USN										

Sub:	Wireless and Cellular Communication					Sub Code: 18EC81
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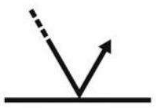
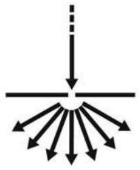
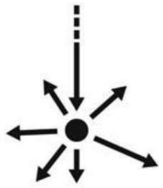
1 a	Explain Path Loss model for free space propagation along with equations. 6 • Path loss is defined as the difference (in dB) between the effective transmitted power and the received power • Free-space path loss is defined as the path loss of the free-space model $PL(dB)$= 10 $log(Pt/Pr)$=−10 $log[(GtGr\lambda 2)/(4\pi)\ 2d2$ • The gain of an antenna G is related to its affective aperture Ae by G = 4πAe / λ2 where Ae is related to the physical size of the antenna, λ is related to the carrier frequency (λ = c/f = 2πc / ωc) where f is carrier frequency in Hertz , c is speed of light in meters/sec o ωc is carrier frequency in radians per second • Friis free space model is only valid to predict "Pr" for the values of "d" which are in far-field from transmitting antenna • The far-field or Fraunhofer region of a transmitting antenna is defined as the region beyond the far-field distance df given by: df= $2D2/\lambda$, D is the largest physical dimension of the antenna o Additionally df>>D
1 b.	Explain briefly three basic propagation mechanisms. eflection, diffraction, and scattering are the three basic propagation mechanisms that impact propagation in mobile communication systems. Reflection occurs when a propagating electromagnetic wave impinges upon an obstacle that has very large dimensions compared to the wavelength of the propagating wave. During propagation, the electromagnetic wave usually encounters obstacles such as: walls, buildings, Earth's surface, water surface, etc. Depending on the reflecting surface (obstacle surface), the reflected signal suffers additional attenuation. If there is a large number of reflected signals at the receiving point, the received signal is very unstable.  Diffraction happens when, on the radio path between the transmitter and the receiver, the edge of an impenetrable obstacle, ie. an obstacle of sharp edges, is encountered, and the dimension of the object is significantly larger than the wavelength of the emitted electromagnetic wave. In diffraction, the electromagnetic wave "breaks" on the sharp edge of the obstacle and continues to propagate further weakened. The diffraction phenomenon enables signal transmission from transmitter to receiver,

although there is an impenetrable obstacle between them and there is no direct optical visibility.



Scattering occurs when the medium through which the wave propagates consists of objects with dimensions that are small compared to the wavelength, and where the number of obstacles per unit volume are large. When an electromagnetic wave encounters such objects, the reflected energy is scattered in all directions. Obstacles of this type can be: traffic signs, lampposts, leaves, etc.



The multi path propagation effect is a propagation phenomenon which implies that electromagnetic waves reach the receiving antenna by multiple paths.

1 c. If a transmitter produces 50 watts of power, express the transmit power in units of (a) dBm, and (b) dBW. If 50 watts 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.

(a) To express the transmit power of 50 watts in dBm:

1 watt = 30 dBm

*So, 50 watts = 10 * log10(50 watts) dBm*

*= 10 * log10(50000 mW) dBm*

= 47 dBm (rounded to the nearest whole number)

Therefore, the transmit power of 50 watts can be expressed as 47 dBm.

(b) To express the transmit power of 50 watts in dBW:

1 watt = 0 dBW

*So, 50 watts = 10 * log10(50 watts) dBW*

*= 10 * log10(50000 mW) dBW*

= 17 dBW (rounded to the nearest whole number)

Therefore, the transmit power of 50 watts can be expressed as 17 dBW.

To find the received power in dBm at a free space distance of 100 m from the antenna, we can use the Friis transmission equation:

Pr = Pt + Gt + Gr - L

where:

Pr is the received power

Pt is the transmitted power (in watts)

Gt is the transmitter antenna gain

Gr is the receiver antenna gain

L is the path loss in dB

Since the antenna is a unity gain antenna, Gt and Gr are both 0 dBi (or 2.15 dBi in linear scale). Also, at a free space distance, the path loss L can be calculated using:

$L = 20 \log10(4\pi d/\lambda)$

where:

d is the distance between the transmitter and receiver (in meters)

λ is the wavelength (in meters), which can be calculated as λ = c/f where c is the speed of light (299,792,458 m/s) and f is the frequency (in Hz)

So, for a carrier frequency of 900 MHz:

$\lambda = c/f = 299{,}792{,}458 \text{ m/s} / 900{,}000{,}000 \text{ Hz} = 0.3331 \text{ meters}$

Thus, $L = 20 \log10(4\pi * 100 / 0.3331) = 92.45$ *dB*

Substituting the values into the Friis transmission equation, we get:

$Pr = 50 + 0 + 0 - 92.45 = -42.45$ *dBm (rounded to two decimal places)*

Therefore, the received power in dBm at a free space distance of 100 m from the antenna is -42.45 dBm.

To find the received power at a distance of 10 km, we can use the same formula with a distance of 10,000 m:

$L = 20 \log10(4\pi * 10000 / 0.3331) = 152.45$ *dB*

Substituting the values into the Friis transmission equation, we get:

$Pr = 50 + 0 + 0 - 152.45 = -102.45$ *dBm (rounded to two decimal places)*

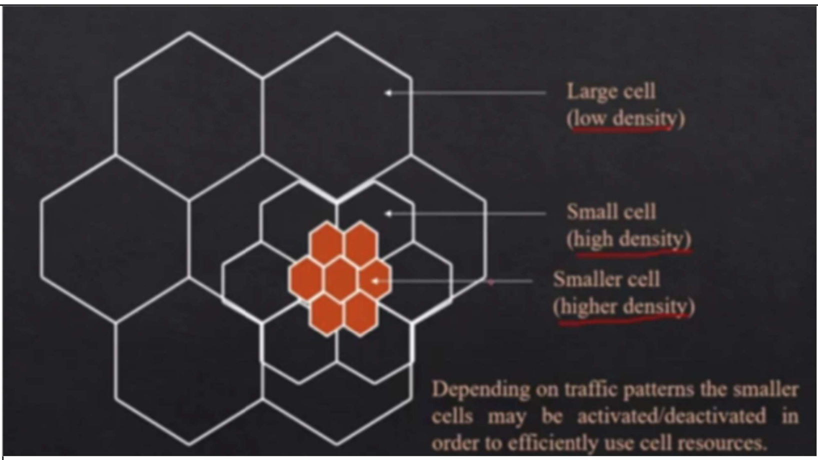
Therefore, the received power in dBm at a distance of 10 km is -102.45 dBm.

2 a. Distinguish between delay spread and coherence bandwidth.

a) Delay Spread

- The delay spread is the amount of time that elapses between the echo (typically the first line-of-sight component and the last arriving) path.
- It is also referred to as the Multipath Intensity Profile, or power delay profile as it measures the multipath richness of channel.
- It specifies the duration of the channel impulse response $h(\tau, t)$.

