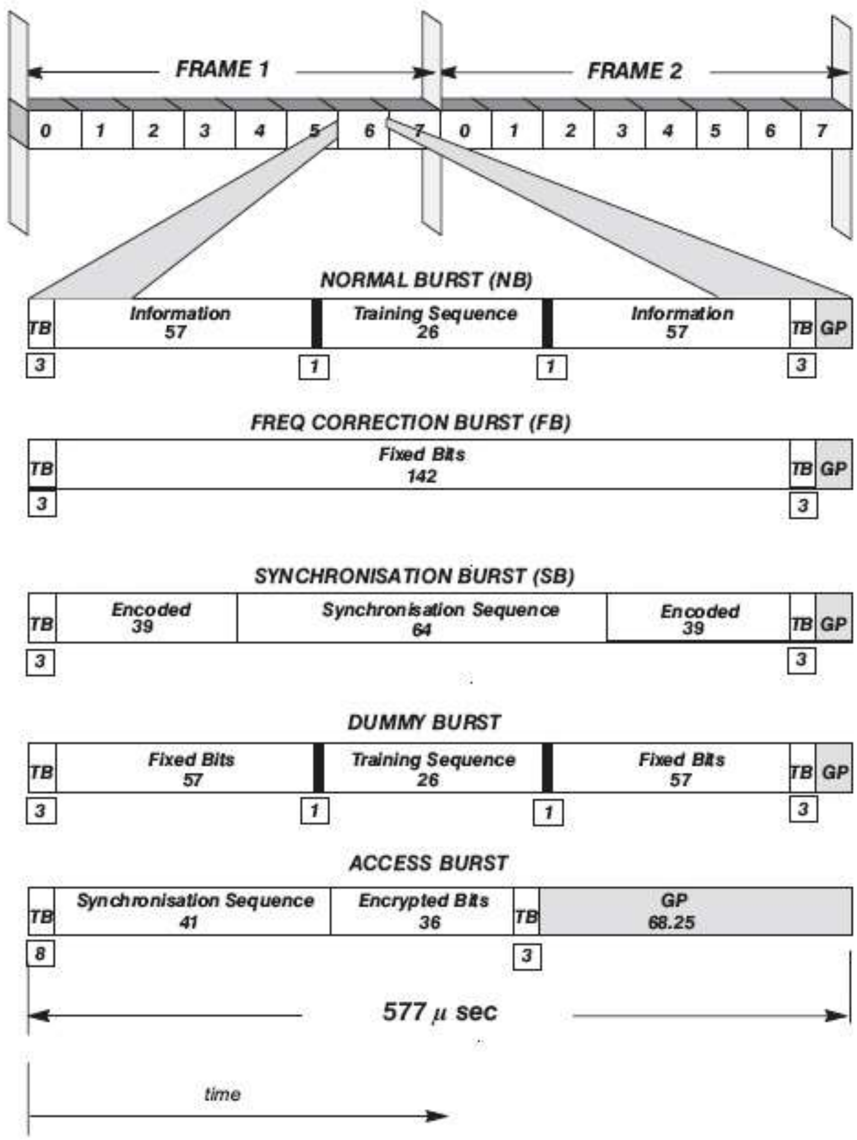


Figure: GSM Burst Types

8.4 Signal Processing in GSM

The GSM physical layer is nothing but the modules through which speech will pass through before they are transmitted.

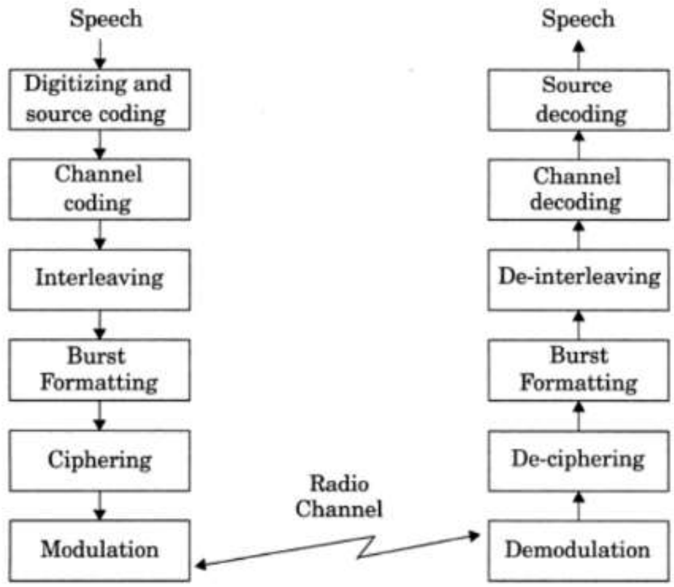


Figure: Speech Processing in GSM

- ***Speech coding*** block uses 13 kbps RELP (Residually Excited Linear Predictive Coder).
- ***Channel coding*** block uses convolution coding of rate 1/2 with constraint length of 5.
- ***Interleaving*** block does diagonal interleaving: after 456 encoded bits in 20ms duration are broken into 57 bits sub-blocks.
- ***Ciphering*** block uses A3 and A5 encryption algorithms. Encryption is changed call by call to enhance privacy.
- ***Burst assembly*** block frames the burst as required by GSM frame structure. The same is modulated and Gaussian filtered
- ***Modulation*** block minimizes the occupied bandwidth using GMSK modulation with BT of 0.3.

