2
UNIT

Wireless Channel Modelling

CONTENTS

PART-1

Wireless Channel Modelling, AWGN Channel, Rayleigh Channel, Rician Fading Channel.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 2.1. Write a short note on AWGN channel.

Answer

1. The simplest mathematical model of the radio channel is the additive white Gaussian noise (AWGN) channel. It is a very good model for physical reality as long as the thermal noise at the receiver is the only source of disturbance.
2. The AWGN channel model can be characterized as follows :

i. The noise is additive :

1. The received signal equals the transmit signal plus some noise, where the noise is statistically independent of the signal.
2. The noise $w(t)$ is an additive random disturbance of the useful signal $s(t)$; that is, the received signal is given by

$$r(t) = s(t) + w(t)$$

ii. The noise is white :

1. It has constant PSD. Therefore, the autocorrelation of the noise in time domain is zero for any non-zero time offset.
2. The one-sided PSD is usually denoted by N_0. Thus, $N_0/2$ is the two-sided PSD and WN_0 is the noise inside the noise bandwidth W.
3. The unit of N_0 is W/Hz,. Usually, N_0 is written as dBm/Hz.

iii. The noise samples have a Gaussian distribution.

1. The Gaussian PDF with variance σ^2 is given by

$$p(x) = \frac{1}{\sqrt{2\pi\sigma^2}} e^{-(x-m)^2/2\sigma^2}$$

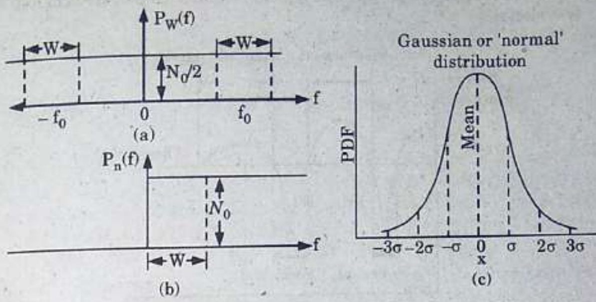

Fig. 2.1.1. Noise representation (a) Two-sided PSD (b) Equivalent one-sided PSD (c) Typical gaussian distribution with zero mean and σ standard deviation.

2. The output of every (linear) noise measurement is a zero-mean Gaussian random variable that does not depend on the time instant when the measurement is done.
3. The AWGN model is a mathematical function, because it implies that the total power (*i.e.*, the PSD integrated over all frequencies) is infinite. Thus, a time sample of white noise has infinite average power, which is certainly not a physically reasonable property.
4. The mean of the multiplication of two noise samples (autocorrelation) is

$$E\{\omega(t_1)\,\omega(t_2)\} = \frac{N_0}{2}\,\delta(t_1 - t_2)$$

where $\delta(t)$ represents the unit impulse at time t.

Que 2.2. Discuss in detail Rayleigh fading channel.

Answer

1. When the waves of multipath signals are out of phase, there can be a reduction of the signal strength at the receiver. This may result in deep fading.
2. The basic model of Rayleigh fading assumes a received multipath signal to consist of a large number of reflected waves with independent and identically distributed (i.i.d) in phase and quadrature amplitudes.
3. Rayleigh distribution is a good model for channel propagation there is no strong LOS path from the transmitter to the receiver and where the base station is hidden behind a building several blocks away and the arriving signal is bouncing off many scattering objects in the local area.
4. In the time domain, Rayleigh fading looks like periodic peaks of 10 dB or less interspersed between deep troughs of 40 dB or more. These deep

fades (nulls in signal power) will typically occur at separations of half a wavelength.

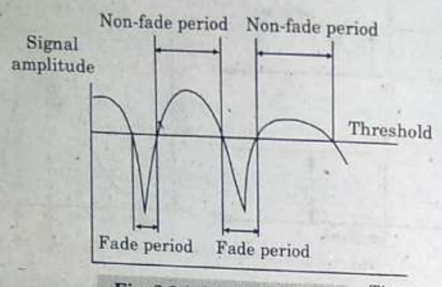

Fig. 2.2.1. Rayleigh fading.

5. We model the case in which the transmitted signal reaches a stationary receiver via multiple paths, but the path differences are due to only local reflections.
6. The Rayleigh distribution is plotted in Fig. 2.2.2. The median of the distribution is $R = R_{rms}$ which implies that there is constructive interference ($R > R_{rms}$) for 50 % of locations and destructive interference ($R < R_{rms}$) for 50 % of locations.
7. The mean value of Rayleigh distribution is given by

$$E[\alpha] = \alpha\sqrt{\frac{\pi}{2}} = 1.2533\,\sigma$$

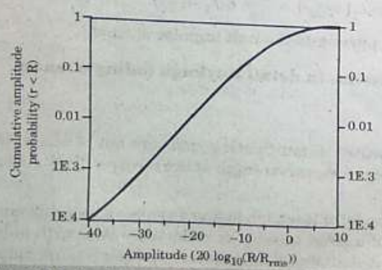

Fig. 2.2.2. Rayleigh amplitude distribution (R/R_{rms}).

8. The variance of Rayleigh distribution is given by

$$\sigma_\alpha^2 = E[\alpha^2] - E^2[\alpha] = 0.4292\,\sigma^2$$

9. The RMS value of the envelope is the square root of the mean square, *i.e.*, $\sqrt{2}\sigma$, where σ is the standard deviation and the median value of α is $1.177\,\sigma$.
10. If the set of reflected waves are dominated by one strong component, Rician fading is a more appropriate model.

Que 2.3. **Describe Rician fading channel.**

Answer

1. The Rayleigh fading model assumes that all paths are relatively equal, *i.e.*, there is no dominant path. Despite the fact that Rayleigh fading is the most popular model, occasionally there is a direct line of sight path in mobile radio channels and in indoor wireless as well.
2. The presence of a direct path is usually required to close the link in satellite communications. In this case, the reflected paths tend to be weaker than the direct path, and we model the complex envelope as

$$\vec{E} = E_0 + \int_{n=1}^{N} E_n e^{j\theta_n}$$

where, the constant term represents the direct path and the summation represents the collection of reflected paths. This model is referred to as a Rician fading model.

3. The analysis proceeds in a manner similar to that of the Rayleigh fading case, but with the addition of a constant term.
4. A key factor in the analysis is the ratio of the power in the direct path to the power in the reflected paths. This ratio is referred to as the Rician K-factor, defined as

$$K = \frac{s^2}{\sum_{n=1}^{N}|E_n|^2}$$

where $s^2 = |E_0|^2$. The Rician K-factor is often expressed in dB.

5. The calculation of the amplitude density function in the Rician fading case is more involved than with Rayleigh fading, so we merely give the result here. We have

$$f_R(r) = \frac{r}{\sigma^2} e^{-(r^2+s^2)/2\sigma^2} I_0\left(\frac{rs}{\sigma^2}\right) r \ge 0$$

6. The amplitude distributions of the Rician fading channel for different K factors are shown in Fig. 2.3.1. Deep fades are clearly less probable than with the Rayleigh channel, and the probability of their occurrence decreases as the K factor increases.

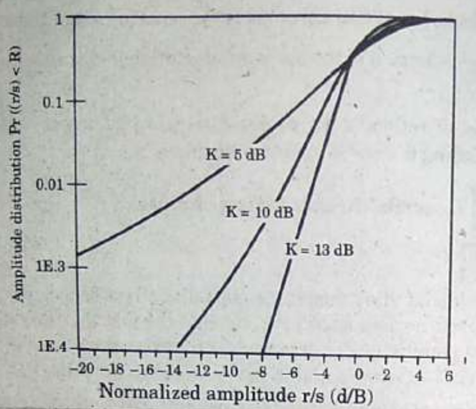

Fig. 2.3.1. Amplitude distribution for the rician channel.

PART-2

Nakagami Fading Channel, Okumura and Hata Path Loss Model.

Questions-Answers
Long Answer Type and Medium Answer Type Questions

Que 2.4. **Write a short note on Nakagami fading channel.**

1. Besides Rayleigh and Rician fading, a refined model suggested for the PDF of signal amplitude exposed to mobile fading is the Nakagami fading model. The distribution of the amplitude and signal power can be used to find the probabilities on signal outages.
2. In wireless communication the main role of the Nakagami model can be summarized as follows :
   i. It describes the amplitude of the received signal after maximum ratio diversity combining.
   ii. The sum of multiple *i.i.d.* Rayleigh fading signals has a Nakagami-distributed signal amplitude.
   iii. Nakagami distribution matches some empirical data better than the other models.
   iv. Nakagami fading occurs for multipath scattering with relatively large delay time spreads and with different clusters of reflected waves.

   v. The Rician and Nakagami models behave approximately equivalently near their mean value.

Que 2.5. **Explain the outdoor models given below :**

1. Durkin's model

2. Okumura model.

AKTU 2014-15, Marks 05

OR

Explain the different outdoor models are given below :

i. Hata path loss model.

ii. Okumura model.

AKTU 2018-19, Marks 10

Answer

a. Okumura's model :

1. This model is widely used for signal prediction in urban areas.
2. It is applicable in the frequency range from 150 MHz to 1920 MHz.
3. It can be used for distances from 1 km to 100 km from transmitter to receiver.
4. It can be applied for base station antenna heights ranging from 30 meter to 1000 meter.
5. To determine the path loss using Okumura's model, the equation can be expressed as:

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

where, L_{50} = 50[th] percentile (*i.e.* median) value of propagation path loss.

L_F = Free space propagation loss.

A_{mu} = Median attenuation relative to free space.

$G(h_{te})$ = Base station antenna height gain factor.

$G(h_{re})$ = Mobile antenna height gain factor.

6. This model is completely based on measured data and does not provide any analytical explanation.
7. This model is considered as best in terms of accuracy in path loss prediction in cluttered environments.
8. The main drawback of Okumura's model is that is does not response sufficiently quick to rapid changes in radio path profile or in terrain. Therefore, this model is good in urban and suburban areas but not as good in rural areas.

b. Hata model :

1. Hata model does not account for any of the path specific corrections used in Okumura's model.

2. Hata's model is an experimental formulation of the geographical path loss data provided by Okumura.
3. This model is valid from 150 MHz to 1500 MHz.
4. Hata's model is applicable in urban, suburban and open areas.
5. This model is well suited for large cell mobile systems, but not for personal communications systems (PCS) which have cells of the order of 1 km radius.
6. This model predicts the median path loss in three types of environment: urban, suburban and rural areas. The median path loss in dB for these three environments are given by,

$$L_{50} = A + B \log_{10} d \quad \text{(dB) Urban}$$
$$L_{50} = A + B \log_{10} d - C \text{ (dB) Suburban}$$
$$L_{50} = A + B \log_{10} d - D \text{ (dB) Open}$$

where d is the range in kilometers from BS to MS.

c. **Durkin's model :**
1. This model was developed by Durkin and Edward.
2. This model consists of a computer simulator, for predicting field strength over irregular terrain.
3. This was adopted by Joint Radio Committee (JRC) in the UK for the estimation of mobile coverage area.
4. The simulator predicts large scale phenomena (*i.e.*, path loss) and the losses caused by obstacles in a radio path.
5. The execution of path loss simulator consists two parts :
i. The first part accesses the topographic data base of a proposed service area and reconstructs the ground profile information.
ii. The second part calculates the expected path loss along the radial.

