Wireless Communication

PART A-UNIT 1

1. What is meant by frequency reuse?
2. What are the trends in cellular radio systems?
3. What do you mean by forward and reverse channel?
4. What is the function of control channel? What are the types?
5. What is channel assignment? What are the types?
6. What is fixed channel assignment?
7. What is dynamic channel assignment?
8. Define MS, BS and MSC.
9. Define had off and mode of hand off.
10. Write the types of hand off.
11. Define Cell, Cluster.
12. What do you mean by foot print and dwell time?
13. What are the major types of cellular interference?
14. What are the techniques used to expand the capacity of cellular system?
15. Define frequency reuse ratio.
16. Define FDMA, TDMA and CDMA.
17. Define Grade of service.
18. What is blocked call clear system(BCC)?
19. What is blocked call delay system?
20. Define cell splitting.
21. What is sectoring?
22. What are the features of TDMA?
23. What are the features of FDMA?
24. **Differentiate cellular telephony and cordless telephony. Nov. 2011**
25. **When does a WLAN become a personal area network (PAN)? Nov. 2011**
26. **What are the different types of multiple access schemes? May 2012**
27. **Mention the significance of frequency reuse in cellular networks. May 2012**
28. **What is flat fading? Nov. 2012**
29. **Define signal to self-interference ratio. Nov. 2012**
30. **What are the three most important effects of small scale multipath propagation. Nov/Dec. 2013**
31. **What is a multiple access technique? Nov/Dec. 2013**

PART B-UNIT 1

1. i. Compare and contrast wired and wireless communication. (8) Nov. 2011
   ii. Discuss briefly about the requirements of services for a wireless system.(8)
2. i. Discuss in detail the constructive and destructive interferences. (8) Nov. 2011
   ii. Explain how Inter Symbol Interference is caused and how it is eliminated. (8)
3. i. Explain in detail Wide Area Data Services and Broadband Wireless Access services offered to wireless networks. (10) May 2012
   ii. What are paging systems? Explain. (6)
4. i. With a neat block diagram, explain the cellular network architecture.(10) May 2012
   ii. Explain any one type of Multiple Access scheme. (6)
5. i. Explain about the factors that influence small scale fading. (10) Nov 2012

   ii. Find the average fade duration for threshold levels $\rho = 0.01$, $\rho = 0.1$ and $\rho = 1$, when the Doppler frequency is 200 Hz.

6. i. Write a note on Noise and Interference Limited Systems. (8) Nov. 2012
   ii. Discuss the principles of cellular networks. (8)

7. i. Discuss the types of services, requirements, spectrum limitations and noise considerations of wireless communication. Nov/Dec. 2013.
   ii. Explain the principle of cellular networks and various types of Handoff techniques. Nov/Dec. 2013.

2MARKS:

UNIT 1

1.1.What is meant by frequency reuse?

If an area is served by a single Base Station, then the available spectrum can be divided into *N* frequency channels that can serve *N* users simultaneously. If more than *N* users are to be served, multiple BSs are required, and frequency channels have to be reused in different locations. Since spectrum is limited, the *same* spectrum has to be used for *different* wireless connections in different locations. This method of reusing the frequency is called as frequency reuse.

1.2. What are the trends in cellular radio systems?

The trends in personal cellular radio systems are:

i. PCS – Personal Communication Services
ii. PCN – Personal Communication Networks

1.3. What do you mean by forward and reverse channel?

Forward channel is a radio channel used for transmission of information from base station to mobile.Reverse channel is a radio channel used for transmission from mobile to base station.

1.4. What is the function of control channel? What are the types?

The function of control channel is to transmit call setup, call request, call initiation and Control. There are two types of control channels,

i. Forward control channel
ii. Reverse control channel

1.5. What is channel assignment? What are the types?

For efficient utilization of radio spectrum a frequency reuse scheme with increasing capacity and minimizing interference is required. For this channel assignment is used. The types of channel assignment are:

i. Fixed channel assignment
ii. Dynamic channel assignment.

1.6. What is fixed channel assignment?

If the channels in each cell are allocated to the users within the cell, it will be called as fixed channel assignment. If all channels are occupied, the call will be blocked.

1.7. What is dynamic channel assignment?

If the voice channels are not allocated permanently in a cell, it will be called as dynamic channel assignment. In this assignment, channels are dynamically allocated to users by the MSC.

1.8. Define MS, BS and MSC.

MS – Mobile station. A station in the cellular radio service intended for use.

BS – Base Station. A fixed station in a mobile radio system used for radio communication with MS.

MSC – Mobile Switching Centre. Mobile switching centre coordinates the routing of calls in large service area. It connects the base station and mobiles to PSTN. It is also called as MTSO(Mobile telephone switching office.

1.9. Define hand off and mode of hand off.

A handoff refers to the process of transferring an active call or data session from one cell in a cellular network to another or from one channel in a cell to another. A well-implemented handoff is important for delivering uninterrupted service to a caller or data session user. Modes of hand off are:

i. MCHO – Mobile Controlled Hand off
ii. NCHO – Network Controlled Hand off
iii. MAHO – Mobile Assisted Hand off

1.10. Write the types of hand off.

Types of handoff are:

i. Hard hand off – Mobile monitors BS and new cell is allocated to a call with strong signal.

ii. Soft hand off – MS with 2 or more calls at the same time and find which is the strongest signal BS, the MSC automatically transfers the call to that BS.

1.11**. Define Cell, Cluster.**

For a large geographic coverage area, a high powered transmitter therefore has to be used. But a high power radio transmitter causes harm to environment. Mobile communication thus calls for replacing the high power transmitters by low power transmitters by dividing the coverage area into small segments, called cells.

Each cell uses a certain number of the available channels and a group of adjacent cells together use all the available channels. Such a group is called a cluster.

1.12. What do you mean by foot print and dwell time?

The region over which the signal strength lies above this threshold value x dB is known as the coverage area of a BS and it must be a circular region, considering the BS to be isotropic radiator. Such **a circle, which gives this actual radio coverage, is called the foot print of a cell**.The time over which a call may be maintained within a cell without hand off is called the dwell time.

1.13. What are the major types of cellular interference?

The major types of cellular interferences are as follows

i. CCI – Co-channel interference is the interference between signals from co-channel cells.
ii. ACI – Adjacent channel interference resulting from signals which are adjacent in frequency to the desired signal.

1.14. What are the techniques used to expand the capacity of cellular system?

Cell splitting, Sectoring, Coverage Zone approaches are the techniques used to expand the capacity of cellular system.

Cell splitting – Cell-splitting is a technique which has the capability to add new smaller cells in specific areas of the system. i.e. divide large cell size into small size.

Sectoring – use of directional antennas to reduce Co-channel interference.

Coverage Zone approaches – large central BS is replaced by several low power transmitters on the edge of the cell.

1.15. What is frequency reuse ratio?

If the cell size and the power transmitted at the base stations are same then co-channel interference will become independent of the transmitted power and will depend on radius of the cell (R) and the distance between the interfering co-channel cells (D). If D/R ratio is increased, then the effective distance between the co-channel cells will increase and interference will decrease. The parameter Q is called the frequency reuse ratio and is related to the cluster size. For hexagonal geometry

$$Q = \frac{D}{R}$$

$$= \frac{\text{Distance between centres of the nearest co-channel cells}}{\text{Radius of the cell}}$$

From the above equation, small of `Q' means small value of cluster size `N' and increase in cellular capacity.

1.16. Define FDMA, TDMA and CDMA.

FDMA - the total bandwidth is divided into non-overlapping frequency subbands.

TDMA – divides the radio spectrum into time slots and in each slot only one user is allowed to either transmit or receive.

CDMA – many users share the same frequency same tome with different coding.

1.17. Define Grade of service.

Grade of service is defined as the measure of the ability of a user to access a trunked system during the busiest hour.

1.18. What is blocked call clear system (BCC)?

In a system, a user is blocked without access by a system when no channels are available in the system. The call blocked by the system is cleared and the user should try again .This is called BCC system.

1.19. What is blocked call delay system?

If a channel is not available immediately, the call request may be delayed until a channel becomes available. This is called as blocked call delay system.

1.20. Define cell splitting.

Cell splitting is the process of subdividing congested cells into smaller cells each with its own base stations and a corresponding reduction in antenna height and transmitter power. It increases the capacity of cellular system.

1.21. What is sectoring?

Sectoring is a technique for decreasing co-channel interference and thus increasing the system performance by using directional antennas.

1.22. What are the features of TDMA?

Features of TDMA are:

i. TDMA shares a single carrier frequency with several users, where each user makes use of non overlapping time slots.

ii. Data transmission occurs in bursts.
iii. Handoff process is much simpler
iv. Duplexers are not required, since transmission and reception occurs at different time slots.

1.23. What are the features of FDMA?

Features of FDMA are:

i. FDMA channel carries only one phone circuit at a time
ii. The bandwidth of FDMA channels are relatively narrow as each channel supports only one circuit per carrier.

Part A- UNIT 2

1. What are the propagation mechanisms of EM waves?
2. What is the significance of propagation model?
3. What do you mean by small scale fading?
4. What are the factors influencing small scale fading?
5. Define large scale propagation.
6. Differentiate the propagation effects with mobile radio.
7. Define Doppler shift.
8. Differentiate time selective and frequency selective channel.
9. Define coherence time and coherence bandwidth.
10. What do you mean by WSSUS channels?
11. What is free space propagation model?
12. Define EIRP.
13. Explain path loss?
14. What is intrinsic impedance& Brewster angle?
15. What is scattering?
16. Define radar cross section?
17. Name some of the outdoor propagation models?
18. Define indoor propagation models?
19. Mention some indoor propagation models?
20. What are merits and demerits of Okumara's model?
21. List the advantages and disadvantages of Hata model?
22. What is the necessity of link budget?
23. **Distinguish between narrow band and wideband systems. Nov. 2012**

24. What is link budget calculation? Nov. 2012
25. List the different types of wireless channels. May 2012
26. What is frequency selective fading? How to avoid fading problem? May 2012
27. Compute the Rayleigh distance of a square antenna with 20 dB gain. Nov. 2011
28. List out any two properties of wide band channel. Nov. 2011.
29.State the difference between narrowband and wideband systems. Nov./Dec.2013.
30.Find the far field distance for an antenna with maximum dimension of 1m and operating frequency of 900 MHZ. Nov./Dec.2013.

PART B-UNIT II

1. i. Describe any two methods of diffraction by multiple screens. (8) Nov. 2011
   ii. Discuss about ultra wide band channel. (8)
2. i. Compare coherence bandwidth and coherence time. (8) Nov. 2011
   ii. Discuss the mathematical formulation for narrowband and wideband system, with relevant figures. (8)
3. i. Explain the free space path loss and derive the gain expression. (8) May 2012
   ii. Describe in detail Two Ray Model propagation mechanism. (8)
4. i. Define the following: Auto correlation, Cross correlation and Power spectral density for narrow band fading model. (8) May 2012
   ii. What is the need for link calculation? Explain with suitable example. (8)
5. i. How the received signal strength is predicted using the free space propagation model? Explain. (10) Nov. 2012
   ii. Find the far-field distance for an antenna with maximum dimension of 1 m and operating frequency of 900 MHz. (6)
6. i. With system theoretic description, explain the characteristics of time-dispersive channels. (8)
   ii. Explain the three basic propagation mechanisms in a mobile communication system. (8)

7.a.i. Briefly explain the factors that influence small scale fading.(8). Nov./Dec.2013.

ii.If a transmitter produces 50 W of power , express the transmit power in units of dBM and dBW. If 50 W is applied to a unity gain antenna with a 900 MHz carrier frequency, find the received power in dBM at a free space distance of 100 m from the antenna. What is Pr (10 km)? assume unity gain for the receiver antenna.(8). Nov./Dec.2013.

b. i. Briefly explain the three basic propagation mechanisms which impact propagation in a mobile communication system (8). Nov./Dec.2013.

ii. What is Brewster angle? Calculate the Brewster angle for a wave impinging on ground having a premitivity of 4. Nov./Dec.2013.

UNIT 2

2.1. What are the propagation mechanisms of EM waves?

The four propagation mechanisms of EM waves are

i. Free space propagation
ii. Reflection
iii. Diffraction
iv. Scattering

2.2. What is the significance of propagation model?

The major significance of propagation model are:

i. Propagation model predicts the parameter of receiver.
ii. It predicts the average received signal strength at a given distance from the transmitter.

2.3. What do you mean by small scale fading?

Rapid fluctuations of the amplitude, phase as multipath delays of a radio signal over a short period of time is called small scale fading.

2.4. What are the factors influencing small scale fading?

The factors which influence small scale fading are:

Multipath propagation, Speed of the mobile, Speed of surrounding objects and the transmission bandwidth of the signal.

2.5. When does large scale propagation occur?

Large scale propagation occurs due to general terrain and the density and height of buildings and vegetation, large scale propagation occurs.

2.6. Differentiate the propagation effects with mobile radio.

Slow Fading	Fast Fading
Slow variations in the signal strength.	Rapid variations in the signal strength.
Mobile station (MS) moves slowly.	Local objects reflect the signal causes fast fading.
It occurs when the large reflectors and diffracting objects along the transmission paths are distant from the terminal. Eg. Rayleigh fading, Rician fading and Doppler shift	It occurs when the user terminal (MS) moves for short distances.

2.7. Define Doppler shift.

If the receiver is moving towards the source, then the zero crossings of the signal appear faster and the received frequency is higher.The opposite effect occurs if the receiver is moving away from the source. The resulting chance in frequency is known as the Doppler shift (f_D).

$$F_D = f_r – f_0 = -f_0V/C$$

Where f_0 -> transmission frequency

f_r -> received frequency

2.8. Differentiate time selective and frequency selective channel.

The gain and the signal strength of the received signal are time varying means then the channel is described as time selective channel. The frequency response of the time selective channel is constant so that frequency flat channel. The channel is time invariant but the impulse response of the channel show a frequency-dependent response so called frequency selective channel.

2.9. Define coherence time and coherence bandwidth.

Coherence time is the maximum duration for which the channel can be assumed to be approximately constant. It is the time separation of the two time domain samples. Coherence bandwidth is the frequency separation of the two frequency domain samples.

2.10. What do you mean by WSSUS channels?

In multipath channels, the gain and phase shift at one delay are uncorrelated with another delay is known as uncorrelated scattering of WSSUS.

2.11. What is free space propagation model?

The free space propagation model is used to predict received signal strength, when unobstructed line-of-sight path between transmitter & receiver. Friis free space equation is given by,

$$P_{\mathrm{RX}}(d) = P_{\mathrm{TX}} G_{\mathrm{TX}} G_{\mathrm{RX}} \left(\frac{\lambda}{4\pi d} \right)^2$$

The factor $(\lambda/4\pi d)^2$ is also known as the free space loss factor.

2.12.Define EIRP.

EIRP (Equivalent Isotropically Radiated Power) of a transmitting system in a given direction is defined as the transmitter power that would be needed, with an isotropic radiator, to produce the same power density in the given direction.

EIRP=PtGt

Where Pt-transmitted power in W

Gt-transmitting antenna gain

2.13. Explain path loss.

The path loss is defined as the difference (in dB) between the effective transmitted power and the received power. Path loss may or may not include the effect of the antenna gains.

$$PL(dB) = 10 \log P_t/P_r.$$

2.14. What is intrinsic impedance and Brewster angle?

Intrinsic impedance is defined by the ratio of electric to magnetic field for a uniform plane wave in the particular medium.

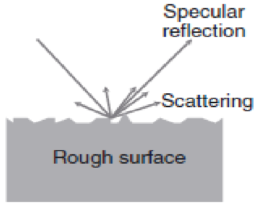
Brewster angle is the angle at which no reflection occurs in the origin. Brewster angle

is denoted by θ_B as shown below,

$$sin(\theta_B) = \sqrt{\frac{\epsilon_1}{\epsilon_1 + \epsilon_2}}.$$

2.15. What is scattering?

When a radio wave impinges on a rough surface, the reflected energy is spread out in all directions due to scattering.



2.16. Define radar cross section.

Radar Cross Section of a scattering object 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 & has units of squares meters

2.17. Name some of the outdoor propagation models?

Some of the commonly used outdoor propagation models are

i. Longely-Rice model
ii. Durkin's model
iii. Okumura model.

2.18. Define indoor propagation models.

The indoor propagation models are used to characterizing radio propagation inside the buildings. The distances covered are much smaller, and the variability of the environment is much greater for smaller range of Transmitter and receiver separation distances. Features such as lay-out of the building, the construction materials, and the building type strongly influence the propagation within the building.

2.19. Mention some indoor propagation models?

Some of the indoor propagation models are:

i. Long –distance path loss model
ii. Ericession multiple break point model
iii. Attenuation factor model.

2.20.What are merits and demerits of Okumara's model?

Merits:

Accuracy in parameter prediction.

Suitable for modern land mobile radio system.

Urban, suburban areas are analyzed.

Demerits:

Rural areas are not analyzed.

Analytical explanation is not enough.

2.21.List the advantages and disadvantages of Hata model?

Advantages: Suitable for large cell mobile system. Cell radius on the order of 1km is taken for analysis.

Disadvantages: Not suitable for PCS model. This model does not have any path specific correction.

2.22.What is the necessity of link budget?

The necessities of link budget are:

i. A link budget is the clearest and most intuitive way of computing the required Transmitter power. It tabulates all equations that connect the Transmitter power to the received SNR
ii. It is reliable for communications.
iii. It is used to ensure the sufficient receiver power is available.
iv. To meet the SNR requirement link budget is calculated.

PART A -Unit 3

1. List the advantages of digital modulation techniques.
2. What are the factors that influence the choice of digital modulation?
3. Define power efficiency and bandwidth efficiency.
4. What is QPSK?
5. Define offset QPSK and π/4 differential QPSK.
6. What is meant by MSK?
7. List the salient features of MSK scheme.
8. Why GMSK is preferred for multiuser, cellular communications?
9. How can we improve the performance of digital modulation under fading channels?
10. Write the advantages of MSK over QPSK.
11. Define M-ary transmission system?

12. What is quadrature modulation?
13. What is QAM?
14. Define QPSK?
15. What is linear modulation?
16. Define non linear modulation?
17. What is the need of Gaussian filter?
18. Mention some merits of MSK
19. Give some examples of linear modulation?
20. What are the techniques used to improve the received signal quality?
21. What is the need of equalization?
22. What is diversity?
23. Define spatial diversity?
24. Define STCM.
25. Define adaptive equalization?
26. Define training mode in an adaptive equalizer?
27. What is tracking mode in an adaptive equalizer?
28. Write a short note on linear equalizers and non linear equalizers?
29. Why non linear equalizers are preferred?
30. What are the nonlinear equalization methods used?
31. What are the factors used in adaptive algorithms?
32. Define diversity concept?
33. **Draw the mathematical link model for analysis of modulation schemes. Nov. 2011**
34. **What is OQPSK? Nov. 2011**
35. **List the advantages of QPSK. May 2012**
36. **Differentiate between MSK and GMSK. May 2012**
37. **Find the 3-dB bandwidth for a Gaussian low pass filter used to produce 0.25 GMSK with a channel data rate of R_b = 270 Kbps. What is the 90 % power bandwidth in the RF channel? Nov. 2012**
38. **What is slotted frequency hopping? Nov. 2012**
39. **Give the expression for bit error probability of Gaussian minimum shift keying modulation. Nov./Dec.2013.**
40. **What is fading and Doppler spread? Nov./Dec.2013.**

PART B-UNIT III

UNIT 3

1. Compute the ratio of signal power to adjacent channel interference when using (i) raised cosine pulses (ii) root raised cosine pulses with $\propto = 0.5$, when two considered signals have center frequencies 0 and 1.25 / T. (16) Nov. 2011

2. i. Discuss in detail any two demodulation techniques of minimum shift keying method. (8) Nov. 2011

   ii. Explain in detail about optimum receiver structure for non-coherent detection. (8)

3. Explain with neat signal diagrams, the modulation and demodulation technique of QPSK. (16) May 2012

4. i. Describe with a block diagram, offset – Quadrature phase shift keying and its advantages. (8) May 2012

   ii. Explain the concept of GMSK and mention its advantages. (8)

5. i. Briefly explain the structure of a wireless communication link. (6) Nov. 2012

   ii. With block diagram, explain the MSK transmitter and receiver. Derive an expression for MSK and its power spectrum. (10) Nov. 2012

6. Derive an expression for:

   i. M-ary Phase Shift Keying and (8)

   ii. M-ary Quadrture amplitude modulation. (8)

   Also derive an expression for their bit error probability.

7. i. Explain the Nyquist criterion for ISI cancellation. (8) Nov./Dec.2013.

   ii. Explain the performance of digital modulation in slow flat-fading channels.(8) Nov./Dec.2013.

8. i. Explain the QPSK transmission and detection techniques. (8) Nov./Dec.2013.

   ii. with transfer function, explain the raised cosine roll off filter. (8) Nov./Dec.2013.

UNIT 3

3.1.List the advantages of digital modulation techniques.

The advantages of digital modulation techniques are:

i. Immunity to channel noise and external interference.
ii. Flexibility operation of the system.
iii. Security of information.
iv. Reliable since digital circuits are used.
v. Multiplexing of various sources of information into a common format is possible.
vi. Error detection and correction is easy.

3.2.What are the factors that influence the choice of digital modulation?

The factors that influence the choice of digital modulation are:

i. Low BER at low received SNR.
ii. Better performance in multipath and fading conditions.
iii. Minimum bandwidth requirement.
iv. Better power efficiency.
v. Ease of implementation and low cost.

3.3.Define power efficiency and bandwidth efficiency.

Power efficiency describes the ability of a modulation technique to preserve the fidelity of the digital message at low power levels.

$\eta_p = E_b/N_0$ = Bit energy / Noise power spectral density

Ability of a modulation scheme to accommodate data within a limited bandwidth is called bandwidth efficiency.

$\eta_B = R/B$ = Datarate / Bandwidth in bps/Hz

3.4. What is QPSK?

The Quadrature Phase Shift Keying (QPSK) is a 4-ary PSK signal. The phase of the carrier in the QPSK takes 1 of 4 equally spaced shifts.

Two successive bits in the data sequence are grouped together.

1 symbol = 2 bits

This reduces bit rate and bandwidth of the channel.

Coherent QPSK = 2 x coherent BPSK system

The phase of the carrier takes on one of four equally spaced values such as $\pi/4$, $3\pi/4$, $5\pi/4$ and $7\pi/4$.