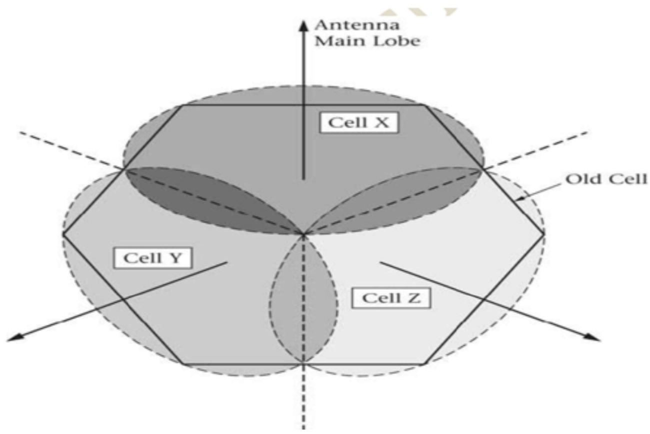
	Delay spread can be quantified through different metrics, although the most common one is the root mean square, the mean delay of the channel is : $$\mu_\tau = \frac{\int_0^\infty \Delta\tau A_\tau(\Delta\tau)d(\Delta\tau)}{\int_0^\infty A_\tau(\Delta\tau)d(\Delta\tau)}$$ And the rms delay spread is given by : $$\tau_{\text{rms}} = \sqrt{\frac{\int_0^\infty (\Delta\tau - \mu_\tau)^2 A_\tau(\Delta\tau)d(\Delta\tau)}{\int_0^\infty A_\tau(\Delta\tau)d(\Delta\tau)}}$$ The delay spread can be found by channel autocorrelation function A ($\Delta\tau$, 0) by setting $\Delta t = 0$. b) Coherence bandwidth B c - • Bandwidth over which the channel transfer function remains virtually constant. • Channel is considered relatively constant over the transmit bandwidth – happens if the transmission BW < 'coherence' BW (i.e. Bc of the channel). • A signal sees a narrowband channel if the bit duration >> inter-arrival time of reflected waves - ISI is small.
2 b.	Distinguish between Doppler spread and coherence time. c) Doppler Spread • In channels where transmitter and receiver move relative to each other the signal frequency is shifted depending on the velocity and as a result the spectrum of the narrowband signal transmitted widens. • This so-called Doppler effect can be observed on passing cars, moving stars and wireless communications. d) Coherence Time • It gives the time period over which the channel is significantly correlated i.e. the time duration over which two received signals have a strong potential for amplitude correlation. • Coherence time is actually a statistical measure of the time duration over which the channel impulse response is essentially invariant. Mathematically $$\lvert t_1 - t_2 \rvert \le T_c \Rightarrow \mathbf{h}(t_1) \approx \mathbf{h}(t_2)$$ $$\lvert t_1 - t_2 \rvert > t_c \Rightarrow \mathbf{h}(t_1) \text{ and } \mathbf{h}(t_2) \text{ are uncorrelated}$$ Coherence time and doppler spread inversely related $$T_c \approx \frac{1}{f_D}$$
2 c.	Explain the analysis of cellular systems. (a) Cell splitting



- **It 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).
- The increased no. of cells would increase the no. of clusters which in turn would increase the no. of channels reused and capacity.
- Each cell is divided into six new smaller cells with approximately one-quarter the area of the larger cells and use the same channel as shown in figure 3.4.
- To preserve the overall system frequency reuse plan, the transmit power of these cells must be reduced by a factor of approximately 16 or 12dB.

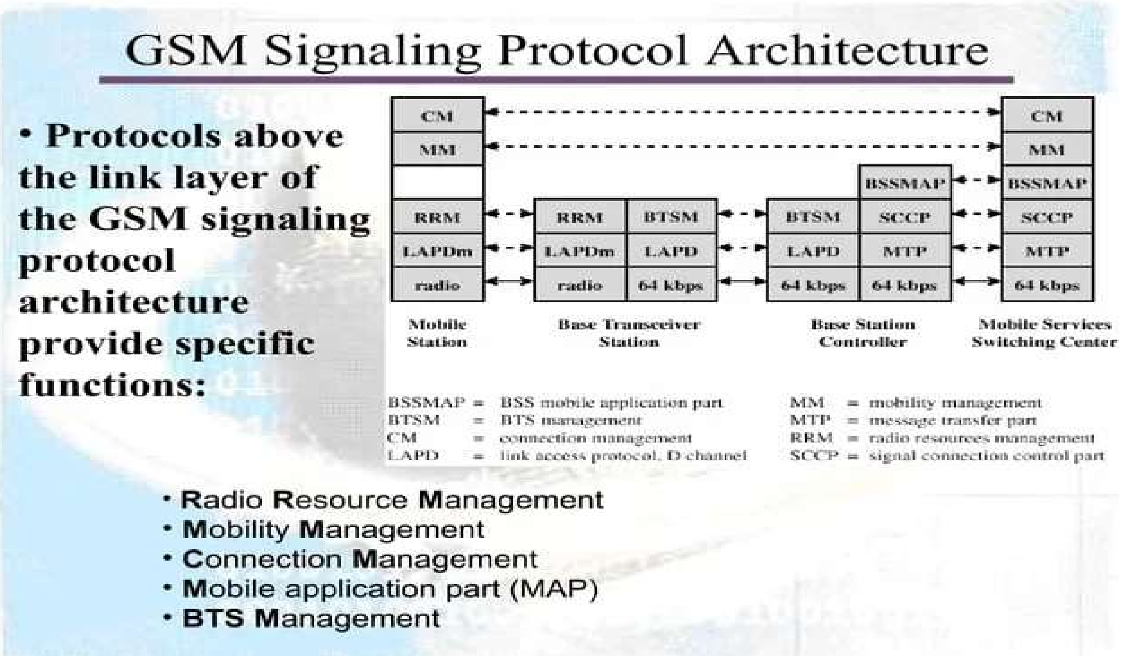
(b) Cell sectoring

Cell Sectoring - Uses directional antennas to effectively split a cell into 3 or sometimes 6 new cells.



- Reuse Factor/ frequency reuse ratio : **$Q = D/R = (3N)^{1/2}$**
- 3 directional antennas with 120^{o} beamwidth to illuminate the entire area previously services by omnidirectional antenna
- **Cell Sectoring** provides co-channel interference reduction, hence S/I ratio increases.
- It does not require new cell sites and additional antennas and triangular mounting only.

3 Describe GSM protocols and signaling model with a neat diagram. 10



GSM architecture is a layered model that is designed to allow communications between two different systems. The lower layers assure the services of the upper-layer protocols. Each layer passes suitable notifications to ensure the transmitted data has been formatted, transmitted, and received accurately.

Layer 1 − The physical layer. It uses the channel structures over the air interface.

Layer 2 − The data-link layer. Across the Um interface, the data-link layer is a modified version of the Link access protocol for the D channel (LAP-D) protocol used in ISDN, called Link access protocol on the Dm channel (LAP-Dm). Across the A interface, the Message Transfer Part (MTP), Layer 2 of SS7 is used.

Layer 3 − GSM signalling protocol's third layer is divided into three sublayers −

- Radio Resource Management (RR),
- Mobility Management (MM), and
- Connection Management (CM).

MS to BTS Protocols

The RR layer is the lower layer that manages a link, both radio and fixed, between the MS and the MSC. For this formation, the main components involved are the MS, BSS, and MSC. The responsibility of the RR layer is to manage the RR-session, the time when a mobile is in a dedicated mode, and the radio channels including the allocation of dedicated channels.

The MM layer is stacked above the RR layer. It handles the functions that arise from the mobility of the subscriber, as well as the authentication and security aspects.

The CM layer is the topmost layer of the GSM protocol stack. This layer is responsible for Call Control, Supplementary Service Management, and Short Message Service Management.

BSC Protocols

The BSC uses a different set of protocols after receiving the data from the BTS. The Abis interface is used between the BTS and BSC. At this level, the radio resources at the lower portion of Layer 3 are changed from the RR to the Base Transceiver Station Management (BTSM). The BTS management layer is a relay function at the BTS to the BSC.

The RR protocols are responsible for the allocation and reallocation of traffic channels between the MS and the BTS. These services include controlling the initial access to the system, paging for MT calls, the handover of calls between cell sites, power control, and call termination.

MSC Protocols

At the MSC, starting from the BSC, the information is mapped across the A interface to the MTP Layers 1 through 3. Here, Base Station System Management Application Part (BSS MAP) is said to be the equivalent set of radio resources. The relay process is finished by the layers that are stacked on top of Layer 3 protocols, they are BSS MAP/DTAP, MM, and CM.

3 b Explain the various logical channels used in GSM.