PART-3

Channel Modeling : Stochastic, Flat Fading, Wideband Time-Dispersive Channel.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 2.6. **What do you understand by channel modelling ?**

Answer

1. A channel can be modelled by two approaches: physical and statistical. Reflection, multipath, attenuation, and so on are considered in the physical approach, whereas input-output elements and transaction probabilities are considered in the statistical approach.
2. Perfect channel modelling is required to study or analyse a wireless system. This involves the identification of various channel properties, of which one or more must be considered at a time for modelling the channel.
3. Some important properties of a channel are summarized as follows:
i. Channels may be time varying or static: The effect of mobility is that the channel varies with the users location and time, which results in rapid fluctuations of the received power.
ii. Channels may be time dispersive or non-dispersive. Due to dispersion, pulse spreading will be observed, which will result in the intersymbol interference (ISI) effect.
iii. Channels may be linear or non-linear.
iv. All channels act as low-pass filter under certain conditions as they show the pulse spreading effect.
v. Channels may be fast fading or slow fading, and may be frequency selective or flat fading.
4. Channel models are primarily divided into two main categories:
i. Outdoor channel model (*e. g.*, Longley-Rice model, Okumura-Hata model, Nakagami model, and Rician model)
ii. Indoor channel model (*e. g.*, Rayleigh model)

Que 2.7. **What are the factors influencing small scale fading ? Show that the mobile radio channel can be modelled as a linear filter with time varying impulse response.**

OR

Derive the impulse response model of multipath channel.

AKTU 2014-15, Marks 05

Answer

1. The small scale variations of a mobile radio signal can be directly related to the impulse response of the mobile channel.
2. The impulse response is a wideband channel characterization and contains all information necessary to simulate or analyze any type of radio transmission through the channel.
3. To show that a mobile radio channel may be modeled as a linear filter with a time varying impulse response, consider the case where time

variation is due to only receiver motion in space. This is shown in Fig. 2.7.1.

Fig. 2.7.1. The mobile radio channel as a function of time and space.

4. Consider the receiver moves along the ground at some constant velocity v. For a fixed position d, the channel between the transmitter and the receiver can be modeled as a linear time invariant system.
5. The impulse response of the linear time invariant channel should be a function of the position of the receiver. That is, the channel impulse response can be expressed as $h(d, t)$.
6. Let $x(t)$ represent the transmitted signal, then the received signal $Y(d, t)$ at position d can be expressed as a convolution of $x(t)$ with $h(d, t)$.

$$Y(d, t) = x(t) \otimes h(d, t) = \int_{-\infty}^{\infty} x(\tau) h(d, t-\tau) d\tau \quad ...(2.7.1)$$

For a causal system, $h(d, t) = 0$ for $t < 0$, thus eq. (2.7.1) reduces to

$$Y(d, t) = \int_{-\infty}^{t} x(\tau) h(d, t-\tau) d\tau \quad ...(2.7.2)$$

7. Since the receiver moves along the ground at a constant velocity v, the position of the receiver can be expressed as

$$d = vt \quad ...(2.7.3)$$

Substituting eq. (2.7.3) in eq. (2.7.2), we obtain

$$Y(vt, t) = \int_{-\infty}^{t} x(\tau) h(vt, t-\tau) d\tau \quad ...(2.7.4)$$

8. Since v is a constant, $Y(vt, t)$ is just a function of t. Therefore, eq. (2.7.4) can be expressed as

$$Y(t) = \int_{-\infty}^{t} x(\tau) h(vt, t-\tau) d\tau = x(t) \otimes h(vt, t)$$

$$= x(t) \otimes h(d, t) \quad ...(2.7.5)$$

9. From eq. (2.7.5), it is clear that the mobile radio channel can be modeled as a linear time varying channel, where the channel changes with time and distance.
10. The received signal can be expressed as a convolution of the transmitted signal $x(t)$ with the channel impulse response.

$$Y(t) = \int_{-\infty}^{\infty} x(\tau) h(t, \tau) d\tau$$

$$Y(t) = x(t) \otimes h(t, \tau)$$

i. **For bandpass channel impulse response model :**

If the $x(t)$ is the transmitted bandpass waveform and $h(t, \tau)$ channel impulse response then $Y(t)$ will be :

$$Y(t) = x(t) \otimes h(t, \tau)$$

where,

$$h(t, \tau) = \text{Re}\{h_b(t, \tau)\, e^{j\omega ct}\}$$

$$Y(t) = \text{Re}\{r(t)\, e^{j\omega ct}\}$$

$$Y(t) = x(t) \otimes h(t)$$

ii. **For baseband equivalent channel impulse response model.**

1. If the multipath channel is assumed to be a bandlimited bandpass channel, which is reasonable then $h(t, \tau)$ may be equivalently described by a complex baseband impulse response $h_b(t, \tau)$ with the input and output being the complex envelope representations of the transmitted and received signals, respectively. That is

$$r(t) = c(t) \otimes \frac{1}{2} h_b(t, \tau) \quad ...(2.7.6)$$

where $c(t)$ and $r(t)$ are the complex envelopes of $x(t)$ and $Y(t)$, defined as

$$x(t) = \text{Re}\{c(t) \exp(j2\pi f_c t)\}$$

$$Y(t) = \text{Re}\{r(t) \exp(j2\pi f_c t)\}$$

2. Since the received signal in a multipath channel consists of a series of attenuated, time delayed, phase shifted replicas of the transmitted signal, the baseband impulse response of a multipath channel can be expressed as

$$h_b(t, \tau) = \sum_{i=0}^{N-1} a_i(t, \tau) \exp[j(2\pi f_c \tau_i(t) + \phi_i(t, \tau))] \delta(\tau - \tau_i(t))$$

3. If the channel impulse response is assumed to be time invariant, or is at least wide sense stationary over a small scale time or distance interval, then the channel impulse response may be simplified as

$$h_b(\tau) = \sum_{i=0}^{N-1} a_i \exp(j\theta_i) \delta(\tau - \tau_i)$$

4. For small-scale channel modeling, the power delay profile of the channel is found by taking the spatial average of $|h_b(t; \tau)|^2$ over a local area. The received power delay profile in a local area is given by

$$P(\tau) \approx \overline{K |h_b(t;\tau)|^2}$$

where the bar represents the averages over the local area and many snapshots of $|h_b(t; \tau)|^2$ are averaged over local area (small-scale) to provide a single time-variant multipath power delay profile $P(\tau)$.

5. The gain K relates the transmitted power in the probing pulse $p(t)$ to the total power received in a multipath delay profile.

Que 2.8. **Explain in brief stochastic radio channel modelling.**

Answer

1. In principle, the following three different domains determine radio signal transmission:
   i. Physical conditions selected or operational scenario.
   ii. Dispersion phenomena of wave propagation.
   iii. Transceiver characteristics.
2. The operation scenario implies some fundamental data such as frequency range, system bandwidth and environments, (urban, rural, indoor etc).
3. Dispersion in frequency, time, direction, and polarization is a crucial aspect of radio communication.
4. Transceiver characteristics contributing to the stochastic radio channel modelling (SRCM) are the parameters that describe the mobile terminal (MT) movement and the antenna configuration with its radiation pattern and diversity properties.
5. The deterioration of the transmission quality is also strongly dependent on the signal processing, that is modulation, coding detection and so on.
6. The various parameters extracted for SRCM are complex amplitude, delay, incidence direction and doppler shift of the impinging wave components.
7. A stochastic radio channel model has been developed in order to simulate realistic channel impulse responses according to a wide range of possible physical situations within a given category of environments.
8. If the channel is estimated on the basis of channel statistics, it is called the blind method of channel estimation. The complexity of this method is very high.
9. The transceiver characteristics as well as nearly all dispersion effects that are the different phenomena of multipath propagation and short and long term fluctuations are implemented in the SRCM.

Que 2.9. **Discuss in detail flat fading channel modelling.**

Answer

1. In a flat fading channel, the received signal $r(t)$ is obtained by the addition of the transmitted signal $s(t)$ multiplied by a time-varying attenuation $\alpha(t)$ and the noise contribution $w(t)$:

$$r(t) = \alpha(t)\, s(t) + w(t)$$

2. α is usually follows a Rayleigh distribution

$$\text{pdf}(\alpha) = \left(\frac{\alpha}{\sigma^2}\right) \times \exp\left(-\frac{\alpha^2}{2\sigma^2}\right) \text{ for } 0 < \alpha < \infty$$

where σ^2 is the variance of Gaussian process.

3. If there is one dominant contribution, the distribution of α is a Rice-distributed variable characterized by

$$\text{pdf}(\alpha) = (\alpha/\sigma^2) \times \exp[-(\alpha^2 + A^2)/2\sigma^2] \times I_o(2A/\sigma^2)$$
$$\text{for } 0 < \alpha < \infty$$

where, $I_o(x)$ is the modified Bessel function of the first kind and zero order. The parameter A is the amplitude of the dominant component.

The Rice parameter K is defined as $\left(\frac{A^2}{2\sigma^2}\right)$.

4. If the amplitude fading is Rician, the joint PDF of amplitude and phase Ψ is

$$\text{pdf}_{\alpha\psi} = \left(\frac{\alpha}{2\pi\sigma^2}\right) \times \exp[-(\alpha^2 + A^2 - 2\alpha A\cos\psi)/2\sigma^2]$$

5. Nakagami-M distribution is given by

$$\text{pdf}_{\alpha}(\alpha) = \frac{2}{\Gamma(m)}\left(\frac{m}{\Omega}\right)^m \alpha^{2m-1} \exp\left(-\frac{m}{\Omega}\alpha^2\right)$$

for $\alpha \ge 0$ and $m \ge 1/2$.
Here m is the shape factor. For $m = 1$, this distribution reduces to the Rayleigh distribution.

6. The parameter Ω is the mean square value $\Omega = \alpha^2$ and the parameter m is given by

$$m = \frac{\Omega^2}{(\alpha^2 - \Omega)^2}$$

7. Nakagami and Rice distribution are quite similar and each can be approximately converted to the other for $m \ge 1$:

$$m = \frac{(K+1)^2}{(2K+1)} \quad \text{and} \quad K = \frac{\sqrt{m^2 - m}}{m - \sqrt{m^2 - m}}$$

8. The movement of the mobile station leads to a frequency shift of the arriving waves (Doppler effect). If the sinusoidal wave of frequency f_c is transmitted, the spectrum of the received signal is

$$Y(f) \propto \left[pdf_\gamma(\gamma) G(\gamma) + pdf_\gamma(-\gamma) G(-\gamma) \frac{1}{\sqrt{\left(\frac{v}{c} f_c\right)^2 - (f - f_c)^2}} \right]$$

Here, f is the variable for the frequency, $\frac{v}{c} f_c$ is the extreme new frequency due to Doppler effect as other than f_c and it varies in the range $-\frac{v}{c} f_c < f < \frac{v}{c} f_c$ and 0 elsewhere, v is the velocity or speed of movement, γ is the angle between the directions of incidence of the move and the direction of movement, and $G(\gamma)$ is the antenna pattern.

