GSM RF INTERVIEW QUESTIONS

1. What are the three services offered by GSM? Explain each of them briefly.

GSM services are categorized in three teleservices, bearer, and supplementary services.

A. Teleservices (communicate with other subscribers).
B. Bearer service (provides the underlaying network capacity necessary for transmission to occur between two points in the same or different networks). The bearer services describe what the network can offer (e.g. speech, data and fax).
C. Supplementary service is optional which subscriber can subscribe for free. Ex: call forwarding, call waiting,

2. Which uplink/downlink spectrum is allocated to GSM-900?

3. Which uplink/downlink spectrum is allocated to DCS-1800?

4. How many carrier frequencies are there in GSM-900/DCS-1800? How much is the separation between the carrier frequencies?

System	**P-GSM 900**	**E-GSM 900**	**GSM 1800**	**GSM 1900**
Frequencies: • Uplink • Downlink	890-915 MHz 935-960 MHz	880-915 MHz 925-960 MHz	1710-1785 MHz 1805-1880 MHz	1850-1910 MHz 1930-1990 MHz
Wavelength	~ 33 cm	~ 33 cm	~ 17 cm	~ 16 cm
Bandwidth	25 MHz	35 MHz	75 MHz	60 MHz
Duplex Distance	45 MHz	45 MHz	95 MHz	80 MHz
Carrier Separation	200 kHz	200 kHz	200 kHz	200 kHz
Radio Channels[1]	125	175	375	300
Transmission Rate	270 kbits/s	270 kbits/s	270 kbits/s	270 kbits/s

5. What is Ciphering? Why do we need it? Name the algorithm(s) used in it?

The purpose of ciphering is to encode the burst so that it cannot be interpreted by any device other than the intended receiver. The ciphering algorithm in GSM is called the A5 algorithm. It does not add bits to the burst, meaning that the input and output to the ciphering process is the same as the input.

6. What is Authentication? Why do we need it? Name the algorithm(s) used in it?

Authentication is the process to confirm that this user belong to the network. The Authentication algorithm in GSM is called the A8 algorithm.

7. What is equalization? Why do we need it?

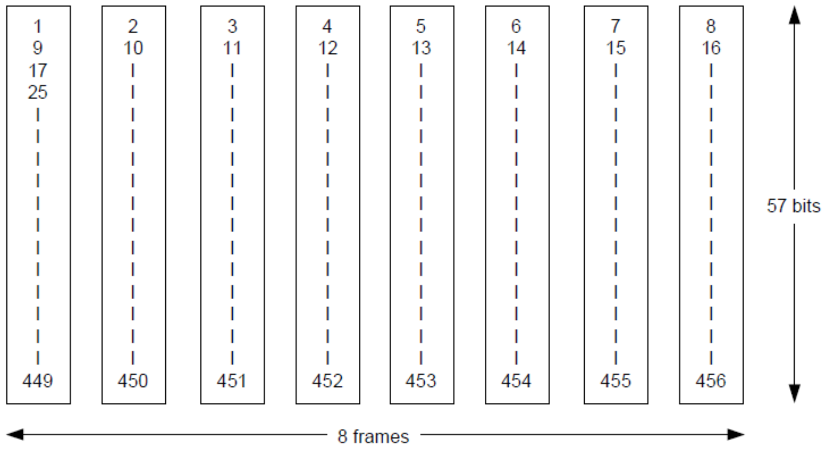
Adaptive equalization is a solution specifically designed to counteract the problem of time dispersion. It works as follows:

1. A set of predefined known bit patterns exist, known as training sequences. These are known to the BTS and the MS (programmed at manufacture). The BTS instructs the MS to include one of these in its transmissions to the BTS.
2. The MS includes the training sequence (shown in the figure as "S") in its transmissions to the BTS. However, due to the problems over the radio path, some bits may be distorted.
3. The BTS receives the transmission from the MS and examines the training sequence within it. The BTS compares the received training sequence with the training sequence which it had instructed the MS to use. If there are differences between the two, it can be assumed that the problems in the radio path affected these bits must have had a similar affect on the non-training sequence bits.
4. The BTS begins a process in which it uses its knowledge of what happened the training sequence to correct the other bits of the transmission.

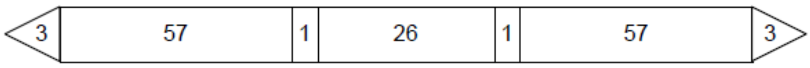
8. What is interleaving? Why do we need it?