The categories of logical channels is given below:

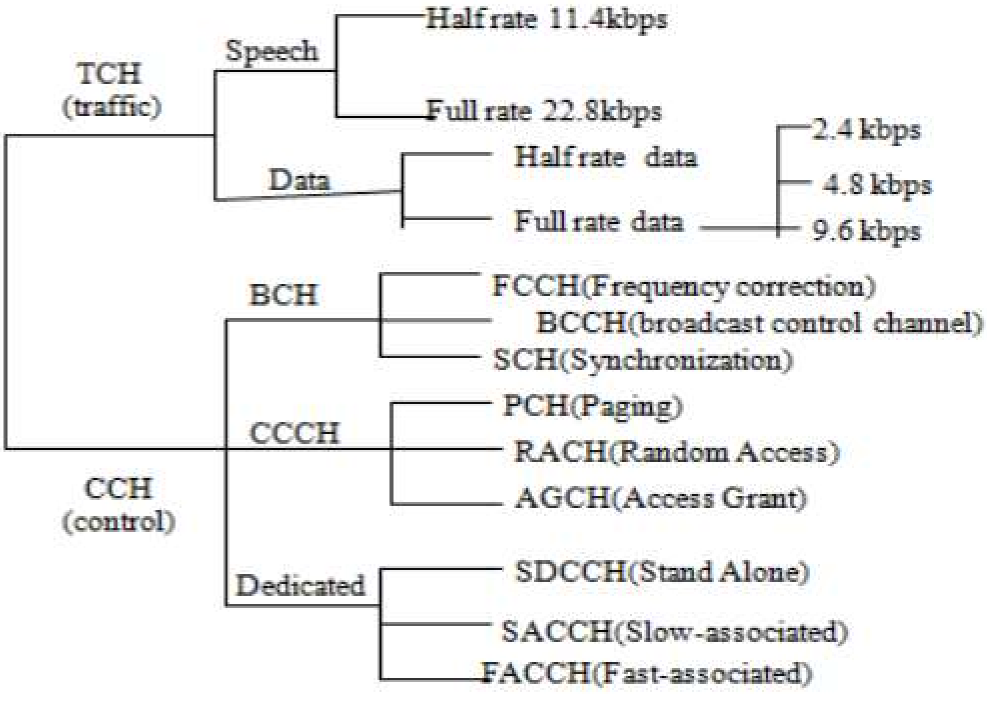


Figure 5: Types of logical channels

There are two types of GSM logical channels, called *Traffic CHannels* (TCH) and *Control CHannels* (CCH). Traffic channels carry digitally encoded user speech or user data and have identical functions and formats on both the forward and reverse link. Control channels carry signaling and synchronizing commands between the Base station and the mobile station. Certain types of control channels are defined for just the forward or reverse link.

1. GSM traffic channels: GSM traffic channels may be either full-rate or half-rate and may carry either digitized speech or user data. Full rate channels are those in which a user gets a time slot per frame. Half rate channels are those in which the user gets a time slot per alternate frame. More number of users i.e. 16 users can get the same channel simultaneously at a compromise of data rate. Whether a physical channel will be used as full rate or half rate totally depends upon the service provider who will decide according to the congestion in the network.

a) Full-Rate TCH

Full-Rate Speech Channel (TCH/FS) :The full-rate speech channel carries user speech which is digitized at a raw data rate of 13 Kbps. With GSM channel coding added to the digitized speech, the full-rate speech channel carries 22.8 Kbps.Full Rate data channels are used in three configurations and are termed as TCH/F9.6, TCH/F4.8 and TCH/F2.4.

b) Half-Rate TCH

The following half-rate speech and data channels are supported:

- Half-Rate Speech Channel (TCH/HS): The half-rate speech channel has been designed to carry digitized speech which is sampled at a rate half that of the full-rate channel. With GSM channel coding added to the digitized speech, the half-rate speech channel will carry 11.4 Kbps. Half rate channels are of two types : TCH/H4.8 and TCH/H2.4
- Half-Rate Data channels: These channels carry user data at 4.8 and 2.4 Kbps respectively. With additional forward error correction coding applied by the GSM standard, the data is sent at 11.4 Kbps for both the channels.

2. GSM Control Channels (CCH): There are three main control channels in the GSM system. These are the

i) Broadcast CHannel (BCH), ii) The Common Control CHannel (CCCH), and the iii) Dedicated Control CHannel (DCCH)

Each control channel consists of several logical pieces of information which is discussed in detail:

i) Broadcast Channels (BCH): 34 ARFCNs out of 124 are defined as Broadcast control channels. BCH is sent only on the TS0 (Time slot 0) of each frame. The other 7 time slots of each frame can be used for sending user traffic. Unlike TCHs which are duplex, BCHs only use the forward link. BCH carries information About frequency, time synchronization, occasionally monitored by neighbouring mobiles for MAHO decisions.

Three types of broadcast channels exist which are:

a) Broadcast Control CHannel (BCCH) : The BCCH is a forward control channel that is used to broadcast information such as cell and network identity, and operating characteristics of the cell (current control channel structure, channel availability, and congestion). The BCCH also broadcasts a list of channels that are currently in use within the cell. Frame 2 through frame 5 in a control multiframe (4 out of every 51 frames) contain BCCH data.

b) Frequency Correction CHannel (FCCH): The FCCH is a special data burst which occupies TS 0 for the very first GSM frame (frame 0) and is repeated every ten frames within a control channel multiframe. The FCCH allows each subscriber unit to synchronize its internal frequency standard (local oscillator) to the exact frequency of the Base Station.

c) Synchronization CHannel (SCH): SCH is broadcasted in TS 0 of the frame immediately following the FCCH frame and is used to time synchronize the serving Base Station while allowing each mobile to frame synchronize with the BS. The Frame Number (FN), which ranges from 0 to 2,715,647 (26 frames ×× 51 multiframes ×× 2048superframe), is sent with the base station identity code (BSIC) during the SCH burst. The BSIC is uniquely assigned to each BTS in a GSM system. Since a mobile may be as far as 30 km away from a serving BS, it is often necessary to adjust the timing of a particular mobile user such that the received signal at the base station is synchronized with the Base Station clock. The BS issues coarse timing advancement commands to the mobile stations over the SCH,as well. The SCH is transmitted once every ten frames within the control channel multiframe, as shown in Figure 2(a).

ii) Common Control CHannels (CCCH): On the broadcast (BCH) ARFCN,the common control channels occupy TS 0 of every GSM frame that is not otherwise used by the BCH or the Idle frame. CCCH consists of three different channels:

a) Paging CHannels (PCH):

- Notifies a mobile about an incoming call.
- Transmits the IMSI of the target subscriber.
- Provides the cell broadcast ASCII text msgs.

b) Random Access CHannels (RACH):

- It is reverse link channel
- To acknowledge the page message from the PCH or request for a call setup.

c) Access Grant CHannel(AGCH):

- It is forward link channel.
- It is a response to the RACH message from mobile.
- It conveys the information about time slot and ARFCN allotted to a mobile station for authentication process or for starting the communication.

iii) Dedicated Control CHannels (DCCH): There are three types of dedicated control channels in GSM, and, like traffic channels they are bidirectional and have the same format and function on both the forward and reverse links. Like TCHs, DCCHs may exist in any time slot and on any ARFCN except TS0 of the BCH ARFCN.

a) Stand alone Dedicated Control CHannel (SDCCH)

- Carries data following connection of mobile.
- Ensures mobile and BS connectivity during authentication and alert messages.

b) Slow Associated Control CHannel (SACCH)

- To acknowledge a page message from PCH

- Transmitted during the 13th frame and 26th frame
- SACCH is used to send slow but regularly changing data like transmit power level instructions.

c) Fast Associated Control CHannel (FACCH)

- Carries urgent messages (HO requests)
- Gains access to a time slot by stealing frames from the traffic channel
- The FACCH gains access to a time slot by "stealing" frames from the traffic channel to which it is assigned. This is done by setting two special bits, called stealing bits, in a TCH forward channel burst. If the stealing bits are set, the time slot is known to contain FACCH data, not a TCH, for that frame.