9. For the case when the waves are all incident horizontally and are uniformly distributed in azimuth and the antenna has a uniform pattern in azimuth, we get

$$v(f) \propto \frac{1}{\sqrt{\left(\frac{v}{c} f_c\right)^2 - (f - f_c)^2}}$$

10. The Rayleigh or Rice fading is small-scale fading since it describes the variation of the amplitude within an area of about ten wavelengths.

Que 2.10. Write a short note on wideband Time-dispersive channel modelling.

Answer

1. Most of the latest communication systems are digital and wideband, such as code division multiple access (CDMA) and orthogonal frequency division multiplexing (OFDM) explained.
2. At the same time, wireless channel is a dielectric medium, and hence, the refractive index of the channel, phase velocity, and wave and media propagation constants are the key parameters.
3. Therefore, over a wide range of frequencies in the transmission bandwidth, frequency-dependent performance is obtained.
4. Wideband measurements are much more complicated than simple field strength (*i.e.*, narrowband) measurement.
5. From the measurements, the channel models are derived, which should fulfil the following two criteria :
   i. They must be simple enough to allow an analytical computation of basic system performance.
   ii. They must be very close to the physical reality; in other words, performance computed by these models must be close to the performance measured in actually existing mobile radio channels.
6. These requirements are contradictory.
7. Models based on multipath delay and hence phase addition are not applied directly to CDMA system, because in CDMA, the rake receiver handles multipath.

VERY IMPORTANT QUESTIONS

Following questions are very important. These questions may be asked in your SESSIONALS as well as UNIVERSITY EXAMINATION.

Q. 1. Write a short note on AWGN channel.
Ans. Refer Q. 2.1.

Q. 2. Describe Rician fading channel.
Ans. Refer Q. 2.3.

Q. 3. Explain the different outdoor models are given below :
i. Hata path loss model.
ii. Okumura model.
Ans. Refer Q. 2.5.

Q. 4. What are the factors influencing small scale fading ? Show that the mobile radio channel can be modelled as a linear filter with time varying impulse response.
Ans. Refer Q. 2.7.

Q. 5. Discuss in detail flat fading channel modelling.
Ans. Refer Q. 2.9.

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3
UNIT

Vocoders and Modulation Techniques

CONTENTS

PART-1

Theory of Vocoders, Types of Vecoders.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 3.1. What is the basic principle of vocoders ? Write down the advantages and disadvantages of vocoders.

Answer

1. Vocoders are a class of speech coding systems that analyze the voice signal at the transmitter, transmit parameters derived from the analysis, and then synthesize the voice at the receiver using those parameters.
2. All vocoders systems attempt to model the speech generation process as a dynamic system and try to quantify certain physical constraints of the system.

Speech generation model :

1. Fig. 3.1.1 shows the traditional speech generation model that is the basis of all vocoding systems. The sound generating mechanism forms the source and is linearly separated from the intelligence modulating vocal tract filter which forms the system.

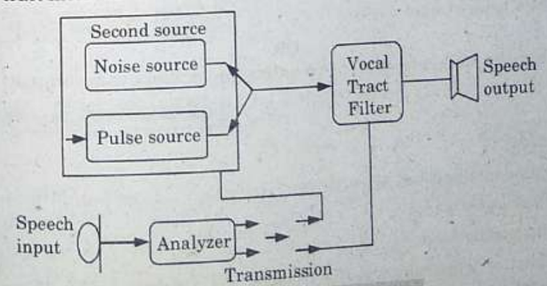

Fig. 3.1.1. Speech generation model.

2. The speech signal is assumed to be of two types: voiced and unvoiced. Voiced sound ("*m*", "*n*", "*v*" pronunciations) are a result of quasiperiodic vibrations of the vocal chord and unvoiced sounds ("*f*", "*s*", "*sh*" pronunciations) are fricatives produced by turbulent air flow through a constriction.

3. The parameters associated with this model are the voice pitch, the pole frequencies of the modulating filter, and the corresponding amplitude parameters.
4. When the signal is voiced, the output of the filter provides a voltage which is proportional to the voice frequency. This frequency is the pitch of the voice.
5. When the speech is unvoiced, the output of the filter is a smaller voltage than we encounter for voiced speech. Using a detector we can then determine by noting the output of filter at detector side whether the speech is voiced or unvoiced.
6. At the vocoder receiver, the signal is demultiplexed and decoded and converted back into analog form. Corresponding to each filter-rectifier combination at the encoder, a balanced amplitude modulator and band-pass filter is provided at the decoder.
7. When the speech is unvoiced, the sound is live noise, while pitch is provided by detector if the speech is voiced.

Advantages of vocoders :
1. Low bit rate.
2. Can be classified into the frequency domain and time domain subclasses.

Disadvantages of vocoders :
1. Much more complex.
2. Less robust.
3. Relatively low synthetic speech quality.

Que 3.2. What is the basic mechanism of vocoder and explain any two type of vocoders ?

AKTU 2014-15, Marks 05

OR

Explain the different type of vocoders in wireless communication.

AKTU 2016-17, Marks 15

Answer

A. Basic mechanism of vocoders : Refer Q. 3.1, Page 3–2A, Unit-3.

B. Types of vocoder :

i. Channel vocoder :
1. The channel vocoder was the first system to analyze speech. These are based on frequency domain coding.
2. It divides the speech signal into a number of frequency bands. Each band is then sampled, encoded and multiplexed. Sampling is done at every 10-30 milliseconds.

ii. Formant vocoder :
1. The word formant indicates to the pole frequencies which correspond to the resonant frequencies of the vocal tract.
2. Theoretically, these vocoders operate at lower bit rate than the channel vocoder. These vocoders can generate the bit rate lower than 1200 bit/sec.
3. A formant vocoder must be able to identify at least three formants for representing the speech signal. But this vocoder faces some difficulty to compute the accurate position of formants therefore these are not very popular.

iii. Cepstrum vocoder :
1. The excitation of signal and vocal tract spectrum are separated in this filter by inverse Fourier transformation. This separation is done to produce the cepstrum of the signal.
2. Linear filtering is used to filter the vocal tract cepstral coefficient from the excitation coefficient. In the receiver side, the cepstral coefficients are Fourier transformed and then convolved with an excitation signal (random noise) to reproduce the original speech.

iv. Voice excited vocoder :
1. Voice excited vocoder quality is superior to other traditional pitch excited vocoders. Here, a pitch signal is generated at the synthesizer by rectifying, bandpass filtering and clipping the baseband signal.
2. The bit rate of these vocoders is 7200 bits/sec to 9600 bits/sec.

Que 3.3. Draw the block diagram of LPC system and explain it.

AKTU 2015-16, Marks 05

Answer

1. Linear predictive coder (LPCs) belongs to the time domain class of vocoders. This class of vocoders attempts to extract the significant feature of speech from the time waveform. With LPC, it is possible to transmit good quality voice at 4.8 kbps and poorer quality voice at even lower rates.
2. In linear predictive coding system, the transfer function is described by

$$H(z) = \frac{G}{1+\sum_{k=1}^{M} b_k Z^{-k}}$$

where, G is a gain of the filter and z^{-1} represents a unit delay operation.
3. The excitation to this filter is either a pulse at the pitch frequency or random white noise depending on whether the speech segment is voiced or unvoiced.

4. The coefficients of the all pole filter are obtained in the time domain using linear prediction techniques.

5. The LPC system transmits only selected characteristics of the error signal. The parameters include the gain factor, pitch information and the voiced/ unvoiced decision information, which allow approximation of the correct error signal.

6. At the receiver, the received information about the error signal is used to determine the appropriate excitation for synthesis filter. That is, the error signal is the excitation to the decoder. The synthesis filter is designed at the receiver using the received predictor coefficients.

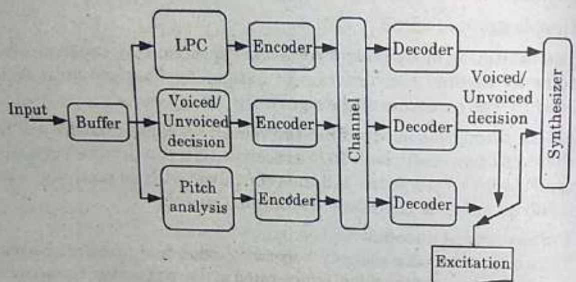

Fig. 3.3.1. Block diagram of a LPC coding system.

Advantages of non-uniform quantization :

i. Minimize the distortion of a signal with a given PDF.
ii. Reduces errors like slop over distortion.

PART-2

Spread Spectrum Modulation, Pseudo-Noise Codes with Properties and Code Generation Mechanisms, DSSS and FHSS Systems.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 3.4. **Write a short note on spread spectrum modulation (SSM).**

Answer

1. Spread spectrum transmission offers the following three main advantages over fixed frequency transmission :
i. Spread spectrum signals are highly resistant to noise and interference.
ii. Spread spectrum signals are difficult to intercept.
iii. Spread spectrum transmissions can share a frequency band with many types of conventional transmission with minimal interference. As a result, bandwidth can be utilized more efficiently.
2. Spread spectrum techniques also meet the following objectives :
i. Operation with a low energy spectral density.
ii. Multiple access capability without external control.
iii. Security.
iv. Anti-jamming capability.
v. Multipath protection.
vi. Ranging.
3. One way of classifying spread spectrum systems is as follows :
a. **Averaging system :** In this system, interference takes place because the interference can be averaged over a large time interval.
b. **Avoidance system :** In this system, reduction of interference occurs because the signal is made to avoid the interference for a large fraction of time.
4. The spread spectrum technique is a wideband modulation technique. The two major techniques used in spread spectrum systems are direct sequence (DS) and frequency hopping (FH).
5. The following are the other spread spectrum modulation (SSM) techniques :
i. Time hoping spread spectrum (THSS).
ii. Hybrid methods.
iii. Chirped spread spectrum (CSS).
6. A DS system is an averaging system whereas FH, time hopping (TH), and chirping systems are avoidance systems.

Que 3.5. **What is PN sequence ? Draw suitable PN sequence generator and prove the properties of PN sequence.**

AKTU 2017-18, Marks 10

Answer

1. In a DS spread spectrum (DSSS) system, a unique code is used to spread and despread the signal this unique code is the DS, known as the

pseudo-noise (PN) code. The PN code, while concatenated, appears as a random sequence to an unauthentic user.

2. In the case of FH or TH, the PN sequence is used to generate the hopping frequency or the hopping time slots.

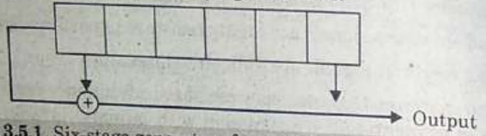

Fig. 3.5.1. Six-stage generator of a maximum length PN sequence.

3. A PN code is periodic. A digital shift register circuit with output feedback can generate a sequence with long period and low susceptibility to structural identification by an outsider.
4. The bit rate of a PN code is called the chip rate ($1/t_{chip}$), the smallest time increment in the sequences of certain period or duration is t_{chip} and is known as a time chip. The total period consists of N_C time chips.
5. The most widely known binary PN sequences are the maximum length shift register sequences (m-sequences). Such a sequence which can be generated by an m-stage shift register with suitable feedback connection, has a length $L = 2^m - 1$ bits, Fig. 3.5.1 shows a shift register for $m = 6$ and $L = 63$.