8.5 CDMA Digital Cellular Standard (IS – 95)

Introduction/Background Concept

- Second generation digital wireless cellular communication system.
- Allows each user within a cell to use the same radio channel, and users in the adjacent cells also use the same radio channels, since this is a direct sequence spread spectrum (DSSS) system.
- Band of operation 800 MHz:
  - Reverse Link (Uplink): 824 MHz to 849 MHz
  - Forward Link (Downlink): 869 MHz to 894 MHz
- Forward and reverse link separation: 45 MHz with Frequency Division Duplexing (FDD).
- User data is spread to a channel chip rate of 1.228 Mcps.
- Speech Coder: Qualcomm Code Excited Linear Predictive (QCELP) Coder
- Channel Coder: Convolution Encoder
- Modulation Technique: Forward Link (QPSK) and Reverse Link (Offset QPSK)
- RAKE receiver are used to resolve and combine multipath components, thereby reducing the degree of fading.
- A combination of open and closed loop power control is used to adjust the transmit power of users so that the base station receives each user with the same received power.

Channels in CDMA (IS-95)

- The IS-95 CDMA system is unique in that its forward and reverse links have different link structures.
- The forward link consists of four types of logical channels: pilot, sync, paging and traffic channels. There are one pilot channel, one sync channel, up to seven paging channels, and several traffic channels. Each of these forward link channels is first spread orthogonally by its Walsh function, then it is spread by a quadrature pair of PN sequences. All Channels are added together to form the composite spread spectrum signal to be transmitted on the forward link.
- The reverse link consists of two types of logical channels: access and traffic channels. Each of these reverse link channels is spread orthogonally by a unique long PN sequence; hence, each channel is identified using the distinct long PN code.

1. Forward CDMA Channel

- The IS-95 CDMA system uses a 64x64 Hadamard matrix to generate 64 Walsh functions that are orthogonal to each other, and each of the logic channels on the forward link is identified by its assigned Walsh function.

- IS-95 uses a set of 64 fixed length Walsh codes to spread forward physical channels. For example, Walsh code 0 is assigned to the pilot channel, code 32 to the sync channel, code 1 to 7 to paging channels, and the rest to the forward traffic channels.

1.1 Pilot Channel

- The pilot channel is identified by the Walsh function 0 (W_0). The channel itself contains no baseband information. The baseband sequence is a stream of 0s that are spread by Walsh function, which is also a sequence of all 0s. The resulting sequence (still all 0s) is then spread, or multiplied, by a pair of quadrature PN sequences (at the rate of 1.228 Mcps).
- After PN spreading, baseband filters are used to shape the digital pulses.
- The pilot channel is transmitted continuously by the base station sector. It provides the mobile with timing and phase reference.

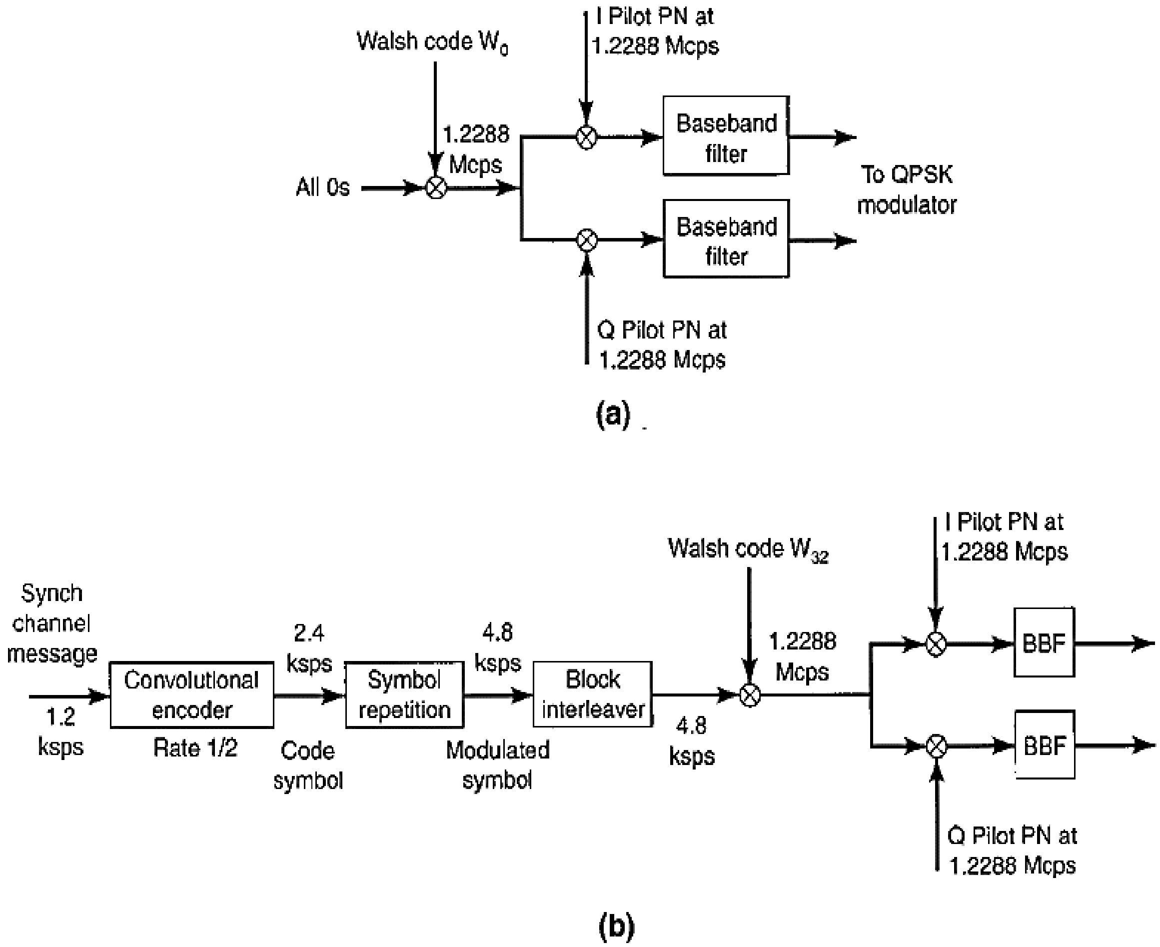


Figure: (a) Pilot Channel (b) Sync Channel

1.2 Sync Channel

- Unlike the pilot channel, the sync channel carries baseband information. The information is contained in the sync channel message that notifies the mobile of important information about system synchronization and parameters.
- The baseband information is error protected and interleaved. It is then spread by Walsh function 32 and further spread by the PN sequence that is identified with the serving sector.
- The baseband information is spread at the rate of 1.2 kbps.