3.5. Define offset QPSK and π/4 differential QPSK.

In offset QPSK the amplitude of data pulses are kept constant. The time alignment of the even and odd bit streams are offset by one bit period in offset QPSK.

In $\pi/4$ QPSK, signaling points of the modulated signal are selected from two QPSK constellations which are shifted by $\pi/4$ with respect to each other. It is differentially encoded and detected so called $\pi/4$ differential QPSK.

3.6. What is meant by MSK?

A continuous phase FSK signal with a deviation ratio of one half is referred to as MSK. It is a spectrally efficient modulation scheme.

3.7. List the salient features of MSK scheme.

Salient features of MSK are:

i. It has constant envelope, smoother waveforms than QPSK.
ii. Relatively narrow bandwidth.
iii. Coherent detection suitable for satellite communications.
iv. Side lobes are zero outside the frequency band, so it has resistance to co-channel interference.

3.8. Why GMSK is preferred for multiuser, cellular communication?

It is a simple binary modulation scheme.

Premodulation is done by Gaussian pulse shaping filter, so side lobe levels are much reduced. GMSK has excellent power efficiency and spectral efficiency than FSK.

For the above reasons GMSK is preferred for multiuser, cellular communication.

3.9.How can we improve the performance of digital modulation under fading channels?

By the using of diversity technique, error control coding and equalization techniques performance of the digital modulation under fading channels are improved.

3.10.Write the advantages of MSK over QPSK.

Advantages of MSK over QPSK:

i. In QPSK the phase changes by 90degree or 180 degree .This creates abrupt amplitude variations in the waveform, Therefore bandwidth requirement of QPSK is more filters of other methods overcome these problems , but they have other side effects.
ii. MSK overcomes those problems. In MSK the output waveform is continuous in phase hence there are no abrupt changes in amplitude.

3.11.Define M-ary transmission system?

In digital modulations instead of transmitting one bit at a time, two or more bits are transmitted simultaneously. This is called M-ary transmission.

3.12.What is quadrature modulation?

Sometimes two or more quadrature carriers are used for modulation. It is called quadrature modulation.

3.13.What is QAM?

At high bit rates a combination of ASK and PSK is employed in order to minimize the errors in the received data. This method is known as “Quadrature Amplitude Modulation”.

3.14.**Define QPSK**

QPSK is defined as the multilevel modulation scheme in which four phase shifts are used for representing four different symbols.

3.15. What is linear modulation?

In linear modulation technique the amplitude of the transmitted signal varies linearly with the modulating digital signal. In general, linear modulation does not have a constant envelope.

3.16. Define non linear modulation.

In the non linear modulation the amplitude of the carrier is constant, regardless of the variation in the modulating signals.

Non-linear modulations may have either linear or constant envelopes depending on whether or not the baseband waveform is pulse shaped.

3.17. What is the need of Gaussian filter?

Need for Gaussian Filter:

i. Gaussian filter is used before the modulator to reduce the transmitted bandwidth of the signal.
ii. It uses less bandwidth than conventional FSK.

3.18. Mention some merits of MSK.

Merits of MSK:

i. Constant envelope
ii. Spectral efficiency
iii. Good BER performance
iv. Self-synchronizing capability
v. MSK is a spectrally efficient modulation scheme and is particularly attractive for use in mobile radio communication systems.

3.19. Give some examples of linear modulation.

Examples of linear modulation:

i. Pulse shaped QPSK
ii. OQPSK

3.20. What are the techniques used to improve the received signal quality?

Techniques such as,

- Equalization
- Diversity
- Channel coding

are used to improve the received signal quality.

3.21. What is the need of equalization?

Equalization can be used to compensate the Inter Symbol Interference created by multipath within time dispersion channel.

3.22. What is diversity?

Diversity is used to compensate the fading channel impairments and is usually implemented by using two or more receiving antennas. Diversity improves transmission performance by making use of more than one independently faded version of the transmitted signal.

3.23. Define spatial diversity.

The most common diversity technique is spatial diversity, whereby multiple antennas are strategically spaced and connected to a common receiving system. While one antenna sees a signal null, one of the other antenna may sees a signal peak, and the receiver is able to select the antenna with the best signals at any time.

3.24. Define STCM.

Channel coding can also be combined with diversity a technique called Space-Time Coded Modulation. The space-time coding is a bandwidth and power efficient method for wireless communication.

3.25. Define adaptive equalization?

To combine Inter Symbol Interference, the equalizer coefficients should change according to the channel status so as to break channel variations. Such an equalizer is called an adaptive equalizer since it adapts to the channel variations.

3.26. Define training mode in an adaptive equalizer?

First, a known fixed length training sequence is sent by the transmitter then the receivers equalizers may adapt to a proper setting of minimum bit error detection where the training sequence is a pseudo random binary signal or a fixed and prescribed bit pattern.

3.27. What is tracking mode in an adaptive equalizer?

Immediately following this training sequence the user data is sent and the adaptive equalizer at the receiver utilizes a recursive algorithm to evaluate the channel and estimate filter coefficients to compensate for the distortion created by multipath in the channel.

3.28. Write a short note on linear equalizers and non linear equalizers?

Linear equalizers: If the output d(t) is not used in the feedback path to adapt the equalizer. This type of equalizers is called linear equalizer.

Nonlinear equalizers: If the output d(t) is fed back to change the subsequent outputs of the equalizers is called non linear equalizers.

3.29. Why non linear equalizers are preferred?

The linear equalizers are very effective in equalizing channels where ISI is not severe.The severity of the ISI is directly related to the spectral characteristics. In this case that there are spectral noise in the transfer function of the effective channel, the additive noise at the receiver input will be dramatically enhanced by the linear equalizer. To overcome this problem non linear equalizers are used.

3.30. What are the nonlinear equalization methods used?

Commonly used non linear equalization methods are:

i. Decision feedback equalization
ii. Maximum likelihood symbol detection
iii. Maximum likelihood sequence estimation

3.31. What are the factors used in adaptive algorithms?

Rate of convergence

Mis adjustments

Computational complexity

3.32. Define diversity concept.

If one radio path undergoes a deep fade, another independent path may have a strong signal. By having more than one path to select from, both the instantaneous and average SNRs at the receiver may be improved often by as much as 20dB to 30dB. The principle of diversity is to ensure that the same information reaches the receiver on statistically independent channels.

PART A -Unit 4

1. How the link performance can be improved?
2. Why diversity and equalization techniques are used?
3. What is diversity?
4. Differentiate selection diversity and combining diversity.
5. Define switched diversity.
6. Define feedback or scanning diversity.
7. Define temporal diversity.

8. What is meant by frequency diversity?
9. Differentiate micro and macro diversity.
10. What is transmit diversity?
11. What is an equalizer?
12. What is linear and non-linear equalizer?
13. **Assume four branch diversity is used, where each branch receives an independent Rayleigh fading signal. If the average SNR is 20 dB, determine the probability that the SNR will drop below 10 dB. Compare this with the case of a single receiver without diversity. Nov. 2012**
14. **Define coding gain. Nov. 2012**
15. **List the different types of speech coding techniques. May 2012**
16. **State the significance of linear and decision feedback equalizer. May 2012**
17. **Mention any four common methods of micro diversity. Nov. 2011**
18. **Define hamming distance and Euclidean distance between two codes. Nov. 2011.**
19. **What is Diversity? Nov./Dec.2013.**
20. **What is Equalization? Nov./Dec.2013.**

PART B -UNIT IV

1. Explain the viterbi decoding scheme if the decoder input sequence is '010 000 100 001 011 110 001'. (16) Nov. 2011
2. i. With a neat block diagram, discuss the structure of a decision feedback equalizer. (8) Nov. 2011

   ii. Discuss linear predictive vocoder with block diagram. (8)
3. i. With a neat block diagram, explain the principle of diversity. (8) May 2012

   ii. Explain in detail Decision feedback equalizer. (8)
4. i. Explain any one method of channel coding. (8) May 2012

   ii. What are the advantages of speech coding? Explain any one technique of speech coding. (8)
5. Explain in detail about: Nov. 2012
   i. Polarization diversity (6)
   ii. Time diversity (5)
   iii. Frequency diversity (5)
6. i. Explain the basic idea about linear and behind decision feedback equalizers and derive an expression for its minimum mean square error. (8) Nov. 2012

   ii. With a suitable diagram, explain the channel coding and speech coding techniques. (8)
7. i. Explain in detail about Linear Equalizers (8). Nov./Dec.2013.

ii. Explain in detail about non linear equalizers. (8) Nov./Dec.2013.

8. i. With block diagram, explain the operation of a RAKe receiver. (8) Nov./Dec.2013.
ii. Briefly explain the frequency domain coding for speech signals. (8). Nov./Dec.2013.

UNIT 4

4.1. How the link performance can be improved?

Link performance can be improved by various techniques such as

i. Equalization
ii. Diversity
iii. Channel coding

4.2. Why diversity and equalization techniques are used?

To reduce ISI, Equalization technique is used. Diversity is used to reduce fading effects.

4.3.What is diversity?

Signal is transmitted by more than one antenna via channel. It ensures that the same information reaches the receiver on statistically independent channels.

4.4.Differentiate selection diversity and combining diversity.

Selection Diversity	Combining Diversity
The best signal is selected and processed while all other signals are discarded.	All signals are combined before processing and the combined signal is decoded.
Simple circuits are used.	At individual receiver, phasing circuits are

	needed.
None of the signal is not in acceptable SNR.	It works well.

4.5. Define Switched Diversity

If the signal level falls below the threshold, then the receiver switches to a new antenna which is called as switched diversity.

4.6. Define feedback or scanning diversity.

All the signals are scanned in a fixed sequence until one signal is found to be above a predetermined threshold.

4.7. Define temporal diversity.

Wireless propagation channel is time variant, so for sufficient decorrelation, the temporal distance between antennas must be atleast the half of maximum Doppler frequency.

4.8.What is meant by frequency diversity?

Correlation is increased by transmitting information on more than one carrier frequency. Frequencies are separated by more than one coherence bandwidth of the channel. So the signals will not experience same fades.

4.9.Differentiate micro and macro diversity.

Micro diversity	**Macro diversity**
Used to reduce small scale fading effects.	Used to reduce large scale fading effects.
Multiple reflection causes deep fading. This effect is reduced.	Deep shadow causes fading. This effect is reduced.

BS-MS are separated by small distance.	BS-MS are separated by large distance.

4.10.What is transmit diversity?

Diversity effect is achieved by transmitting signals from several transmit antenna.

4.11.What is an equalizer?

Equalizer is a linear pulse shaping circuit which is used to reduce ISI.

4.12.What is linear and non-linear equalizer?

Linear equalizer: the current and past values of the received signal are linearly weighted by the filter coefficients and summed to produce the output. No feedback path is used. Simple and easy to implement. Not suitable for severely distorted channel. Noise power signal is enhanced.

Nonlinear equalizer: If the past decisions are correct, then the ISI contributed by present symbol can be cancelled exactly, feedback path is used. Suitable for severely distorted channel. Noise power signal is not enhanced. Complex in structure.

channels with low SNR. Suffers from error propagation.

PART A -Unit 5

1. Write the two types of spread spectrum.
2. What do you mean by spread spectrum?
3. What is PN sequence?
4. When is the PN sequence called as maximal length sequence?
5. Write the properties which a PN sequence should have.
6. Define chip duration and chip rate.
7. What do you mean by processing gain of a spread spectrum?
8. List the advantages and disadvantages of DS-SS.

9. Define jamming and jamming margin.
10. What is meant by anti-jamming?
11. List the advantages and disadvantages of FH-SS.
12. List the types of FH-SS.
13. Compare slow and fast FH-SS.
14. Compare DS-SS and FH-SS.
15. State the principles of CDMA.
16. How the capacity can be increased in CDMA?
17. Write short notes on OFDM.
18. Why cyclic prefix?
19. Write the goals of GSM standard.
20. What is W-CDMA?
21. What are the services offered by GSM?
22. **Discuss the principle of OFDM modulation scheme. Nov. 2011**
23. **Give three important functional blocks of GSM system. Nov. 2011**
24. **State the effects of multipath propagation on CDMA. May 2012**
25. **List a few wireless network standards. May 2012**
26. **What is duplexing? Nov. 2012**
27. **What is the speech codes used in IS-95 system? Why?**
28. **What is a PN sequence? Give its significance in spread spectrum modulation technique. Nov./Dec.2013.**
29. **What is DECT? Nov./Dec.2013.**

PART B-UNIT 5

1. i. Explain the principle of direct sequence spread spectrum technique.(8) Nov. 2011

   ii. Discuss some methods to increase the capacity of wireless communication system. (8)

2. i. Explain in detail about the GSM logic channels. (8) Nov. 2011

   ii. Explain the block diagram of IS-95 transmitter. (8)

3. Explain: Code Division Multiple Access (CDMA) and compare its performance with TDMA. (16) May 2012

4. What is orthogonal frequency division multiplexing? Explain OFDM technique and mention its merits, demerits and application. (16) May 2012

5. i. Discuss in detail about cellular code division multiple access systems with neat diagrams. (8) Nov. 2012

   ii. Write a short notes on transceiver implementation. (8)

6. i. Explain with neat diagram of orthogonal frequency division multiplexing.

   (8) Nov. 2012

   ii. Write a note on second generation and third generation wireless networks and standards. (8)

7. Discuss in detail about second generation(2G) and third generation (3G) wireless networks and standards. (16). Nov./Dec.2013.

8.i. Explain in detail about direct sequence spread spectrum technique (8) Nov./Dec.2013.

ii. Explain in detail about frequency hopped spread spectrum technique. (8). Nov./Dec.2013.

UNIT 5

5.1. Write the two types of spread spectrum?

Types of spread spectrum are:

Direct Sequence Spread Spectrum (DS-SS)

Frequency hop spread spectrum (FH-SS)

5.2. What do you mean by spread spectrum?

Spread spectrum multiple access uses signals which have a transmission bandwidth whose magnitude is greater than the minimum required RF bandwidth. A pseudo noise (PN) sequence converts a narrowband signal to a wideband noise like signal before transmission

5.3. What is PN sequence?

Pseudo noise sequence is a coded sequence of 1's and 0's with autocorrelation properties.

5.4. When is the PN sequence called as maximal length sequence?

When the pseudo-noise sequence generated by linear feedback shift register has the length (N) of 2^m-1 where m is number of stages in shift register is called maximal length sequence.

5.5. Write the properties which a PN sequence should have.

Properties of PN sequence are:

i. Balance property
ii. Run property
iii. Correlation property

5.6. Define chip duration and chip rate.

The duration of every bit in PN sequence is known as chip duration. The number of bits (chips) per second is called chip rate.

5.7. What do you mean by processing gain of a spread spectrum?

$$\text{Processing gain} = \frac{\text{Bandwidth of spreaded data signal}}{\text{Bandwidth of unspreaded data signal}}$$

$$= \frac{\text{Bit Duration}}{\text{Chip duration}} = \frac{\text{Bandwidth}}{\text{Information rate}}$$

5.8.List the advantages and disadvantages of DS-SS.

Advantages of DS-SS:

i. The performance of DS-SS in presence of noise is superior to FH-SS.
ii. Good antijamming capability.
iii. Low multipath interference.

Disadvantages of DS-SS:

i. Poor synchronization.
ii. Requires large bandwidth.
iii. Long acquisition time so that the system is slow.

5.9. Define jamming and jamming margin.

Jamming is a multitone or powerful broad band noise. It is the ratio of the average interference power and the signal power.

Jamming margin in dB as the difference between the processing gain in dB and minimum SNR in dB.

5.10. What is meant by anti-jamming?

With the help of spread spectrum method, the transmitted signals are spread over the mid frequency band. Hence these signals appear as noise. Then it becomes difficult for the jammers to attack our signal. This method is called antijamming.

5.11.List the advantages and disadvantages of FH-SS.

Advantages of FH-SS:

i. High processing gain than DS-SS.
ii. Shorter acquisition time makes the system fast.

Disadvantages of FH-SS:

i. FH-SS requires large bandwidth.

ii. Circuit used for FH-SS is complex. Expensive frequency synthesizers are required.

5.12.List the types of FH-SS.

Types of FH-SS are:

i. Slow frequency hopping
ii. Fast frequency hopping

5.13.Compare slow and fast FH-SS.

Slow FH-SS	**Fast FH-SS**
More than one symbol is transmitted per hop.	One symbol is transmitted with more than one hops.
Chip rate is equal to the symbol rate.	Chip rate is equal to the hop rate.
Same carrier frequency is used to transmit one or more symbols.	One symbol is transmitted over multiple carriers in different hops.

5.14.Compare DS-SS and FH-SS.

DS-SS	**FH-SS**
PN sequence is multiplied with narrow band signal.	Data bits are transmitted in different frequency slots which are changed by PN sequence.
Modulation used is BPSK-coherent.	Modulation used is M-ary FSK noncoherent.

	Faster than DS-SS.
Fixed chip rate.	Variable chip rate.
Long acquisition time is required.	Short acquisition time.
Effect of distance is high.	Effect of distance is less.

5.15. State the principles of CDMA.

Principles of CDMA:

i. Many users share the same frequency.
ii. Each user is assigned a different spreading code.

5.16. How the capacity can be increased in CDMA?

Capacity in CDMA can be increased by

i. Quiet periods during speech transmission is shared by many users.
ii. Flexible data rate.
iii. Soft capacity.
iv. Error Correction coding used.

5.17.Write short notes on OFDM.

OFDM splits the information into N parallel streams which are modulated by N distinct carriers and then transmitted. In order to separate the subcarriers by the receiver, they have to be orthogonal.

5.18. Why cyclic prefix?

In delay dispersive channel, inter carrier interference occur. To overcome the effect of inter carrier interference and ISI, cyclic prefix is introduced. It is a cyclically extended guard interval whereby each symbol sequence is preceded by a periodic extension of the sequence itself.

5.19. Write the goals of GSM standard.

Better and more efficient technical solution for wireless communication. Single standard was to be realized all over Europe enabling roaming across borders.

5.20. What is W-CDMA?

It is a 3G wireless standard for cellular telephony. It provides better efficiency, higher peak rates upto 2 Mbps. Bandwidth of 5 MHz. Supports multimedia applications.

5.21.What are the services offered by GSM?

Services offered by GSM are:

i. Telephone services
ii. Bearer or Data services
iii. Supplementary services

WIRELESS COMMUNICATION

PART B

UNIT 1

7. i. Compare and contrast wired and wireless communication. (8) Nov. 2011
   ii. Discuss briefly about the requirements of services for a wireless system.(8)
8. i. Discuss in detail the constructive and destructive interferences. (8) Nov. 2011
   ii. Explain how Inter Symbol Interference is caused and how it is eliminated. (8)
9. i. Explain in detail Wide Area Data Services and Broadband Wireless Access services offered to wireless networks. (10) May 2012
   ii. What are paging systems? Explain. (6)
10. i. With a neat block diagram, explain the cellular network architecture.(10) May 2012
    ii. Explain any one type of Multiple Access scheme. (6)
11. i. Explain about the factors that influence small scale fading. (10) Nov 2012

    ii. Find the average fade duration for threshold levels ρ = 0.01, ρ = 0.1 and ρ = 1, when the Doppler frequency is 200 Hz.

12. i. Write a note on Noise and Interference Limited Systems. (8) Nov. 2012

ii. Discuss the principles of cellular networks. (8)

1.Compare and contrast wired and wireless communication (8) Nov. 2011

Wired communication	Wireless communication
The communication takes place over a more or less stable medium like copper wires or optical fibers. The properties of the medium are well defined and time-invariant.	Due to user mobility as well as multipath propagation, the transmission medium varies strongly with time.
The range over which communications can be performed without repeater stations is mostly limited by attenuation by the medium (and thus noise); for optical fibers, the distortion of transmitted pulses can also limit the speed of data transmission.	The range that can be covered is limited both by the transmission medium (attenuation, fading, and signal distortion) and by the requirements of spectral efficiency (cell size).

Increasing the transmission capacity can be achieved by using a different frequency on an existing cable, and/or by stringing new cables.	Increasing the transmit capacity must be achieved by more sophisticated transceiver concepts and smaller cell sizes (in cellular systems), as the amount of available spectrum is limited.
Interference and crosstalk from other users either do not happen or the properties of the interference are stationary.	Interference and crosstalk from other users are inherent in the principle of cellular communications. Due to the mobility of the users, they also are time-variant.
The delay in the transmission process is also constant, determined by the length of the cable and the group delay of possible repeater amplifiers.	The delay of the transmission depends partly on the distance between base station and Mobile Station (MS), and is thus time-variant.
The *Bit Error Rate* (BER) decreases strongly (approximately exponentially) with increasing *Signal-to-Noise Ratio* (SNR). This means that a relatively small increase in transmit power can greatly decrease the error rate.	For simple systems, the average BER decreases only slowly (linearly) with increasing average SNR. Increasing the transmit power usually does not lead to a significant reduction in BER. However, more sophisticated signal processing helps.
Due to the well-behaved transmission medium, the quality of wired transmission is generally high.	Due to the difficult medium, transmission quality is generally low unless special measures are used.
Jamming and interception of dedicated links with wired transmission is almost impossible without consent by the network operator	Jamming a wireless link is straightforward, unless special measures are taken. Interception of the on-air signal is possible. Encryption is therefore necessary to prevent unauthorized use of the information.
Establishing a link is location based. In other words, a link is established from one outlet to another, independent of which *person* is connected to the outlet.	Establishing a connection is based on the (mobile) equipment, usually associated with a specific person. The connection is not associated with a fixed location.
Power is either provided through the communications network itself (e.g., for traditional landline telephones), or from traditional power mains (e.g., fax). In neither case is energy consumption a major concern for the designer of the device.	MSs use rechargeable or one-way batteries. Energy efficiency is thus a major concern.

2.Explain about the factors that influence small scale fading. (10) Nov 2012

Many physical factors in the radio propagation channel influence small scale fading. These include the following:

Multipath propagation — The presence of reflecting objects and scatterers in the channel creates a constantly changing environment that dissipates the signal energy in amplitude, phase, and time. These effects result in multiple versions of the transmitted signal that arrive at the receiving antenna, displaced with respect to one another in time and spatial orientation. The random phase and amplitudes of the different multipath components cause fluctuations in signal strength, thereby inducing small-scale fading, signal distortion, or both. Multipath propagation often lengthens the time required for the baseband portion of the signal to reach the receiver which can cause signal smearing due to intersymbol interference.