Autocorrelation :

1. The autocorrelation should be maximal for the DS or PN codes so that correct PN signal can be identified at the receiver from the numerous coexisting signals. The autocorrelation function of a typical PN sequence is shown in Fig. 3.5.2.

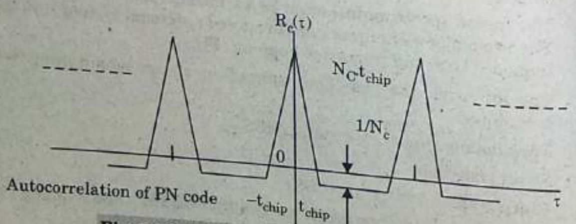

Fig. 3.5.2. Autocorrelation function (normalized) for the codes with respect to time.

2. The autocorrelation of the spreading waveform (PN signal) $c(t)$ is represented mathematically as

$$R_a(\tau) = \frac{1}{T_{code}} \int_0^{\infty} c(t)c(t+\tau)\, dt \qquad ...(3.5.1)$$

where, $T_{code} = N_c\, t_{chip}$ is the code period and τ represents a time shift variable.

Cross-correlation :

1. Signals of different users have different spreading codes. The cross-correlation between the signals of two codes i and j is given by

$$R_c(\tau) = \frac{1}{T_{code}} \int_0^{\infty} c_i(t)\, c_j(t+\tau)\, dt$$

which is equal to the autocorrelation, if $i = j$.

2. It is desirable to have poor cross-correlation between two different codes so that the unwanted code can be rejected easily by the receiver of the CDMA system.

Properties of PN codes :

i. **Balance property :** In each period of the sequence, the number of binary 1's differs from that of binary 0's by at most one digit. Consider a typical PN code :
0001 0011 0101 111 (seven 0's and eight 1's meet the balance condition)

ii. **Run length property :** Among the runs of 1's and 0's in each period, it is desirable that about one half of the runs of each type are of length one, one-fourth are of length two, one-eight are of length three and so on. Consider the same code again :
Number of runs = 8

$$\frac{000}{3}\ \frac{1}{1}\ \frac{00}{2}\ \frac{11}{2}\ \frac{0}{1}\ \frac{0}{1}\ \frac{1111}{4}$$

iii. **Autocorrelation property :** The autocorrelation function of a maximal length sequence is periodic and binary valued. We can state the autocorrelation function as

$$R_a(\tau) = \frac{1}{N_c}\text{[Number of agreements } (a) - \text{Number of disagreements } (d) \text{ in one full period]}$$

$$\begin{array}{l} 000100110101111 \\ \underline{100010011010111} \\ daaddadadddda\,a\,a \end{array}$$

$$R_c(\tau) = -\frac{1}{15}$$

Que 3.6. **Draw and explain the transmitter and receiver block diagrams of a DSSS system.**

OR

What are the different types of vocoders and describe direct sequence spread spectrum.

AKTU 2018-19, Marks 10

Answer

A. **Different types of vocoders :** Refer Q. 3.2, Page 3–3A, Unit-3.

B. **Direct sequence spread spectrum system :** In a DSSS system, the user signal is multiplied by a PN code sequence of high bandwidth. The resulting coded signal is transmitted over the radio channel. DS is responsible for the spreading of the bandwidth.

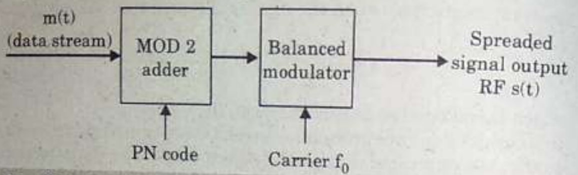

Fig. 3.6.1. Transmitter simplified diagram for biphase modulation.

Transmitter and receiver :

1. In the transmitter (Fig. 3.6.1) of a DSSS system, a MOD 2 adder is used for biphase modulation.
2. While using quadriphase modulation, two MOD 2 adders are used with two alternate chips available from the PN code generator.
3. Two balanced modulators are fed with 90° phase shifted carriers (similar to QPSK generation). Adding both the signals, the SSM RF output is obtained.
4. The diagram of a receiver is shown in Fig. 3.6.2. The receiver for a spread signal must perform three distinct functions: detection of the presence of a signal, carrier removal and despreading or demodulation using a PN sequence.
5. Detection of signal and despreading operations can be either active or passive.

i. **Active method :** It involves searching for the signals presence in both the time and frequency domains and tracking the sequence after it has been acquired, despreading the signal with the correlator, and demodulating the signal in the usual way, that is using a PN sequence again.

ii. **Passive method :** It is required to search the signal in terms of its carrier frequency to be detected, because the passive system will respond whenever the signal occurs. Despreading is accomplished in a matched filter rather than a correlator. Demodulation is performed in the usual manner as mentioned previously.

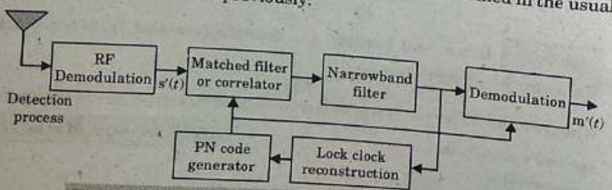

Fig. 3.6.2. Receiver carrier demodulation and despreading of SSM signal to get original data.

Que 3.7. **With the help of block diagram and suitable expressions explain the generation and reception of direct sequence spread spectrum (DSSS) signal using BPSK modulation.**

AKTU 2017-18, Marks 10

Answer

1. Direct sequence spread spectrum systems are much more efficient in bandwidth and power utilization.
2. DSSS has become the prominent CDMA technology in advanced wireless communication system.
3. Direct sequence spread spectrum expands the traditional narrowband signal by utilizing a spreading signal $c(t)$. This technology is more suitable for integration with bandwidth efficient linear modulation such as QAM/PSK.
4. In this technique spectrum spreading is achieved by a PN sequence, also known as PN code or PN chip.
5. The PN sequence is mostly binary, consisting of 1 s and 0 s, which are represented by polar signaling of + 1 and – 1. In order to minimize interference and to facilitate chip synchronization PN sequence uses its properties of autocorrelation and cross-correlation.

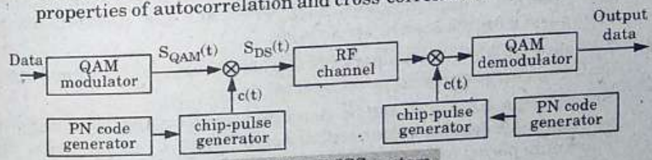

Fig. 3.7.1. DSSS system.

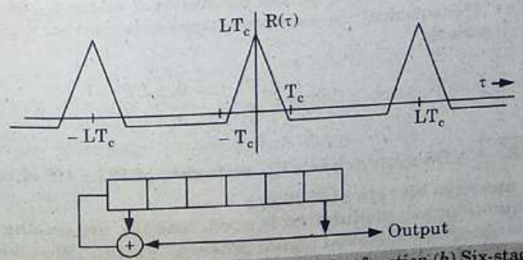

Fig. 3.7.2. (*a*) PN sequence autocorrelation function (*b*) Six-stage generator of a maximum length PN sequence.

6. From the Fig. 3.7.2, it is shown that the original data signal is linearly modulated into QAM signal $S_{QAM}(t)$.

7. Instead of transmitting this signal directly over its required bandwidth, DSSS modifies the QAM signal by multiplying the $c(t)$ (spreading chip signal) with the QAM narrow band signal.
8. Though the signal carrier frequency remains unchanged at ω_c, then the new signal after spreading becomes

$$S_{DS}(t) = S_{QAM}(t)\, c(t) \quad \text{...(3.7.1)}$$

9. Hence, the transmitted signal $S_{DS}(t)$ is a product of two signals whose spread bandwidth is equal to the bandwidth sum of the QAM signal $S_{QAM}(t)$ and the spreading signal $c(t)$.

i. PN sequence generation :

1. A good PN sequence $c(t)$ is characterized by an autocorrelation *i.e.*, similar to that of a white noise.
2. This indicates that autocorrelation function of a PN sequence should be high for $\tau = 0$ and low for all $\tau \neq 0$ as shown in Fig. 3.7.1(*a*).
3. Moreover in CDMA applications several users share the same band using different PN sequence.
4. Hence, it is necessary that the cross-correlation should be small among different pairs of PN sequences in order to reduce mutual interference.
5. A PN code is periodic. A digital shift register circuit with output feedback can generate a sequence with long period and low susceptibility to structural identification by an outsider.
6. The most widely known binary PN sequences are the maximum length shift register sequences (m-sequences).
7. Such a sequence, which can be generated by an m-stage shift register with suitable feedback connection, has a length $L = 2^m - 1$ bits, the maximum period for such a finite state machine.
8. Fig. 3.8.2(*b*) shows a shift register encoder for $m = 6$ and $L = 63$. For such "short" PN sequences, the autocorrelation function is nearly an impulse and is periodic.

$$R_c(\tau) = \int_0^{T_c} c(t)c(t+\tau)d\tau = \begin{cases} LT_c & \tau = 0, \pm LT_c,... \\ -T_c & \tau \neq 0, \pm LT_c,... \end{cases}$$

Que 3.8. **A DS system has a PN code rate of 192×10^6 chips and a binary message bit rate at 7500 bps.**
a. If quadriphase modulation is used, find the processing gain.
b. Assuming the received signal power is 4×10^{-14} W and the one-sided noise spectral density level N_o is 1.6×10^{-20} W/Hz, find the SNR in the input bandwidth of the receiver.

Answer

Given : $t_{chip} = \dfrac{1}{192 \times 10^6} = 0.0052\ \mu s$, $t_m = \dfrac{1}{7500} = 0.133$ ms,
P_r = Received power = 4×10^{-14} W, One sided noise spectral density = 1.6×10^{-20} W/Hz

To Find : Processing gain, SNR.

a. For quadriphase modulation

$$PG = \frac{t_m}{t_{chip}} = 0.133 \times 10^{-3}/0.0052 \times 10^{-6} = 25.577$$

Processing gain in decibels = $10 \log_{10} (25.577) = 44.08$ dB

b. For biphase modulation

1. Signal bandwidth $= \dfrac{2}{t_{chip}} = \dfrac{2}{0.0052} \times 10^6 = 384.6$ MHz

Noise power = Noise spectral density × Bandwidth
$= 1.6 \times 10^{-20} \times 384.6 \times 10^6$
$= 615 \times 10^{-14}$ W

$$SNR = \frac{P_r}{N_o} = 4 \times 10^{-14}/615 \times 10^{-14} = 0.0065$$

In decibel scale, SNR = $10 \log_{10} (P_r/N_o) = -21.86$ dB

2. For quadriphase modulation.

Signal bandwidth $= \dfrac{1}{t_{chip}} = 1/0.0052 \times 10^{-6} = 192.3$ MHz

Noise power = N_o × Bandwidth = $1.6 \times 10^{-20} \times 192.3 \times 10^6$
$= 307.7 \times 10^{-14}$ W

$$SNR = \frac{P_r}{N_o} = 4 \times 10^{-14}/307.7 \times 10^{-14} = 0.013$$

In decibel scale, SNR = $10 \log_{10} (P_r/N_o) = -18.86$ dB

Que 3.9. **Write short note Near-far problem.**

Answer

1. Near-far problem in a spread spectrum system relates to the problem of very strong signals at the receiver swamping out the effects of weaker signals.
2. Consider one transmitter is near the receiver; the other is far away from the receiver. If both the transmitters transmit simultaneously with equal powers, then the receiver will receive more power from the nearer transmitter. This creates a difficulty in detecting the signal from the farther transmitter.