For proper understanding of the reader, a control channel multiframe in the forward and reverse channel is as shown in Figure 2. Unlike traffic multiframe, a control channel mutiframe has 51 TDMA frames and is of 235 ms duration. This is done intentionally to ensure that any GSM subscriber (whether in the serving or adjacent cell) will be certain to receive the SCH and FCCH transmissions from the BCH, no matter what particular frame or time slot they are using.

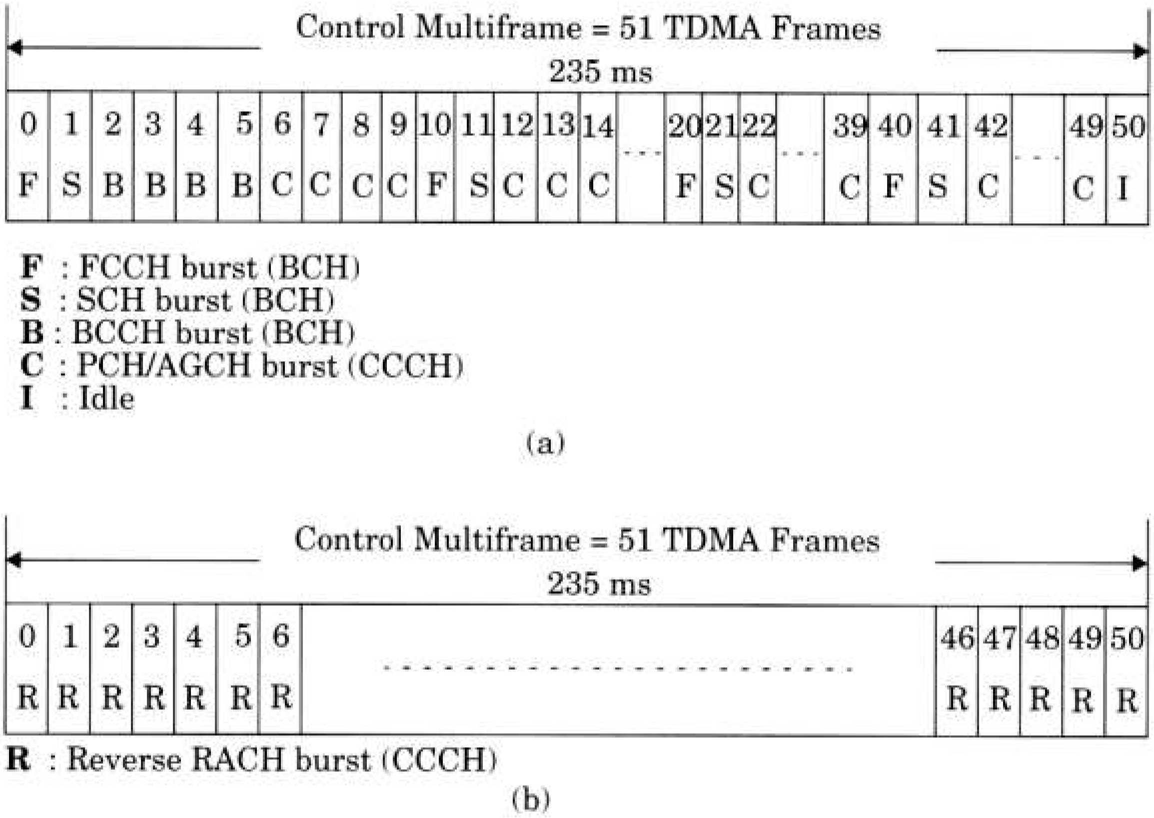


Figure 6 a) The TS0 time slots of every frame in a control channel multiframe of the forward link. b) The TS0 time slots of every frame in a control channel multiframe of the reverse link link

4a List out the ten operations in call set up in GSM system. Explain in detail authentication and ciphering mode operation.

Call set-up within GSM consists of the following main steps:

- RR connection establishment
- Service request
- Authentication
- Ciphering mode setting
- IMEI Check
- TMSI-Reallocation
- Call initiation
- Assignment of a traffic channel
- User alerting
- Call accepted

1. To start a call, the MS sends a request to the Base Station Subsystem(BSS), for creating a radio channel between them. The BSS reserves a channel for the MSS and assigns a frequency and time slot for that channel. this channel is used by the MS to communicate with the BSS.

2. A confirmation is sent by the MS implying that the radio channel has been established and the BSS can now create a connection to the MSC.

3. Next step after connection setup, the subscriber needs to be authenticated, which is done by verifying the number stored in the SIM card, and the subscriber's authenticity is thus verified. After the verification, a connection is set up between the MS and Mobile Service Center(MSC).

4. The MSC encrypts the messages which are transmitted to the BTS so that data cannot be read or deleted by hackers. The BTS then forwards the message to the MS. The BTS also works with encrypted data. It both receives and sends encrypted data.

5. Before initiating the call setup, the MSC authenticates the service requested by the subscriber. This information about subscribers is checked in the Virtual Location Register, which stores temporary subscriber data like location, preferences, and allowed services. Once the VLR sends a green signal to the request from MS, the MSC can start the call setup.

6. The MSC reserves a voice channel between the MSC and the BSS. The change to voice mode is notified by the BSS, and the MS returns a confirmation message. The MSC is in charge of setting up the call to the called number. When the call is received in the public switched telephone network(PSTN), the MSC is notified that the person whom we are calling is receiving the call in form of a ringtone as means of notification, at which point the originating MS receives a caller tune as a notification.