The aim of interleaving is to distribute subblocks of data obtained by channel coding in such a way that one data block is distributed over several TDMA frames.

The channel coder provides 456 bits for every 20 ms of speech. These are interleaved, forming eight blocks of 57 bits each, as shown in the figure below.



In any one burst, there is space for two of these blocks. (The remaining bits are explained later in this book.) Thus, if one burst transmission is lost, there is a 25% BER for the entire 20 ms of speech (2/8 = 25%).



As shown in the diagram, the front and end 3 tail bits delimit the burst; the 26 bits are training sequence bits; and the bit at both sides of the training bits are used as “bit stealing flags”.

Second level of interleaving

If only one level of interleaving is used, a loss of this burst results in a total loss of 25%. This is too much for the channel decoder to correct. A second level of interleaving can be introduced to further reduce the possible BER to 12.5%. Instead of sending two blocks of 57 bits from the same 20 ms of speech within one burst, a block from one 20 ms and a block from another 20 ms are sent together. This causes a delay in the system, because the MS must wait for the next 20 ms of speech. However, the system can now afford to lose a whole burst because the loss only affects 12.5% of the bits from each speech frame. This rate can be corrected by a channel decoder.

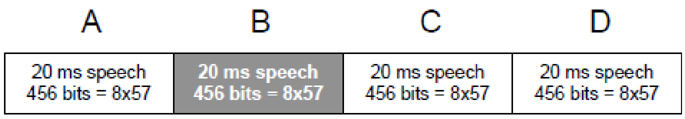
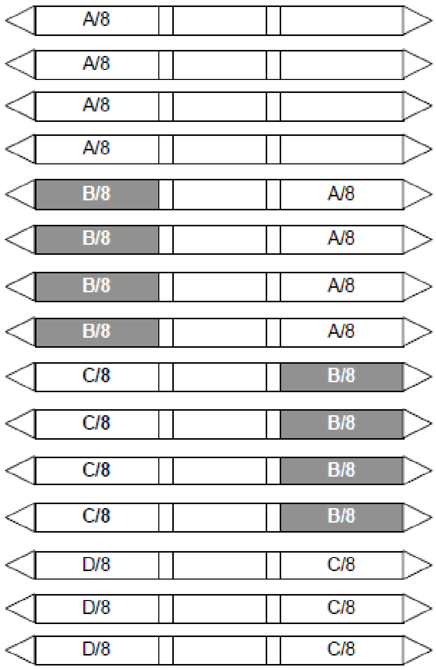


Figure 3-32 Speech frame



10. Explain Speech Coding?

Speech Coding: Instead of using 13 bits per sample as in A/D conversion, GSM speech coding uses 260 bits. This calculates as 50 x 260 = 13 kbits/s. This provides a speech quality which is acceptable for mobile telephony and comparable with wireline PSTN phones.

The voice compression coding technique is widely used in the modern digital communication systems. In this technique, a voice coder is used to set up a model to simulate the voice and noise produced by human vocal organs. The parameters to form the model will be transmitted through the TCH channels.

The voice coder is based on the residual excited linear prediction (REIP) coder. Moreover, the long term predictor (LTP) is used to enhance the compression effect. LTP can make the coding of residual data more advantageous by removing the vowels from the voice. With 20ms as the unit, the voice coder outputs 260bits after compressed coding. Therefore, the code rate is 13kbps. According to the different

classes of the importance of the information, the output bits can be classified into three categories: 50 very important bits, 132 important bits and 78 ordinary bits.
Comparing with the traditional PCM line on which the voice is coded directly and transmitted (64kbps), the 13kbps voice rate of the GSM system is much lower. The more advanced voice coder in the future can further reduce the rate to 6.5kbps (half-rate voice coding).