3. As one transmission's signal is the other's noise, the SNR for the farther transmitter must be much higher.
4. If the nearer transmitter transmits a signal that is orders of magnitude higher than the farther transmitter, then the SNR for the farther transmitter may be below the required value, making the signal undetectable and the farther transmitter may just as well not transmit. This effectively jams the communication channel.
5. In short, the near-far problem is one of detecting and receiving a weaker signal among the stronger signals. In order to maintain the strength of the received signal level at the base station, power control is employed in CDMA systems.

Que 3.10. Explain in detail frequency hopping spread spectrum (FHSS) system.

Answer

1. In frequency hopping systems, the transmitter changes the carrier frequency according to a certain hopping pattern, meaning that the frequency is constant in each time chip but changes from chip to chip.
2. There are two kinds of FH – slow FH (SFH) and fast FH (FFH).
3. In SFH, one or more data bits are transmitted within one hop; that is the hoping rate is less than the message bit rate. An advantage is that coherent data detection is possible. Systems using slow hopping often also employ (burst) error-control coding to restore loss of (multiple) bits in one hope.
4. In FFH, one data bit is divided over multiple hops, that is, FH rate is greater than the message bit rate. In fast hopping, coherent signal detection is difficult, and seldom used Mostly, FSK or MFSK modulation is used.
5. SFH is a popular technique for wireless local area networks (LANs). FFH is adopted in bluetooth.
6. Consider a fast hop system in which there are k frequency hops in every message bit duration t_m. Thus, the chip duration is

$$t_{chip} = \frac{t_m}{k} = 1, 2, 3, ...$$

7. The number of frequencies is in a power of two because it is generated by a PN sequence generator control, and PN sequences are related to powers of two. The ML sequence that is, c chips, will produce $M = 2^c$ frequencies for each distinct combination of these digits.
8. As shown in Fig. 3.10.1, one bit comes from the message and $c - 1$ bit come from the PN code generator. The $c - 1$ bit from the PN code generator then hop this FSK signal over the range of possible frequencies.
9. There is a frequency multiplier K at the output of the system. It is to increase the bandwidth and thereby increase the processing gain.

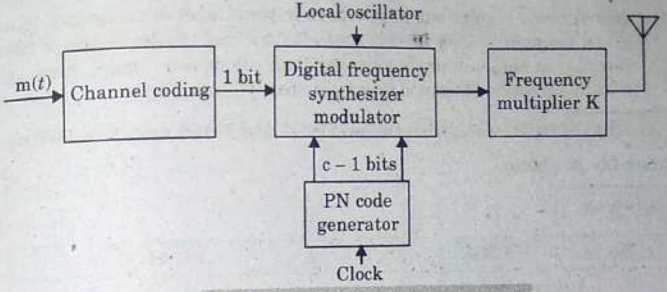

Fig. 3.10.1. FHSS generator diagram.

10. Considering again the fast hopping case, if M frequencies are separated by $f_1 = 1/t_{chip} = k/t_m$, then the signal bandwidth is given by

$$B_s = KMf_1 = \frac{KM}{t_{chip}}$$

Hence, the processing gain is calculated as

$$PG = \frac{B_s}{B_m} = \frac{KM / t_{chip}}{1/t_m} = \frac{kKM / t_m}{1/t_m} = kKM$$

11. A non-coherent FHSS receiver is shown in Fig. 2.10.2. The locally generated frequency hop signal is multiplied by the incoming signal in a mixer.

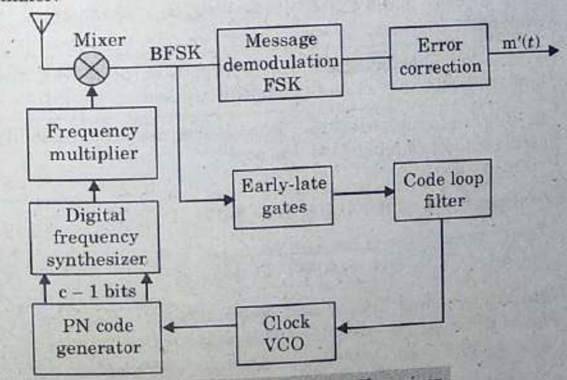

Fig. 3.10.2. Non-coherent FH receiver.

12. If the two are in step, the result will be a normal BFSK signal, which is demodulated in the usual way.
13. Error corrections made to improve the BER response. The output of the mixer is also applied to the early and late gates that produce an error signal to control the clock frequency. This keeps the locally generated frequency hop pattern in synchronism with the incoming signal.

14. Interference results whenever there is simultaneous occupancy of a given frequency slot; in this case, it does not matter much if the interfering signal is much stronger than the desired signal, because within a fraction of time, a new frequency will come up.

Que 3.11. **Differentiate between DSSS and FHSS systems. Define near-far problem.**

Answer

S.No.	DSSS	FHSS
1.	A direct sequence spread spectrum (DSSS) system spreads the baseband data by directly multiplying the baseband data pulses with a pseudo-noise (PN) sequence that is produced by a pseudo-noise code generator.	Frequency hopping involves a periodic change of transmission frequency. A frequency hopping signal obtained by a sequence of modulated data bursts with time-varying, pseudo-random carrier frequencies (called hopset).
2.	Spreading process is complex.	Spreading process is simple.
3.	DSSS systems always use the total bandwidth available.	FHSS systems use only a portion of the total band at any time.
4.	More resistant to fading and multipath effects.	Less resistant to fading and multipath effects.
5.	Rejection capabilities to cancel out interferences and noise are good.	Interference rejection capability is poor.

Near-far problem : Refer Q. 3.9, Page 3–12A, Unit-3.

PART-3

Time Hopping and Hybrid Spread System : Multi-carrier Modulation Techniques, Zero Intersymbol Interference Communication Techniques, Detection Strategies.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 3.12. **Explain time hopping spread spectrum (THSS) system.**

Answer

1. The TH concepts can be understood with the help of Fig. 3.12.1.

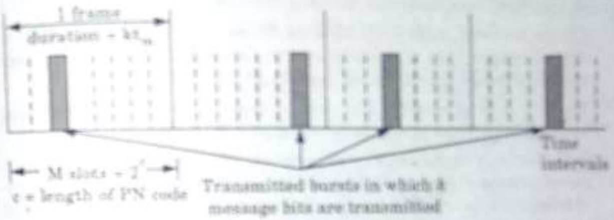

Fig. 3.12.1. TH Concept.

2. The time axis is divided into intervals known as frames, and each frame is subdivided into M time slots as shown in Fig. 3.12.1. The slots and length of the m-sequence are related to each other by the relation $M = 2^r$
3. During each frame, one and only one time slot is modulated with a message by any reasonable modulation method.
4. The particular time slot chosen for a given frame is selected by means of a PN code generator. All the message bits accumulated in the previous frame are transmitted in a burst during the selected time slot.
5. The frame duration T_f, the number of message bits k, and the message bit duration t_m are related to each other by

$$T_f = kt_m$$

6. The width of each time slot in a frame is T_f/M and the width of each bit in the time slot is T_f/kM or simply t_m/M.

Processing gain :

$$PG = \begin{cases} B_s / B_m = 2t_m / (t_m / M) = 2M & ;\text{ for biphase modulation} \\ t_m / (t_m / M) = M & ;\text{ for quadriphase modulation} \end{cases}$$

7. This indicates that the transmitted signal bandwidth is $2M$ times the message bandwidth, and hence, the processing gain of the TH system is twice the number of time slots in each frame when biphase modulation is used and half this when quadriphase modulation is used.
8. Interference among simultaneous users in a TH system can be minimized by coordinating the times at which each user can transmit a signal. This also avoids the near-far problem. The acquisition time is similar to that of DS systems for a given bandwidth. Implementation is simpler than in an FH system.

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Que 3.13. Write short note on hybrid spread spectrum systems.

Answer

1. The use of hybrid techniques attempt to capitalize upon the advantages of a particular method while avoiding the disadvantages.
2. DS, on one hand, suffers heavily from the near-far effect, which makes this technique hard to apply to systems without the ability of power control. On the other hand, its implementation is inexpensive.
3. The PN code generators are easy to implement and the spreading operation itself can be simply performed by XOR ports.
4. FH effectively suppresses the near-far effect and reduces the need for power control. However, implementation of the (fast) hopping frequency synthesizer required for a reasonable spreading gain is more problematic in terms of higher silicon cost and increased power consumption.
5. Applying both techniques allows for combining their advantages while reducing the disadvantages. This results in a reasonable nearfar resistance at an acceptable hardware cost.
6. Many different hybrid combinations are possible, some of which are PN/FH, PN/TH, FH/TH and PN/FH/TH.
7. While designing a hybrid system, the designer should decide FFH or SFH is to be applied. FFH increases the cost of the frequency synthesizer but provides more protection against the near-far effect.
8. SFH combines a less expensive synthesizer with a poor near-far rejection and the need for a more powerful error-correction scheme (several symbols are lost during a hit jamming).

Que 3.14. Discuss in brief multi-carrier modulation (MCM).

Answer

1. In multi-carrier modulation, multiple orthogonal carriers are used to send the information signal. The main attraction of MCM is its good ISI mitigation property. Higher bit rates are more vulnerable to ISI.
2. MCM splits the high bit rate stream into many lower bit rate streams, each stream being sent using an independent carrier frequency, if for example, n symbols/s have to be transmitted, each subcarrier transmits n/c symbols/s with c being the number of subcarriers. One symbol could for example represent 2 bit as in QPSK.
3. In a multi-carrier modulation, symbols are assigned to multiple orthogonal carriers and all the carriers are transmitted in a combined manner; hence, the carriers are called subcarriers.

4. After symbol assignment, these subcarriers are combined by the inverse fast Fourier transform (IFFT) technique. This combined signal is known as the OFDM baseband. For symbol mapping, conventional M-ary techniques (QPSK and 16 QAM) are used.
5. Fig. 3.14.1 shows the superposition of orthogonal frequencies. The maximum of one subcarrier frequency appears exactly at a frequency where all other subcarriers equal zero.

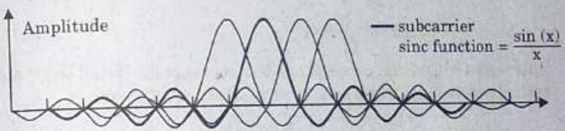

Fig. 3.14.1. Superposition of orthogonal frequencies.