Speed of the mobile — The relative motion between the base station and the mobile results in random frequency modulation due to different Doppler shifts on each of the multipath components. Doppler shift will be positive or negative depending on whether the mobile receiver is moving toward or away from the base station.

Speed of surrounding objects — If objects in the radio channel are in motion, they induce a time varying Doppler shift on in multipath components. If the surrounding objects move at a greater rate than the mobile, then this effect dominates the small-scale fading. Otherwise, motion of surrounding objects may be ignored, and only the speed of the mobile need be considered.

The transmission bandwidth of the signal — If the transmitted radio signal bandwidth is greater than the "bandwidth" of the multipath channel, the received signal will be distorted, but the received signal strength will not fade much over a local area (i.e., the small-scale signal fading will not be significant). As will be shown, the bandwidth of the channel can be quantified by the coherence bandwidth which is related to the specific multipath structure of the channel. The coherence bandwidth is a measure of the maximum frequency difference for which signals are still strongly correlated in amplitude. If the transmitted signal has a narrow bandwidth as compared to the
channel, the amplitude of the signal will change rapidly, but the signal will not be distorted in time. Thus, the statistics of small-scale signal strength and the likelihood of signal smearing appearing over small-scale distances are very much related to the specific amplitudes and delays of the multipath channel, as well as the bandwidth of the transmitted signal.

3.Write a note on Noise and Interference Limited Systems. (8) Nov. 2012

Noise-Limited Systems

Wireless systems are required to provide a certain minimum transmission quality . This transmission quality in turn requires a minimum *Signal-to-Noise Ratio* (SNR) at the receiver (RX). Consider now a situation where only a single BS transmits, and a Mobile Station (MS) receives; thus, the performance of the system is determined only by the strength of the (useful) signal and the noise. As the MS moves further away from the BS, the received signal power decreases, and at a certain distance, the SNR does not achieve the required threshold for reliable communications. Therefore, the range of the system is noise limited; equivalently, we can call it *signal power limited* . Depending on the interpretation, it is too much noise or too little signal power that leads to bad link quality.

$$P_{RX} = P_{TX}G_{RX}G_{TX}(\lambda / 4\pi d)^2 \quad \text{--- (1)}$$

where G_{RX} and G_{TX} are the gains of the receive and transmit antennas, respectively, λ is the wavelength, and P_{TX} is the transmit power

The noise that disturbs the signal can consist of several components, as follows

***Thermal noise*:** The power spectral density of thermal noise depends on the environmental temperature T_e that the antenna "sees." The temperature of the Earth is around 300 K, while the temperature of the (cold) sky is approximately $T_e \approx 4$K (the temperature in the direction of the Sun is of course much higher). As a first approximation, it is usually assumed that the environmental temperature is isotropically 300 K. Noise power spectral density is then

$$N_0 = k_B T_e \quad \text{--- (2)}$$

where k_B is Boltzmann's constant, $k_B = 1.38 \cdot 10^{-23}$ J/K, and

the noise power is $P_n = N_0 B$---(3)
where B is RX bandwidth (in units of Hz). It is common to write Eq. (.2) using logarithmic units (power P expressed in units of dBm is 10 log10 (P/1 mW)) $N_0 = -174$ dBm/Hz --- (4)

This means that the noise power contained in a 1-Hz bandwidth is −174 dBm. The noise power contained in bandwidth B is

$-174 + 10 \log10(B)$ dB.---(5)
The logarithm of bandwidth B, specifically $10 \log10(B)$, has the units dBHz.

***Man-made noise*:** We can distinguish two types of man-made noise:

(a) *Spurious emissions*: Many electrical appliances as well as radio transmitters (TXs) designed for other frequency bands have spurious emissions over a large bandwidth that includes the frequency range in which wireless communications systems operate. For urban outdoor environments, car ignitions and other impulse sources are especially significant sources of noise. In contrast to thermal noise, the noise created by impulse sources decreases with Frequency At 150 MHz, it can be 20 dB stronger than thermal noise; at 900 MHz, it is typically 10 dB stronger. At Universal Mobile Telecommunications System (UMTS) frequencies, Note that frequency regulators in most countries impose limits on "spurious" or "out-of-band" emissions for *all* electrical devices. Furthermore, for communications operating in licensed bands, such spurious emissions are the only source of man-made noise. It lies in the nature of the license (for which the license holder usually has paid) that no other intentional emitters are allowed to operate in this band. In contrast to thermal noise, man-made noise is not necessarily Gaussian distributed. However, as a matter of convenience, most system-planning tools, as well as theoretical designs, assume *Gaussianity* anyway.

(b**)** ***Other intentional emission sources***: Several wireless communications systems operate in unlicensed bands. In these bands, everybody is allowed to operate (emit electromagnetic radiation) as long as certain restrictions with respect to transmit power, etc. are fulfilled. The most important of these bands is the 2.45-GHz Industrial, Scientific, and Medical (ISM) band. The amount of interference in these bands can be considerable.

Receiver noise: The amplifiers and mixers in the RX are noisy, and thus increase the total noise power. This effect is described by the noise figure F, which is defined as the SNR at the RX input (typically after down conversion to baseband) divided by the SNR at the RX output. As the amplifiers have gain, noise added in the later stages does not have as much of an impact as noise added in the first stage of the RX. Mathematically, the total noise figure F_{eq} of a cascade of components is

$$F_{eq} = F_1 + \frac{F_2 - 1}{G_1} + \frac{F_3 - 1}{G_1 G_2} + \cdot\ \cdot\ \cdot$$

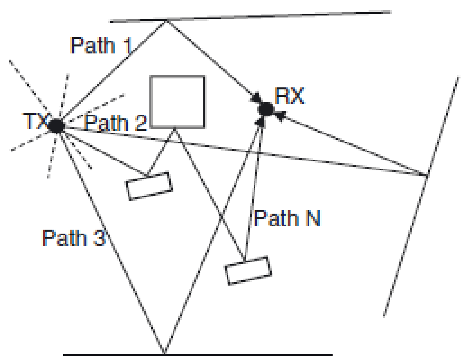
where F_i and G_i are noise figures and noise gains of the individual stages in absolute units (not in decibels (dB)). Note that for this equation, passive components, like attenuators with gain $m < 1$, can be interpreted as *either* having a noise figure of $F = 1/m$ and unit gain of $G = 1$, *or* unit noise figure $F = 1$, and gain $G = m$.

Interference-Limited Systems

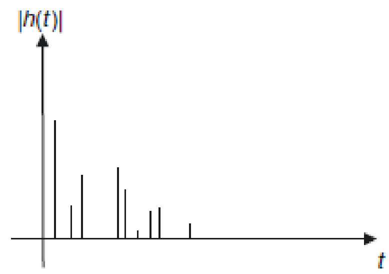
Consider now the case that the interference is so strong that it completely dominates the performance, so that the noise can be neglected. Let a BS cover an area (cell) that is approximately described by a circle with radius R and center at the location of the BS. Furthermore, there is an interfering TX at distance D from the "desired" BS, which operates at the same frequency, and with the same transmit power. How large does D have to be in order to guarantee satisfactory transmission quality 90% of the time, assuming that the MS is at the cell boundary (worst case)? As a first approximation, we treat the interference as Gaussian. This allows us to treat the interference as equivalent noise, and the minimum SIR, SIR_{min}, takes on the same values as SNR_{min} in the noise-limited case One difference between interference and noise lies in the fact that interference suffers from fading, while the noise power is typically constant (averaged over a short time interval). For determination of the fading margin, we thus have to account for the fact that (i) the desired signal is weaker than its median value during 50% of the time and (ii) the interfering signal is stronger than its median value 50% of the time. Mathematically speaking, the cumulative distribution function of the SIR is the probability that the ratio of two random variables is larger than a certain value in x% of all cases (where x is the percentage of locations in which transmission quality is satisfactory), As a first approximation, we can add the fading margin for the desired signal (i.e., the additional power we have to transmit to make sure that the desired signal level exceeds a certain value, x%, of the time, instead of 50%) and the fading margin of the interference –i.e., the power *reduction* to make sure that the interference exceeds a certain value only $(100 - x)$% of the time, instead of 50% of the time. This results in an overestimation of the true fading margin. Therefore, if we use that value in system planning, we are on the safe side.

4. Explain how Inter- Symbol Interference is caused and how it is eliminated. (8) Nov. 2011

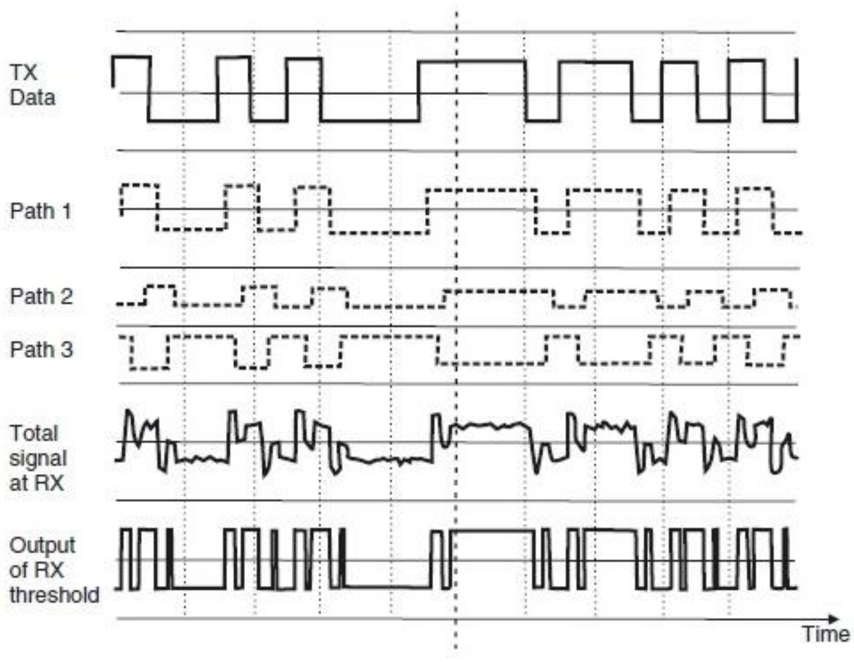
The runtimes for different MPCs are different. This can lead to different phases of MPCs, which lead to interference in narrowband systems. In a system with large bandwidth, and thus good resolution in the time domain,3 the major consequence is signal dispersion: in other words, the impulse response of the channel is not a single delta pulse but rather a sequence of pulses (corresponding to different MPCs), each of which has a distinct arrival time in addition to having a different amplitude and phase). This signal dispersion leads to Inter-Symbol Interference (ISI) at the RX. MPCs with long runtimes, carrying information from bit k, and MPCs with short runtimes, carrying contributions from bit $k + 1$ arrive at the RX at the same time, and interfere with each other. Assuming that no special measures4 are taken, this ISI leads to errors that cannot be eliminated by simply increasing the transmit power, and are therefore often called *irreducible errors*.



Multi path components with different runtimes



Channel impulse response



Intersymbol interference.

ISI is essentially determined by the ratio between symbol duration and the duration of the impulse response of the channel. This implies that ISI is not only more important for higher data rates but also for multiple access methods that lead to an increase in transmitted *peak* data rate (e.g., time division multiple access,). Finally, it is also noteworthy that ISI can even play a role when the duration of the impulse response is *shorter* (but not *much* shorter) than bit duration.

5. Explain any one type of Multiple Access scheme. (6) May 2012

Frequency Division Multiple Access (FDMA)

Frequency division multiple access (FDMA) assigns individual channels to individual users. It can be seen from Figure 1 that each user is allocated a unique frequency band or channel. These channels are assigned on demand to users who request service. During the period of the call, no other user can share the same frequency band. In FDD systems, the users are assigned a channel as a pair of frequencies; one frequency is used for the forward channel, while the other frequency is used for the reverse channel. The features of FDMA are as follows:

• The FDMA channel carries only one phone circuit at a time.
If an FDMA channel is not in use, then it sits idle and cannot be used by other users to increase or share capacity. It is essentially a wasted resource.

• After the assignment of a voice channel, the base station and the mobile transmit simultaneously and continuously.

• The bandwidths of FDMA channels are relatively narrow (30 kHz) as each channel supports only one circuit per carrier that is, FDMA is usually implemented in narrowband systems.

• The symbol time is large as compared to the average delay spread. This implies that the amount of intersymbol interference is low and, thus, little or no equalization is required in FDMA narrowband systems.

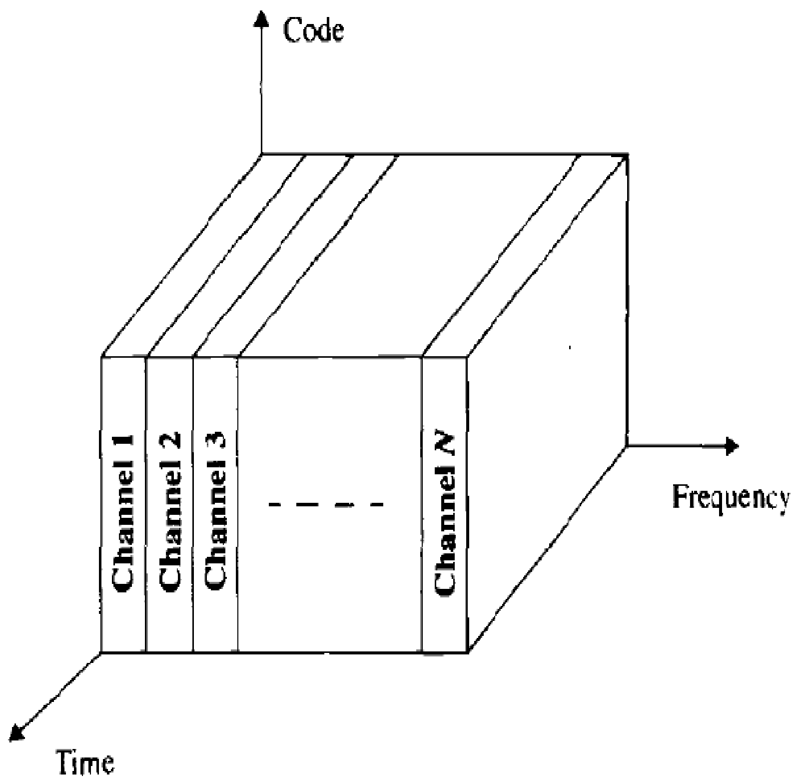
• The complexity of FDMA mobile systems is lower when compared to TDMA systems, though this is changing as digital signal processing methods improve for TDMA.

• Since FDMA is a continuous transmission scheme, fewer bits are needed for overhead purposes (such as synchronization and framing bits) as compared to TDMA.

• FDMA systems have higher cell site system costs as compared to TDMA systems, because of the single channel per carrier design, and the need to use costly band pass filters to eliminate spurious radiation at the base station.
• The FDMA mobile unit uses duplexers since both the transmitter and receiver operate at the same time. This results in an increase in the cost of FDMA subscriber and base stations.

• FDMA requires tight RF filtering to minimize adjacent channel interference.

Nonlinear Effects in FDMA— In a FDMA system, many channels share the same antenna at the base station. The power amplifiers or the power combiners, when operated at or near saturation for maximum power efficiency, are nonlinear. The nonlinearities cause signal spreading in the frequency domain and generate inter-modulation (IM) frequencies. IM is undesired RF radiation which can interfere with other channels in the FDMA systems, spreading of the spectrum results in adjacent-channel interference. Inter-modulation is the generation of undesirable harmonics. Harmonics generated outside the mobile radio band cause interference to adjacent services, while those present inside the band cause interference to other users in the mobile system



Frequency Division Multiple Access

6.What are paging systems? Explain. (6) May 2012

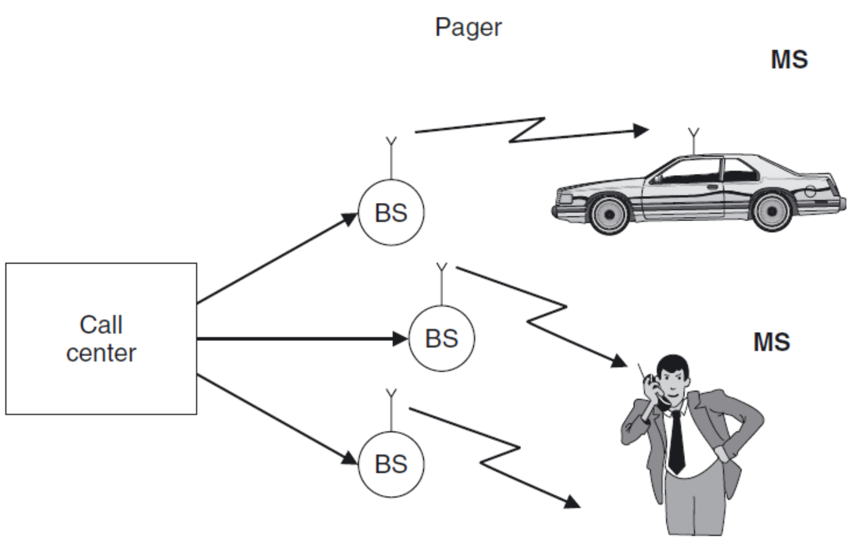
Similar to broadcast, paging systems are unidirectional wireless communications systems. They are characterized by the following properties .

- The user can only receive information, but cannot transmit. Consequently, a "call" (message) can only be initiated by the call center, not by the user.

- The information is intended for, and received by, only a single user.
- The amount of transmitted information is very small. Originally, the received information

Consisted of a single bit of information, which indicated to the user that “somebody has sent you a message.” The user then had to make a phone call (usually from a payphone) to the call center, where a human operator repeated the content of the waiting message. Later, paging systems became more sophisticated, allowing the transmission of short messages (e.g., a different phone number that should be called, or the nature of an emergency). Still, the amount of information was rather limited.

Due to the unidirectional nature of the communications, and the small amount of information, the bandwidth required for this service is small. This in turn allows the service to operate at lower carrier frequencies – e.g., 150MHz – where only small amounts of spectrum are available. As we will see later on, such lower carrier frequencies make it much easier to achieve good coverage of a large area with just a few transmitters. Pagers were very popular during the 1980s and early 1990s. For some professional groups, like doctors, they were essential tools of the trade, allowing them to react to emergencies in shorter time. However, the success of cellular telephony has considerably reduced their appeal. Cellphones allow provision of all the services of a pager, plus many other features as well. The main appeal of paging systems, after the year 2000, lies in the better area coverage that they can achieve.

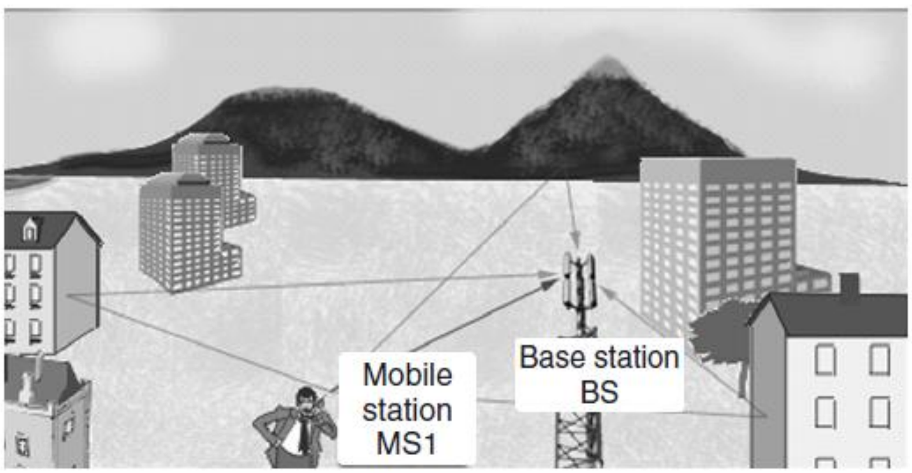


Principle of a pager.

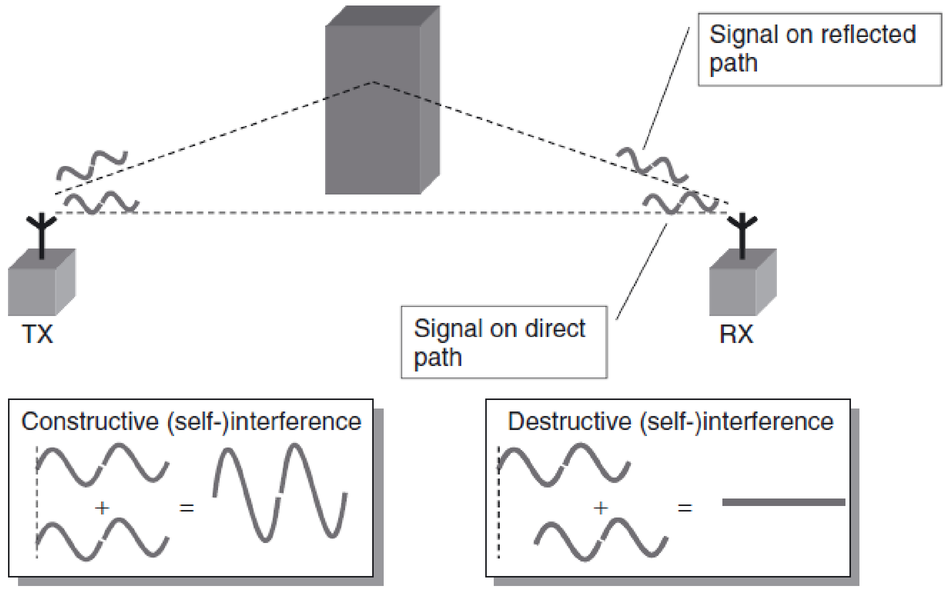
Pagers were very popular during the 1980s and early 1990s. For some professional groups, like doctors, they were essential tools of the trade, allowing them to react to emergencies in shorter time. However, the success of cellular telephony has considerably reduced their appeal. Cellphones allow provision of all the services of a pager, plus many other features as well. The main appeal of paging systems, after the year 2000, lies in the better area coverage that they can achieve.