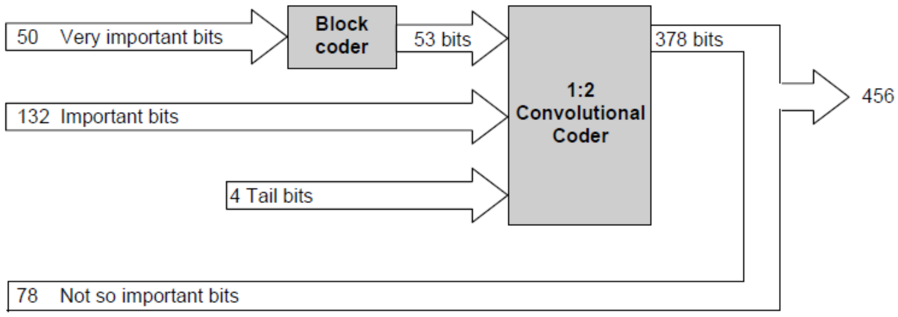
The coding mode is called Regular Pulse Excited-Long Term Prediction (RPE-LTP). It works as follow: 8KHZ of sampling is performed first, then divided into frames with 20ms; every frame has 4 sub-frames; the duration of every sub-frame is 5ms; and the pure bit rate is 13kbit/s.

11. What is channel coding?

The data to be transmitted over the interface must be specially protected against corruption due to the effects of fading and co-channel interference. Additional check bits are generated for this purpose and permit the detection of transmission errors and to a certain extent the reconstruction of the original data.

The channel coding process increases the bit rate from 13 Kbit/s to 22.8 Kbit/s through adding protection to the Class-I bits. Convolution coding and addition of a CRC (Cyclic Redundancy Check) result in 456 bits coming out during the 20 ms speech data block period (= 22.8 Kbit/s).

Channel coding in GSM uses the 260 bits from speech coding as an input and outputs 456 encoded bits.
The 260 bits are split according to their relative importance:
· Block 1: 50 very important bits
· Block 2: 132 important bits and
· Block 3: 78 not so important bits



12. What do you mean by Frequency re-use?

An operator purchases some frequency band. This band is divided into channels (200 KHz). To cover the whole country or city, the operator tend to reuse the channels after some distance "D" which at this distance the interference can be under control.

13. What is Cell Splitting?

Unfortunately, the design requires cells of different sizes; one radius for urban, another for suburban, etc. There is a method for merging different sized grids, called cell splitting. Cell splitting provides a mathematical transition from one morphological region to another.

Splitting is accomplished by centering the next smaller grid on the corner or the side of the next larger hexagon. Figure 6 demonstrates how the side-split and corner-split are accomplished. The cell splits also help the engineer maintain the C/I ratio required by different technologies.

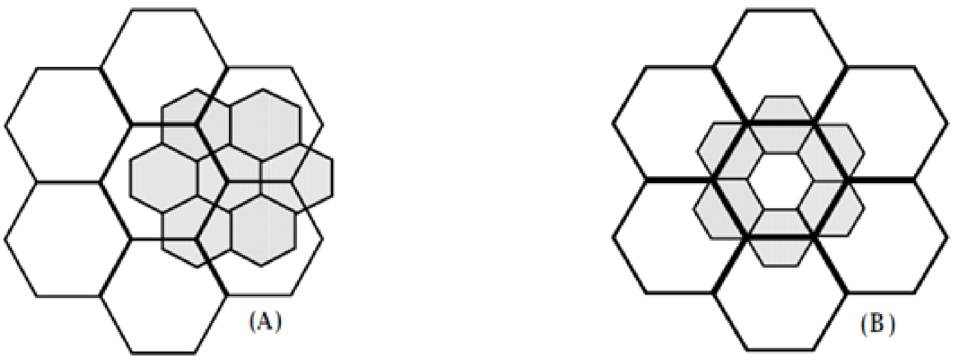


Figure : An Example of a Corner Split (A) and a Side Split (B)

14. Name the interfaces between a) BTS and MS b) BTS and BSC c) BSS and MSC?

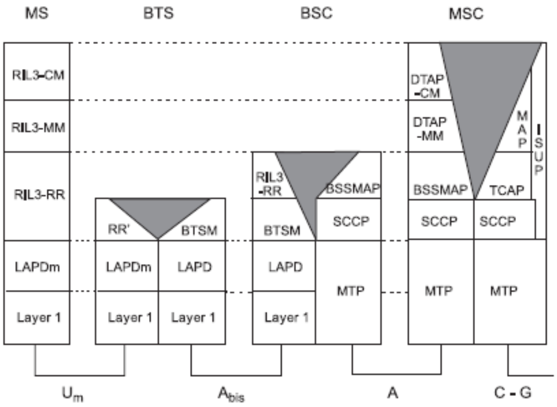
A) U_m interface.

b) A_{bis} interface.

c) A interface

15. What are LAPD and LAPDm?

Layer 2: LAPDm (modified version of ISDN LAPD protocol)



16. What is WPS?

http://wps.ncs.gov/index.html

17. What is MA?

Mobile Allocation (all frequency available for frequency hopping in the cell).