6. Using this scheme, frequency selective fading only influences some subcarriers, and not the whole signal-an additional benefit of MCM.
7. Typically, MCM transmits symbols with guard spaces between single symbols or groups of symbols. This helps the receiver to handle multi-path propagation.
8. OFDM is special method of implementing MCM using orthogonal carriers. Computationally, this is a very efficient algorithm based on fast Fourier transform (FFT) for modulation/demodulation.
9. If additional error-control coding across the symbols in different subcarriers is applied, the system is referred to as COFDM (coded OFDM).

$$s_k(t) = \begin{cases} \cos(2\pi k \Delta f t) & ; \quad 0 < 1 < T_s, k = 1, 2, 3, \ldots N_c \\ 0 & ; \quad \text{(otherwise)} \end{cases} \quad \ldots(3.14.1)$$

where, $\Delta f = \dfrac{1}{T_s}$

is the subcarrier spacing, T_s is the symbol duration, N_c is the number of subcarriers, and $(N_c + 1)\Delta f$ is the transmission bandwidth of the OFDM baseband signal.

Que 3.15. Explain nyquist criterion for ISI cancellation.

Answer

1. Nyquist was the first to solve the problem of overcoming intersymbol interference while keeping the transmission bandwidth low.
2. He observed that the effect of ISI could be completely nullified if the overall response of the communication system is designed so that at every sampling instant at the receiver, the response due to all symbols except the current symbol is equal to zero.
3. Consider the impulse response

$$h_{\text{eff}}(t) = \frac{\sin(\pi t / T_s)}{(\pi t)/T_s} \quad ...(3.15.1)$$

This impulse response satisfies the Nyquist condition for ISI cancellation.

4. Therefore if the overall communication system can be modelled as a filter with the impulse response of eq. (3.15.1), it is possible to eliminate the effect of ISI.
5. The transfer function of the filter is given by

$$H_{\text{eff}}(f) = \frac{1}{f_s}\,\text{rect}\left(\frac{f}{f_s}\right) \quad ...(3.15.2)$$

This transfer function corresponds to a rectangular "brick-wall" filter with absolute band width $f_s/2$, where f_s is the symbol rate.

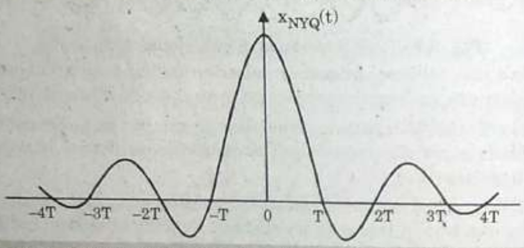

Fig. 3.15.1. Nyquist ideal pulse shape for zero intersymbol interference.

6. Nyquist also proved that any filter with a transfer function having a rectangular filter of bandwidth $f_0 \geq 1/2T_s$, convolved with any arbitrary even function $Z(f)$ with zero magnitude outside the passband of the rectangular filter, satisfies the zero ISI condition.
7. Mathematically, the transfer function of the filter which satisfies the zero ISI condition can be expressed as

$$H_{\text{eff}}(f) = \text{rect}\left(\frac{f}{f_0}\right) \otimes Z(f)$$

where, $Z(f) = Z(-f)$ and $Z(f) = 0$ for $|f| \geq f_0 \geq 1/2T_s$.

8. Expressed in terms of the impulse response, the Nyquist criterion states that any filter with an impulse response

$$h_{\text{eff}}(t) = \frac{\sin(\pi t / T_s)}{\pi t}\, z(t)$$

can achieve ISI cancellation. Filters which satisfy the Nyquist criterion are called Nyquist filters.

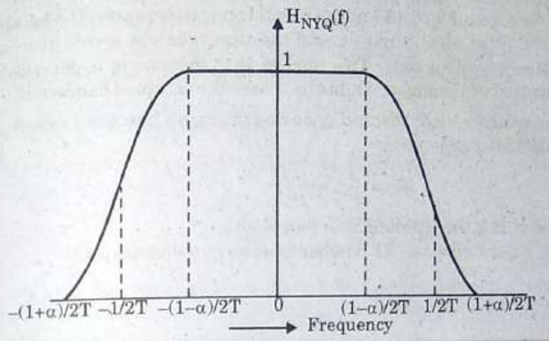

Fig. 3.15.2. Transfer function of a Nyquist pulse-shaping filter at baseband.

Que 3.16. **Write a short note on raised cosine rolloff filter.**

Answer

1. The most popular pulse shaping filter used in mobile communications is the raised cosine filter.
2. A raised cosine filter belongs to the class of filters which satisfy the Nyquist criterion.
3. The transfer function of a raised cosine filter is given by

$$H_{RC}(f) = \begin{cases} 1 & ; \quad 0 \leq |f| \leq \dfrac{(1-\alpha)}{2T_s} \\ \dfrac{1}{2}\left[1 + \cos\left[\dfrac{\pi|f|\cdot 2T_s - 1 + \alpha}{2\alpha}\right]\right] & ; \quad \dfrac{(1-\alpha)}{2T_s} \leq |f| \leq \dfrac{(1+\alpha)}{2T_s} \\ 2 & ; \quad |f| > \dfrac{(1+\alpha)}{2T_s} \end{cases}$$

where, α is the rolloff factor which ranges between 0 and 1.

4. When $\alpha = 0$, the raised cosine rolloff filter corresponds to a rectangular filter of minimum bandwidth. The corresponding impulse response of the filter is given by

$$h_{RC}(t) = \frac{\sin\left(\dfrac{\pi t}{T_s}\right)}{\pi t} \cdot \frac{\sin\left(\dfrac{\pi \alpha t}{T_s}\right)}{1 - \left(\dfrac{4\alpha t}{2T_s}\right)}$$

5. The impulse response of the cosine rolloff filter at baseband is plotted in Fig. 3.16.1, for various values of α. Notice that the impulse response decays much faster at the zero-crossings when compared to the "brick-wall" filter ($\alpha = 0$).

6. As seen from Fig. 3.16.1 as the rolloff factor α increases, the bandwidth of the filter also increases, and the time sidelobe levels decrease in adjacent symbol slots. This implies that increasing α decreases the sensitivity to timing jitter, but increases the occupied bandwidth.
7. The symbol rate R_s that can be passed through a baseband raised cosine rolloff filter is given by

$$R_s = \frac{1}{T_s} = \frac{2B}{1+\alpha}$$

where, B is the absolute filter bandwidth.

8. For RF systems, the RF passband bandwidth doubles and

$$R_s = \frac{B}{1+\alpha}$$

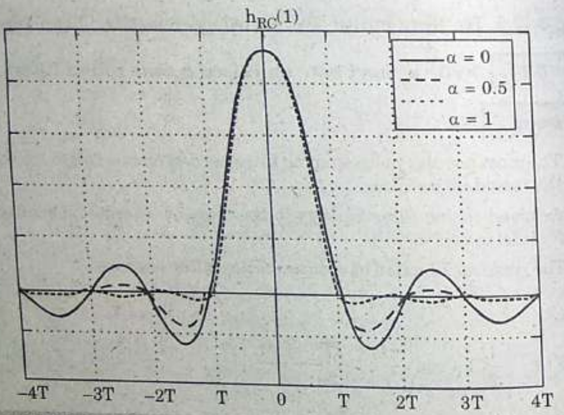

Fig. 3.16.1. Impulse response of a raised cosine rolloff filter at baseband.

Que 3.17. **Explain various pulse shaping techniques for zero intersymbol interference.**

Answer

Various pulse shaping techniques for zero ISI :

i. **Nyquist criterion for ISI cancellation :** Refer Q. 3.15, Page 3–18A, Unit-3.

ii. **Raised cosine rolloff filter :** Refer Q. 3.16, Page 3–20A, Unit-3.

iii. **Gaussian pulse shaping filter :**

1. It is also possible to use non-Nyquist techniques for pulse shaping. Unlike Nyquist filters which have zero crossings at adjacent symbol peaks and a truncated transfer function, the Gaussian filter has a smooth rise to a transfer function with no zero-crossings.

2. The impulse response of the Gaussian filter gives rise to a transfer function that is highly dependent upon the 3-dB bandwidth.
3. The Gaussian lowpass filter has a transfer function given by

$$H_G(f) = \exp(-\alpha^2 f^2)$$

4. The parameter α is related to B, the bandwidth of the baseband Gaussian shaping filter.

$$\alpha = \frac{\sqrt{\ln 2}}{\sqrt{2}B} = \frac{0.5887}{B}$$

5. As α increases, the spectral occupancy of the Gaussian filter decreases and time dispersion of the applied signal increases. The impulse response of the Gaussian filter is given by

$$h_G(t) = \frac{\sqrt{\pi}}{\alpha}\exp\left(-\frac{\pi^2}{\alpha^2}t^2\right)$$

6. Fig. 3.17.1 shows the impulse response of the baseband Gaussian filter for various values of 3-dB bandwidth-symbol time product (BT_s).

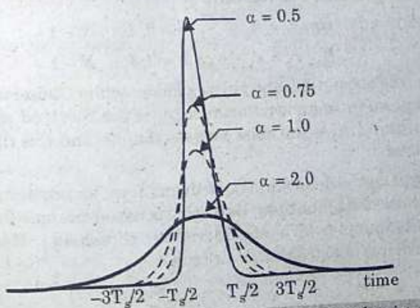

Fig. 3.17.1. Impulse response of a Gaussian pulse-shaping filter.

7. Since the Gaussian pulse-shaping filter does not satisfy the Nyquist criterion for ISI cancellation, reducing the spectral occupancy creates degradation in performance due to increased ISI. Thus, a tradeoff is made between the desired RF bandwidth and the irreducible error due to ISI of adjacent symbols when Gaussian pulse shaping is used.
8. Gaussian pulses are used when cost and power efficiency are major factors and the bit error rates due to ISI are deemed to be lower than what is nominally required.

Que 3.18. **Write a short note on detection strategies for a signal.**

Answer

1. The expected signal must be detected from among the various coexisting signals at the receiving end. Detection theory, or signal detection theory, is well established and is a means to quantify the ability to differentiate between information bearing energy patterns and random energy patterns (such as noise) that distract from the information.
2. The basis for the signal detection theory is that nearly all reasoning and decision-making takes place in the presence of some uncertainty.
3. Mostly, the threshold-based detection method is used but according to the theory of detection, there are a number of determiners of how a detecting system will detect a signal and where its threshold level will be.
4. The aim of the detection problem is to determine whether there exists a signal or not.
5. Suppose there are two states H_0 and H_1 at detection hypothetically, level, only two states are possible-the presence of the required signal within noise or only noise with no desired signal; mathematically this can be represented as follows :

$$H_0 : r_k = n_k ; \quad k = 0, 1, \dots N-1$$
$$H_1 : r_k = s_k + n_k ; \; k = 0, 1, \dots N-1$$

where, the noise is assumed to be additive white Gaussian noise (AWGN) with known or unknown variance, r_k is the received signal sample at time instant k, n_k is the noise process sample, and s_k is the known signal waveform.