7. Discuss in detail the constructive and destructive interferences. (8) Nov. 2011

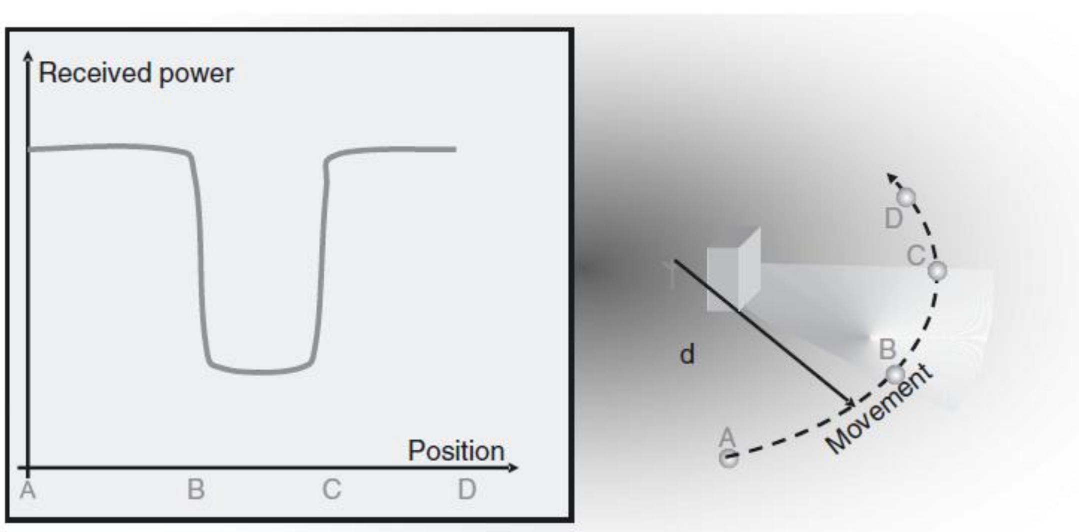
For wireless communications, the transmission medium is the radio channel between transmitter TX and receiver RX. The signal can get from the TX to the RX via a number of different propagation paths. In some cases, a Line Of Sight (LOS) connection might exist between TX and RX. Furthermore, the signal can get from the TX to the RX by being reflected at or diffracted by different *Interacting Objects* (IOs) in the environment: houses, mountains (for outdoor environments), windows, walls, etc. The number of these possible propagation paths are very large. As shown in figure 1 , each of the paths has a distinct amplitude, delay (runtime of the signal), direction of departure from the TX, and direction of arrival; most importantly, the components have different phase shifts with respect to each other. In the following, we discuss some implications of the multipath propagation for system design.



Multipath propagation.



Principle of small-scale fading.



The principle of shadowing.

Fading

A simple RX cannot distinguish between the different *Multi Path Components* (MPCs); it just adds them up, so that they interfere with each other. The interference between them can be constructive or destructive, depending on the phases of the MPCs, (Figure.2). The phases, in turn, depend mostly on the run length of the MPC, and thus on the position of the Mobile Station (MS) and the IOs. For this reason, the interference, and thus the amplitude of the total signal, changes with time if either TX, RX, or IOs is moving. This effect – namely, the changing of the total signal amplitude due to interference of the different MPCs – is called *small-scale fading*.

At 2-GHz carrier frequency, a movement by less than 10 cm can already effect a change from constructive to destructive interference and vice versa. In other words, even a small movement can result in a large change in signal amplitude. A similar effect is known to all owners of car radios – moving the car by less than 1m (e.g., in stop-and-go traffic) can greatly affect the quality of the received signal. For cellphones, it can often be sufficient to move one step in order to improve signal quality. As an additional effect, the amplitudes of each separate MPC change with time (or with location). Obstacles can lead to a shadowing of one or several MPCs. Imagine, e.g., the MS in Figure3 that at first (at position A) has LOS to the Base Station (BS). As the MS moves behind the high-rise building (at position B), the amplitude of the component that propagates along the direct connection (LOS) between BS and MS greatly decreases. This is due to the fact that the MS is now in the radio shadow of the high-rise building, and any wave going through or around that building is greatly attenuated – an effect called *shadowing*. Of course, shadowing can occur not only for an LOS component but also for *any* MPC. Note also that obstacles do not throw "sharp" shadows: the transition from the "light" (i.e., LOS) zone to the "dark" (shadowed) zone is gradual. The MS has to move over large distances (from a few meters up to several hundreds of meters) to move from the light to the dark zone. For this reason, shadowing gives rise to *large-scale fading*.

8.Discuss briefly about the requirements of services for a wireless system.(8) Nov. 2011
Requirements for the Services

Data Rate

Data rates for wireless services span the gamut from a few bits per second to several gigabit per second, depending on the application *Sensor networks* usually require data rates from a few bits per second to about 1 kbit/s. Typically, a sensor measures some critical parameter, like temperature, speed, etc., and transmits the current value (which corresponds to just a few bits) at intervals that can range from milliseconds to several hours. Higher data rates are often required for the central nodes of sensor networks that collect the information from a large number of sensors and forward it for further processing. In that case, data rates of up to 10Mbit/s can be required. These "central nodes" show more similarity to WLANs or fixed wireless access.

• *Speech communications* usually require between 5 and 64 kbit/s depending on the required quality and the amount of compression.

Elementary data services require between 10 and 100 kbit/s. One category of these services uses the display of the cellphone to provide Internet-like information. Another type of data service provides a wireless mobile connection to laptop computers. In this case, speeds that are at least comparable with dial-up (around 50 kbit/s) are demanded by most users, though elementary services with 10 kbit/s

Communications between computer peripherals and similar devices: for the replacement of cables that link computer peripherals, like mouse and keyboard, to the computer (or similarly for cellphones), wireless links with data rates around 1Mbit/s are used. The functionality of these links is similar to the previously popular infrared links, but usually provides higher reliability.

High-speed data services: WLANs and 3G cellular systems are used to provide fast Internet access, with speeds that range from 0.5 to 100 Mbit/s (currently under development).

Personal Area Networks (PANs) is a newly coined term that refers mostly to the range of a wireless network (up to 10m), but often also has the connotation of high data rates (over 100 Mbit/s), mostly for linking the components of consumer entertainment systems (streaming video from computer or DVD player to a TV) or high-speed computer connections (wireless Universal Serial Bus (USB)).

Range and Number of Users

Another distinction among the different networks is the range and the number of users that they serve. By "range," we mean here the distance between one transmitter and receiver. The coverage area of a system can be made almost independent of the range, by just combining a larger number of BSs into one big network.

Body Area Networks (BANs) cover the communication between different devices attached to one body – e.g., from a cellphone in a hip holster to a headset attached to the ear. The range is thus on the order of 1m. BANs are often subsumed into PANs.
Personal Area Networks include networks that achieve distances of up to or about 10 m, covering the "personal space" of one user. Examples are networks linking components of computers and home entertainment systems. Due to the small range, the number of devices within a PAN is small, and all are associated with a single "owner." Also, the number of overlapping PANs (i.e.sharing the same space or room) is small – usually less than five. That makes cell planning and multiple access much simpler.

WLANs, as well as cordless telephones cover still larger ranges of up to 100 m. The number of users is usually limited to about 10. When much larger numbers occur (e.g., at conferences or meetings), the data rates for each user decrease. Similarly, cordless phones have a range of up to 300m and the number of users connected to one BS is of the same order as for WLANs.

Cellular systems have a range that is larger than, e.g., the range of WLANs. Microcells typically cover cells with 500m radius, while macrocells can have a radius of 10 or even 30 km.

Fixed wireless access services cover a range that is similar to that of cellphones – namely, between 100m and several tens of kilometers. Also, the number of users is of a similar order as for cellular systems.

Satellite systems provide even larger cell sizes, often covering whole countries and even continents.
Cell size depends critically on the orbit of the satellite: geostationary satellites provide larger cell sizes (1,000-km radius) than LEOs.

Mobility
Wireless systems also differ in the amount of mobility that they have to allow for the users. The ability to move around while communicating is one of the main charms of wireless communication for the user. Still, within that requirement of mobility, different grades exist.

Fixed devices are placed only once, and after that time communicate with their BS, or with each other, always from the same location. The main motivation for using wireless transmission techniques for such devices lies in avoiding the laying of cables. Even though the devices are not mobile, the propagation channel they transmit over can change with time.

Nomadic devices: nomadic devices are placed at a certain location for a limited duration of time (minutes to hours) and then moved to a different location. This means that during one "drop" (placing of the device), the device is similar to a fixed device. However, from one drop to the next, the environment can change radically. Laptops are typical examples: people do not operate their laptops while walking around, but place them on a desk to work with them.

Low mobility: many communications devices are operated at pedestrian speeds. Cordless phones, as well as cellphones operated by walking human users are typical examples. The effect of the low mobility is a channel that changes rather slowly, and – in a system with multiple BSs.

High mobility usually describes speed ranges from about 30 to 150 km/h. Cellphones operated by people in moving cars are one typical example.

Extremely high mobility is represented by high-speed trains and planes, which cover speeds between 300 and 1000 km/h. These speeds pose unique challenges both for the design of the physical layer (Doppler shift, see Chapter 5) and for the handover between cells.

Energy Consumption
Energy consumption is a critical aspect for wireless devices. Most wireless devices use (one-way or rechargeable) batteries, as they should be free of *any* wires – both the ones used for communication and the ones providing the power supply. *Rechargeable batteries*: nomadic and mobile devices, like laptops, cellphones, and cordless phones, are usually operated with rechargeable batteries. Standby times as well as operating times are one of the determining factors for customer satisfaction. Energy consumption is determined on one hand by the distance over which the data have to be transmitted (remember that a minimum SNR has to be maintained), and on the other hand, by the amount of data that are to be transmitted (the SNR is proportional to the energy per bit).
One-way batteries: sensor network nodes often use one-way batteries, which offer higher energy density at lower prices. Furthermore, changing the battery is often not an option; rather, the sensor including the battery and the wireless transceiver is often discarded after the battery has run out.

Power mains: BSs and other fixed devices can be connected to the power mains. Therefore, energy efficiency is not a major concern for them. It is thus desirable, if possible, to shift as much functionality (and thus energy consumption) from the MS to the BS.

Use of Spectrum
Spectrum can be assigned on an exclusive basis, or on a shared basis. That determines to a large

degree the multiple access scheme and the interference resistance that the system has to provide:

Spectrum dedicated to service and operator: in this case, a certain part of the electromagnetic spectrum is assigned, on an exclusive basis, to a service provider. A prime point in case is cellular telephony, where the network operators buy or lease the spectrum on an exclusive basis (often for a very high price). Due to this arrangement, the operator has control over the spectrum and can plan the use of different parts of this spectrum in different geographical regions, in order to minimize interference.

Spectrum allowing multiple operators:

◦ *Spectrum dedicated to a service*: in this case, the spectrum can be used only for a certain service (e.g., cordless telephones in Europe and Japan), but is not assigned to a specific operator. Rather, users can set up qualified equipment without a license. Such an approach does not require (or allow) interference planning. Rather, the system must be designed in such a way that it avoids interfering with other users in the same region

◦ *Free spectrum*: is assigned for different services as well as for different operators. The ISM band at 2.45 GHz is the best known example – it is allowed to operate microwave ovens, WiFi LANs, and Bluetooth wireless links, among others, in this band. Also for this case, each user has to adhere to strict emission limits, in order not to interfere too much with other systems and users. However, coordination between users (in order to minimize interference) becomes almost impossible – different systems cannot exchange coordination messages with each other, and often even have problems determining the exact characteristics (bandwidth, duty cycle) of the interferers.

Direction of Transmission

Simplex systems send the information only in one direction – e.g., broadcast systems and pagers.

• *Semi-duplex systems* can transmit information in both directions. However, only one direction is allowed at any time. Walkie-talkies, which require the user to push a button in order to talk, are a typical example. Note that one user must signify (e.g., by using the word “over”) that (s)he has finished his/her transmission; then the other user knows that now (s)he can transmit. • *Full-duplex systems* allow simultaneous transmission in both directions – e.g., cellphones and cordless phones.

• *Asymmetric duplex systems*: for data transmission, we often find that the required data rate in one direction (usually the downlink) is higher than in the other direction. However, even in this case, full duplex capability is maintained.

Service Quality

The requirements for service quality also differ vastly for different wireless services. The first main indicator for service quality is *speech quality* for speech services and *file transfer speed* for data services. Speech quality is usually measured by the *Mean Opinion Score* (MOS). It represents the average of a large number of (subjective) human judgments (on a scale from 1 to 5) about the quality of received speech . The speed of data transmission is simply measured in bit/s – obviously, a higher speed is better. An even more important factor is the availability of a service. For cellphones and other speech services, the *service quality* is often computed as the complement of "fraction of blocked calls. plus 10 times the fraction of dropped calls.

Explain in detail Wide Area Data Services and Broadband Wireless Access services offered to wireless networks. (10) May 2012

Wireless Local Area Networks

The functionality of Wireless Local Area Networks (WLANs) is very similar to that of cordless phones – connecting a single mobile user device to a public landline system. The "mobile user device" in this case is usually a laptop computer and the public landline system is the Internet. As in the cordless phone case, the main advantage is convenience for the user, allowing mobility. Wireless LANs can even be useful for connecting fixed-location computers (desktops) to the Internet, as they save the costs for laying cables to the desired location of the computer. A major difference between wireless LANs and cordless phones is the required data rate. While cordless phones need to transmit (digitized) speech, which requires at most 64 kbit/s, wireless LANs should be at least as fast as the Internet that they are connected to. For consumer (home) applications, this means between 700 kbit/s (the speed of DSLs in the U.S.A.) and 3–5 Mbit/s (speed of cable providers in the U.S.A. and Europe) to ≥20 Mbit/s (speed of DSLs in Japan). For companies that have faster Internet connections, the requirements are proportionately higher. In order to satisfy the need for these high data rates, a number of standards have been developed, all of which carry the identifier IEEE 802.11. The original IEEE 802.11 standard enabled transmission with 1Mbit/s, the very popular 802.11b standard (also known under the name WiFi) allows up to 11 Mbit/s and the 802.11a standard extends that to 55 Mbit/s. Even higher rates are realized by the 802.11n standard that was introduced in 2008/2009. WLAN devices can, in principle, connect to any BS (access point) that uses the same standard. However, the owner of the access point can restrict the access – e.g., by appropriate security settings.

Personal Area Networks

When the coverage area becomes even smaller than that of WLANs, we speak of *Personal Area Networks* (PANs). Such networks are mostly intended for simple "cable replacement" duties. For example, devices following the *Bluetooth* standard allows to connect a hands-free headset to a phone without requiring a cable; in that case, the distance between the two devices is less than a meter. In such applications, data rates are fairly low (<1Mbit/s). Recently, wireless communications between components in an entertainment system (DVD player to TV), between computer and peripheral devices (printer, mouse), and similar applications have gained importance, and a number of standards for PANs have been developed by the IEEE 802.15 group. For these applications, data rates in excess of 100 Mbit/s are used.

Networks for even smaller distances are called *Body Area Networks* (BANs), which enable communications between devices located on various parts of a user's body. Such BANs play an increasingly important role in the monitoring of patients' health and of medical devices (e.g., pacemakers).

Fixed Wireless Access

Fixed wireless access systems can also be considered as a derivative of cordless phones or WLANs, essentially replacing a dedicated cable connection between the user and the public landline system.

The main difference from a cordless system is that (i) there is no mobility of the user devices and (ii) the BS almost always serves multiple users. Furthermore, the distances bridged by fixed wireless access devices are much larger (between 100m and several tens of kilometers) than those bridged by cordless telephones. The purpose of fixed wireless access lies in providing users with telephone and data connections without having to lay cables from a central switching office to the office or apartment the user is in. Considering the high cost of labor for the cable-laying operations, this can be an economical approach. However, it is worth keeping in mind that most buildings, especially in the urban areas of developed countries, are already supplied by some form of cable – regular telephone cable, cable TV, or even optical fiber. Rulings of the telecom regulators in various countries have stressed that incumbent operators (owners of these lines) have to allow competing companies to use these lines. As a consequence, fixed wireless access has its main market for covering rural areas, and for establishing connections in developing countries that do not have any wired infrastructure in place. In general, the business cases for fixed wireless has been disappointing. The IEEE 802.16 (WiMAX) standard tries to alleviate that problem by allowing some limited mobility in the system, and thus blurs the distinction from cellular telephony.

Unit II – Part B

1. Compare coherence Bandwidth and Coherence time (8) Nov. 2011

Coherence Bandwidth

The delay spread is a natural phenomenon caused by reflected and scattered propagation paths in the radio channel, the coherence bandwidth B_c, is a defined relation derived from the rms delay spread. Coherence bandwidth is a statistical measure of the 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); In other words, coherence bandwidth is the range of frequencies over which two frequency components have a strong potential for amplitude correlation. Two sinusoids with frequency separation greater than 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

$B_c = (1 /50\ \sigma_\tau)$ ----------------------- (1)

If the definition is relaxed so that the frequency correlation function is above 0.5. then the coherence bandwidth is approximately

$B_c = (1 /50\sigma_\tau)$----------------------------(2)

It is important to note that an exact relationship between coherence bandwidth and rms delay spread does not exist, and equations (1) and (2) are "ball park estimates".

Coherence time T_c:

Coherence time T_c, is the time domain dual of Doppler spread and is used to characterize the time varying nature of the frequency dispersiveness of the channel in the time domain. The Doppler spread and coherence time are inversely proportional to one another. That is,

$T_c = (1/f_m)$ ------------------(3)

Coherence time is actually a statistical measure of the time duration over which the channel impulse response is essentially invariant, and quantifies the similarity of the channel response at different times. In other words, coherence time is the time duration over which two received signals have a strong potential for amplitude correlation. If the reciprocal bandwidth of the baseband signal is greater than the coherence time of the channel, then the channel will change during the transmission of the baseband message, thus causing distortion at the receiver. If the coherence time is defined as the time over which the time correlation function is above 0.5, then the coherence time is approximately

$T_c = (9 / 16\pi f_m)$ ------------------(4)

where f_m is the maximum Doppler shift given by $f_m = v/\lambda$.

---

2. Describe any two methods of diffraction by multiple screens. (8) Nov. 2011

Diffraction by Multiple Screens

Diffraction by a single screen is a problem that has been widely studied, because it is amenable to closed-form mathematical treatment, and forms the basis for the treatment of more complex problems. However, in practice, we usually encounter situations where *multiple IOs* are located between TX and RX. Such a situation occurs, e.g., for propagation over the rooftops of an urban environment. (See figure 1) Such a situation can be well approximated by diffraction by multiple screens. Unfortunately, diffraction by multiple screens is an extremely challenging mathematical problem, and – except for a few special cases – no exact solutions are available.

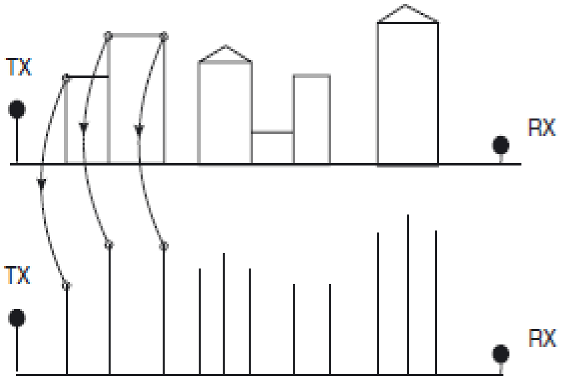


Fig.1 Approximation of multiple buildings by a series of screens.

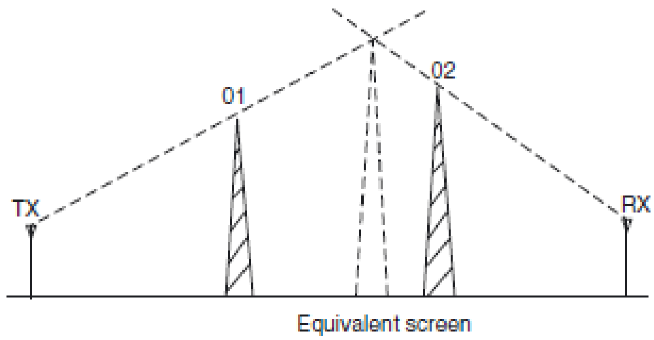


Fig. 2 Equivalent screen after Bullington.

Bullington's Method

Bullington's method replaces the multiple screens by a single, "equivalent" screen. This equivalent screen is derived in the following way: put a tangential straight line from the TX to the real obstacles, and select the steepest one (i.e., the one with the largest elevation angle), so that all obstacles either touch this tangent, or lie below it. Similarly, take the tangents from the RX to the obstacles, and select the steepest one. The equivalent screen is then determined by the intersection of the steepest TX tangent and the steepest RX tangent (see Figure 2).

The major attraction of Bullington's method is its simplicity. However, this simplicity also leads to considerable inaccuracies. Most of the physically existing screens do not impact the location of the equivalent screen. Even the highest obstacle might not have an impact. Consider Figure 2 if the highest obstacle lies between screens 01 and 02, it could lie below the tangential lines, and thus not influence the "equivalent" screen, even though it is higher than either screen 01 or screen 02. In reality, these high obstacles *do* have an effect on propagation loss, and cause an additional attenuation. The Bullington method thus tends to give optimistic predictions of the received power.

The Epstein–Petersen Method

The low accuracy of the Bullington method is due to the fact that only two obstacles determine the equivalent screen, and thus the total diffraction coefficient. This problem can be somewhat mitigated by the Epstein–Petersen method [Epstein and Peterson 1953]. This approach computes the diffraction losses for each screen separately. The attenuation of a specific screen is computed by putting a virtual "TX" and "RX" on the tips of the screens to the left and right of this considered screen (see Figure 3). The diffraction coefficient, and the attenuation, of this one screen can be

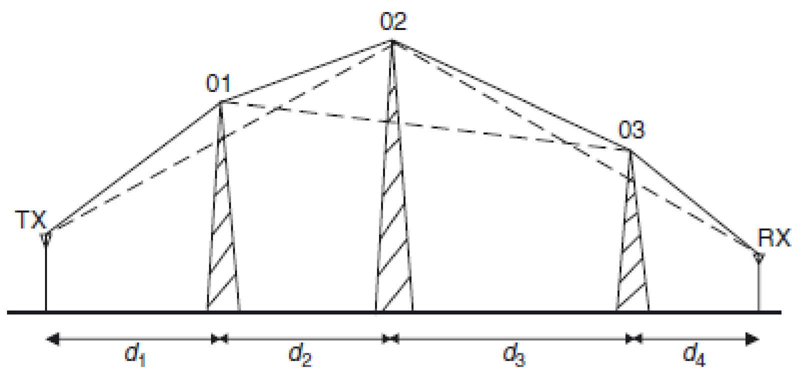


Fig. 3 The Epstein–Petersen method.