18. What is MAIO?

Mobile Allocation Index offset. (The offset from the initial point in an array of frequency)

19. What is the difference between Synthesized Frequency Hopping and Base Band Frequency Hopping?

A. ***In baseband hopping***, the transmitter will change its frequency on frame basis.

B. ***In synthesizer hopping***, the transmitter will change its frequency on time slot basis. That is why they also said it is fast hopping.

20. What is Cycling Frequency Hopping?

The hopping sequence occurs in a uniform manner. (Not random).

21. What is HSN? How do we apply it?

Hopping sequence number, if its value (0) cycle hopping. Otherwise it is random hopping.

22. What is DTX? Why is it used?

Discontinuous Transmission (DTX) is a mechanism which allows the radio transmitter to be switched off most of the time during speech pauses.

DTX may be applied independently to each direction, so that the control of DTX must take into account two components:

· The uplink mode

· The downlink mode.

DTX can be enabled or disabled for the uplink and/or downlink mode on a per-cell basis.

Reasons for DTX When DTX is applied, actual transmission on the radio path is reduced. This will cause a decrease of the interference level in co-channel cells (using the same frequency). Another advantage will appear when using DTX in the uplink mode: it saves battery power for the mobile station. However, a disadvantage of the DTX mode is that it slightly deteriorates the quality of transmission. Note that transmitting in DTX mode does not save timeslots on the air-interface.

23. What is DRX? Why do we need it?

Discontinuous reception is method used to conserve power at the MS. The paging channel, used by the BTS to signal an incoming call, is structured into subchannels. Each MS is assigned one of these sub-channels and needs to listen only to its own sub-channel. In the time between successive paging sub-channels, the mobile can go into “sleep mode”, when almost no power is used.

24. What is the gross data rate of GSM?

270kbps.

25. What is Erlangs? What is meant by GoS?

Traffic refers to the usage of channels and is usually thought of as the holding time per time unit (or the number of “call hours” per hour) for one or several circuits (trunks or channels). Traffic is measured in Erlangs (E), for example, if one subscriber is continuously on the telephone, this would generate one call hour per hour or 1 E of traffic. The traffic one cell can carry depends on the number of traffic channels available and the amount of congestion that is acceptable (to both the customer and the provider), the so-called Grade of Service (GoS).

26. We use two different bands for GSM/DCS communications; GSM900 and DCS-1800. Which one is the better of the two in terms of coverage?

G900 is better. Due to path loss formula as frequency is increased, the losses which the signal will encounter will be more.

27. What is TA? Why do we need TA?

Time advance (alignment) process The RF communication experiences a propagation delay over the distance between the BTS and the MS. In order to synchronize the MS to the BTS, a timing advance is used to align the time slots arriving at the BTS receiver:

1. The BTS measures the reception time of the incoming MS burst
2. The BTS requests the MS to advance its transmission to compensate for the delay over the distance. A 6-bit number indicates how many bits the MS must advance its transmission.
3. The time advance value TA can have a value between 0 and 63 bit lengths, which corresponds to a delay of between 0 and 233 ms.
4. This leads to a maximum mobile range of 35 km, which is rather determined by the TA than by the signal strength.

28. What is meant by Location Area?

29. What is location update? Why do we need location update?

A Location Area (LA) is defined as a group of cells. Within the network, a subscriber's location is known by the LA which they are in. The identity of the LA in which an MS is currently located is stored in the VLR. When an MS crosses a boundary from a cell belonging to one LA into a cell belonging to another LA, it must report its new location to the network1. When an MS crosses a cell boundary within an LA, it does need to report its new location to the network. When there is call for an MS, a paging message is broadcast within all cells belonging to an LA.