1.3 Paging Channel

- Similar to sync channel, the paging channel also carries baseband information, but unlike the sync channel, it transmits at higher rates (4.8 or 9.6 kbps).
- Once the mobile acquires timing and synchronization using the sync channel, the mobile begins to monitor the paging channel. Although there can be up to seven paging channels per sector, each mobile only monitors one paging channel.

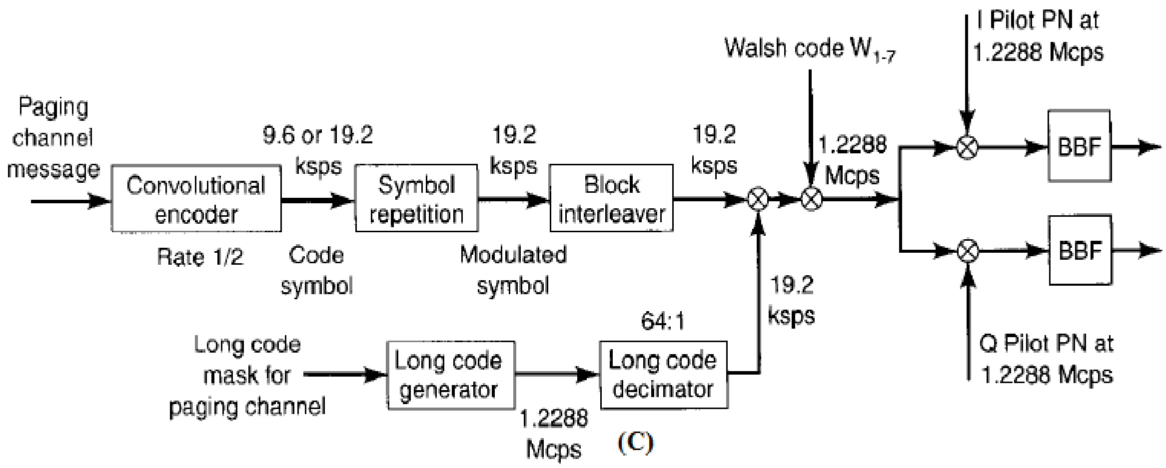


Figure: (C) Paging Channel

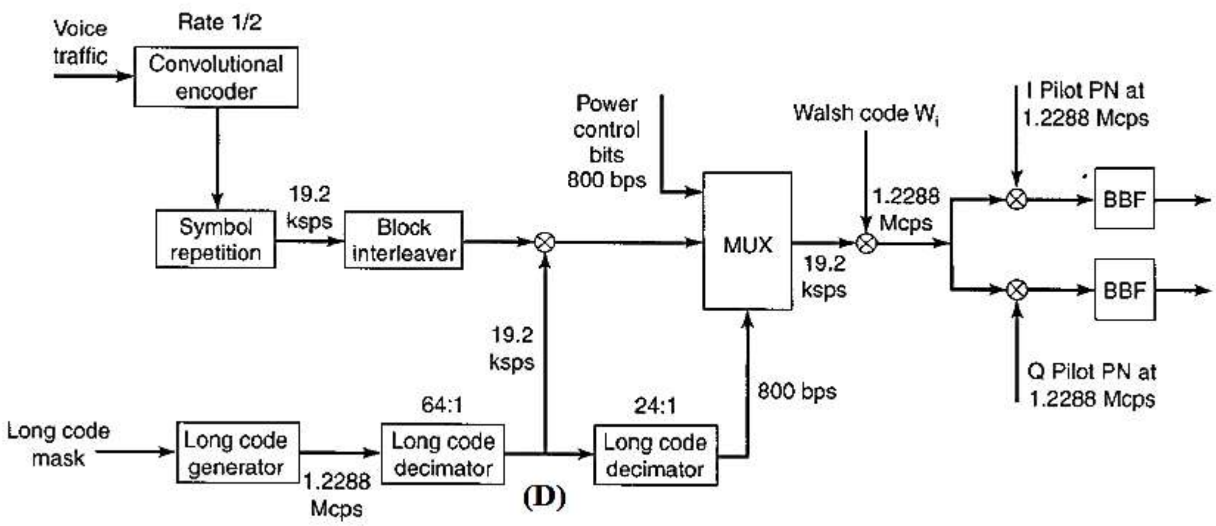


Figure: (D) Forward Traffic Channel

1.4 Traffic Channel

- The forward traffic channel is used to transmit user data and voice; signaling messages are also sent over the traffic channel.
- The structure of traffic channel is similar to that of paging channel, the only difference is that the forward traffic channel contains multiplexed PCBs.
- The base station sends the power control commands to the mobile using the forward link. These power control commands are in the form of *Power Control Bits* (PCBs). The amount of mobile power increase and power decrease per PCB is nominally +1dB and -1dB respectively.