Attenuations by the different screens are then added up (on a logarithmic scale). The method thus includes the effects of *all* screens. Despite this more refined modeling, the method is still only approximate. It uses the diffraction attenuation that is based on the assumption that the RX is in the far field of the screen. If, however, two of the screens are close together, this assumption is violated, and significant errors can occur. The inaccuracies caused by this "far-field assumption" can be reduced considerably by the *slope diffraction method*. In this approach, the field is expanded into a Taylor series. In addition to the zeroth-order term (far field), which enforces continuity of the electrical field at the screen, also the first-order term is taken into account, and used to enforce continuity of the first derivative of the field. This results in modified coefficients *A* and *D*, which are determined by recursion equations

The Bullington method is independent of the number of screens, and thus obviously gives a wrong functional dependence.

The Epstein–Petersen method adds the attenuations *on a logarithmic scale* and thus leads to an exponential increase of the total attenuation on a linear scale.

---

3. Discuss about wide band Model .

Tapped Delay Line Models

The most commonly used wideband model is an N-tap Rayleigh-fading model. This is a fairly generic structure, and is basically just the tapped delay line structure with the added restriction that the amplitudes of all taps are subject to Rayleigh fading. Adding an LOS component does not pose any difficulties; the impulse response then just becomes

$$h(t,\tau) = a_0\delta(\tau - \tau_0) + \sum_{i=1}^{N} c_i(t)\delta(\tau - \tau_i)$$

where the LOS component a_0 does *not* vary with time, while the $c_i(t)$ are zero-mean complex Gaussian random processes, whose autocorrelation function is determined by their associated Doppler spectra (e.g., Jakes spectra). In most cases, $\tau_0 = \tau_1$, so the amplitude distribution of the first tap is Rician. The model is further simplified when the number of taps is limited to $N = 2$, and no LOS component is allowed. This is the simplest stochastic fading channel exhibiting delay dispersion,and thus very popular for theoretical analysis. It is alternatively called the *two-path channel*, *two delaychannel*, or *two-spike channel*. Another popular channel model consists of a purely deterministic LOS component plus *one* fading tap $(N = 1)$ whose delay τ_0 can differ from τ_1. This model is widely used for satellite channels – in these channels, there is almost always an LOS connection, and the reflections from buildings near the RX give rise to a delayed fading component. The channel reduces to a flat-fading Rician channel when $\tau_0 = \tau_1$.

Models for the Power Delay Profile

It has been observed in many measurements that the Power Delay Profile (PDP) can be approximated by a one-sided exponential function

$$P_h(\tau) = P_{sc}(\tau) = \begin{cases} \exp(-\tau/S_\tau) & \tau \geq 0 \\ 0 & \text{otherwise} \end{cases} \qquad \text{--------------- (1)}$$

In a more general model (see also Section 7.3.3), the PDP is the sum of several delayed exponential functions, corresponding to multiple *clusters* of Interacting Objects (IOs)

$$P_h(\tau) = \sum_{\ell} \frac{P_l^c}{S_{\tau,l}^c} P_{sc}(\tau - \tau_{0,l}^c) \quad \text{-------------------------(2)}$$

Where

P_l^c, $\tau_{0,l}^c$, $S_{\tau,l}^c$ are the power, delay, and delay spread of the *l*th cluster, respectively. The sum of all cluster powers has to add up to the narrowband power described.For a PDP in the form of Eq. (1), the rms delay spread characterizes delay dispersion. In the case of multiple clusters, Eq. (2), the rms delay spread is defined mathematically, but often has a limited physical meaning. Still, the vast majority of measurement campaigns available in the literature use just this parameter for characterization of delay dispersion.Typical values of the delay spread for different environments are as follows:

• *Indoor residential buildings*: 5–10 ns are typical; but up to 30 ns have been measured.

• *Indoor office environments*: these show typical delay spreads of between 10 and 100 ns, but even 300 ns have been measured. Room size has a clear influence on delay spread. Building size and shape have an impact as well.

• *Factories and airport halls*: these have delay spreads that range from 50 to 200 ns.

• *Microcells*: in microcells, delay spreads range from around 5–100 ns (for LOS situations) to 100–500 ns (for non-LOS).

• *Tunnels and mines*: empty tunnels typically show a very small delay spread (on the order of 20 ns), while car-filled tunnels exhibit larger values (up to 100 ns).

• *Typical urban and suburban environments*: these show delay spreads between 100 and 800 ns, although values up to 3 µs have also been observed.

• *Bad Urban* (*BU*) *and Hilly Terrain* (*HT*) *environments*: these show clear examples of multiple clusters that lead to much larger delay spreads. Delay spreads up to 18 µs, with cluster delays of up to 50 µs, have been measured in various European cities, while American cities show somewhat smaller values. Cluster delays of up to 100 µs occur in mountainous terrain.

4. Discuss about ultra wide band channel. (8) Nov.2011.

Ultra Wideband Channels

UWB Signals with Large Relative Bandwidth

The above models are wideband in the sense that they model the delay dispersion caused by multipath propagation. However, they are still based on the following two assumptions.

1. The reflection, transmission, and diffraction coefficients of the IOs are constant over the considered bandwidth.
2. The relative bandwidth of the system (bandwidth divided by carrier frequency) is *much* Smaller than unity.

Note that these conditions are met for the bandwidth of most currently used wireless systems. However, in recent years, a technique called *UltraWide Band* (UWB) transmission has gained increased interest. UWB systems have a relative bandwidth of more than 20%. In that case, the different frequency components contained in the transmitted signal "see" different propagation environments. For example, the diffraction coefficient of a building corner is different at 100MHz compared with 1 GHz; similarly, the reflection coefficients of walls and furniture can vary over the bandwidth of interest. Channel impulse realization is then given by

$$h(\tau) = \sum_{i=1}^{N} a_i \chi_i(\tau) \otimes \delta(\tau - \tau_i) \qquad \text{--------------------------- (1)}$$

where $\chi_i(\tau)$ denotes the distortion of the *i*th MPC by the frequency selectivity of IOs. One example for a distortion of a short pulse by diffraction by a screen is shown in Figure 1.

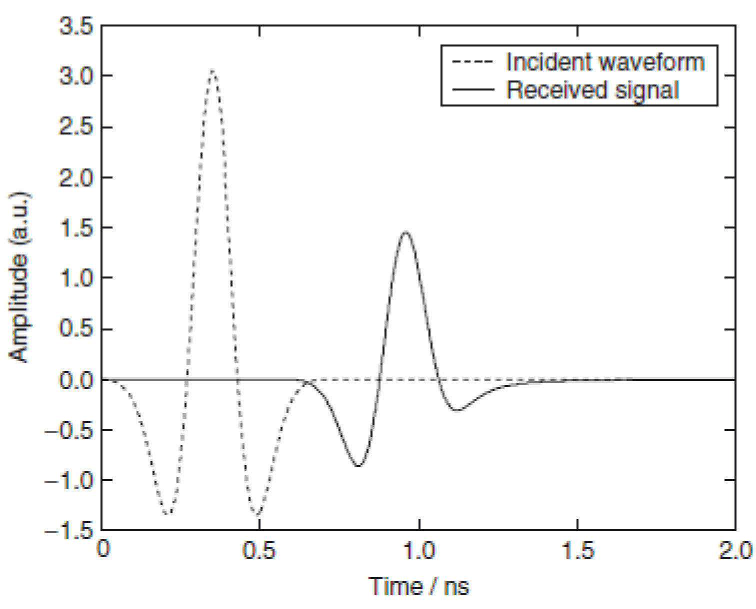


Fig.1 UWB pulse diffracted by a semi-infinite screen.

For UWB systems, propagation effects can also show frequency dependence. path loss is a function of frequency if the antennas have constant gain. Similarly, diffraction and reflection are frequency dependent. Thus, the higher frequency components of the transmitted signal are usually attenuated more strongly by the combination of antenna and channel. Also, this effect leads to a distortion of individual MPCs since *any* frequency dependence of the transfer function leads to delay dispersion, and thus distortion of an MPC. As a consequence of the distortion of the frequency dependence, statistical channel models
also change.

---

5. What is the need for link calculation? Explain with suitable example. (8) May 2012

Link Budget

A link budget is the clearest and most intuitive way of computing the required TX power. It tabulates all equations that connect the TX power to the received SNR. As most factors influencing the SNR enter in a multiplicative way, it is convenient to write all the equations in a logarithmic form – specifically, in dB. It has to be noted, however, that the link budget gives only an approximation
(often a worst case estimate) for the total SNR, because some interactions between different effects are not taken into account.

Before showing some examples, the following points should be stressed:

- The attenuation due to propagation effects, between TX and RX. For the purpose of this chapter, we use a simple model, the so-called "breakpoint" model. For distances $d < d$ break, the received power is proportional to d^{-2}, according to Eq. (3.1). Beyond that point, the power is proportional to d^{-n}, where n typically lies between 3.5 and 4.5. The received power is thus

$$P_{RX}(d) = P_{RX}(d_{\text{break}}) \left(\frac{d}{d_{\text{break}}} \right)^{-n} \text{ for } d > d_{\text{break}} \qquad \text{-------------------(1)}$$

Wireless systems, especially mobile systems, suffer from temporal and spatial variations of the transmission channel (*fading*). In other words, even if the distance is approximately constant, the received power can change significantly with small movements of the TX and/or RX. The power computed from Eq. (1) is only a *mean* value; the ratio of the transmit power to this mean received power is also known as the *path loss* (inverse of the path gain). If the mean received power is used as the basis for the link budget, then the transmission
quality will be above the threshold only in approximately 50% of the times and locations. This

is completely unacceptable quality of service. Therefore, we have to add a *fading margin*, which makes sure that the minimum received power is exceeded in at least, e.g., 90% of all cases (see Figure 1). The value of the fading margin depends on the amplitude statistics of the fading

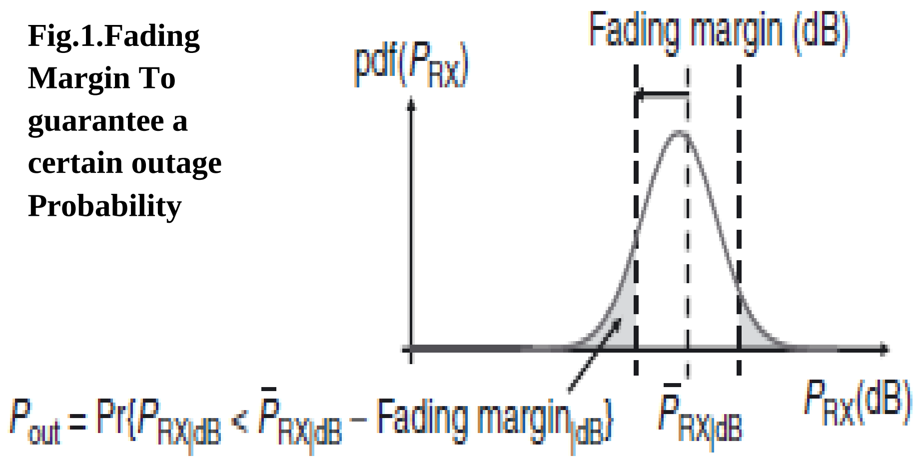


Fig.1.Fading Margin To guarantee a certain outage Probability

Uplink (MS to BS) and downlink (BS to MS) are reciprocal, in the sense that the voltage and currents at the antenna ports are reciprocal (as long as uplink and downlink use the same carrier frequency). However, the noise figures of BSs and MSs are typically quite different. As MSs have to be produced in quantity, it is desirable to use low-cost components, which typically have higher noise figures. Furthermore, battery lifetime considerations dictate that BSs can emit more power than MSs. Finally, BSs and MSs differ with respect to antenna diversity, how close they are to interferers, etc. Thus, the link budgets of uplinks and downlinks are different.

Example:

Consider the downlink of a GSM system .The carrier frequency is 950MHz and the RX sensitivity is (according to GSM specifications) −102 dBm. The output power of the TX amplifier is 30 W. The antenna gain of the TX antenna is 10 dB and the aggregate attenuation of connectors, combiners, etc. is 5 dB. The fading margin is 12 dB and the breakpoint *d*break is at a distance of 100 m. What distance can be covered?

TX side:

TX power P_{TX}	30W	45 dBm
Antenna gain G_{TX}	10	10 dB
Losses (combiner, connector, etc.)	L_f	−5dB

EIRP (Equivalent Isotropically Radiated Power) 50 dBm

RX side:
RX sensitivity P_{min} −102 dBm
Fading margin 12 dB
Minimum RX power (mean) −90 dBm

Admissible path loss (difference EIRP and min. RX power) 140 dB
Path loss at $d$break = 100m $[\lambda/(4\pi d)]^2$ 72 dB
Path loss beyond breakpoint $\propto d^{-n}$ 68 dB

Depending on the path loss exponent,
$n = 1.5 \ldots 2.5$ (line-of-sight)[3]
$n = 3.5 \ldots 4.5$ (non-line-of-sight)
we obtain the coverage distance,

$d$cov = 100 · 1068/(10n)m (3.9)

If, e.g., $n = 3.5$, then the coverage distance is 8.8 km. This example was particularly easy, because RX sensitivity was prescribed by the system specifications.

6.How the received signal strength is predicted using the free space propagation model? Explain.(10) Nov. 2012

The free space propagation model is used to predict received signal strength when the transmitter and receiver have a clear, unobstructed line-of-sight path between them. Satellite communication systems and microwave line-of-sight radio links typically undergo free space propagation. As with most large-scale radio wave propagation models, the free space model predicts that received power decays as a function of the T-R separation distance raised to some power (i.e. a power law function). The free space power received by a receiver antenna which is separated from a radiating transmitter antenna by a distance d, is given by the Friis free space equation,

$$P_r(d) = \frac{P_t G_t G_r \lambda^2}{(4\pi)^2 d^2 L} \qquad (1)$$

where P_t is the transmitted power, $P_r(d)$ is the received power which is a function of the T-R separation, G_T is the transmitter antenna gain, G_r is the receiver antenna gain, d is the T-R separation distance in meters, L is the system loss factor not related to propagation (L $\geq$1 1), and X is the wavelength in meters. The gain of an antenna is related to its effective aperture, A_e is

$$G = \frac{4\pi A_e}{\lambda^2} \qquad \text{\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_} \ (2)$$

effective aperture, A_e is related to the physical size of the antenna, and λ is related to the carrier frequency by

$$\lambda = \frac{c}{f} = \frac{2\pi c}{\omega_c} \qquad \text{\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_} \ (3)$$

where f is the carrier frequency in Hertz, ω_c is the carrier frequency in radians per second, and c is the speed of light given in meters/s. The values for P_t and P_r must be expressed in the same units, and G_t and G_r are dimensionless quantities. The miscellaneous losses L (L _ 1) are usually due to transmission line attenuation, filter losses, and antenna losses in the communication system. A value of L = I indicates no loss in the system hardware.

The Friis free space equation of (1) shows that the received power falls off as the square of the T-R separation distance. This implies that the received power decays with distance at a rate of 20 dB/decade. An isotropic radiator is an ideal antenna which radiates power with unit gain uniformly in all directions, and is often used to reference antenna gains in wireless systems.

The effective isotropic radiated power (EIRP) is defined as

$$EIRP = P_t\,G_t \qquad \text{\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_} \ (4)$$

direction of maximum antenna gain, as compared to an isotropic radiator. In practice, effective radiated power (ERP) is used instead of EIRP to denote the maximum radiated power as compared to a half-wave dipole antenna (instead of an isotropic antenna), Since a dipole antenna has a gain of 1.64 (2.15 dB above an isotrope), the ERP will be 2.15 dB smaller than the EIRP for the same transmission system. In practice, antenna gains are given in units of dBi (dB gain with respect to an isotropic source) or dBd (dB gain with respect to a half-wave dipole)

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

$$PL\,(\mathrm{dB}) = 10\log\frac{P_t}{P_r} = -10\log\left[\frac{G_t G_r \lambda^2}{(4\pi)^2 d^2}\right] \qquad (5)$$

When antenna gains are excluded, the antennas are assumed to have unity gain, and path loss is given by

$$PL\,(\mathrm{dB}) = 10\log\frac{P_t}{P_r} = -10\log\left[\frac{\lambda^2}{(4\pi)^2 d^2}\right] \qquad (6)$$

The Friis free space model is only a valid predictor for P_r, for values of d which are in the far-field of the 'transmitting antenna. The far-field, or Fraunhofer region, of a transmitting antenna is defined as the region beyond the farfield 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

$$d_f = \frac{2D^2}{\lambda} \qquad (7\ a)$$

where D is the largest physical linear dimension of the antenna. Additionally, to be in the far-field region, d_f must satisfy

$$d_f \gg D \qquad (7b)$$

and

$$d_f \gg \lambda \qquad (7c)$$

Furthermore, it is clear that equation (1) does not hold for d = 0. For this reason, large-scale propagation models use a close-in distance, d_0, as a known received power reference point. The received power, $P_r(d)$, at any distance d > d0, may be related to at d0. The value (d_0) may be predicted from equation (1), or may be measured in the radio environment by taking the average received power at many points located at a close-in radial distance d_0 from the transmitter. The reference distance must be chosen such that it lies in the far-field region, that is, d0 _ d1, and d0 is chosen to be smaller than any practical distance used in the mobile communication system.

Thus, using equation (1), the received power in free space at a distance greater than d_0 is given by

$$P_r(d) = P_r(d_0)\left(\frac{d_0}{d}\right)^2 \qquad d \geq d_0 \geq d_f \qquad (8)$$

In mobile radio systems, it is not uncommon to find that P_t may change by many orders of magnitude over a typical coverage area of several square kilometers. Because of the large dynamic range of received power levels, often dBm or dBW units are used to express received power levels. Equation(8) may be expressed in units of dBm or dBW by simply taking the logarithm of both sides and multiplying by 10. For example, that P_r if is in units of dBm, the received power is given by

$$P_r(d)\ \text{dBm} = 10\log\left[\frac{P_r(d_0)}{0.001\ \text{W}}\right] + 20\log\left(\frac{d_0}{d}\right) \qquad d \geq d_0 \geq d_f \qquad (9)$$

Where $P_r(d_0)$ The reference distance d_0 for practical systems using low-gain antennas in the 1-2 GHz region is typically chosen to be 1 m in indoor environments and 100 m or 1 km in outdoor environments, so that the numerator in equations (8) and (9) is a multiple of 10. This makes path loss computations easy in dB units.

UNIT IV Part B

1. With a neat block diagram, discuss the structure of a decision feedback equalizer. (8)Nov. 2011

2. Explain in detail Decision feedback equalizer. (8) May 2012

Decision Feedback Equalizer

The basic idea behind decision feedback equalization is that once an information symbol has been detected and decided upon, the 1ST that it induces on future symbols can be estimated and subtracted out before detection of subsequent symbols [Pro891. The DFE can be realized in either the direct transversal form or as a lattice filter. The direct form is shown in Figure 6.8. It consists of a feedforward filter (FFF) and a feedback filter (FBF). The FBF is driven by decisions on the output of the detector, and its coefficients can be adjusted to cancel the ISI on the current symbol from past detected symbols. The equalizer has N1 + N2 + I taps in the feed forward filter and N3 taps in the feedback filter, and its output can be expressed as:

$$\hat{d}_k = \sum_{n=-N_1}^{N_2} c_n^* y_{k-n} + \sum_{i=1}^{N_3} F_i d_{k-i}$$

------(1)

where c^*_n, and y_n, are tap gains and the inputs, respectively, to the forward filter, F_1 are tap gains for the feedback filter, and d_1 (i c_k) is the previous decision made on the detected signal. That is, once d_k is obtained using equation (1),d_k is decided from it. Then, d_k along with previous decisions d_{k-1},d_{k-2}, arefed back into the equalizer, and $\hat{d}_{k+1}$ is obtained using equation (1)

The minimum mean squared error a DFE can achieve is

$$E[|e(n)|^2]_{min} = \exp\left\{\frac{T}{2\pi}\int_{-\pi/T}^{\pi/T} \ln\left[\frac{N_0}{|F(e^{j\omega T})|^2 + N_0}\right] d\omega\right\}$$

___________ (2)

It can be shown that the minimum MSE for a DFE in equation (2) is always smaller than that of an LTE

$$E[|e(n)|^2] = \frac{T}{2\pi}\int_{-\pi/T}^{\pi/T} \frac{N_0}{|F(e^{jwT})|^2 + N_0} d\omega$$

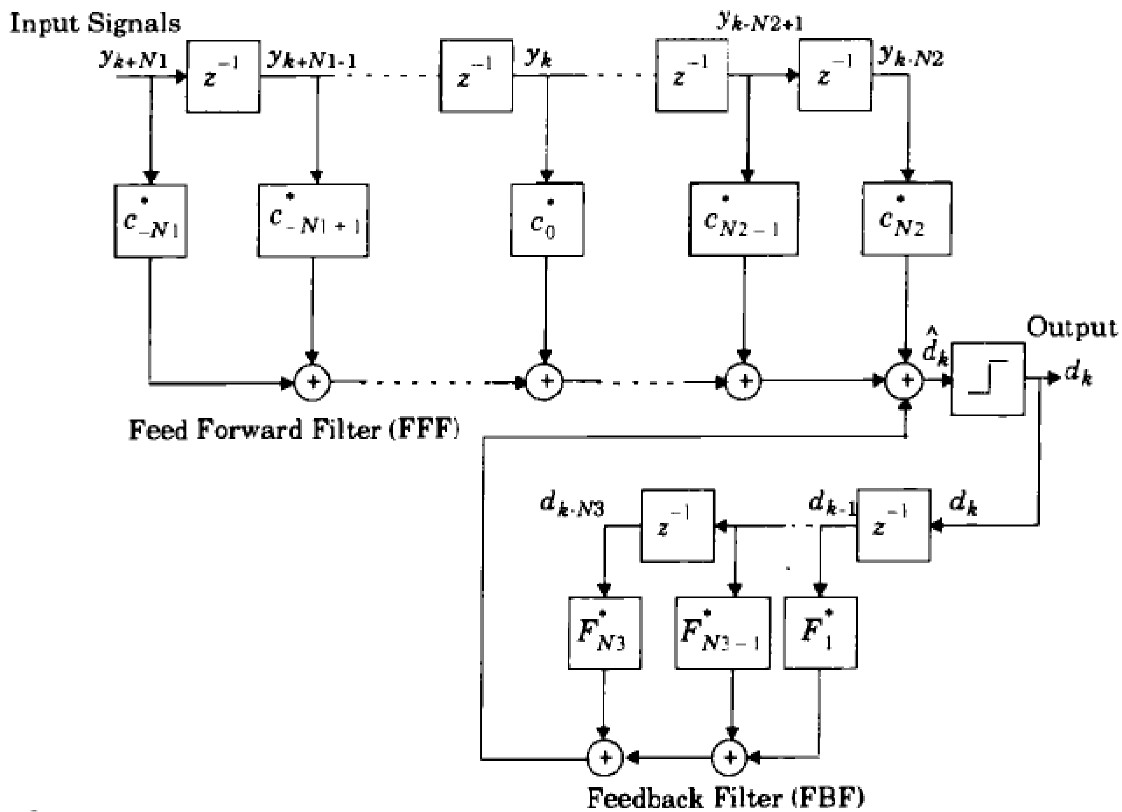
________________ (3)

in equation (3)

unless

$|F(e^{j\omega T})|$ is a constant (i.e, when adaptive equalization is not needed). If there are nulls

In $|F(e^{j\omega T})|$ a DFE has significantly smaller minimum MSE than an LTE. Therefore,an LTE is well behaved when the channel spectrum is comparatively flat, but if the channel is severely distorted or exhibits nulls in the spectrum, the performance of an LTE deteriorates and the mean squared error of a DFE is much better than a LTE. Also, an LTE has difficulty equalizing a non minimum phase channel, where the strongest energy arrives after the first arriving signal component. Thus, a DFE is more appropriate for severely distorted wireless channels.