30. What is meant by IMSI, TMSI, IMEI and MS-ISDN? Why they are needed?

IMSI = International Mobile Subscriber Identity

TMSI = Temporary Mobile Subscriber Identity

IMEI = International Mobile Equipment Identity

MS-ISDN =Mobile Station ISDN Number

A. The **MSISDN** is the directory number allocated to the mobile subscriber. It is dialed to make a telephone call to the mobile subscriber.
B. A MS is identified by its **IMSI**. The IMSI is embodied in the SIM of the mobile equipment. It is provided by the MS anytime it accesses the network.
C. The **TMSI** is an identity alias which is used instead of the IMSI when possible. The use of a TMSI ensures that the true identity of the mobile subscriber remains confidential by eliminating the need to transfer an IMSI code unciphered over a radio link.

   A VLR allocates a unique TMSI code to each mobile subscriber that is operating in its area. This code, which is only valid within the area supervised by the VLR, is used to identify the subscriber in messages to and from the MS. When a change of location area also involves a change of VLR area, a new TMSI code is allocated and communicated to the MS. The MS stores the TMSI on its SIM.
D. **IMEI** codes that identify the mobile equipment deployed in the GSM system.

31. What is ARFCN?

Absolute Radio-Frequency Channel Number (**ARFCN**) is a code that specifies a pair of physical radio carriers and channels used for transmission and reception on the Um interface, one for the uplink signal and one for the downlink signal.

ARFCN Calculator:

http://www.aubraux.com/design/arfcn-calculator.php

32. Explain Power Control?

Power Control enables the mobile station and/or the BTS to increase or decrease the transmission power on a per-radio link basis.

Power Control is separately performed for the uplink and downlink. In both cases the BSC is responsible for initiating Power Control; the mobile station and the BTS adopt transmit power according to the BSC Power Control commands.

Measurements While a mobile station is active on a call, it has the responsibility of providing measurement data about the performance of the air-interface to its serving BTS so that the serving BSC can decide if a power control should be performed. Also the serving BTS measures the performance of the air-interface. Whereas the mobile station measures the performance of the downlink, the BTS measures the performance of the uplink.

Downlink measurements The mobile station measures and reports the following measurements to the BSC regarding the performance of the downlink:

· Strength of the signal being received from its serving BTS (in dBm)

· Quality of the signal being received from its serving BTS (in bit error rate).

Uplink measurements The BTS measures and reports the following measurements to the BSC regarding the performance of the uplink:

· Strength of the signal being received from the mobile station

· Quality of the signal being received from the mobile station.

Periodically measuring The mobile station measures periodically the performance of the downlink, and sends the measurements in the SACCH (Slow Associated Control Channel) via the serving BTS to the BSC every SACCH multi-frame. This corresponds to the transmission of data every 104 TDMA frames or 480 ms. The base station measures the quality of the uplink. Also, it transfers the measurements in the SACCH to the BSC every 480 ms.

Signal strength When the BSC notices that the signal strength of a particular radio link measured on the uplink becomes below the lower pre-defined threshold because the mobile station moves away from the BTS, it sends a Power Control command to the mobile station to increase its transmit power (MS_TXPWR) by a pre-defined step (typically 2 dB).

33. What is the difference between FDD and TDD?

FDD = frequency division duplexing (transmitter and receiver operates on different frequencies)

TDD = Time division Duplexing (transmitter and receiver operates on same frequency).

34. What is an extended cell? How does it impact the system? Channels and TDMA structure?

The current limitation on the range of a GSM cell site to 35km is mandated by the duration of the standard timeslots defined in the GSM specification. The maximum distance is given by the maximum time that the signal from the mobile/BTS needs to reach the receiver of the mobile/BTS on time to be successfully heard. At the air interface the delay between the transmission of the downlink (BTS) and the uplink (mobile) has an offset of 3 timeslots. Until now the mobile station has used a timing advance to compensate for the propagation delay as the distance to the BTS changes. This timing advance is defined in the GSM specification as 64 bits, which gives the theoretical maximum BTS/mobile separation as 35km.

With Extended Range Cell Feature, the BTS is able to receive the uplink signal in two adjacent timeslots instead of one. When the mobile station reaches its maximum timing advance, i.e. maximum range, the BTS expands its hearing window with an internal timing advance that gives the necessary time for the mobile to be heard by the BTS even from the extended distance. This extra advance is the duration of a single timeslot, a 156 bit period.

35. Why do we use Multiple Access Schemes? What is the difference between FDMA, TDMA and CDMA?

Multiple Access schemes allow for many users to access the network.