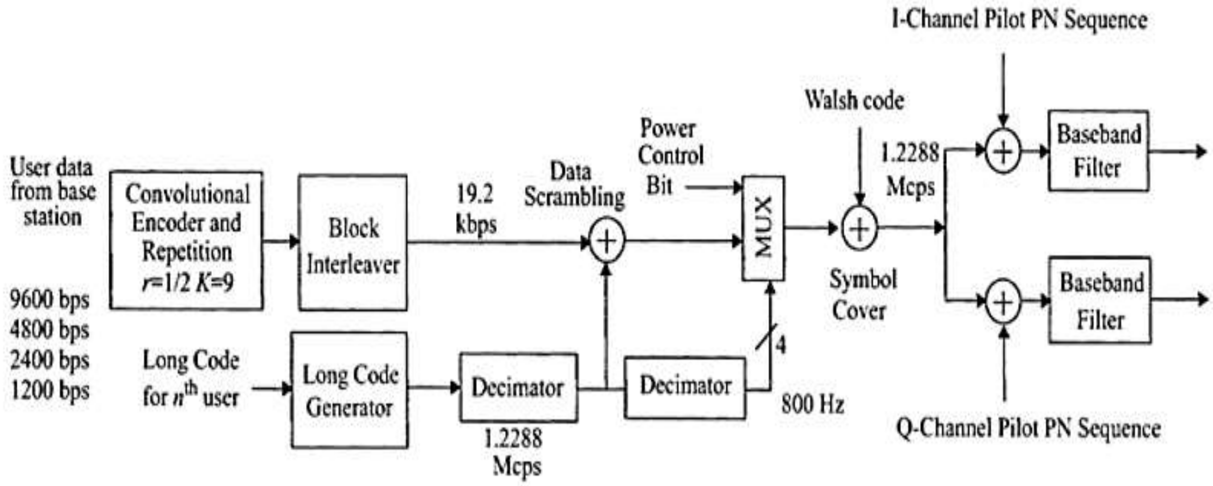


Figure: Forward Channel Modulation Processing in IS-95

2. Reverse CDMA Channel

- The reverse link support two types of logical channels: access channels and traffic channels. Because of the non-coherent nature of the reverse link, Walsh functions are not used for channelization. Instead, long PN sequences are used to distinguish the users from one another.

2.1 Access Channel

- Used by the mobile to communicate with the base station when the mobile does not have a traffic channel assigned. The mobile uses this channel to make call originations and respond to pages and orders.

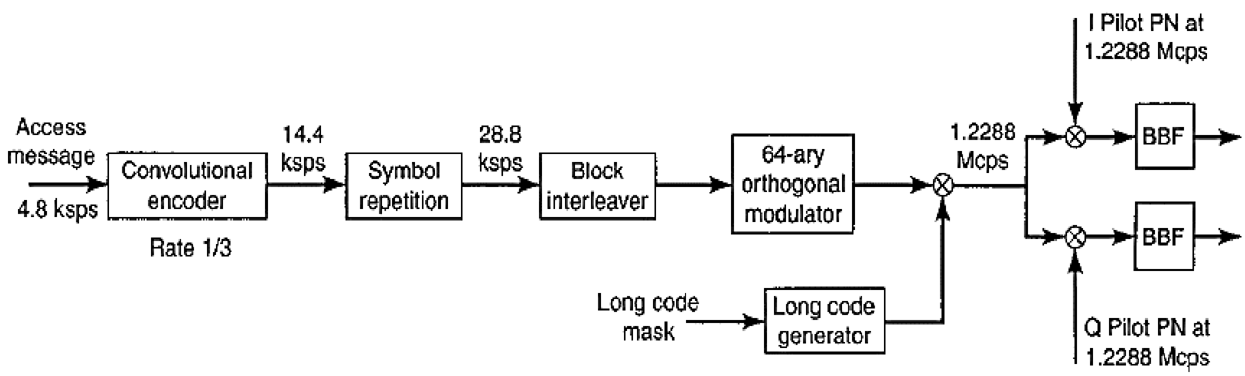


Figure: Reverse Access Channel

- The baseband data rate of the access channel is fixed at 4.8 kbps. –
- The baseband information is first error protected by a convolution encoder.
- The symbol is repeated once yielding a code symbol rate of 28.8 ksps.
- Data is then interleaved to combat fading.
- Following interleaver, the data is coded by a 64-ary orthogonal modulator and mixed with long random code and transmitted with QPSK modulator.

2.2 Reverse Traffic Channel

- The reverse traffic channel is used to transmit user data and voice; signaling messages are also sent over the traffic channel. The reverse traffic channel is similar to that of the access channel. The major difference is that the reverse traffic channel contains a data burst randomizer.
- The orthogonally modulated data is fed into the data burst randomizer. The function of data burst randomizer is to take advantage of the voice activity factor on the reverse link

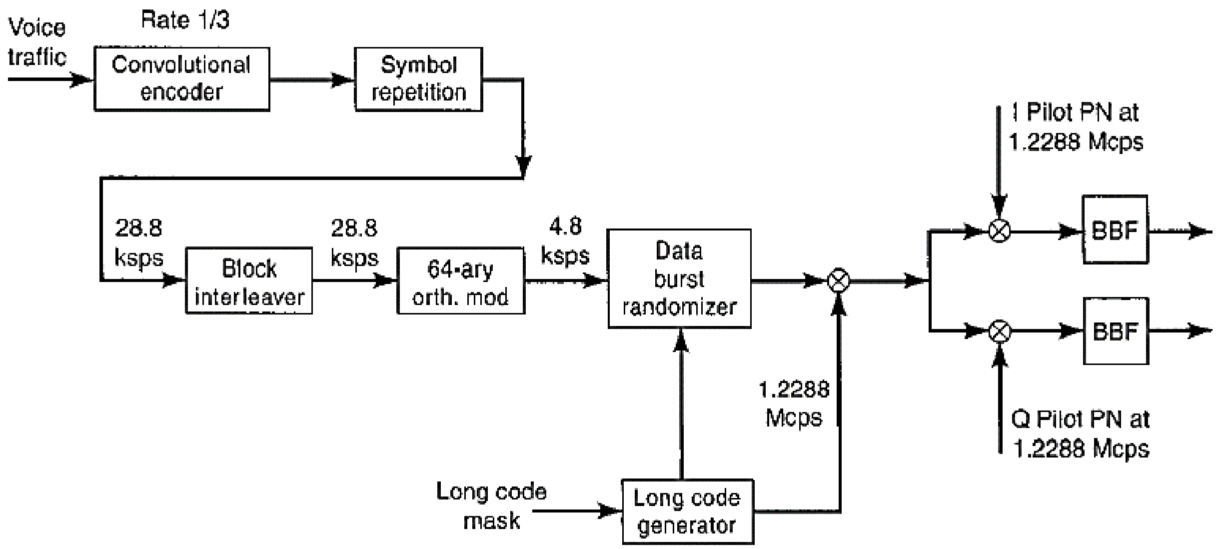


Figure: Reverse Traffic Channel

- .