Decision feedback equalizer (DFE).

The lattice implementation of the DFE is equivalent to a transversal DFE having a feed forward filter of length N1 and a feedback filter of length N2, where N1 > N2. Another form of DFE proposed by Belfiore and Park is called a predictive DFE, and is shown in Figure 6.9. It also consists of a feed forward filter (FFF) as in the conventional DFE. However, the feedback filter (FBF) is driven by an input equence fonned by the difference of the output of the detector and the output of the feed forward filter. Hence, the FBF here is called a noise predictor because it predicts the noise and the residual 151 contained in the signal at the FFF output and subtracts from it the detector output after some feedback delay. The predictive DFE performs as well as the conventional DFE as the limit in the number of taps in the FFF and the FBF approach infinity. The FEF in the predictive DFE can also be realized as a lattice structure The RLS lattice algorithm can be used in this case to yield fast convergence.

3.Explain in detail about: Nov. 2012

Polarization diversity (6)

Time diversity (5)

Frequency diversity (5)

Diversity Techniques

Diversity is a powerful communication receiver technique that provides wireless link improvement at relatively low cost. Unlike equalization, diversity requires no training overhead since a training sequence is not required by the transmitter. Furthermore, there are a wide range of diversity implementations, many which are very practical and provide significant link improvement with little added cost. Diversity exploits the random nature of radio propagation by finding independent (or at least highly uncorrelated) signal paths for communication. In virtually all applications, diversity decisions are made by the receiver, and are unknown to the transmitter. The diversity concept can be explained simply. If one radio path undergoes a deep fade, another independent path may have a strong signal. By having more than one path to select from, both the instantaneous and average SNRs at the receiver may be improved, often by as much as 20 dB to 30 dB.

There are two types of fading — small-scale and large-scale fading. Small-scale fades are characterized by deep and rapid amplitude fluctuations which occur as the mobile moves over distances of just a few wavelengths. These fades are caused by multiple reflections from the surroundings in the vicinity of the mobile. Small-scale fading typically results in a Rayleigh fading distribution of signal strength over small distances. In order to prevent deep fades from occurring, **microscopic diversity techniques** can exploit the rapidly changing signal. For example, the small-scale fading shown in Figure 1 reveals that if two antennas are separated by a fraction of a meter, one may receive a null while the other receives a strong signal. By selecting the best signal at all times, a receiver can mitigate small-scale fading effects (this is called antenna diversity or space diversity). Large-scale fading is caused by shadowing due to variations in both the terrain profile and the nature of the surroundings. In deeply shadowed conditions, the received signal strength at a mobile can drop well below that of free space. In Chapter 3, large-scale fading was shown to be log-normally distributed with a standard deviation of about 10 dB in urban environments. By selecting a base station which is not shadowed when others are, the mobile can improve substantially the average ratio on the forward link. This is called **macroscopic diversity**, since the mobile is taking advantage of large separations between the serving base stations. Macroscopic diversity is also useful at the base station receiver. By using base station antennas that are sufficiently separated in space, the base station is able to improve the reverse link by selecting the antenna with the strongest signal from the mobile.

Polarization Diversity

At the base station, space diversity is considerably less practical than at the mobile because the narrow angle of incident fields requires large antenna spacings. The comparatively high cost of using space diversity at the base station prompts the consideration of using orthogonal polarization to exploit polarization diversity. While this only provides two diversity branches it does allow the antenna elements to be co-located. In the early days of cellular radio, all subscriber units were mounted in vehicles and used vertical whip antennas. 'Today, however, over half of the subscriber units are portable. This means that most subscribers are no longer using vertical polarization due to hand-tilting when the portable cellular phone is used. This recent phenomenon has sparked interest in polarization diversity at the base station. Measured horizontal and vertical polarization paths between a mobile and a base station are reported to be uncorrelated by Lee and Yeh .The decorrelation for the signals in each polarization is caused by multiple reflections in the channel between the mobile and base station antennas. The reflection coefficient for each polarization is different, which results in different amplitudes and phases for each, or at least some, of the reflections. After sufficient random reflections, the polarization state of the signal will be independent of the transmitted polarization. In practice, however, there is some dependence of the received polarization on the transmitted polarization. Circular and linear polarized antennas have been used to characterize multipath inside buildings. When the path was obstructed, polarization diversity was found to dramatically reduce the multipath delay spread without significantly decreasing the received power. While polarization diversity has been studied in the past, it has primarily been used for fixed radio links which vary slowly in time. Line-of-sight microwave links, for example, typically use polarization diversity to support two simultaneous users on the same radio channel. Since the channel does not change much in such a link, there is little likelihood of cross polarization interference. As portable users proliferate, polarization diversity is likely to become more important for improving link margin and capacity.

Time Diversity

Time diversity repeatedly transmits information at time spacings that exceed the coherence time of the channel, so that multiple repetitions of the signal will be received with independent fading conditions, thereby providing for diversity. One modem implementation of time diversity involves the use of the RAKE receiver for spread spectrum CDMA, where the multipath channel provides redundancy in the transmitted message.

RAKE Receiver

In CDMA spread spectrum systems (see Chapter 5), the chip rate is typically much greater than the flat fading bandwidth of the channel. Whereas conventional modulation techniques require an equalizer to undo the inter-symbol interference between adjacent symbols, CDMA spreading codes are designed to provide very low correlation between successive chips. Thus, propagation delay spread in the radio channel merely provides multiple versions of the transmitted signal at the receiver. If these multipath components are delayed in time by more than chip duration, they appear like uncorrelated noise at a CDMA receiver, and equalization is not required. However, since there is useful information in the multipath components. CDMA receivers may combine the time delayed versions of the original signal transmission in order to improve the signal to noise ratio at the receiver. A RAKE receiver does just this — it attempts to collect the time-shifted versions of the original signal by providing a separate correlation receiver for each of the
multipath signals. The RAKE receiver, shown in Figure 1., is essentially a diversity receiver designed specifically for CDMA, where the diversity is provided by the fact that the multipath components are practically uncorrelated from one another when their relative propagation delays exceed a chip period.

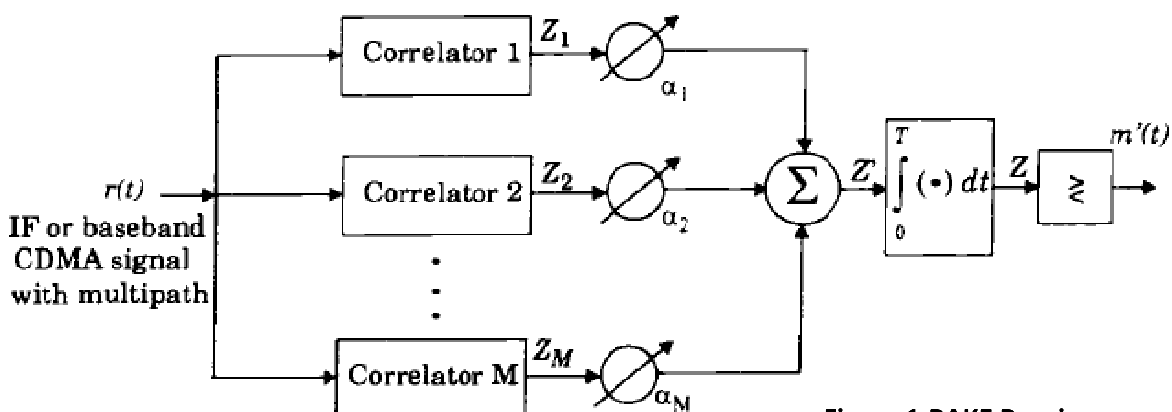


Figure 1.RAKE Receiver

An M-branch (M-finger) RAKE receiver implementation. Each correlator detects a time shifted version of the original CDMA transmission, and each finger of the RAKE correlates to a portion of the signal which is delayed by at least one chip in time from the other fingers.

A RAKE receiver utilizes multiple correlators to separately detect the M strongest multipath components. The outputs of each correlator are weighted to provide a better estimate of the transmitted signal than is provided by a single component. Demodulation and bit decisions are then based on the weighted outputs of the M correlators. The basic idea of a RAKE receiver was first proposed by Price and Green. In outdoor environments, the delay between multipath components is usually large and, if the chip rate is properly selected, the low autocorrelation properties of a CDMA spreading sequence can assure that multipath components will appear nearly uncorrelated with each other.

Frequency Diversity

Frequency diversity transmits information on more than one carrier frequency. The rationale behind this technique is that frequencies separated by more than the coherence bandwidth of the channel will not experience the same fades. Theoretically, if the channels are uncorrelated, the probability of simultaneous fading will be the product of the individual fading probabilities . Frequency diversity is often employed in microwave line-of-sight links which carry several channels in a frequency division multiplex mode (FDM). Due to tropospheric propagation and resulting refraction, deep fading sometimes occurs. In practice, 1:N protection switching is provided by a radio licensee, wherein one frequency is nominally idle but is available on a stand-by basis to provide frequency diversity switching for any one of the N other carriers (frequencies) being used on the same link, each carrying independent traffic. When diversity is needed, the appropriate traffic is simply switched to the backup frequency.

This technique has the disadvantage that it not only requires spare bandwidth but also requires that there be as many receivers as there are channels used for the frequency diversity. However, for critical traffic, the expense may be justified.

4. With a neat block diagram, explain the principle of diversity.(8) May 2012

Principle of Diversity

We treated conventional transceivers that transmit an uncoded bitstream over fading channels. For Additive White Gaussian Noise (AWGN) channels, such an approach can be quite reasonable: the Bit Error Rate (BER) decreases exponentially as the Signal-to-Noise Ratio (SNR) increases, and a 10-dB SNR leads to BERs on the order of 10^{-4}. However, in Rayleigh fading the BER decreases only linearly with the SNR. We thus would need an SNR on the order of 40 dB in order to achieve a 10^{-4} BER, which is clearly unpractical. The reason for this different performance is the fading of the channel: the BER is mostly determined by the probability of channel attenuation being large, and thus of the instantaneous SNR being low. A way to improve the BER is thus to change the effective channel statistics – i.e., to make sure that the SNR has a smaller probability of being low. Diversity is a way to achieve this. The principle of diversity is to ensure that the same information reaches the receiver (RX) on statistically independent channels. Consider the simple case of an RX with two antennas. The antennas are assumed to be far enough from each other that small-scale fading is independent at the two antennas. The RX always chooses the antenna that has instantaneously larger receive power. As the signals are statistically independent, the probability that both antennas are in a fading dip simultaneously is low – certainly lower than the probability that one antenna is in a fading dip. The diversity thus changes the SNR statistics at the detector input.

Diversity reception in a two-state fading channel.

To quantify this effect, let us consider a simple numerical example: the noise power within the RX filter bandwidth is 50 pW, the average received signal power is 1 nW, the SNR is thus 13 dB. In an AWGN channel, the resulting BER is 10−9, assuming that the modulation is differentially detected Frequency Shift Keying (FSK). Now consider a fading channel where during 90% of the time the received power is 1.11 nW, and the SNR is thus 13.5 dB, while for the remainder, it is zero. This means that during 90% of the time, the BER is 10^{-10}; the remainder of the time,

it is 0.5; the average BER is thus

$0.9 \cdot 10^{-10} + 0.1 \cdot 0.5 = 0.05$ -----------------(1)

For the case of two-antenna diversity, the probability that the received signal power is 0 at both antennas simultaneously is $0.1 \cdot 0.1 = 0.01$. The probability that the received power is 1.11nW at both antennas simultaneously is $0.9 \cdot 0.9 = 0.81$; the probability that it is 1.11nW at one antenna and 0 at the other is 0.18. Assuming selection diversity, in both the latter cases, the SNR at the detector is 13.5 dB. The total BER is thus

$0.01 \cdot 0.5 + 0.99 \cdot 10-10 = 0.005$ --------------(2)

0.02 This is approximately the square of the BER for a single-antenna system. If we have three antennas, then the probability that the signal power is 0 at all three antennas simultaneously is 0.13; the total BER is then $0.5 \cdot 0.001 + 0.999 \cdot 10^{-10} = 0.0005$; this is approximately the third power of the BER for a single-antenna system.

5.With a suitable diagram, explain the channel coding and speech coding techniques. (8) Nov.2012.

Fundamentals of Channel Coding

Channel coding protects digital data from errors by selectively introducing redundancies in the transmitted data. Channel codes that are used to detect errors are called error detection codes, while codes that can detect and correct errors are called error correction codes. In 1948, Shannon demonstrated that by proper encoding of the information, errors induced by a noisy channel can be reduced to any desired level without sacrificing the rate of information transfer. Shannon's channel capacity formula is applicable to the AWGN channel and is given by

$$C = B\log_2\left(1 + \frac{P}{N_0 B}\right) = B\log_2\left(1 + \frac{S}{N}\right) \qquad \text{(1)}$$

where C is the channel capacity (bits per second), B is the transmission bandwidth (Hz), P is the received signal power (watts), and N0 is the single-sided noise power density (watts/liz). The received power at a receiver is given as

$$P = E_b R_b \qquad (2)$$

where E_b is the average bit energy, and R_b is the transmission bit rate. Equation (1) can be normalized by the transmission bandwidth and is given by

$$\frac{C}{B} = \log_2\left(1 + \frac{E_b}{N_0}\frac{R_b}{B}\right) \qquad (3)$$

where C/B denotes bandwidth efficiency.
The basic purpose of error detection and error correction techniques is to introduce redundancies in the data to improve wireless link performance. The introduction of redundant bits increases the raw data rate used, in the link, hence increases the bandwidth requirement for a fixed source data rate. This reduces the bandwidth efficiency of the link in high SNR conditions, but provides excellent BER performance at low SNR values.

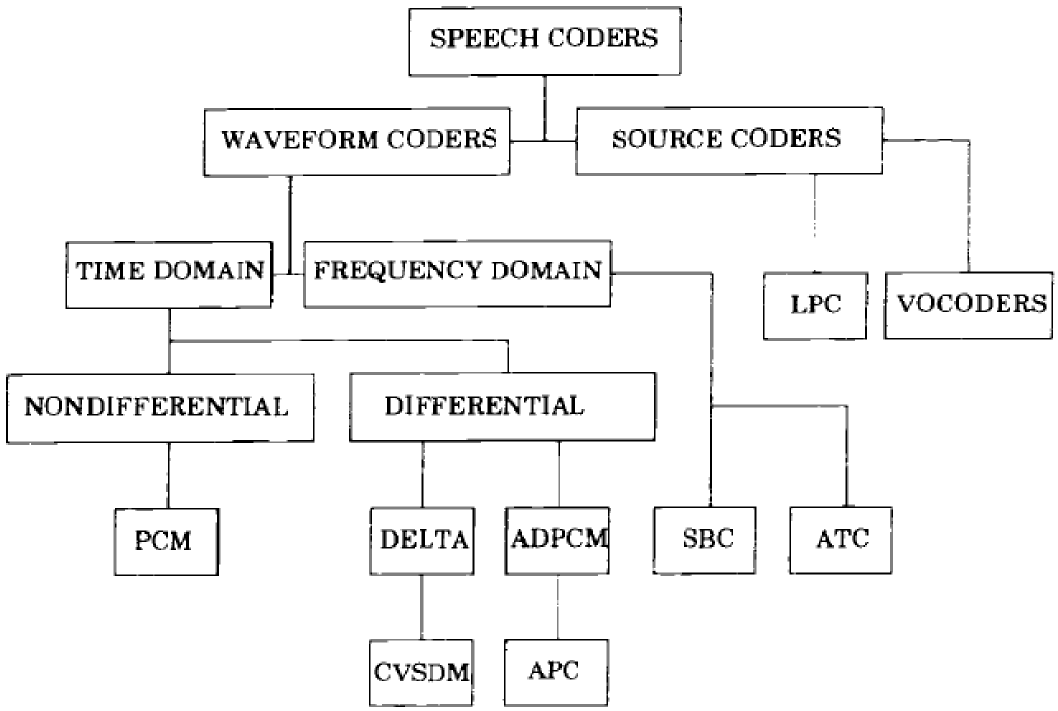
It is well known that the use of orthogonal signaling allows the probability of error to become arbitrarily small by expanding the signal set, i.e., by making the number of waveforms M provided that the SNR per bit exceeds the Shannon limit of SN/lb ≥ —1.6 dB EVit79]. In the limit, Shannon's result indicates that extremely wideband signals could be used to achieve error free communications, as long as sufficient SNR exists. Error control coding waveforms, on the other hand, have bandwidth expansion factors that grow only linearly with the code block length. Error correction coding thus offers advantages in bandwidth limited applications, and also provides link protection in power limited applications. A channel coder operates on digital message (or source) data by encoding the source information into a code sequence for transmission through the channel. There are two basic types of error correction and detection codes: block codes and convolutional codes.

Speech Coding

Speech coders have assumed 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 mobile communication systems, bandwidth is a precious commodity, and service providers are continuously met with the challenge of accommodating more users within a limited allocated bandwidth. Low bit-rate speech coding offers a way to meet this challenge. The lower the bit rate at which the coder can deliver toll quality speech, the more speech channels can be compressed within a given bandwidth. For this reason, manufacturers and service providers are continuously in search of speech codes that will provide toll quality speech at lower bit rates.

In mobile communication systems, the design and subjective test of speech coders has been extremely difficult. Without low data rate speech coding, digital modulation schemes offer little in the way of spectral efficiency for voice traffic. To make speech coding practical, implementations must consume little power and provide tolerable, if not excellent speech quality.

The goal of all speech coding systems is to transmit speech with the highest possible quality using the least possible channel capacity. This has to be accomplished while maintaining certain required levels of complexity of implementation and communication delay. In general, there is a positive correlation between coder bit-rate efficiency and the algorithmic complexity required to achieve it. The more complex an algorithm is, the more its processing delay and cost of implementation. A balance needs to be struck between these conflicting factors, and it is the aim of all speech processing developments to shift the point at which this balance is made towards ever lower bit rates



Hierarchy of speech coders

Speech coders differ widely in their approaches to achieving signal compression. Based on the means by which they achieve compression, speech coders are broadly classified into two categories: Waveform Coders and Vocoders. Waveform coders essentially strive to reproduce the time waveform of the speech signal as closely as possible. They are, in principle, designed to be source independent and can hence code equally well a variety of signals. They have the advantage of being robust for a wide range of speech characteristics and for noisy environments. All these advantages are preserved with minimal complexity, and in general this class of coders achieves only moderate economy in transmission bit rate. Examples of waveform coders include pulse code modulation (PCM), differential pulse code modulation (DPCM), and adaptive differential pulse code modulation (ADPCM), delta modulation (DM), and continuously variable slope delta modulation (CVSDM), and adaptive predictive coding. Vocoders on the other hand achieve very high economy in transmission bit rate and are in general more complex. They are based on using a priori knowledge about the signal to be coded, and for this reason, they are, in general, signal specific.

---

6. Explain any one method of channel coding. (8) May 2012

A channel coder operates on digital message (or source) data by encoding the source information into a code sequence for transmission through the channel. There are two basic types of error correction and detection codes: block codes and convolutional codes.

Block Codes

Block codes are forward error correction (FEC) codes that enable a limited number of errors to be detected and corrected without retransmission. Block codes can be used to improve the performance of a communications system when other means of improvement (such as increasing transmitter power or using a more sophisticated demodulator) are impractical. In block codes, parity bits are added to blocks of message bits to make code- words or code blocks. In a block encoder. k information bits are encoded into n code bits. A total of n — k redundant bits are added to the k information bits for the purpose of detecting and correcting errors. The block code is referred to as an (n,k) code, and the rate of the code is defined as = k/n and is equal to the rate of information divided by the raw channel rate. The ability of a block code to correct errors is a function of the code distance. Many families of codes exist that provide varying degrees of error protection

Besides the code rate, other important parameters are the distance and the weight of a code.

These are defined below.

Distance of a Code — The distance of a codeword is the number of elements in which two codewords and C_i and C_j differ

$$d(C_i, C_j) = \sum_{l=1}^{N} C_{i,l} \oplus C_{j,l} (modulo-q) \qquad \text{(1)}$$

where d is the distance of the codeword and q is the number of possible values of C_i and C_j If the code used is binary, the distance is known as the Hamming distance. The minimum distance d_{min} is the smallest distance for the given set and is given as

$$d_{min} = Min\{d(C_i, C_j)\} \qquad (2)$$

Weight of a Code

The weight of a codeword is given by the number of nonzero elements in the codeword. For a binary code, the weight is basically the number of l's in the codeword and is given as

$$w(C_i) = \sum_{l=1}^{N} C_{i,l} \qquad (3)$$

Properties of Block Codes

Linearity — Suppose C. and C1 are two code words in an (n. k) block code. Let α_1 and α_2 be any two elements selected from the alphabet. Then the code is said to be linear if and only if $\alpha_1 C_1 + \alpha_2 C_2$ is also a code word. A linear code must contain the all-zero code word. Consequently, a constant-weight code is nonlinear.