FDMA= Users access the network through frequency division (separation between users is made through frequency). (1G cellular systems)

TDMA= Users access the network through time division (separation between users is made through time). (2G cellular systems)

CDMA= Users access the network through code division (separation between users is made through code). (3G cellular systems)

36. Which channel(s) is used for SMS?

Either SDCCH or SACCH.

37. Which channel is used by MS to request access to the network?

RACH = random access channel.

38. What is AGCH?

AGCH (Access Grant Channel) assigns a signaling channel (SDCCH) to the MS.

3**9. Why do we need SDCCH?**

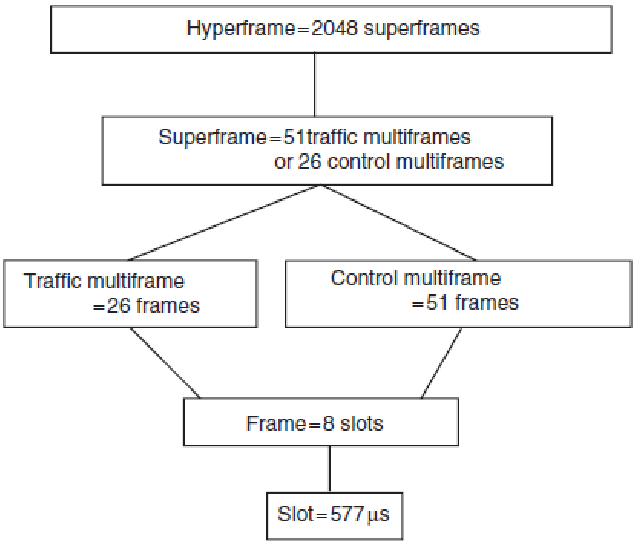
The BTS switches to the assigned SDCCH. The call set-up procedure is performed in idle mode. The BSC assigns a TCH. (SDCCH is also used to transmit text messages).

40. What is a physical channel? How do we differentiate between physical and logical channels?

Each timeslot on a TDMA frame is called a physical channel. Therefore, there are 8 physical channels per carrier frequency in GSM.

Physical channels can be used to transmit speech, data or signaling information. A physical channel may carry different messages, depending on the information which is to be sent. These messages are called logical channels. For example, on one of the physical channels used for traffic, the traffic itself is transmitted using a Traffic Channel (TCH) message, while a handover instruction is transmitted using a Fast Associated Control Channel (FACCH) message.

41. What are TDMA frames, multiframes, superframes and hyperframes?



42. Why do we need FCCH, SCH and BCCH?

FCCH: Identifies BCCH carrier by the carrier frequency and synchronizes with the frequency.

Synchronization Channel (SCH): Transmits information about the TDMA frame structure in a cell (e.g. frame number) and the BTS identity (Base Station Identity Code (BSIC)).

BCCH: Broadcasts some general cell information such as Location Area Identity (LAI), maximum output power allowed in the cell and the identity of BCCH carriers for neighboring cells.

43. Why do we need SACCH?

Instructs the MS the transmitting power to use and gives instructions on timing advance.

44. What is the purpose of PCH and CBCH?

Paging Channel (PCH): Transmits a paging message to indicate an incoming call or short message. The paging message contains the identity number of the mobile subscriber that the network wishes to contact.

Cell Broadcast Channel (CBCH): BS uses this logical channel to transmit short message service cell broadcast.

45. Do we keep BCCH on a hopping radio? Give the reason to support your answer.

No, BCCH is a signaling channel which must be continuously transmitted in a cell.

46. How much delay is present between downlink and uplink frames? Why do we need this delay?

47. Explain the structure of a Traffic Multiframe. Why do we need SACCH and Idle bursts in a traffic multiframe?

Traffic Multiframe Structures - The 26 traffic multiframe structure is used to send information on the traffic channel. The 26 traffic multiframe structure is used to combine user data (traffic), slow control signaling (SACCH), and idle time period. The 12th frame (no. 13) in the 26-frame traffic channel multiframe is used by the ***Slow Associated Control Channel (SACCH)*** which carries link control information to and from the MS–BTS. Each timeslot in a cell allocated to traffic channel usage will follow this format, that is, 12 bursts of traffic, 1 burst of SACCH, 12 bursts of traffic and 1 idle.