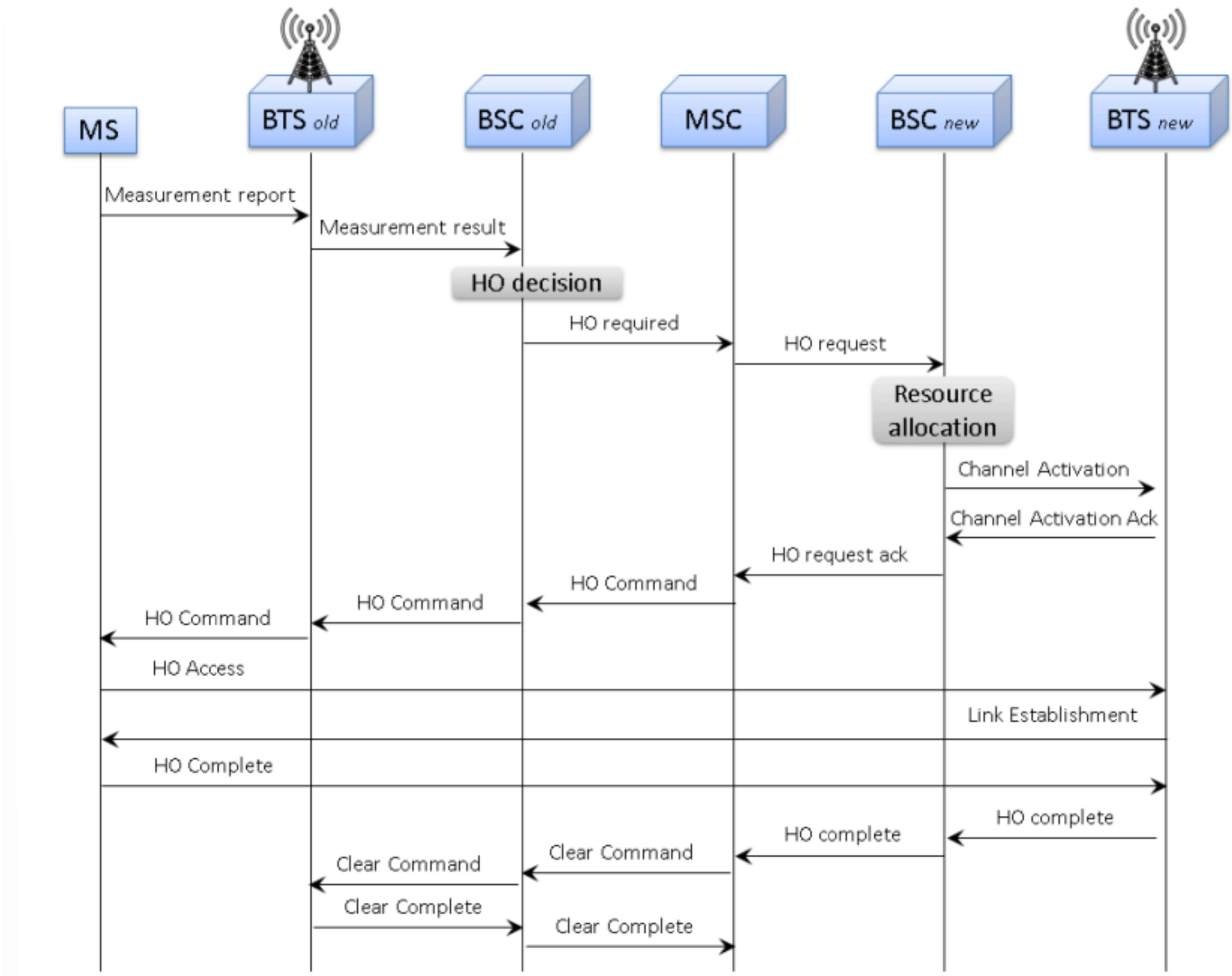
Procedure for Call Establishment

When a call is made to a mobile phone in a GSM network, it is first received by the nearest Base Station System (BSS). The BSS then establishes a connection with the Mobile Switching Center (MSC) which acts as a control center for the entire network. The MSC then sends a paging request to the Mobile Station (MS), which is the mobile phone, to indicate an incoming call. The MS sends an acknowledgment to the MSC, which then sends a setup request to the BSS to establish a connection between the calling and called parties.

Once the connection is established, the audio signals are transmitted between the calling and called parties through the BSS. The MSC continuously monitors the call and ensures that it remains connected until it is terminated either by the calling or called party.

4 b Explain the intra BSC handover operation in GSM.

During Handover management procedure, when a mobile user travels from one area of coverage to another within a call's duration, the call is transferred to the new cell's base station. ❖ Mobile continuously transmits measurement reports (signal strength, distance, traffic volume) ❖ BSS decides when to perform handover and request handover from the MSC ❖ MSC causes the new BSS to prepare a channel for the handover, and frees handover to the mobile as soon as this channel is acknowledged ❖ Mobile station accesses new BSS and receives information about the new air interface (timing advance value, transmitter power level) ❖ Once the mobile can occupy the new channel, resources of the old BSS are released.



5 a. Explain frequency planning issues for inter systems in CDMA

Frequency reuse is the ability to reuse the same radio channel frequency at other cell sites within a cellular system. In the FDMA and TDMA systems frequency planning is an important consideration. The frequencies used in different cells must be planned carefully to ensure signals from different cells do not interfere with each other. In a CDMA system, the same frequency can be used in every cell, because channelization is done using the pseudo-random codes. Reusing the same frequency in every cell eliminates the need for frequency planning in a CDMA system; however, planning of the different pseudo-random sequences must be done to ensure that the received signal from one cell does not correlate with the signal from a nearby cell.

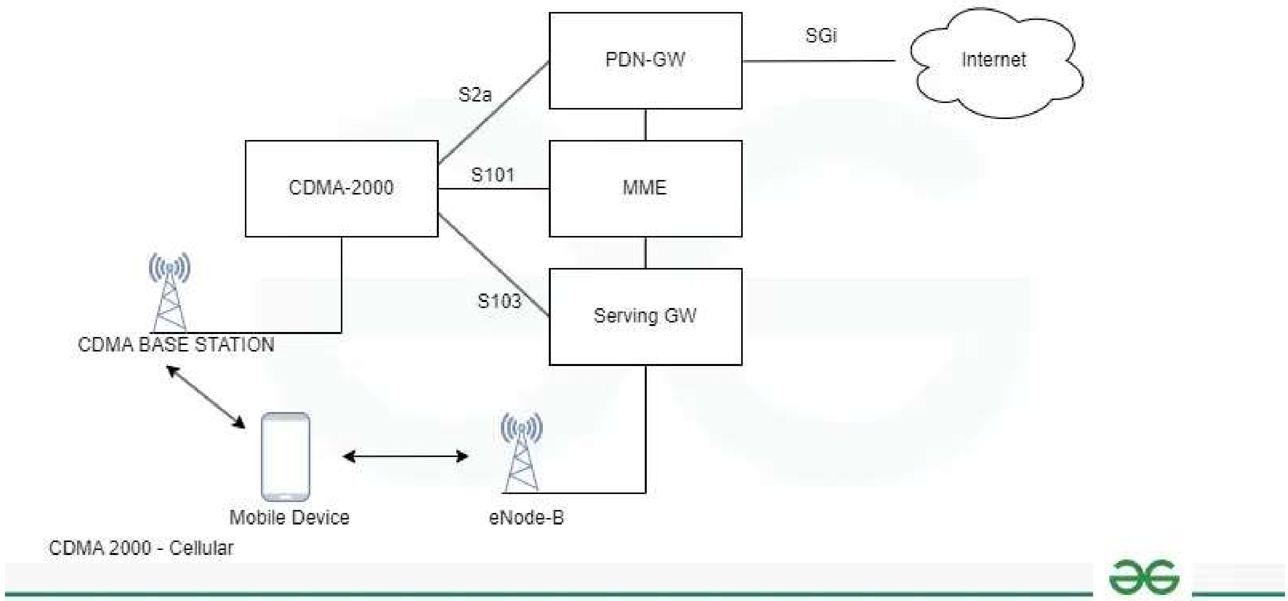
Since adjacent cells use the same frequencies, CDMA systems have the ability to perform soft hand offs. Soft hand offs allow the mobile telephone to communicate simultaneously with two or more cells. The best signal quality is selected until the hand off is complete. This is different from hard hand offs utilized in other cellular systems. In a hard hand off situation, as the mobile telephone approaches a hand off, signal strength may vary abruptly. In contrast, CDMA systems use the soft hand off, which is undetectable and provides a more reliable and higher quality signal.

5 b Explain the network nodes found in CDMA 2000 wireless systems.

CDMA2000 is a 3G network that transmits voice, data, and other information using a set of standards. Additionally, it can be used to access a 2G network because it is backward compatible.

An upgraded air interface in the CDMA2000 system allows CDMA to operate over larger bandwidths for increased capacity and data speeds. Additionally, it preserves backward compatibility with end-user IS-95 CDMA devices currently in use.

Data is sent by CDMA2000 systems using quadrature phase shift keying, or QPSK.



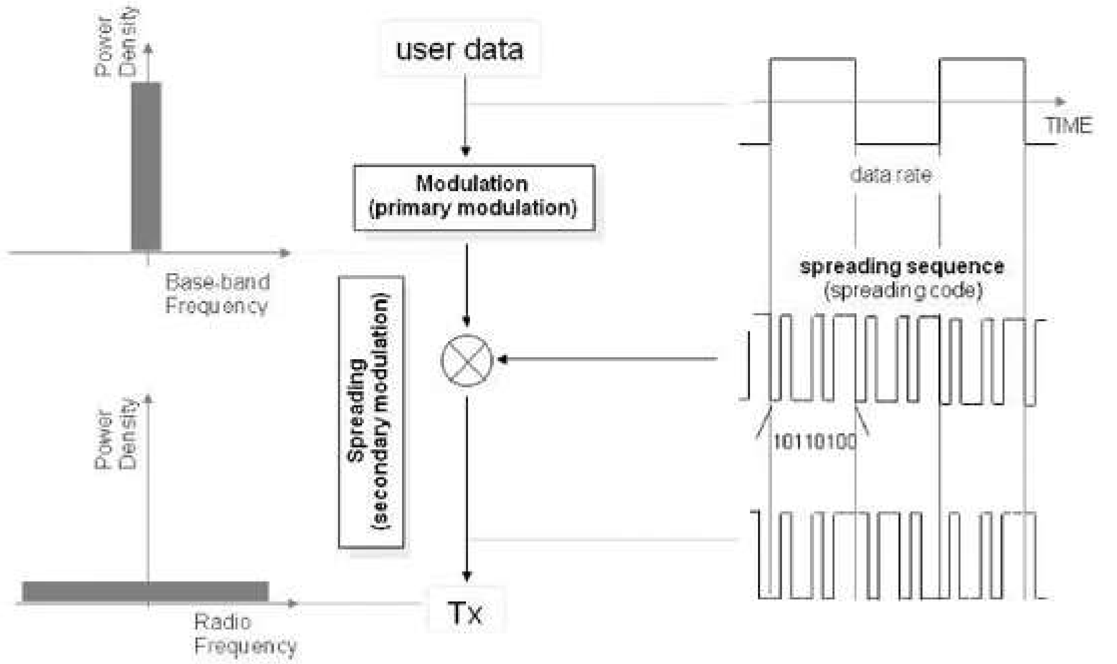
Radio Access Network (RAN)