Sytematic — A systematic code is one in which the parity bits are appended to the end of the information bits. For an (n, k) code, the first k bits are identical to the information bits, and the remaining n — k bits of each code word are linear combinations of the k information bits.

Cyclic — Cyclic codes are a subset of the class of linear codes which satisfies the following cyclic shift property: If C = , c_{n-1}, C_{n-2}, ……..n c0J is a code word of a cyclic code, then [en 2' .., J , obtained by a cyclic shift of the elements
of C, is also a code word. That is, all cyclic shifts of C are code words. As a consequence of the cyclic property, the codes possess a considerable amount of structure which can be exploited in the encoding and decoding operations. Encoding and decoding techniques make use of the mathematical constructs know as finite fields. Finite fields are algebraic systems which contain a finite set of elements. Addition, subtraction, multiplication, and division of finite field elements is accomplished without leaving the set (i.e., the sum/product of two field elements is a field element). Addition and multiplication must satisfy the commutative, associative, and distributive laws.

UNIT 5 Part B

1. Explain: Code Division Multiple Access (CDMA) and compare its performance with TDMA. (16) May 2012

Code Division Multiple Access (CDMA)

In code division multiple access (CDMA) systems, the narrowband message signal is multiplied by a very large bandwidth signal called the spreading signal. The spreading signal is a pseudo-noise code sequence that has a chip rate which is orders of magnitudes greater than the data rate of the message. All users in a CDMA system, as seen from Figure 8.5, use the same carrier frequency and may transmit simultaneously. Each user has its own pseudorandom code word which is approximately orthogonal to all other codewords.

The receiver performs a time correlation operation to detect only the specific desired codeword. All other code words appear as noise due to decorrelation. For detection of the message signal, the receiver needs to know the codeword used by the transmitter. Each user operates independently with no knowledge of the other users. In CDMA, the power of multiple users at a receiver determines the noise floor after decorrelation. If the power of each user within a cell is not controlled such that they do not appear equal at the base station receiver, then the near-far problem occurs.

The near-far problem occurs when many mobile users share the same channel. In general, the strongest received mobile signal will capture the demodulator at a base station. In CDMA, stronger received signal levels raise the noise floor at the base station demodulators for the weaker signals, thereby decreasing the probability that weaker signals will be received. lb combat the near-far problem, power control is used in most CDMA implementations. Power control is provided by each base station in a cellular system and assures that each mobile within the base station coverage area provides the same signal level to the base station receiver. This solves the problem of a nearby subscriber overpowering the base station receiver and drowning out the signals of far away subscribers. Power control is implemented at the base station by rapidly sampling the radio signal strength indicator (RSSI) levels of each mobile and then sending a power change command over the forward radio link. Despite the use of power control within each cell, out-of-cell mobiles provide interference which is not under the control of the receiving base station.

The features of CDMA including the following:

- Many users of a CDMA system share the same frequency. Either TDD or FDD may be used.

- Unlike TDMA or FDMA, CDMA has a soft capacity limit. Increasing the Thus, there is no absolute limit on the number of users in CDMA. Rather, the system performance gradually degrades for all users as the number of users is increased, and improves as the number of users is decreased. Multipath fading may be substantially reduced because the signal is spread over a large spectrum. If the spread spectrum bandwidth is greater than the coherence bandwidth of the channel, the inherent frequency diversity will mitigate the effects of small-scale fading.

- Channel data rates are very high in CDMA systems. Consequently, the symbol (chip) duration is very short and usually much less than the channel delay spread. Since PN sequences have low autocorrelation, multipath which is delayed by more than a chip will appear as noise. A RAKE receiver can be used to improve reception by collecting time delayed versions of the required signal.

- Since CDMA uses co-channel cells, it can use macroscopic spatial diversity to provide soft handoff. Soft handoff is performed by the MSC, which can simultaneously monitor a particular user from two or more base stations. The MSC may chose the best version of the signal at any time without switching frequencies.

- Self-jamming is a problem in CDMA system. Self-jamming arises from the fact that the spreading sequences of different users are not exactly orthogonal, hence in the despreading of a particular PN code, non-zero contributions to the receiver decision statistic for a desired user arise from the transmissions of other users in the system.

- The near-far problem occurs at a CDMA receiver if an undesired user has a high detected power as compared to the desired user.

Time Division Multiple Access (TDMA)

Time division multiple access (TDMA) systems divide the radio spectrum into time slots, and in each slot only one user is allowed to either transmit or receive. It can be seen from Figure 8.3 that each user occupies a cyclically repeating time slot, so a channel may be thought of as articular time slot that reoccurs every frame, where N time slots comprise a frame. TDMA systems transmit data in a buffer-and-burst method, thus the transmission for any user is noncontinuous. This implies that, unlike in FDMA systems which accommodate analog FM, digital data and digital modulation must be used with TDMA.

The transmission from various users is interlaced into a repeating frame structure as shown in Figure 8.4. It can be seen that a frame consists of a number of slots. Each frame is made up of a preamble, an information message, and tail bits. In TDMAITDD, half of the time slots in the frame information message would be used for the forward link channels and half would be used for reverse link channels. In TDMAIFDD systems, an identical or similar frame structure would be used solely for either forward or reverse transmission, but the carrier frequencies would be different for the forward and reverse links. In general, TDMAJFDD systems intentionally induce several time slots of delay between the forward and reverse time slots of a particular user, so that duplexers are not required in the subscriber unit.

In a TDMA frame, the preamble contains the address and synchronization information that both the base station and the subscribers use to identify each other. Guard times are utilized to allow synchronization of the receivers between different slots and frames. Different TDMA wireless standards have different TDMA frame structures, and some are described in Chapter 10.

The features of TDMA include the following:

- TDMA shares a single carrier frequency with several users, where each user makes use of nonoverlapping time slots. The number of time slots per frame depends on several factors, such as modulation technique, available bandwidth, etc.
- Data transmission for users of a TDMA system is not continuous, but occurs in bursts. This results in low battery consumption, since the subscriber transmitter can be turned off when not in use (which is most of the time).
- Because of discontinuous transmissions in TDMA, the handoff process is much simpler for a subscriber unit, since it is able to listen for other base stations during idle time slots. An enhanced link control, such as that provided by mobile assisted handoff (MAHO) can be carried out by a subscriber by listening on an idle slot in the TDMA frame.
- TDMA uses different time slots for transmission and reception, thus duplexers are not required. Even if FDD is used, a switch rather than a duplexer inside the subscriber unit is all that is required to switch between transmitter and receiver using TDMA.
- Adaptive equalization is usually necessary in TDMA systems, since the transmission rates are generally very high as compared to FDMA channels.
- In TDMA, the guard time should be minimized. If the transmitted signal at the edges of a time slot are suppressed sharply in order to shorten the guard time, the transmitted spectrum will expand and cause interference to adjacent channels.
- High synchronization overhead is required in TDMA systems because of burst transmissions. TDMA transmissions are slotted, and this requires the receivers to be

synchronized for each data burst. In addition, guard slots are necessary to separate users, and this results in the TDMA systems having larger overheads as compared to FDMA.

- TDMA has an advantage in that 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 slots based
  on priority.

2. **What is orthogonal frequency division multiplexing? Explain OFDM technique and mention its merits, demerits and application. (16) May 2012.**

3.Explain with neat diagram of orthogonal frequency division multiplexing. (8)Nov. 2012.

Orthogonal Frequency Division Multiplexing (OFDM) is a modulation scheme that is especially suited for high-data-rate transmission in delay-dispersive environments. It converts a high-rate data stream into a number of low-rate streams that are transmitted over parallel, narrowband channels that can be easily equalized.

Principle of Orthogonal Frequency Division Multiplexing

OFDM splits a high-rate data stream into N parallel streams, which are then transmitted by modulating N distinct carriers (henceforth called *subcarriers* or *tones*). Symbol duration on each subcarrier thus becomes larger by a factor of N. In order for the receiver to be able to separate signals carried by different subcarriers, they have to be orthogonal. Conventional Frequency Division Multiple Access (FDMA), and depicted again in **Figure 1,** can achieve this by having large (frequency) spacing between carriers. This, however, wastes precious spectrum. A much narrower spacing of subcarriers can be achieved. Specifically, let subcarriers be at the frequencies $f_n = nW/N$, where n is an integer, and W the total available bandwidth; in the most simple case, $W = N/T_s$. We furthermore assume for the moment that modulation on each of the subcarriers is Pulse Amplitude Modulation (PAM) with rectangular basis pulses. We can then easily see that subcarriers are mutually orthogonal, since the relationship

$$\int_{iT_s}^{(i+1)T_S} \exp(j2\pi f_k t)\exp(-j2\pi f_n t)\,dt = \delta_{nk} \qquad \text{(1)}$$

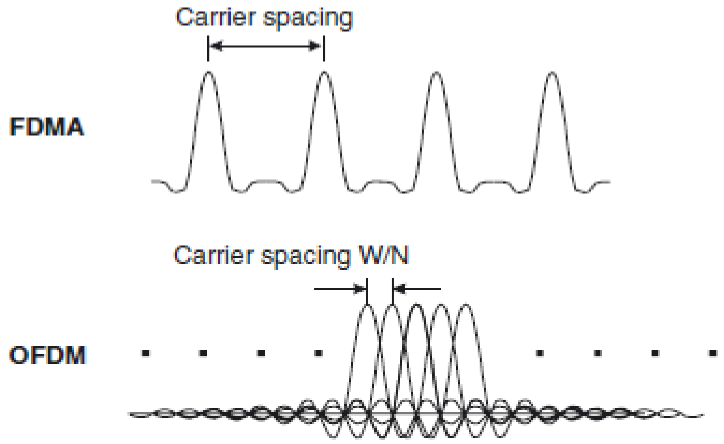


Figure.1.

Principle behind orthogonal frequency division multiplexing: N carriers within a bandwidth of W.

Implementation of Transceivers

OFDM can be interpreted in two ways: one is an "analog" interpretation following from the picture of Figure .2a. we first split our original data stream into N parallel data streams, each of which has a lower data rate. We furthermore have a number of local oscillators (LOs) available, each of which oscillates at a frequency $fn = nW/N$, where $n = 0, 1, \ldots, N - 1$. Each of the parallel data streams then modulates one of the carriers. This picture allows an easy understanding of the principle, but is ill suited for actual implementation – the hardware effort of multiple local oscillators is too high.

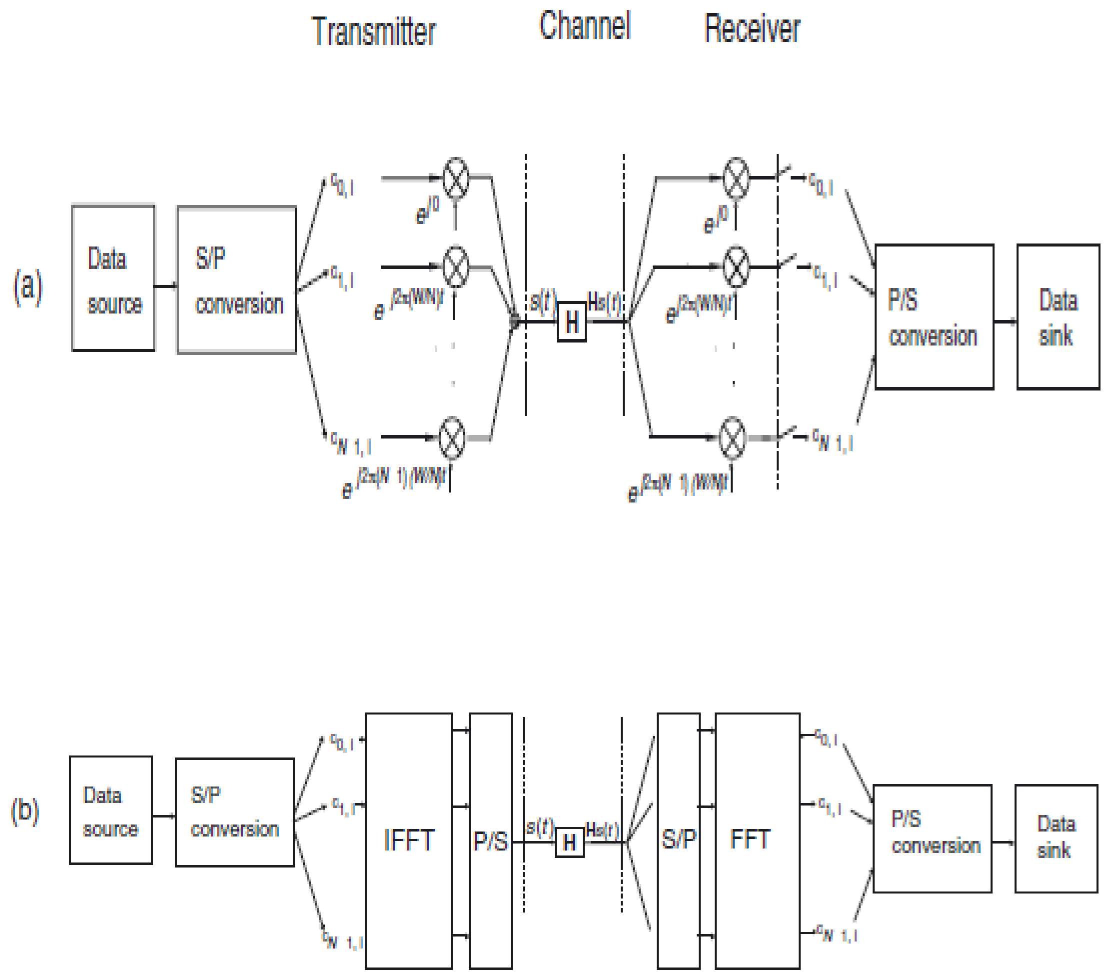


Fig.2. Transceiver structures for orthogonal frequency division multiplexing in purely analog technology (a), and using inverse fast Fourier transformation (b).

An alternative implementation is *digital* . It first divides the transmit data into blocks of N symbols. Each block of data is subjected to an *Inverse Fast Fourier Transformation* (IFFT), and then transmitted (see Figure.2b). This approach is much easier to implement with integrated circuits. In the following, we will show that the two approaches are equivalent. Let us first consider the analog interpretation. Let the complex transmit symbol at time instant i on the nth carrier be $c_{n,i}$. The transmit signal is then

$$s(t) = \sum_{i=-\infty}^{\infty} s_i(t) = \sum_{i=-\infty}^{\infty} \sum_{n=0}^{N-1} c_{n,i} g_n(t - iT_S) \qquad (2)$$

where the basis pulse *gn(t)* is a normalized, frequency-shifted rectangular pulse:

$$g_n(t) = \begin{cases} \frac{1}{\sqrt{T_S}} \exp\left(j2\pi n \frac{t}{T_S}\right) & \text{for } 0 < t < T_S \\ 0 & \text{otherwise} \end{cases} \qquad (3)$$

consider the signal only for $i = 0$, and sample it at instances $t_k = kT_s/N$

$$s_k = s(t_k) = \frac{1}{\sqrt{T_S}} \sum_{n=0}^{N-1} c_{n,0} \exp\left(j2\pi n \frac{k}{N}\right) \qquad (4)$$

Now, this is nothing but the inverse Discrete Fourier Transform (DFT) of the transmit symbols. Therefore, the transmitter can be realized by performing an *Inverse Discrete Fourier Transform*(IDFT) on the block of transmit symbols (the blocksize must equal the number of subcarriers).In almost all practical cases, the number of samples *N* is chosen to be a power of 2, and theIDFT is realized as an IFFT. In the following, we will only speak of IFFTs and Fast Fourier Transforms (FFTs).

Note that the input to this IFFT is made up of *N* samples (the symbols for the different subcarriers), and therefore the output from the IFFT also consists of *N* values. These *N* values now have to be transmitted, one after the other, as temporal samples – this is the reason why we have a P/S (Parallel to Serial) conversion directly after the IFFT. At the receiver, we can reverse the process: sample the received signal, write a block of *N* samples into a vector – i.e., an S/P (Serial to Parallel) conversion – and perform an FFT on this vector. The result is an estimate ˜ *cn* of the original data c_n. Analog implementation of OFDM would require multiple LOs, each of which has to operate with little phase noise and drift, in order to retain orthogonality between the different subcarriers. This is usually not a practical solution. The success of OFDM is based on the above-described digital implementation that allows an implementation of the transceivers that is much simpler and cheaper. In particular, highly efficient structures exist for the implementation of an FFT (so-called “butterfly structures”), and the computational effort (per bit) of performing an FFT increases only with log (*N*). OFDM can also be interpreted in the time–frequency plane. Each index *i* corresponds to a (temporal) pulse; each index *n* to a carrier frequency. This ensemble of functions spans a grid in the time–frequency plane.

4. Explain the principle of direct sequence spread spectrum technique.(8) Nov. 2011

Direct Sequence Spread Spectrum (DS-SS):

A direct sequence spread spectrum (DS-SS) system spreads the baseband data by directly multiplying the baseband data pulses with a pseudo-noise sequence that is produced by a pseudo-noise code generator. A single pulse or symbol of the PN waveform is called a chip. Figure 1 shows a functional block diagram of a DS system with binary phase modulation. This system is one of the most widely used direct sequence implementations. Synchronized data symbols, which may be information bits or binary channel code symbols, are added in modulo-2 fashion to the chips before being phase modulated. A coherent or differentially coherent phase-shift keying (PSK) demodulation may be used in the receiver.

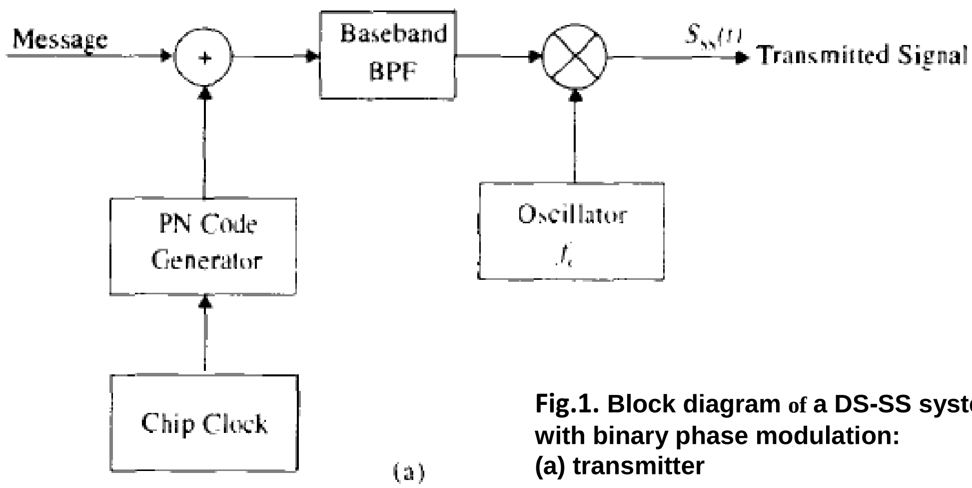


Fig.1. Block diagram of a DS-SS system with binary phase modulation:
(a) transmitter

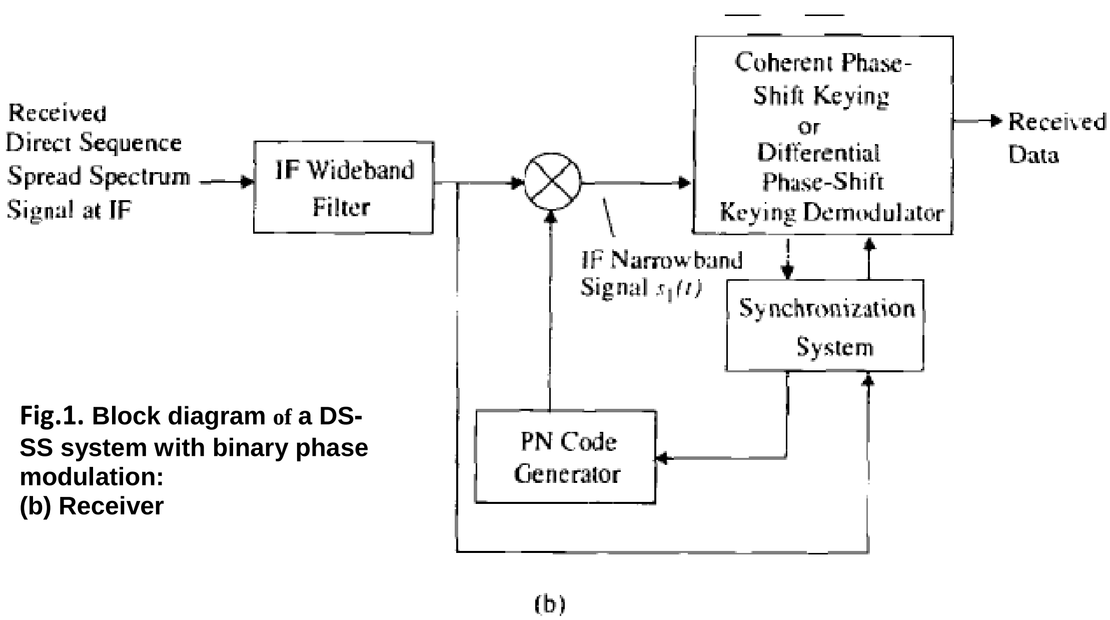


Fig.1. Block diagram of a DS-SS system with binary phase modulation:
(b) Receiver

The received spread spectrum signal for a single user can be represented as

$$S_{ss}(t) = \sqrt{\frac{2E_s}{T_s}} m(t) p(t) \cos(2\pi f_c t + \theta) \qquad \text{\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_ (1)}$$

where m(t) is the data sequence, p(t) is the PN spreading sequence, f_c is the carrier frequency, and θ is the carrier phase angle at t = 0. The data waveform is a time sequence of non overlapping rectangular pulses, each of which has amplitude equal to +1 or -1. Each symbol in m(t) represents a data symbol and has duration Each pulse in p(t) represents a chip, is usually rectangular with an amplitude equal to +1 or -1, and has a duration of The transitions of the data symbols and chips coincide such that the ratio Tç to is an integer. If is the bandwidth of and B is the bandwidth of $S_{SS}(t)$,and B is the bandwidth due to m(t) cos2πf$_c$(t) ,the spreading due to p(t) gives W_{ss} >B.