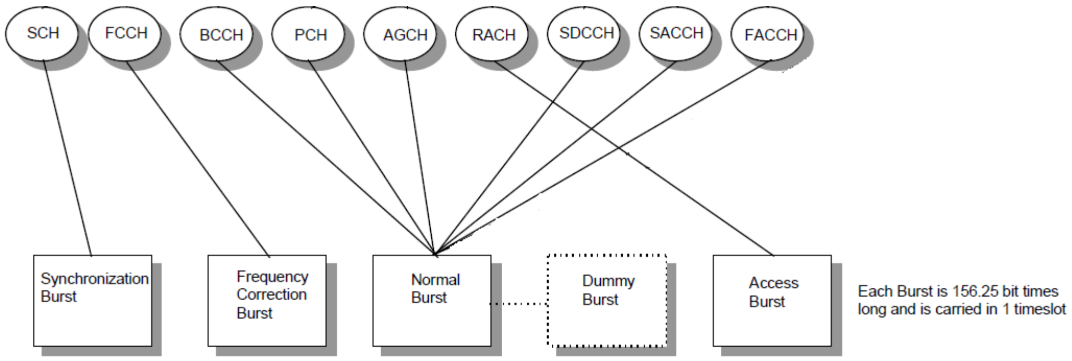
Idle Bursts: The idle time period allows a mobile device to perform other necessary operations such as monitoring the radio signal strength level of a beacon channel from other cells. The time interval of a 26 frame traffic multiframe is 6 blocks of speech coder data (120 msec).

48. How is a FACCH formed? When is a FACCH used?

Fast Associated Control Channel (FACCH): Transmits handover information.

49. What are bursts? Explain various types of bursts, Radio Propagation and Antennas?

Bursts



Radio Propagation

It is how the radio propagates from the receiver till reaching the receiver, maybe the signal encounter obstacles, Knife Edge…etc.

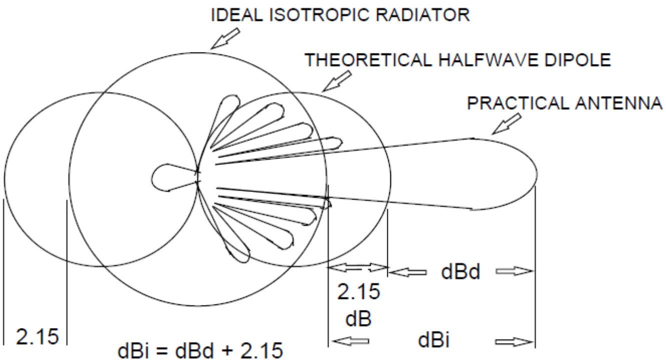
Wave propagation is difficult, no one can argue against that. Nevertheless, it is necessary, for example when planning mobile telephony systems, to define methods for predicting propagation in an area. Different models for estimating signal strength are used for this purpose. They might be simple expressions, suitable for rough assessments or complicated algorithms that demand considerable computation time. The simplest model for wave propagation is the free space case.

Antennas

Isotropic antenna: An isotropic antenna is a completely non-directional antenna that radiates equally in all directions. Since all practical antennas exhibit some degree of directivity, the isotropic antenna exists only as a mathematical concept. The isotropic antenna can be used as a reference to specify the gain of a practical antenna.
The gain of an antenna referenced isotropically is the ratio between the power required in the practical antenna and the power required in an isotropic antenna to achieve the same field strength in the desired direction of the measured practical antenna. The directive gain in relation to an isotropic antenna is called dBi.

Half-wave dipole antenna: A half-wave dipole antenna may also be used as a gain reference for practical antennas. The half-wave dipole is a straight conductor cut to one-half of the electrical wavelength with the radio frequency signal fed to the middle of the conductor. Directive gain in relation to a dipole is expressed in units of “dBd”.

For a dipole and an isotropic antenna with the same input power, the energy is more concentrated in certain directions by the dipole. The difference in directive gain between the dipole and the isotropic antenna is 2.15 dB. Figure illustrates the differences in gain between the isotropic, dipole, and practical antenna. The vertical pattern (Figure) of the practical antenna is that of a directional antenna.



50. What is VSWR? Why do we need it?

Standing wave ratio (**SWR**) is the ratio of the amplitude of a partial standing wave at an antinode (maximum) to the amplitude at an adjacent node (minimum), in an electrical transmission line. The SWR is usually defined as a voltage ratio called the **VSWR**, for *voltage standing wave ratio*. SWR is used as an efficiency measure for transmission lines, electrical cables that conduct radio frequency signals, used for purposes such as connecting radio transmitters and receivers with their antennas, and distributing cable television signals.