Third-generation networks such as the CDMA2000 radio access network (RAN) enable complex mobile multimedia applications. A radio access network (RAN) is a key element of contemporary telecommunications that uses radio links to link devices to other areas of a network. It consists of software interfaces, antennas, radio and baseband units. Another name for a RAN is a radio network. It is a part of CDMA networks, such as CDMA2000 and IS-95.

CDMA2000 Core Network

CDMA2000 is a 3G network that transmits voice, data, and other information using a set of standards. Additionally, it can be used to access a 2G network because it is backward compatible.

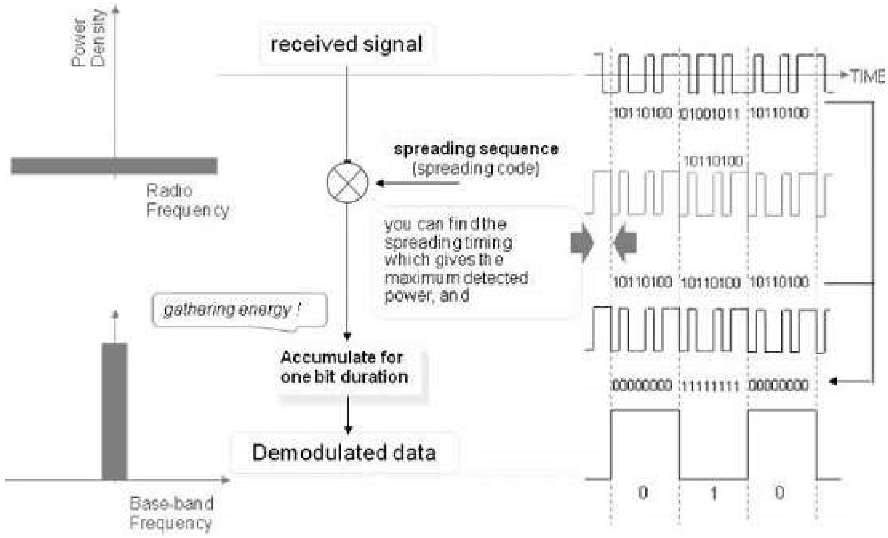
	An upgraded air interface in the CDMA2000 system allows CDMA to operate over larger bandwidths for increased capacity and data speeds. Additionally, it preserves backward compatibility with end-user IS-95 CDMA devices currently in use.
6. a	Explain the basic spectrum spreading operation in CDMA. Spread spectrum is a form of wireless communications in which the frequency of the transmitted signal is deliberately varied resulting higher bandwidth. •Spread-spectrum is apparent in the Shannon and Hartley channel-capacity theorem − $C = B \times \log2 (1 + S/N)$ •In the given equation, \`C' is the channel capacity in bits per second (bps), which is the maximum data rate for a theoretical bit-error rate (BER). 'B' is the required channel bandwidth in Hz, and S/N is the signal-to-noise power ratio. •Spread spectrum uses wideband, noise-like signals that are hard to detect, intercept, or demodulate. Additionally, spread-spectrum signals are harder to jam (interfere with) than narrow band signals. •Since spread-spectrum signals are so wide, they transmit at a much lower spectral power density, measured in watts per hertz, than narrow band transmitters. Spread-spectrum and narrowband signals can occupy the same band, with little or no interference. This capability is the main attraction for all the interest in spread spectrum today. Following are the two types of spread spectrum techniques − •Direct Sequence and •Frequency Hopping. **Direct Sequence (DS)** Direct Sequence Code Division Multiple Access (DS-CDMA) is a technique to multiplex users by different codes. In this technique, the same bandwidth is used by different users. Each user is assigned with one its own spreading code. These sets of codes are divided into two classes − •Orthogonal Codes and •Non-Orthogonal Codes Walsh sequences come into the first category which is Orthogonal Codes whereas other sequences i.e. PN, Gold, and Kasami are shift register sequences.



Orthogonal codes are assigned to the users, the output of the correlator in the receiver will be zero except the desired sequence. In synchronous direct sequence, the receiver receives the same code sequence

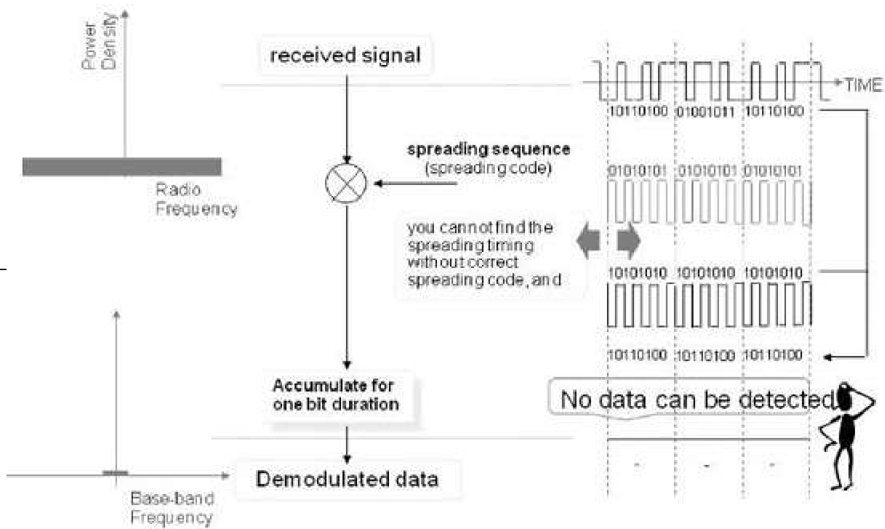
which was transmitted so that there is no time shift between the users.