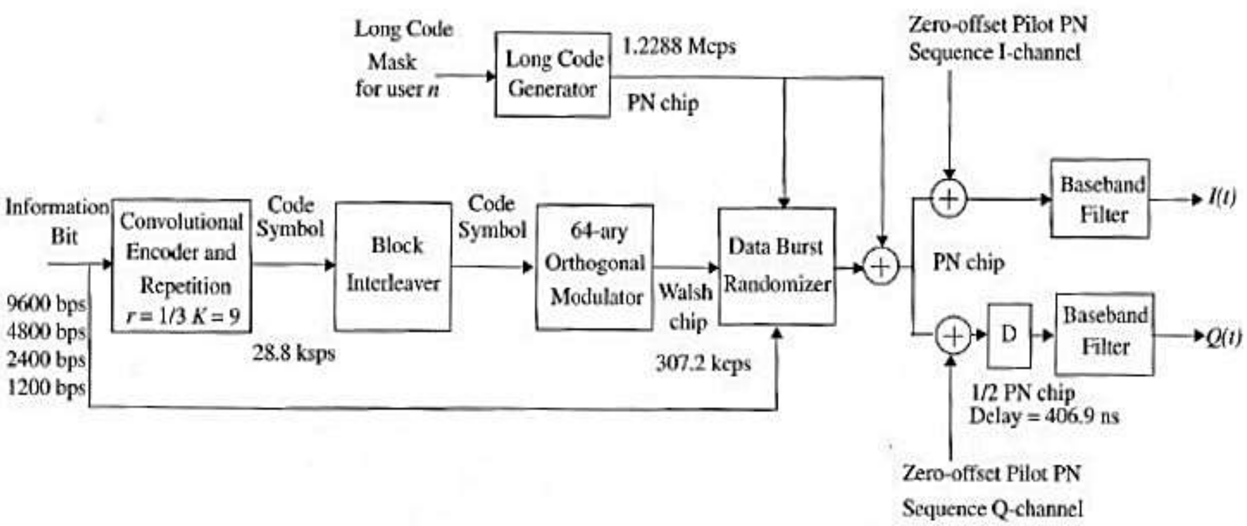


Figure: Reverse IS-95 Channel Modulation Process for a Single User

Modulator

The output of logical channel is fed into the modulator. The gain of each logical channel, including pilot, sync, paging and all traffic channels is first adjusted by the gain control function. The gain of each channel dictates how much power is to be transmitted for that channel. The gains for the individual traffic channels are dynamically changing (i.e. they are controlled by the forward power control process).

After the channel gains are adjusted, the signals are coherently added together to form the composite spread spectrum signal. After the summation, both the '*I*' and '*Q*' paths are up converted by the respective carriers. The up converted signals then are added together to form the final pass band QPSK signal.

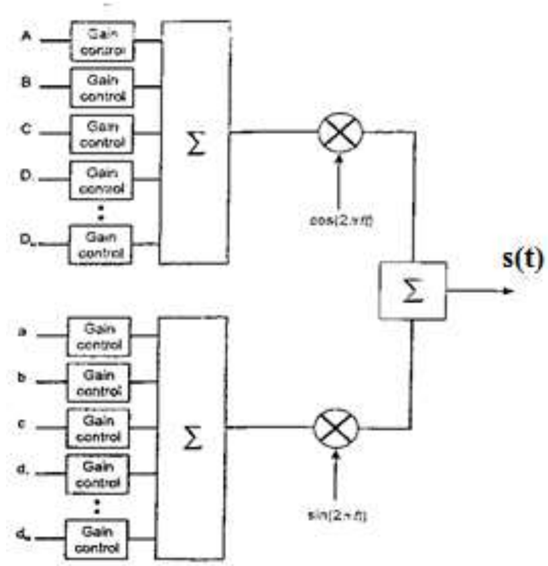


Figure: Modulator (IS – 95)

Walsh Codes

Various physical channels may exist at any time on a radio interface. To separate these channels at the receiver, they are separated with Walsh codes at the transmitter. These codes are formed by the rows of an N x N square matrix, whose entries are either 0 or 1. Usually $N = 2^n$, where n is an integer. They are orthogonal because if a 0 is mapped to -1 and 1 to 1, then the sum of the term by term products of any two rows of this matrix is 0. This matrix is also known as the *Hadamard Matrix.*

Hadamard Codes

Hadamard codes are in the form of a matrix called Hadamard matrix. Each row of the matrix represents a code word. A Hadamard matrix is an N x N matrix of 1s and 0s such that each row differs from any other row in exactly N/2 locations. One row contains all zeroes with the remainder containing N/2 zeroes and N/2 ones. The minimum distance, $d_{min} = N/2$ for these codes. For N = 2, the Hadamard matrix is: $H = \begin{bmatrix} 0 & 0 \\ 0 & 1 \end{bmatrix}$

Scrambling Codes

In CDMA, each bit time is subdivided into '*m*' short intervals called chips. Typically there are 64 or 128 chips per bit. Each station is assigned a unique *m*-bit chip sequence. To transmit a 1 bit, a station sends its chip sequence. To transmit a 0 bit, it sends the one's complement of its chip sequence. No other patterns are permitted. Thus for *m* = 8, if station A is assigned the chip sequence 00011011, it sends a 1 bit by sending 00011011 and a 0 bit by sending 11100100. Increasing the amount of information to be sent from b bits/sec to *mb* chips/sec can only be done if the bandwidth available is increased by a factor of *m*, making CDMA a form of spread spectrum communication (assuming no changes in the modulation or encoding techniques). In order to protect the signal, the chip sequence code used is pseudo-random, it appears random, but is actually deterministic, so that the receiver can reconstruct the code for synchronous detection. This pseudo-random code is also called pseudo-noise (PN) sequence.