A problem with transmission lines is that impedance mismatches in the cable tend to reflect the radio waves back toward the source end of the cable, preventing all the power from reaching the destination end. SWR measures the relative size of these reflections. An ideal transmission line would have an SWR of 1:1, with all the power reaching the destination and no reflected power. An infinite SWR represents complete reflection, with all the power reflected back down the cable. The SWR of a transmission line is measured with an instrument called an SWR meter, and checking the SWR is a standard part of installing and maintaining transmission lines.

The voltage standing wave ratio is then equal to:

$$VSWR = \frac{V_{\max}}{V_{\min}} = \frac{1+\rho}{1-\rho}.$$

Where (ρ) is the magnitude of reflection coefficient.

51. What do you mean by EIRP?

Effective isotropic radiated power (EIRP) is the amount of power that a theoretical isotropic antenna (that evenly distributes power in all directions) would emit to produce the peak power density observed in the direction of maximum antenna gain. EIRP can take into account the losses in transmission line and connectors and includes the gain of the antenna. The EIRP is often stated in terms of decibels over a reference power emitted by an isotropic radiator with equivalent signal strength. The EIRP allows comparisons between different emitters regardless of type, size or form.

52. What is Polarization? What are the types of polarization?

The polarization indicates the plane in which the wave is vibrating. The polarization plane is taken to be that of the electric component.

Vertical and horizontal are the simplest forms of polarization, and they both fall into a category known as linear polarization. However, it is also possible to use circular polarization. This has a number of benefits in areas such as satellite applications, where it helps to overcome the effects of propagation anomalies, ground reflections and the spin that occur on many satellites. Circular polarization is a little more difficult to visualize than linear polarization; however, it can be imagined by visualizing a signal propagating from an antenna that is rotating. Another form of polarization is known as elliptical polarization.

53. What is fading?

Fading is the variation of the received signal with time, it occurs due to propagation distance.

Short term (fast) fading: caused by multipath propagation.
Long term (slow) fading: caused by shadowing.

54. What is Rayleigh Fading?

Rayleigh fading is a reasonable model when there are many objects in the environment that scatter the radio signal before it arrives at the receiver.

Rayleigh fading models assume that the magnitude of a signal that has passed through such a communications channel will vary randomly, or fade, according to a Rayleigh distribution the radial component of the sum of two uncorrelated Gaussian random variables.

Rayleigh fading is most applicable when there is no dominant propagation along a line of sight between the transmitter and receiver. If there is a dominant line of sight, Rician fading may be more applicable.

55. What is multipath fading?

Multipath fading is receiving multiple copies of the signal at receiver due to reflections. The copies reach the receiver with different phases, so summation either constructive or destructive. This affects the quality of received signal in terms of BER.

56. How can we minimize multipath fading?

By using channel equalizer.

57. What are the different types of diversity?

A. Space Diversity.
B. Polarization Diversity.
C. Time Diversity.
D. Frequency Diversity.

58. Explain various types of Antenna Diversity?

Antenna diversity increases the received signal strength by taking advantage of the natural properties of radio waves. There are two primary diverstiy methods: space diversity and polarization diversity.

Space Diversity: Increased received signal strength at the BTS may be achieved by mounting two receiver antennae instead of one. If the two Rx antennae are physically separated, the probability that both of them are affected by a deep fading dip at the same time is low. At 900 MHz, it is possible to gain about 3 dB with a distance of five to six meters between the antennae. At 1800 MHz the distance can be shortened because of its decreased wavelength.

By choosing the best of each signal, the impact of fading can be reduced. Space diversity offers slightly better antenna gain than polarization diversity, but requires more space.

Polarization Diversity: With polarization diversity the two space diversity antennae are replaced by one dual polarized antenna. This antenna has normal size but contains two differently polarized antenna arrays. The most common types are vertical/horizontal arrays and arrays in 45 degree slant orientation. The two arrays are connected to the respective Rx branches in the BTS. The two arrays can also be used as combined Tx/Rx antennas. For most applications, the difference between the diversity gain for space diversity and polarization diversity is negligible, but polarization diversity reduces the space required for antenna.

59. Explain Frequency Diversity.

By using frequency hopping sequence.

60. Explain Time Diversity.

By using code interleave technology.