Demodulating DS Signals - 1

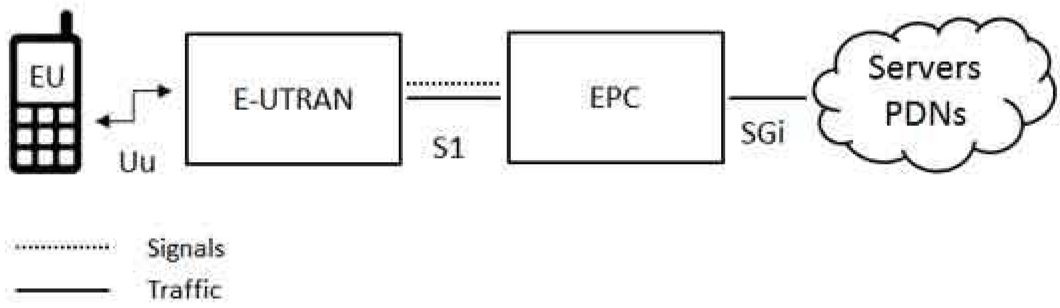


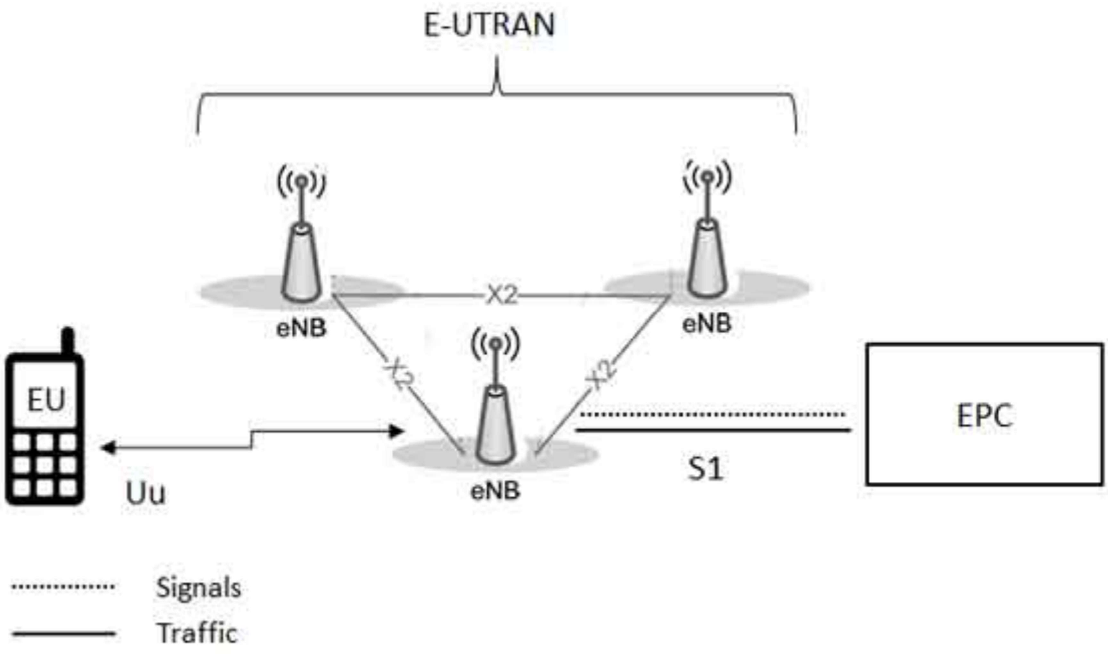
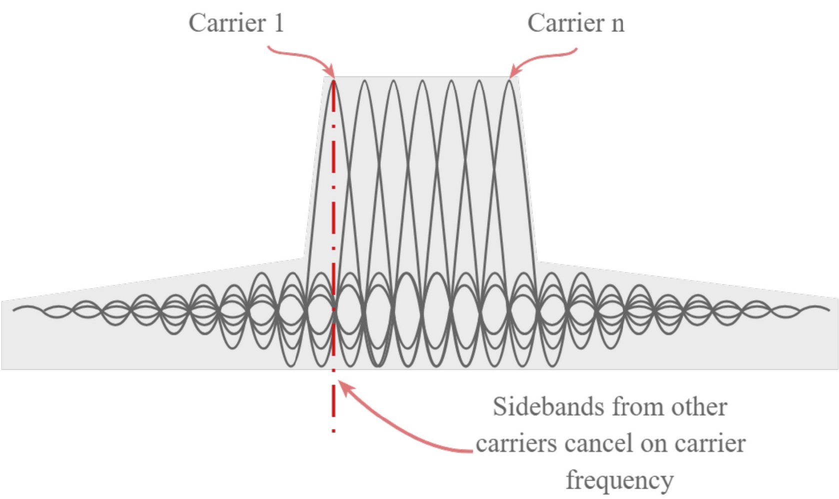
multiple codes were used at the time of transmission to the received signal. It is further demodulated by multiplying the code that was used at the time of this transmission, called the reverse diffusion (de-spreading).

Demodulating DS Signals – 2



	Features of Spread Spectrum The power density of Spread Spectrum signals could be lower than the noise density. This is a wonderful feature that can keep the signals protected and maintain privacy. By spreading the spectrum of the transmitted signal, one can reduce its power density such that it becomes less than the power density of the noise. In this way, it is possible to hide the signal in the noise. It can be demodulated if you know the code that was used to send the signal. In case the code is not known, then the received signal will remain hidden in the noise even after the demodulation.
6b	Explain the generation of the pilot channel signal. **Forward Channel** The Forward channel is the direction of the communication or mobile-to-cell downlink path. It includes the following channels − Pilot channel Sync Channel Paging Channel Forwarding traffic channel Pilot Channel − Pilot channel is a reference channel. It uses the mobile station to acquire the time and as a phase reference for coherent demodulation. It is continuously transmitted by each base station on each active CDMA frequency. And, each mobile station tracks this signal continuously.
7	List the advantages of OFDM leading to its selection for LTE and explain. Some of the main advantages of OFDM/OFDMA are as follows. ❒Simple Equalization: OFDM bypasses the need for a long equalizer as in single-carrier systems and requires a computational load of only 1 division operation per subcarrier as the equalization part. This turns out to be a huge savings in complexity for high rate wireless communication systems in which the symbol time is short resulting in long channel response. ❒Hardware Implementation: In an OFDM system, modulation at transmit side is performed through an inverse Fast Fourier Transform (iFFT) block and the demodulation at receive side is done through a Fast Fourier Transform (FFT) block, both of which are hardware optimized leading to simpler implementations. ❒Intercell Interference: Orthogonality among the subcarriers not only helps in preventing Inter-Symbol Interference (ISI) from multiple copies of the same signal but also helps in avoiding interference among transmissions from neighbouring cells. ❒Integration with MIMO: Since 1995, Multiple Input Multiple Output (MIMO) systems have become an integral part of infrastructure based wireless systems. When multiple antennas transmit or receive signals, there is interference among their transmissions as well known as inter-antenna interference. Like multiple copies of the same signal, OFDM helps in equalizing the effect of this interference among antennas through the same equalization strategy. ❒Flexibility: This is perhaps the biggest advantage offered by OFDM. Having a sliced spectrum allows

	the designer to control different parts of the spectrum in an adaptive manner. For example, a higher-order modulation can be sent for good subcarriers and a lower-order modulation for bad subcarriers. Moreover, users can be scheduled on different subcarriers in OFDMA in an adaptive manner according to the quality of service on offer. Most importantly, it can be easily integrated with adaptive modulation and coding strategies.
7b	With a neat block diagram, explain LTE network architecture and describe briefly the new elements provided in it. The high-level network architecture of LTE is comprised of following three main components: • The User Equipment (UE). • The Evolved UMTS Terrestrial Radio Access Network (E-UTRAN). • The Evolved Packet Core (EPC). The evolved packet core communicates with packet data networks in the outside world such as the internet, private corporate networks or the IP multimedia subsystem. The interfaces between the different parts of the system are denoted Uu, S1 and SGi as shown below:  **The User Equipment (UE)** The internal architecture of the user equipment for LTE is identical to the one used by UMTS and GSM which is actually a Mobile Equipment (ME). The mobile equipment comprised of the following important modules: • Mobile Termination (MT) : This handles all the communication functions. • Terminal Equipment (TE) : This terminates the data streams. • Universal Integrated Circuit Card (UICC) : This is also known as the SIM card for LTE equipments. It runs an application known as the Universal Subscriber Identity Module (USIM). A USIM stores user-specific data very similar to 3G SIM card. This keeps information about the user's phone number, home network identity and security keys etc. **The E-UTRAN (The access network)** The architecture of evolved UMTS Terrestrial Radio Access Network (E-UTRAN) has been illustrated below.

	
8a	With the help of neat diagram, explain how the timing and frequency synchronization is performed by the receiver to demodulate an OFDM signal.  OFDM offers many advantages in terms of resilience to fading, reflections and the like. OFDM also offers a high level of spectrum efficiency. However to reap the rewards, it is necessary that the OFDM system operates correctly, and to achieve this, it is necessary for the OFDM synchronization to be effective. There are a number of areas in which the OFDM synchronisation is critical to the operation of the system: •OFDM synchronization in terms of frequency offset: It is necessary that the frequencies are accurately tracked to ensure that orthogonality is maintained. •OFDM synchronisation in terms of clock accuracy: It is necessary that the sampling occurs at the correct time interval to ensure that the samples are synchronized and data errors are minimised. In order to ensure that the OFDM system works to its optimum, it is necessary to ensure that there are

schemes in place to ensure the OFDM synchronization is within the required limits.