Chapter 9 – Wireless System Application and Development

9.1 Wireless Local Loop (WLL)

- ✓ WLL is a system that connects subscribers to the local telephone station wirelessly.
- ✓ Wired technologies responding to need for reliable, high-speed access by residential, business, and government subscribers via ISDN, xDSL and cable modems
- ✓ Increasing interest shown in competing wireless technologies for subscriber access
- ✓ Wireless local loop (WLL) types:
    - o Narrowband – offers a replacement for existing telephony services
    - o Broadband – provides high-speed two-way voice and data service
- ✓ Systems WLL is based on:
    - o Cellular
    - o Satellite (specific and adjunct)
    - o Microcellular
- ✓ Other names
    - o Radio In The Loop (RITL)
    - o Fixed-Radio Access (FRA)

General Architecture of WLL

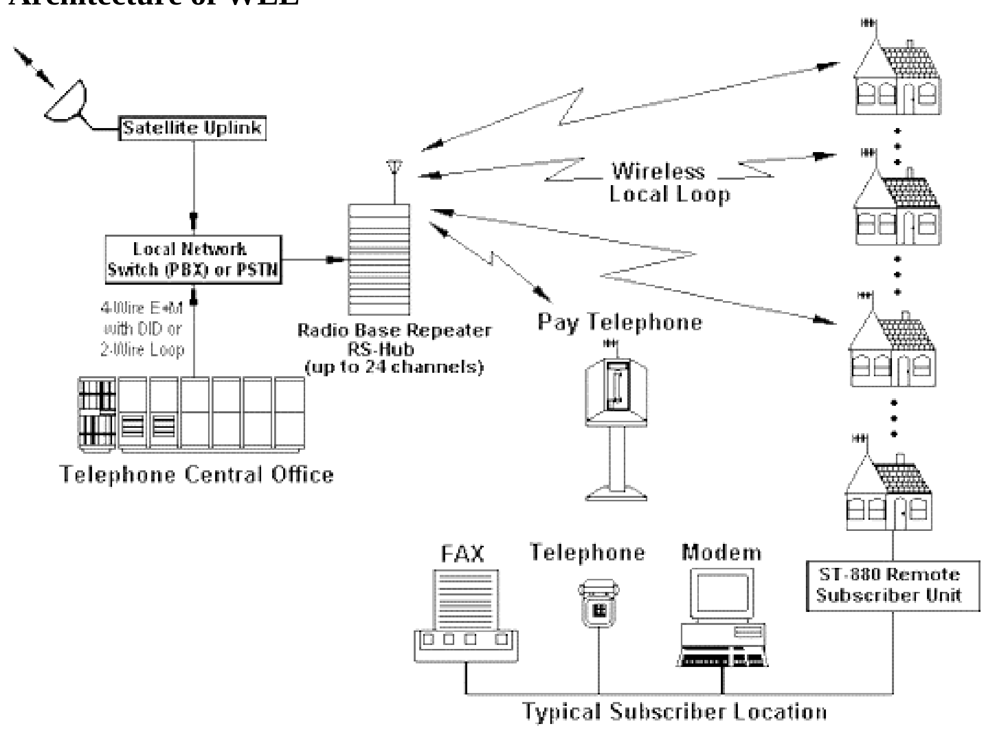


Figure: General Architecture of Wireless Local Loop (WLL)

Connection Setup for WLL

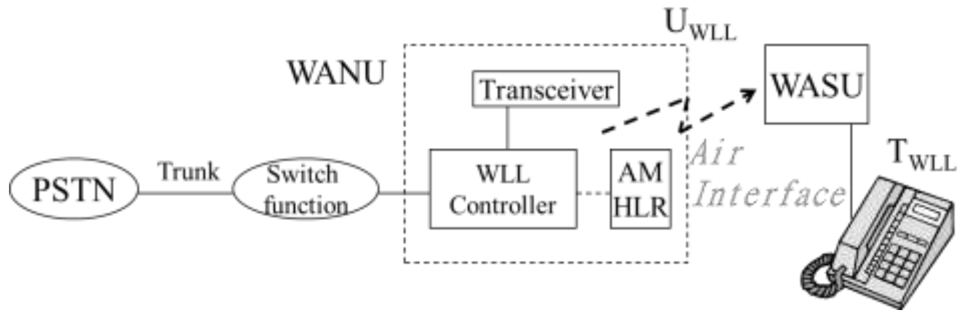


Figure: WLL Connection Setup

Wireless Access Network Unit (WANU)	**Wireless Access Subscriber Unit (WASU)**
- Interface between underlying telephone network and wireless link consists of: ▪ Base Station Transceivers (BTS) ▪ Radio Controller(RPCU) ▪ Access Manager(AM) ▪ Home Location Register(HLR)	▪ located at the subscriber ▪ translates wireless link into a traditional telephone connection

9.2 Mobile IP & Wireless Application Protocol

Why Mobile IP?

- ✓ Enable computers to maintain internet connectivity while moving from one internet attachment point to another.
- ✓ Mobile user's point of attachment changes dynamically and all connections are automatically maintained despite the change.
- ✓ Nomadic users' internet connection is terminated each time user moves and a new connection is initiated when the user dials back in new, temporary IP address is assigned.

Operation of Mobile IP

- ✓ Mobile node is assigned to a particular network – home network.
- ✓ IP address on home network is static – home address.
- ✓ Mobile node can move to another network – foreign network.
- ✓ Mobile node registers with network node on foreign network – foreign agent.
- ✓ Mobile node gives care of address to agent on home network – home agent.