6. The signal can be deterministic with unknown parameters such as arrival time, phase and amplitude. Detection is based on some function T of the received samples, which is compared to a threshold γ. If the threshold is exceeded, it is decided that H_1 is true.

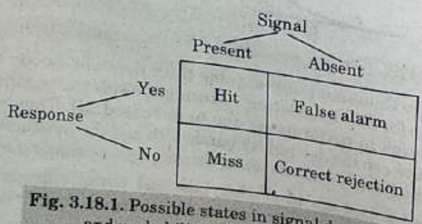

Fig. 3.18.1. Possible states in signal detection and probabilities that may occur.

7. Let r denote the received samples as a column vector. The probability of a false alarm P_{FA} is the probability that H_1 is selected even when H_0 is actually true; that is $P_{FA} = P(T(r) > \gamma, H_0)$

8. The probability of miss P_M is the probability that H_1 is selected when H_1 is true.
9. The probability of detection is $P_D = 1 - P_M$ and is the probability that H_1 is selected when it is actually true; that is $P_D = P(T(r)) > \gamma, H_1)$. The concept is shown in Fig. 3.18.1.

PART-4

Diversity Combining Techniques : Selection Combining, Thresold Combing, Equal Gain Combining, Maximal Ratio Combining, Spatial Diversity and Multiplexing in MIMO System Channel Estimation.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 3.19. **Explain the different type of diversity techniques used in wireless communication system.**

AKTU 2014-15, Marks 05

AKTU 2016-17, 2018-19; Marks 10

Answer

Types of diversity techniques :

i. **Space diversity :**

1. This can be a microscopic or macroscopic diversity technique and is used at the transmitter or the receiver. The signal is transferred over several different propagation paths.
2. In wired transmission, this can be achieved by transmitting via multiple wires.
3. In wireless transmission, it can be achieved by antenna diversity using multiple transmitter antennas (transmit diversity) and/or multiple receiving antennas (receive diversity).
4. If the receiver has multiple antennas, the distance between the receiving antennas is made large enough to ensure independent fading. This arrangement is called space diversity.

ii. **Polarization diversity :**

1. This technique exploits the fact that obstacles scatter waves differently depending on their polarization. Antennas can transmit either a horizontal or a vertical polarized wave.
2. When both waves are transmitted simultaneously, received signals will exhibit uncorrelated fading statistics. This scheme can be considered as a special case of space diversity because separate antennas are used.

3. However, only two diversity branches are available, as there are only two orthogonal polarizations.

iii. Angle diversity :

1. In this technique, directional antennas receive only a fraction of all the scattered energy. As the received signals arrive at the antenna via several paths, each with a different angle of arrival, the signal component can be isolated by using directional antennas.
2. Each directional antenna will isolate a different angular component. Hence, the signals received from different directional antennas pointing at different angles are uncorrelated.

iv. Frequency diversity :

1. In this technique, information is transmitted on more than one carrier frequency, because the frequencies separated by more than the coherence bandwidth of the channel will not experience the same fading.
2. This is often employed in microwave line-of-sight (LOS) links, which carry several channels in a frequency division multiplexing (FDM) mode.

v. Time diversity :

1. In this technique, multiple versions of the same signal are transmitted at different time instants. Alternatively, a redundant forward error-correction code is added, and the message is spread in time by means of bit interleaving before it is transmitted.
2. Thus, error bursts are avoided, which simplifies the error correction. When the same data is sent at different time instants, the received signals can be uncorrelated if the time separations are large enough.
3. Data can be repeatedly transmitted at time spacing that exceeds the coherence time of the channel; hence, multiple repetitions of the signal will be received with independent fading conditions, thereby providing diversity.

Que 3.20. **Explain diversity combining techniques in detail.**

Answer

1. Space diversity reception or combining methods can be classified into four categories :

a. Selection combining :

1. Selection combining is the simplest combining technique. A block diagram of this method is shown in Fig. 3.20.1 where m demodulators are used to provide m diversity branches whose gains are adjusted to provide the same average SNR for each branch.

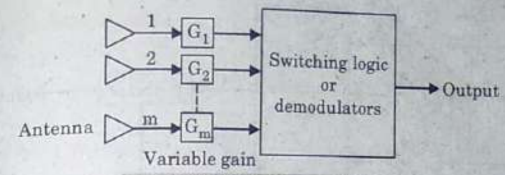

Fig. 3.20.1. Selection combining.

2. The receiver branch having the highest instantaneous SNR is connected to the demodulator.
3. The antenna signals themselves could be sampled and the best one sent to a single demodulator. In practice, the branch with the largest $(S + N)/N$ is used, since it is difficult to measure SNR alone.
4. A practical selection diversity system cannot function on a truly instantaneous basis, but must be designed so that the internal time constants of the selection circuitry are shorter than the reciprocal of the signal fading rate.

b. Feedback or scanning or threshold combining :

1. Threshold combining is very similar to selection diversity except that instead of always using the best of M signals, M signals are scanned in a fixed sequence until one is found to be above a predetermined threshold.
2. This signal is then received until it falls below threshold and the scanning process is again initiated.

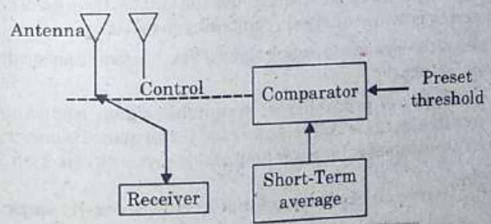

Fig. 3.20.2. Threshold combining.

3. The resulting fading statistics are some what inferior to those obtained by the other methods, but the advantage with this method is that it is very simple to implement only one receiver is required. A block diagram of this method is shown in Fig. 3.20.2.

c. Maximal ratio combining :

1. In this method, the signals from all of the M branches are weighted according to their individual signal voltage to noise power ratios and then summed. Fig. 3.20.3 shows a block diagram of the technique.

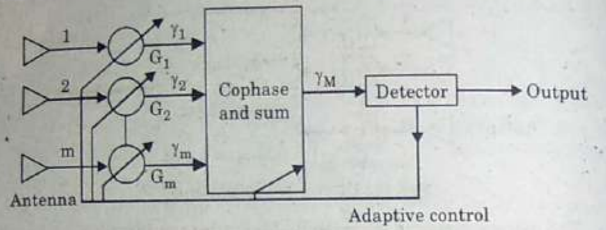

Fig. 3.20.3. Maximal ratio combiner

2. Here, the individual signals must be cophased before being summed (unlike selection diversity) which generally requires an individual receiver and phasing circuit for each antenna element.
3. Maximal ratio combining produces an output SNR equal to the sum of the individual SNRs. Thus, it has the advantage of producing an output with an acceptable SNR even when none of the individual signals are themselves acceptable.
4. This technique gives the best statistical reduction of fading of any known linear diversity combiner. Modern DSP techniques and digital receivers are now making this optimal form of diversity practical.

d. Equal gain combining :

1. In certain cases, it is not convenient to provide for the variable weighting capability required for true maximal ratio combining. In such cases, the branch weights are all set to unity, but the signals from each branch are cophased to provide equal gain combining diversity.
2. This allows the receiver to exploit signals that are simultaneously received on each branch.
3. The possibility of producing an acceptable signal from a number of unacceptable inputs is still retained, and performance is only marginally inferior to maximal ratio combining and superior to selection diversity.

Que 3.21. **Derive an expression for selection diversity improvement in terms of probability of receiving signal using single branch or using *M* branches.**

AKTU 2015-16, 2017-18; Marks 10

Answer

1. Consider M independent Rayleigh fading channels available at a receiver. Each channel is called a diversity branch. Further, assume that each branch has the same average SNR given by

$$SNR = \Gamma = \frac{E_b}{N_0}\overline{\alpha^2} \quad ...(3.21.1)$$

where we assume $\overline{\alpha^2} = 1$.

2. If each branch has an instantaneous $SNR = \gamma_i$, then the pdf of γ_i is

$$p(\gamma_i) = \frac{1}{\Gamma} e^{\frac{-\gamma_i}{\Gamma}} \quad \gamma_i \geq 0 \quad ...(3.21.2)$$

where Γ is the mean SNR of each branch.

3. The probability that a single branch has an instantaneous SNR less than some threshold γ is

$$\Pr[\gamma_i \leq \gamma] = \int_0^{\gamma} p(\gamma_i)\, d\gamma_i = \int_0^{\gamma} \frac{1}{\Gamma} e^{\frac{-\gamma_i}{\Gamma}}\, d\gamma_i = 1 - e^{-\frac{\gamma}{\Gamma}} \quad ...(3.21.3)$$

4. Now, the probability that all M independent diversity branches receive signals which are simultaneously less than some specific SNR threshold γ is

$$\Pr[\gamma_1, \ldots\ldots, \gamma_M \leq \gamma] = (1 - e^{-\gamma/\Gamma})^M = P_M(\gamma) \quad ...(3.21.4)$$

5. $P_M(\gamma)$ in eq. (3.21.4) is the probability of all branches failing to achieve an instantaneous $SNR = \gamma$. If a single branch achieves $SNR > \gamma$, then the probability that $SNR > \gamma$ for one or more branches is given by

$$\Pr[\gamma_i > \gamma] = 1 - P_M(\gamma) = 1 - (1 - e^{-\gamma/\Gamma})^M \quad ....(3.21.5)$$

Eq. (3.21.5) is an expression for the probability of exceeding a threshold when selection diversity is used.

6. For selection diversity, the average SNR is found by first computing the derivative of the CDF $P_M(\gamma)$ in order to find the pdf of γ. The instantaneous SNR when M branches are used.

$$P_M(\gamma) = \frac{d}{d\gamma} P_M(\gamma) = \frac{M}{\Gamma}(1 - e^{-\gamma/\Gamma})^{M-1} e^{-\gamma/\Gamma} \quad ...(3.21.6)$$

7. Then, the mean SNR, $\overline{\gamma}$, may be expressed as

$$\overline{\gamma} = \int_0^{\infty} \gamma p_M(\gamma) d\gamma = \Gamma \int_0^{\infty} Mx(1 - e^{-x})^{M-1} e^{-x}\, dx \quad ...(3.21.7)$$

where $x = \gamma/\Gamma$. Note that Γ is the average SNR for a single branch (when no diversity is used).