It is particularly important that the demodulator in an OFDM receiver is able to synchronize accurately with the carriers within the OFDM signal. Offsets may arise for a number of reasons including any frequency errors between the transmitter and the receiver and also as a result of Doppler shifts if there is movement between the transmitter and receiver.

If the frequency synchronisation is impaired, then the orthogonality of the carriers is reduced within the demodulation process and error rates increase. Accordingly it is essential to maintain orthogonality to reduce errors and maintain the performance of the link.

First look at the way that sampling should occur. With the demodulator in synchronisation, all the contributions from the other carriers sum to zero as shown. On this way all the carriers are orthogonal and the error rate is at its minimum.

If a situation is encountered where the OFDM synchronisation for the frequency aspects are poor, then the demodulator will centre its samples away from the peak of the signal, and also at a point where the contributions from the other signals do not sum to zero. This will lead to a degradation of the signal which could in turn lead to an increase in the number of bit errors.

8 b

What is PAR problem? Explain the methods used for PAR reduction.

PAR problems

The frequency-domain data sequence S = {S(k)} are independent, identically distributed (i.i.d.) random variables and due to the central limit theorem, a small percentage of output samples will take very large magnitudes. This results in the wellknown PAPR problem of OFDM systems. In general, the PAPR (χ) of the time-domain sequence s = {s [n]} is defined as the ratio between the maximum instantaneous power and its average power, that is: χ = P AP R{s} = max |s| 2 E |s| where E{·} denotes expected value. In the literature, the most common way to evaluate the PAPR is to determine the probability that this PAPR exceeds a certain threshold $\chi 0$. This is represented by the Complementary Cumulative Distribution Function (CCDF), which is a random variable, as: CCDF(χ) = P rob(χ ` > $\chi 0$) = 1 − 1(1 − e $\chi 0$) N. CLIPPING TECHNIQUES The simplest PAPR reduction method consists basically in clipping the high parts of the signal amplitude that are outside the allowed region [3]. If the OFDM symbol s is clipped at a level A, then the the clipped signal ˜s is: ˜s = A, |s| ≤ A Aejφ(s) , |s| > A where φ(s) is the phase of s. This technique is the simplest of implementation but it has the following drawbacks: • Clipping causes in-band distortion, which degrades the performance of the BER • Clipping causes out-of-band radiation, resulting in adjacent interference. This can be reduced by filtering, and thus Clipping and Filtering (CF) operation is used in. When the signal passes through the low-pass filtering there is a peak power regrowth. It has been shown that the Nyquist-rate clipping suffers from a much higher peak power regrowth compared to the clipping with oversampling. Thus, the results suggest that for efficient reduction of the peak power, the OFDM should be sufficiently oversampled (i.e. $L \geq 4$) before clipping. • There is the possibility to use iterative CF, but it takes many iterations to reach a desired amplitude level A

There are many PAPR reduction approaches for OFDM systems, as clipping, coding, non-linear companding, ton reservation and ton injection, selective mapping (SLM) and partial transmit sequence (PTS)

9 a

Explain basic design principles followed in LTE specifications.

The LTE standard was designed as a completely new standard, with new numbering and new documentation, and it is not built on the previous versions of 3GPP standards. Earlier elements were brought in only if there was a compelling reason for them to exist in the new standard. The basic design principles that were agreed upon and followed in 3GPP while designing the LTE specifications include:2

- **Network Architecture:** Unlike 3G networks, LTE was designed to support packet-switched traffic with support for various QoS classes of services. Previous generations of networks such as UMTS/HSPA and 1xRTT/EvDO also support packet-switched traffic but this was achieved by subsequent add-ons to the initial version of the standards. For example, HSPA, which is a packet-switched protocol (packet-switched over the air), was built on top of the Release 99 UMTS network and as a result carried some of the unnecessary burdens of a circuit-switched network. LTE is different in the sense that it is a clean slate design and supports packet switching for high data rate services from the start. The LTE radio access network, E-UTRAN, was designed to have the minimum number of interfaces (i.e., the minimum number of network elements) while still being able to provide efficient packet-switched transport for traffic belonging to all the QoS classes such as conversational, streaming, real-time, non-real-time, and background classes.

- **Data Rate and Latency:** The design target for downlink and uplink peak data rates for LTE are 100 Mbps and 50 Mbps, respectively, when operating at the 20MHz frequency division duplex (FDD) channel size. The user-plane latency is defined in terms of the time it takes to transmit a small IP packet from the UE to the edge node of the radio access network or vice versa measured on the IP layer. The target for one-way latency in the user plane is 5 ms in an unloaded network, that is, if only a single UE is present in the cell. For the control-plane latency, the transition time from a camped state to an active state is less than 100 ms, while the transition time between a dormant state and an active state should be less than 50 ms.

- **Performance Requirements:** The target performance requirements for LTE are specified in terms of spectrum efficiency, mobility, and coverage, and they are in general expressed relative to the 3GPP Release 6 HSPA.

- \- **Spectrum Efficiency** The average downlink user data rate and spectrum efficiency target is three to four times that of the baseline HSDPA network. Similarly, in the uplink the average user data rate and spectrum efficiency target is two to three times that of the baseline HSUPA network. The cell edge throughput, measured as the 5th percentile throughput, should be two to three times that of the baseline HSDPA and HSUPA.
- \- **Mobility** The mobility requirement for LTE is to be able to support handoff/mobility at different terminal speeds. Maximum performance is expected for the lower terminal speeds of 0 to 15 km/hr, with minor degradation in performance at higher mobile speeds up to 120 km/hr. LTE is also expected to be able to sustain a connection for terminal speeds up to 350 km/hr but with significant degradation in the system performance.
- \- **Coverage** For the cell coverage, the above performance targets should be met up to 5 km. For cell ranges up to 30 km, a slight degradation of the user throughput is tolerated and a more significant degradation for spectrum efficiency is acceptable, but the mobility requirements should be met. Cell ranges up to 100 km should not be precluded by the specifications.
- \- **MBMS Service** LTE should also provide enhanced support for the Multimedia Broadcast and Multicast Service (MBMS) compared to UTRA operation.
- **Radio Resource Management:** The radio resource management requirements cover various aspects such as enhanced support for end-to-end QoS, efficient support for transmission of higher

	layers, and support for load sharing/balancing and policy management/enforcement across different radio access technologies. ●**Deployment Scenario and Co-existence with 3G:** At a high level, LTE shall support the following two deployment scenarios: ●- Standalone deployment scenario, where the operator deploys LTE either with no previous network deployed in the area or with no requirement for interworking with the existing UTRAN/GERAN (GSM EDGE radio access network) networks. ●- Integrating with existing UTRAN and/or GERAN deployment scenario, where the operator already has either a UTRAN and/or a GERAN network deployed with full or partial coverage in the same geographical area. ●**Flexibility of Spectrum and Deployment:** In order to become a truly global standard, LTE was designed to be operable under a wide variety of spectrum scenarios, including its ability to coexist and share spectrum with existing 3G technologies. Service providers in different geographical regions often have different spectrums in terms of the carrier frequency and total available bandwidth, which is why LTE was designed to have a scalable bandwidth from 1.4MHz to 20MHz. In order to accommodate flexible duplexing options, LTE was designed to operate in both frequency division duplex (FDD) and time division duplex (TDD) modes. ●**Interoperability with 3G and 2G Networks:** Multimode LTE terminals, which support UTRAN and/or GERAN operation, should be able to support measurement of, and handover from and to, both 3GPP UTRAN and 3GPP GERAN systems with acceptable terminal complexity and network performance.