Figure 1(b) illustrates a DS receiver. Assuming that code synchronization has been achieved at the receiver, the received signal passes through the wideband filter and is multiplied by a local replica of the PN code sequence p(t). If p(t) = ±1, then $p^2(t)$ = I, and this multiplication yields the despread signal s(t) given by

$$s_1(t) = \sqrt{\frac{2E_s}{T_s}} m(t) \cos(2\pi f_c t + \theta) \qquad \text{\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_ } \textbf{(2)}$$

at the input of the demodulator. Because s_1 (t) has the form of a BPSK signal, the corresponding demodulation extracts m(t).

Figure 2.a.shows the received spectra of the desired signal and the interference at the output of the receiver wideband filter. Multiplication by the input. The signal bandwidth is reduced to fig.2.b, while the interference energy is spread over a bandwidth exceeding The filtering action of the demodulator removes most of the interference spectrum that does not overlap with the signal spectrum. Thus, most of the original interference energy is eliminated and does not affect the receiver performance. An approximate measure of the interference rejection capability is given by the ratio WSS/B, which is equal to the processing gain defined as

$$PG = \frac{T_s}{T_c} = \frac{R_c}{R_s} = \frac{W_{ss}}{2R_s} \qquad \text{\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_ (3)}$$

The greater the processing gain of the system, the greater will be its ability to suppress in-band interference.

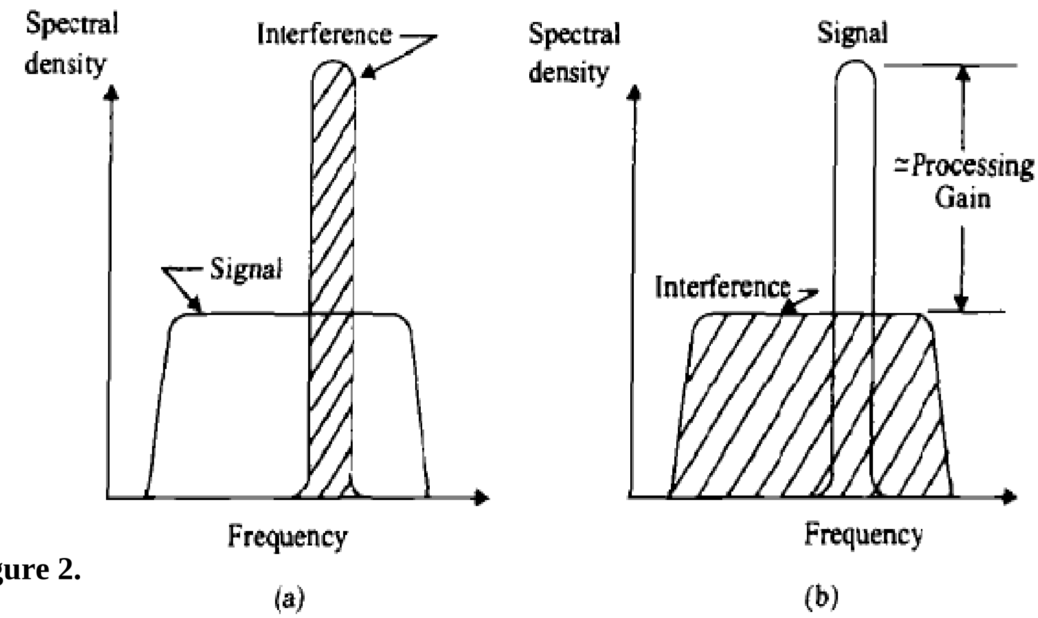


Figure 2.

Spectra of desired received signal with interference: (a) wideband filter output and (b) correlator output after despreading.

5. Explain in detail about the GSM logic channels. (8) Nov. 2011

GSM Logical Channels:

In addition to the actual payload data, GSM also needs to transmit a large amount of signaling information. These different types of data are transmitted via several *logical channels*. The name stems from the fact that each of the data types is transmitted on specific timeslots that are parts of *physical channels*. The first part of this section discusses the kind of data that is transmitted via logical channels. The second part describes the *mapping* of logical channels to physical channels.

Logical Channels of GSM

Traffic CHannels (TCHs)

Payload data are transmitted via the TCHs. The payload might consist of encoded voice data or "pure" data. There is a certain flexibility regarding the data rate: *Full-rate Traffic Channels* (TCH/F) and Half-rate Traffic Channels (TCH/H). Two half-rate channels are mapped to the same timeslot, but in alternating frames.

Full-Rate Traffic CHannels

• Full-rate voice channels: the output data rate of the voice encoder is 13 kbit/s. Channel coding increases the effective transmission rate to 22.8 kbit/s.

• Full-rate data channels: the payload data with data rates of 9.6, 4.8, or 2.4 kbit/s are encoded with Forward Error Correction (FEC) codes and transmitted with an effective data rate of 22.8 kbit/s.

Half-Rate Traffic CHannels

- Half-rate voice channels: voice encoding with a data rate as low as 6.5 kbit/s is feasible. Channel coding increases the transmitted data rate to 11.4 kbit/s.
- Half-rate data channels: payload data with rates of 4.8 or 2.4 kbits/s can be encoded with an FEC code, which leads to an effective transmission rate of 11.4 kbit/s..

Broadcast CHannels (BCHs)

BCHs are only found in the downlink. They serve as *beacon* signals. They provide the MS with the initial information that is necessary to start the establishment of any kind of connection. The MS uses signals from these channels to establish a synchronization in both time and frequency. Furthermore, these channels contain data regarding, e.g., cell identity. As the BSs are not synchronized with respect to each other, the MS has to track these channels not only before a connection is established, but all the time, in order to provide information about possible HOs.

Frequency Correction Channels (FCCHs)

The carrier frequencies of the BSs are usually very precise and do not vary in time, as they are based on rubidium clocks. However, dimension considerations and price considerations make it impossible to implement such good frequency generators in MSs. Therefore, the BS provides the MS with a frequency reference (an un-modulated carrier with a fixed offset from the nominal carrier frequency) via the FCCH. The MS tunes its carrier frequency to this reference; this ensures that both the MS and the BS use the same carrier frequency.

Synchronization Channel (SCH)

In order to transmit and receive bursts appropriately, an MS not only has to be aware of the carrier frequencies used by the BS but also of its frame timing on the selected carrier. This is achieved with the SCH, which informs the MS about the frame number and the *Base Station Identity Code* (BSIC). Decoding of the BSIC ensures that the MS only joins admissible GSM cells and does not attempt to synchronize to signals emitted by other systems in the same band.

Broadcast Control Channel (BCCH)

Cell-specific information is transmitted via the BCCH. This includes, e.g., *Location Area Identity* (LAI),7 maximum permitted signal power of the MS, actual available TCH, frequencies of the BCCH of neighboring BSs that are permanently observed by the MS to prepare for a handover, etc.

Common Control Channels (CCCHs)

Before a BS can establish a connection to a certain MS, it has to send some signaling information to all MSs in an area, even though only one MS is the desired receiver. This is necessary because in the initial setup stage, there is no *dedicated* channel established between the BS and a MS. CCCHs are intended for transmission of information to all MSs.

Paging CHannel (PCH)

When a request – e.g., from a landline – arrives at the BS to establish a connection to a specific MS, the BSs within a location area send a signal to all MSs within their range. This signal contains either the permanent *International Mobile Subscriber Identity* (IMSI) or the *Temporary Mobile Subscriber Identity* (TMSI) of the desired MS. The desired MS continues the process of establishing the connection by requesting (via a Random Access Channel (RACH)) a TCH, as discussed below. The PCH may also be used to broadcast local messages like street traffic information or commercials to all subscribers within a cell. Evidently, the PCH is only found in the downlink.

Random Access CHannel (RACH)

A mobile subscriber requests a connection. This might have two reasons. Either the subscriber wants to initiate a connection, or the MS was informed about an incoming connection request via the PCH. The RACH can only be found in the uplink. ***Access Grant CHannel (AGCH)*** Upon the arrival of a connection request via the RACH, the first thing that is established is a Dedicated Control Channel (DCCH) for this connection. This channel is called the *Standalone Dedicated Control Channel* (SDCCH), which is discussed below. This channel is assigned to the MS via the AGCH, which can only be found in the downlink.

Dedicated Control CHannels (DCCHs)

Similar to the TCHs, the DCCHs are bidirectional – i.e., they can be found in the uplink and downlink. They transmit the signaling information that is necessary during a connection. As the name implies, DCCHs are *dedicated* to one specific connection.

Standalone Dedicated Control CHannel (SDCCH)

After acceptance of a connection request, the SDCCH is responsible for further establishing this connection. The SDCCH ensures that the MS and the BS stay connected during the authentification process. After this process has been finished, a TCH is finally assigned for this connection via the SDCCH.

Slow Associated Control CHannel (SACCH)

Information regarding the properties of the radio link are transmitted via the SACCH. This information need not be transmitted very often, and therefore the channel is called *slow*. The MS informs the BS about the strength and quality of the signal received from serving BSs and neighboring BSs. The BS sends data about the power control and runtime of the signal from the MS to the BS.

Fast Associated Control CHannel (FACCH)

The FACCH is used for HOs that are necessary for a short period of time; therefore, the channel has to be able to transmit at a higher rate than the SACCH. Transmitted information is similar to that sent by the SDCCH. The SACCH is associated with either a TCH or a SDCCH; the FACCH is associated with a TCH.

---

6. Explain the block diagram of IS-95 transmitter. (8) Nov. 2011.

In 1991, the U.S.-based company Qualcomm proposed a system that was adopted by the *Telecommunications Industry Association (U.S.)* (TIA) as *Interim Standard 95*(IS-95). This system became the first commercial Code Division Multiple Access (CDMA) system that achieved wide popularity. In the years after 1992, cellular operators in the U.S.A. started to switch from analog (Advanced Mobile Phone System (AMPS)) to digital communications. While the market remained fragmented, IS-95 was adopted by a considerable number of operators, and by 2005 was used by two of the four major operators in the U.S.A. It also obtained a dominant market position in South Korea.

The original IS-95 system did not fully exploit the flexibility inherent in CDMA systems; however, later refinements and modifications did make the system more flexible, and thus ready for data communications. In the late 1990s, the need for further enhancement of data communications capabilities became apparent. The new third-generation systems needed to be able to sustain high data rates, thus enabling audio- and video-streaming, Web browsing, etc. This would require higher data rates and flexible systems that could easily support multiple data rates with fine granularity. CDMA seemed well suited to this approach, and was chosen by all major manufacturers. However, no unique standard evolved. The IS-95 proponents (mostly U.S.-based) developed the so-called CDMA 2000 standard, which is backward-compatible with IS-95, and allows a seamless transition.

IS-95 is a CDMA system with an additional Frequency Division Multiple Access (FDMA) component. The available frequency range is divided into frequency bands of 1.25 MHz; duplexing is done in the frequency domain. In the U.S.A., frequencies from 1850–1910MHz are used for the uplink, and 1930–1990MHz are used for the downlink band Within each band, traffic channels, control channels, and pilot channels are separated by the different codes (chip sequences) with which they are spread. IS-95 specifies two possible speech coder rates: 13.3 or 8.6 kbit/s. In both cases, coding increases the data rate to 28.8 kbit/s. The signal is then spread by a factor of 64, resulting in a chip rate of 1.2288 Mchip/s. Theoretically, each cell can sustain 64 speech users. In practice, this number is reduced to 12–18, due to imperfect power control, non-orthogonality of spreading codes, etc. The downlink signals generated by one Base Station (BS) for different users are spread by different Walsh–Hadamard sequences (see Section 18.2.6), and thus orthogonal to each other. This puts an upper limit of 64 channels on each carrier. In the uplink, different users are separated by spreading codes that are not strictly orthogonal. Furthermore, interference from other cells reduces signal quality at the BS and Mobile Station

(MS). BSs use transmit powers between 8 and 50 W, depending on the coverage area required. MSs use peak powers of some 200mW; accurate power control makes sure that all signals arriving at the BS have the same signal strength. Traffic channels and control channels are separated by different spreading codes. All BSs are synchronized, using signals from GPS (Global Positioning System) to obtain an accurate system time. This synchronization makes it much easier for the MS to detect signals from different BSs and manage the handover from cell to cell. The requirements for the network and switching system, as well as operating support, servicing, and billing, are quite similar to those in GSM, and will not be repeated here. Similarly, billing considerations are the same as in GSM.

Block diagramof a IS-95 Transmitter.

we have considered the case when the output of the channel coder actually has a data rate of 28.8 kbit/s – i.e., a source rate of 14.4 or 9.6 kbit/s. However, depending on the source, data, a

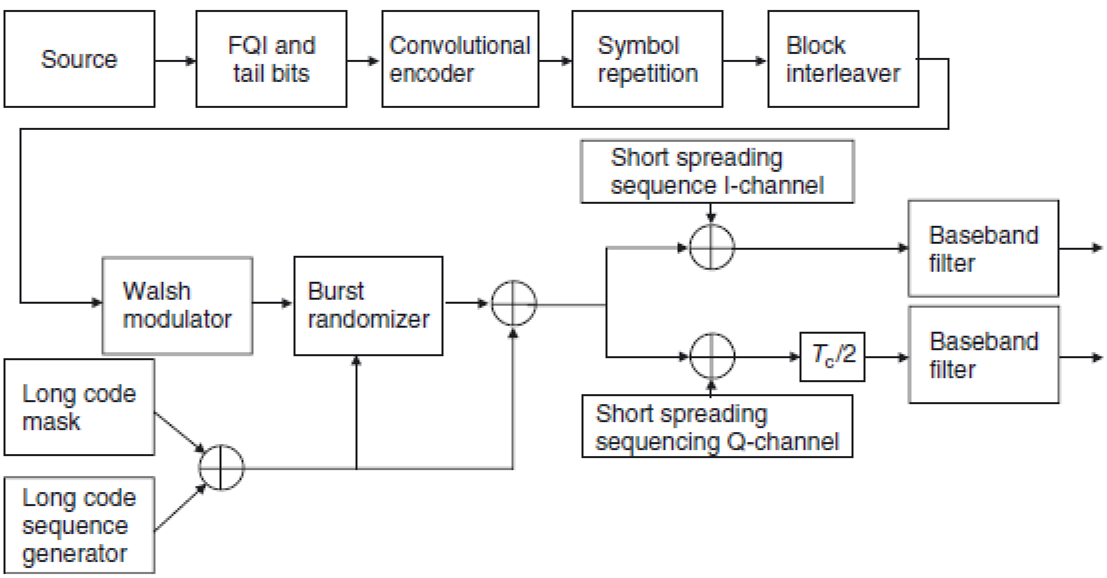


Figure .1 Block diagram of an IS-95 mobile station transmitter.

lower rate (14.4 kbit/s, 7.2 kbit/s, or 3.6 kbit/s) can also be the output of a convolutional encoder. In this case, encoded symbols are repeated (several times, if necessary) until a data rate of 28.8 kbit/s is achieved. It is these repeated data that are sent to the block interleaver for further processing. However, it would waste resources to transmit all of these repeated data at full power. For the uplink, this problem is solved by gating off the transmitter part of the time. If, e.g., the coded data rate is 14.4 kbit/s (source data rate 7.2 kbit/s), then the transmitter is turned on only 1/2 of the time. As a consequence, average transmit power is only 1/2 of the full-data-rate case, and the interference seen by other users is only half as large. Determination of the time for gating off the transmitter is actually quite complicated. The first issue is that gating has to be coordinated with the interleaver. For example, for the 7.2-kbit source data rate mode, each 1.25-ms-long group of output symbols is repeated once. Gating thus eliminates one of these two symbol groups.

7. Write a note on second generation and third generation wireless networks and standards. (8) Nov. 2012

Second Generation Wireless Networks:

Second generation wireless systems employ digital modulation and advanced call processing capabilities. Examples of second generation wireless systems include the Global System for Mobile (GSM), the TDMA and CDMA U.S. digital standards (the Telecommunications Industry Association IS-54 and 15-95 standards), Second Generation Cordless Telephone (CT2), the British standard for cordless telephony, the Personal Access Communications System (PACS) local loop standard, and Digital European Cordless Telephone (DECT), which is the European standard for cordless and office telephony. There are many other second generation systems,

Second generation wireless networks have introduced new network architectures that have reduced the computational burden of the MSC. GSM has introduced the concept of a base station controller (BSC)which is inserted between several base stations and the MSC. In PACS(WACS, the BSC is called a radio port control unit. This architectural change has allowed the data interface between the base station controller and the MSC to be standardized, thereby allowing carriers to use different manufacturers for MSC and BSC components. This trend in standardization and interoperability is new to second generation wireless networks. Eventually, wireless network components, such as the MSC and BSC, will be available as off-the-shelf components, much like their wire-line telephone counterparts. All second generation systems use digital voice coding and digital modulation. The systems employ dedicated control channels within the air interface for simultaneously exchanging voice and control information between the subscriber, the base station, and the MSC while a call is in progress. Second generation systems also provide dedicated voice and signaling trunks between MSCs, and between each MSC and the PSTN.

In contrast to first generation systems, which were designed primarily for voice, second generation wireless networks have been specifically designed to provide paging, and other data services such as facsimile and high-data rate network access. The network controlling structure is more distributed in second generation wireless systems, since mobile stations assume greater control functions. In second generation wireless networks, the handoff process is mobile-controlled and is known as mobile assisted handoff . The mobile units in these networks perform several other functions not performed by first generation subscriber units, such as received power reporting, adjacent base station scanning, data encoding, and encryption.

DECT is an example of a second generation cordless telephone standard which allows each cordless phone to communicate with any of a number of base stations, by automatically selecting the base station with the greatest signal level. In DECT, the base stations have greater control in terms of switching, signaling, and controlling handoffs. In general, second generation systems have been designed to reduce the computational and switching burden at the base station or MSC, while providing more flexibility in the channel allocation scheme so that systems may be deployed rapidly and in a less coordinated manner.

Third Generation Wireless Networks

Third generation wireless systems will evolve from mature second generation systems. The aim of third generation wireless networks is to provide a single set of standards that can meet a wide range of wireless applications and provide universal access throughout the world. In third generation wireless systems, the distinctions between cordless telephones and cellular telephones will disappear, and a universal personal communicator (a personal handset) will provide access to a variety of voice, data, and video communication services. Third generation systems will use the Broadband Integrated Services Digital Network (B-ISDN) to provide access to information networks, such as the Internet and other public and private databases. Third generation networks will carry many types of information (voice, data, and video), will operate in varied regions (dense or sparsely populated regions), and will serve both stationary users and vehicular users traveling at high speeds. Packet radio communications will likely be used to distribute network control while providing a reliable information transfer. The terms Personal Communication System (PCS) and Personal Comrnunication Network (PCN) are used to imply emerging third generation wireless systems for hand-held devices. Other names for PCS include Future Public Land Mobile Telecommunication Systems (FPLMTS) for worldwide use which has more recently been called International Mobile Telecommunication (IMT-2000), and Universal Mobile Telecommunication System (UMTS) for advanced mobile personal services in Europe.

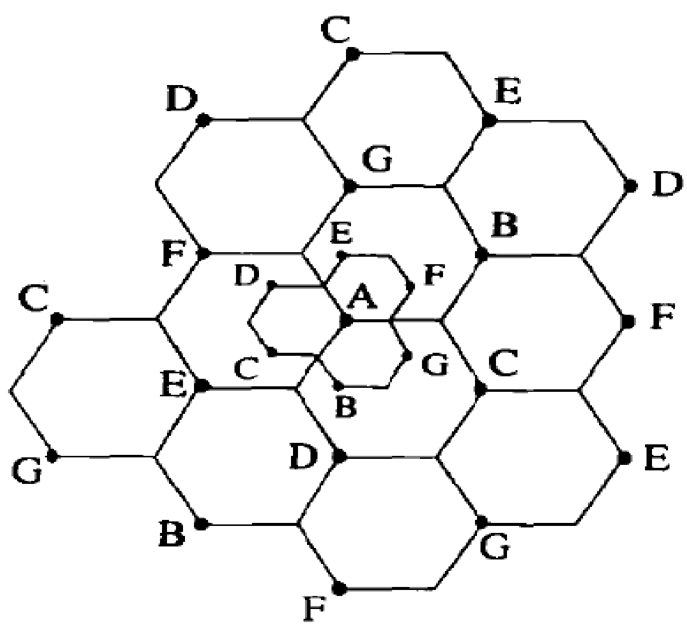
8.Discuss some methods to increase the capacity of wireless communication system.
(8) Nov. 2011

Improving Capacity In wireless communication Systems:

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. At this point, cellular design techniques are needed to provide more channels per unit coverage area. Techniques such as cell splitting, sectoring, and coverage zone approaches are used in practice to expand the capacity of cellular systems. Cell splitting allows an orderly growth of the cellular system. Sectoring uses directional antennas to further control the interference and frequency reuse of channels. The zone microcell concept distributes the coverage of a cell and extends the cell boundary to hard-to-reach places. While cell splitting increases the number of base stations in order to increase capacity, sectoring and zone microcells rely on base station antenna placements to improve capacity by reducing co-channel interference. Cell splitting and zone microcell techniques do not suffer the trunking inefficiencies experienced by sectored cells, and enable the base station to oversee all handoff chores related to the microcells, thus reducing the computational load at the MSC. These three popular capacity improvement techniques will be explained in detail.

Cell Splitting:

Cell splitting is the process of subdividing a congested cell into smaller cells, each with its own base station and a corresponding reduction in antenna height and transmitter power. Cell splitting increases the capacity of a cellular system since it increases the number of times that channels are reused. By defining new cells which have a smaller radius than the original cells and by installing these smaller cells (called microcells) between the existing cells, capacity increases due to the additional number of channels per unit area.



Cell Splitting

Sectoring

Cell splitting achieves capacity improvement by essentially rescaling the system. By decreasing the cell radius R and keeping the co-channel reuse ratio D/R unchanged, cell splitting increases the number of' channels per unit area. However, another way to increase capacity is to keep the cell radius unchanged and seek methods to decrease the D/R ratio. In this approach, capacity improvement is achieved by reducing the number of cells in a cluster and thus increasing the frequency reuse. However, in order to do this, it is necessary to reduce the relative interference without decreasing the transmit power. The co-channel interference in a cellular system may be decreased by replacing a single omni-directional antenna at the base station by several directional antennas, each radiating within a specified sector. By using directional antennas, a given cell will receive interference and transmit with only a fraction of the available co-channel cells. The technique for decreasing co-channel interference and thus increasing system capacity by using directional antennas is called sectoring. The factor by which the co-channel interference is reduced depends on the amount of sectoring used. A cell is normally partitioned into three 120° sectors or six 60° sectors.