Capabilities of Mobile IP

- ✓ Discovery – mobile node uses discovery procedure to identify prospective home and foreign agents.
- ✓ Registration – mobile node uses an authenticated registration procedure to inform home agent of its care of address.
- ✓ Tunneling – used to forward IP datagrams from a home address to a care of address.

9.3 Wireless Application Protocol (WAP)

Open standard providing mobile users of wireless terminals access to telephony and information services.

- Wireless terminals include wireless phones, pagers and personal digital assistants (PDAs).
- Designed to work with all wireless network technologies such as GSM, CDMA and TDMA.
- Based on existing internet standards such as IP, XML, HTML and HTTP
- Includes security facilities.

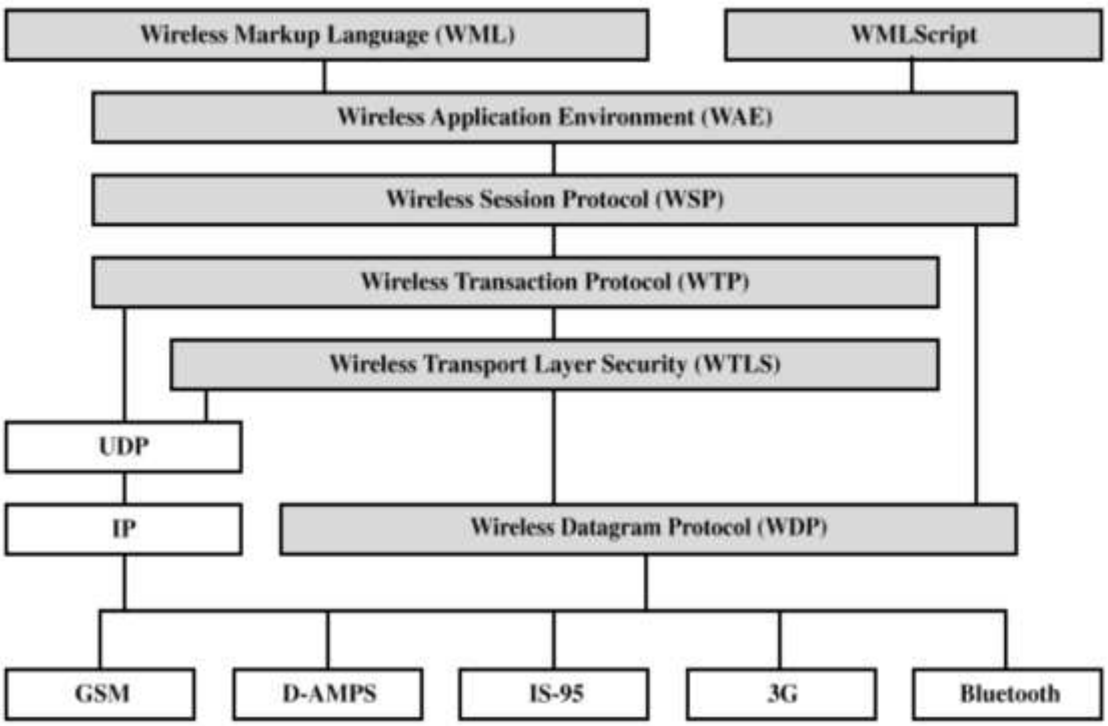


Figure: WAP Protocol Stack

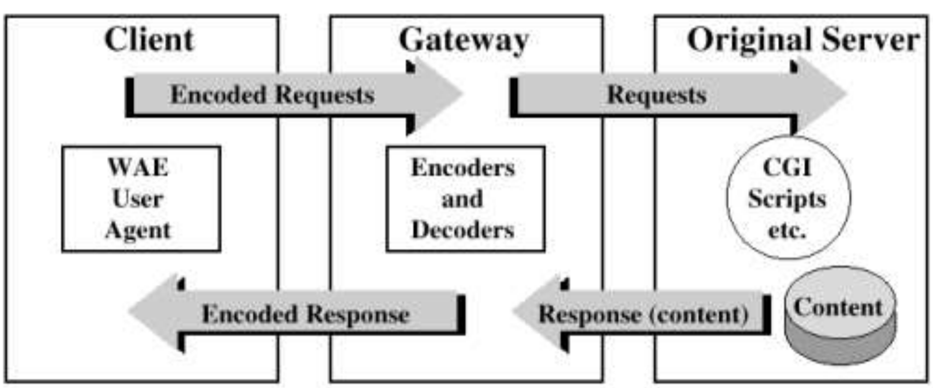


Figure: WAP Programming Model

1. WML (Wireless Markup Language):

- Text and image support – formatting and layout commands
- Deck/card organizational metaphor – WML documents subdivided into cards, which specify one or more units of interaction.
- Support for navigation among cards and decks – includes provisions for event handling; used for navigation or executing scripts.

2. WML Script

- Scripting language for defining script-type programs in a user device with limited processing power and memory.
- Capable of: checking validity of user input before it is sent, accessing device facilities and peripherals and interacting with user without introduction round trips to origin server.
- Features: JavaScript-based scripting language, procedural logic, event based, compiled implementation and integrated to WAE (Wireless application environment)

3. WAE (Wireless Application Environment)

- Specifies an application framework for wireless devices.
- Elements:
  - WAE User Agents – Software that executes in the wireless devices.
  - Content Generators – Applications that produce standard content formats in response to requests from user agents in the mobile terminal
  - Standard Content Encoding – Defined to allow a WAE user agent to navigate web content.
  - Wireless Telephony Application – Collection of telephony-specific extensions for call and feature control mechanisms.

4. WSP (Wireless Session Protocol)

- Transaction-oriented protocol based on the concept of a request and a reply.
- Provides applications with interface for two session services.
  - Connection-oriented session service – operates above reliable transport protocol WTP
  - Connectionless session service – operates above unreliable transport protocol WDP

5. WTP (Wireless Transaction Protocol)

- Lightweight protocol suitable for “thin” clients and over low-bandwidth wireless links
- WTP features:
  - Three classes of transaction services.
  - Optional user-to-user reliability: WTP user triggers confirmation of each received message.
  - Optional out-of-band data on acknowledgements
  - PDU concatenation and delayed acknowledgement to reduce the number of message sent.
- Transaction Classes:
  - Class 0: Unreliable invoke message with no result message.
  - Class 1: Reliable invoke message with no result message.
  - Class 2: Unreliable invoke message with one reliable result message.

6. WTLS (Wireless Transport Layer Security)

- Data Integrity – Ensures that data sent between client and gateway are not modified, using message authentication.
- Privacy – Ensures that the data cannot be read by a third party, using encryption.
- Authentication – Establishes authentication of the two parties, using digital certificates.
- Denial-of-Service Protection – Detects and rejects messages that are replayed or not successfully verified.

7. WDP (Wireless Datagram Protocol)

- Used to adapt higher layer WAP protocol to the communication mechanism used between mobile node and WAP gateway.
- WDP hides details of the various bearer networks from the other layer of WAP.
- Adaptation may include:
  - Partitioning data into segments of appropriate size for the bearer.
  - Interfacing with the bearer network.

9.4 Wireless LAN Technologies

Why would anyone want a wireless LAN?

There are many reasons. An increasing number of LAN users are becoming mobile. These mobile users require that they are connected to the network regardless of where they are because they want simultaneous access to the network. This makes the use of cables, or wired LANs, impractical if not impossible.

Figure: Wireless LAN

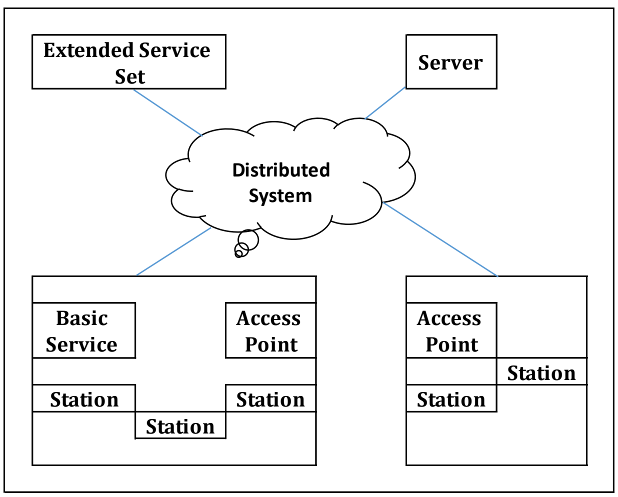


Figure: IEEE 802.11 Architecture

Wireless LANs are very easy to install. There is no requirement for wiring every workstation and every room. This ease of installation makes wireless LANs inherently flexible. If a workstation must be moved, it can be done easily and without additional wiring, cable drops or reconfiguration of the network. Another advantage is its portability. If a company moves to a new location, the wireless system is much easier to move than ripping up all of the cables that a wired system would have snaked throughout the building. Most of these advantages also translate into monetary savings

Categories of Wireless LAN

Wireless LANs are generally categorized according to the transmission technique that is used. All current wireless LAN products fall into one of the following categories.

1. ***Infrared (IR) LANs***: An individual cell of an IR LAN is limited to a single room, because infrared light does not penetrate opaque walls.
2. ***Spread Spectrum (SS) LANs***: This makes use of spread spectrum transmission technology. In most cases, these LANs operate in the ISM (Industrial, Scientific, and Medical) bands, so that no FCC licensing is required for their use in US.
3. ***Narrowband Microwave (NM) LANs***: These LANs operate at microwave frequencies but do not use spread spectrum. Some of these products operate at frequencies that require FCC licensing, while others use one of the unlicensed ISM bands.

	Infrared (IR)		**Spread Spectrum (SS)**		**Radio**
	Diffused IR	Directed Beam IR	Frequency Hopping	Direct Sequence	Narrowband Microwave
Data Rate (Mbps)	1 to 4	1 to 10	1 to 3	2 to 20	10 to 20
Mobility	Stationary/mobile	Stationary/LOS	Mobile	Stationary/mobile	
Range (ft)	50 to 200	80	100 to 300	100 to 800	40 to 130
Detectability	Negligible		Little		Some
Wavelength/Frequency	800 to 900 nm		902 to 928 MHz 2.4 to 2.4835 GHz 5.725 to 5.85 GHz		902 to 928 MHz 5.2 to 5.775 GHz 18.825 to 19.205 GHz
Modulation Technique	ASK		FSK	QPSK	FSK/QPSK
Radiated Power	-		< 1 kW		25mW
Access Method	CSMA	Token Ring, CSMA	CSMA		Reservation ALOHA, CSMA
License Required	No		No		Yes unless ISM

Table: Comparison of Wireless LAN Technologies

IEEE 802.11 (Wireless LAN Standard)

A set of wireless LAN standards has been developed by the IEEE 802.11 committee. The terminology and some of the specific features of 802.11 are unique to this standard and are not reflected in all commercial products. However, it is useful to be familiar with the standard because its features are representative of require wireless LAN capabilities.

The smallest building block of a wireless LAN is a basic service set (BSS), which consists of some number of stations executing the same MAC (Medium Access Control) protocol and competing for access to the same shared medium.
A BSS may be isolated or it may connect to a backbone distribution system through an access point. The access point functions as a bridge.
The MAC protocol may be fully distributed or controlled by a central coordination function housed in the access point. The BSS generally corresponds to what is referred to as a cell in the literature.

An extended service set (ESS) consists of two or more basic service sets interconnected by a distribution system. Typically, the distribution system is a wired backbone LAN. The ESS appears as a single logical LAN to the logical link control level (LLC).