8. Eq. (3.21.7) is evaluated to yield the average SNR improvement offered by selection diversity

$$\frac{\overline{\gamma}}{\Gamma} = \sum_{k=1}^{M} \frac{1}{k} \quad ...(3.21.8)$$

Que 3.22. **Write a short note on spatial diversity in MIMO systems.**

Answer

1. In space diversity, a signal is transferred over several different propagation paths. In wireless transmission, it can be achieved by antenna diversity using multiple transmitter antennas (transmit diversity) and/ or multiple receiving antennas (receiver diversity).
2. Four different types of multi-antenna systems can be categorized based on diversity (input and output refer to the number of antennas):
   a. Single input, single output (SISO) – no diversity.
   b. Single input, multiple outputs (SIMO) – receive diversity.
   c. Multiple inputs, single output (MISO) – transmit diversity.
   d. Multiple inputs, multiple outputs (MIMO) - transmit-receive diversity.
3. The SISO system is very simple and deals with the communication between a transmitter and a receiver. In SISO, error probability is critically damaged by fading.
4. In a SIMO channel, the concept of MRC (maximal ratio combining) is offered as a way to exploit the receive diversity. The error probability achieved by MRC is to be much smaller than that corresponding to a SISO channel.
5. To perform MRC, the receiver has to know the fading, or in other words, the receiver has to have access to the CSI (channel state information). This is usually done by sending some known signal through the channel.
6. When there are l antenna elements in a mobile terminal and one base station antenna element, it makes a MISO channel. In this case, the CIR is an $l \times l$ matrix.
7. When there are l mobile terminals transmitting at a time and K base station antenna elements to receive them all, it makes a MIMO channel. In this case, the CIR is an $l \times k$ matrix that associates a transmission coefficient between each pair of antennas for each multipath component.
8. The MIMO channel brings together and takes full advantage of the transmit and receive diversity. The CSI may not be required in this channel. Given multiple antennas, the spatial dimension can be exploited to improve the BER performance of the wireless link and the data rate depending on the application.

Que 3.23. **Explain the multiplexing in MIMO system.**

AKTU 2018-19, Marks 10

Answer

1. MIMO with spatial multiplexing is a paradigm shift, dramatically changing the perceptions of and responses to multipath propagation (Fig. 3.23.1).

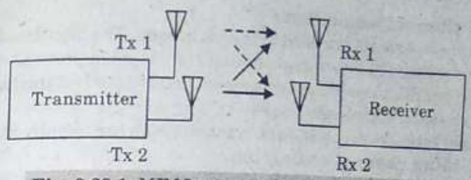

Fig. 3.23.1. MIMO using multiple transmitting and receiving antennas to send multiple signals over the same channel, multiplying spectral efficiency.

2. The information transmitted by both the antennas is different. The underlying mathematical nature of spatial multiplexed MIMO, where data is transmitted over a matrix rather than a vector channel, creates new and enormous opportunities beyond just the added diversity or array gain benefits the spectrum efficiency.
3. As such, this type of MIMO systems can be viewed as an extension of the smart antennas. A strong analogy can be made with CDMA transmission in which multiple user's share the same time of frequency channel, which are mixed upon transmission and recovered through their unique codes.
4. However, the advantage of MIMO is that the unique signatures of the input streams (virtual users) are provided by nature in a close to-orthogonal manner without frequency spreading and hence at no cost of spectrum efficiency.
5. Another advantage of MIMO is the ability to jointly code and decode multiple streams, as they are intended for the same use. The MIMO channel relies on the presence of a rich multipath, which is needed to make the channel spatially selective.

Que 3.24. **Explain in brief channel estimation techniques.**

Answer

1. Channel estimators help in equalization at the receiver end. They usually need some kind of information as a reference, in terms of known training sequences, pilots or some behavioural models or natural constraints.
2. Channel estimation techniques may be divided into three categories :

i. **CSI-based channel estimation :**

1. In this technique, pilots are sent with data symbols. The effects on the pilot symbols represent the CSI indirectly.
2. A pilot is basically a reference carrier/tone or a reference signal/symbol that is known at the receiver end in terms of position of sequence/pattern.
3. It has undergone the most recent channel behaviour along with the other carriers or symbols. Hence, it provides the CSI and is therefore purposely and systematically transmitted and used for channel estimation.

ii. Blind channel estimation :

1. Natural constraints are used in this technique. The likelihood function should be the model to pursue a completely blind approach.
2. In most of these cases, Gaussian assumptions are used for the transmitted data, channel, and received data.
3. Channel estimation is achieved by maximizing the likelihood function.

iii. Semi-blind channel estimation :

1. It is the combination of pilots and constraints. Time-variant and frequency-selective fading channels present a severe challenge to the designer of a wireless communication system.
2. A receiver plays a dual role to tackle this problem : phase correction by channel estimation or equalization and demodulation of the signal.
3. Several choices are possible for the implementation of a receiver depending on the modelling of the channel and the complexity invested in each task.
4. The estimated samples may have an estimation error that depends on the measurement noise, transmitted symbols, properties of the channel, estimation algorithm, and deviation from a time-invariant channel in the estimation interval.

PART-5

Equalization Techniques ; Transversal Filters, Adaptive Equalizers, Zero Forcing Equalizers, Decision Feedback Equalizers and Related Algorithms.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 3.25. **What do you mean by equalization ? Explain equalizer as inverse filter.**

OR

Show that an equalizer is an inverse filter of the channel.

AKTU 2015-16, Marks 05

Answer

A. Equalization :

1. This compensates for intersymbol interference (ISI) created by multipath within time dispersive channels.
2. If the modulation bandwidth exceeds the coherence bandwidth of the radio channel, ISI occurs and modulation pulses are spread in time into adjacent symbols.

3. An equalizer within a receiver compensates for the average range of expected channel amplitude and delay characteristics. Equalizers must be adaptive since the channel is generally unknown and time varying.

B. Equalizer as inverse filter :

1. If $x(t)$ is the original information signal, and $f(t)$ is the combined complex baseband impulse response of the transmitter, channel, and the RF/IF sections of the receiver, the signal received by the equalizer may be expressed as

$$y(t) = x(t) \otimes f^*(t) + n_b(t) \quad ...(3.25.1)$$

where $f^*(t)$ denotes the complex conjugate of $f(t)$
$n_b(t)$ is the baseband noise at the input of the equalizer
and $\otimes$ denotes the convolution operation.

2. If the impulse response of the equalizer is $h_{eq}(t)$, then the output of equalizer is

$$d(t) = x(t) \otimes f^*(t) \otimes h_{eq}(t) + n_b(t) \otimes h_{eq}(t) \quad ...(3.25.2)$$
$$= x(t) \otimes g(t) + n_b(t) \otimes h_{eq}(t)$$

where $g(t)$ is the combined impulse response of the transmitter, channel, RF/IF sections of the receiver, and the equalizer at the receiver.

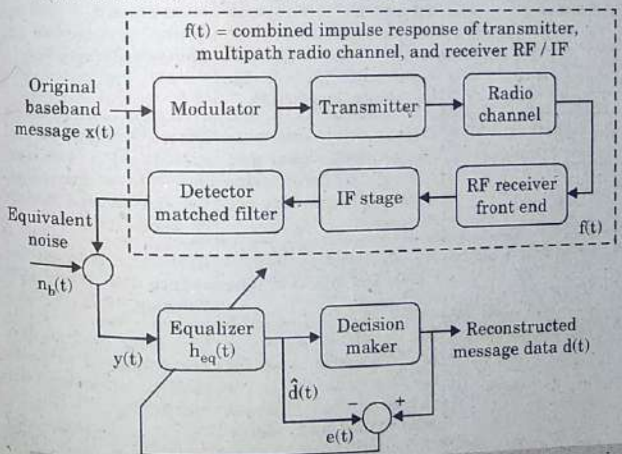

Fig. 3.25.1. Block diagram of a simplified communication's system using an adaptive equalizer at the receive.

3. The complex baseband impulse response of a transversal filter equalizer is given by

$$h_{eq}(t) = \sum_n c_n \delta(t - nT) \quad ...(3.25.3)$$

where, c_n are the complex filter coefficients of the equalizer. The desired output of the equalizer is $x(t)$, the original source data.

4. Assume that $n_b(t) = 0$. Then, in order to force $\hat{d}(t) = x(t)$ in eq. (3.25.2), $g(t)$ must be equal to

$$g(t) = f^*(t) \otimes h_{eq}(t) = \delta(t) \quad ...(3.25.4)$$

5. The goal of equalization is to satisfy eq. (3.25.4) so that the combination of the transmitter, channel and receiver appear to be an all-pass channel.
6. In the frequency domain, eq. (3.25.4) can be expressed as

$$H_{eq}(f)\, F^*(-f) = 1 \quad ...(3.25.5)$$

where, $H_{eq}(f)$ and $F(f)$ are Fourier transforms of $h_{eq}(t)$ and $f(t)$ respectively. Eq. (3.25.5) indicates that an equalizer is actually an inverse filter of the channel.

Que 3.26. **How equalization differs with diversity ?**

Answer

S.No.	Parameter	Equalization	Diversity
1.	Compensation factor	Equalization is used to compensate intersymbol interference (ISI) which exists when modulation bandwidth exceeds the coherence bandwidth.	Diversity is used to compensate fading which occurs due to multi-path reception.
2.	Improvement in performance	Equalization improves the received signal quality and link performance over small-scale time and distances.	Diversity improves the quality of a wireless link without increasing the transmitted power or bandwidth.
3.	Reduction factor	Equalization is used to reduce the effects of time dispersion.	Diversity is used to reduce the depth and duration of the fades.
4.	Requirement	An equalizer within a receiver compensates for the average range of expected channel amplitude and delay characteristics.	Diversity techniques are employed at both base station and mobile resources.

Que 3.27. **Draw the block diagram of survey of equalization and explain it.**

AKTU 2014-15, Marks 05

Answer

1. Equalization techniques can be subdivided into two general categories – linear and non-linear equalization.

2. These categories are determined from how the output of an adaptive equalizer is used for subsequent control (feedback) of the equalizer.
3. In general, the analog signal $\hat{d}(t)$ is processed by the decision making device in the receiver.
4. The decision maker determines the value of the digital data bit being received and applies a thresholding operation in order to determine the value of $d(t)$.
5. If $d(t)$ is not used in the feedback path to adapt the equalizer, the equalization is linear. On the other hand, if $d(t)$ is feedback to change the subsequent outputs of the equalizer, the equalization is non-linear.

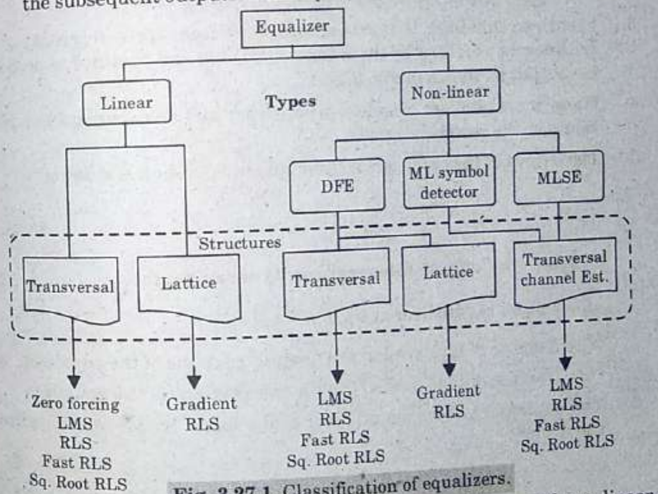

Fig. 3.27.1. Classification of equalizers.

6. Many filter structures are used to implement linear and non-linear equalizers. Further, for each structure, there are numerous algorithms used to adapt the equalizer.

Que 3.28. **Discuss in brief transversal filters.**

Answer

1. The most common equalizer structure is a linear transversal equalizer (LTE). The simplest LTE uses only feed forward taps and the transfer function of the equalizer filter is a polynomial in z^{-1}.
2. This filter has many zeros but poles only at $z = 0$, and is called a finite impulse response (FIR) filter, or simply a transversal filter.

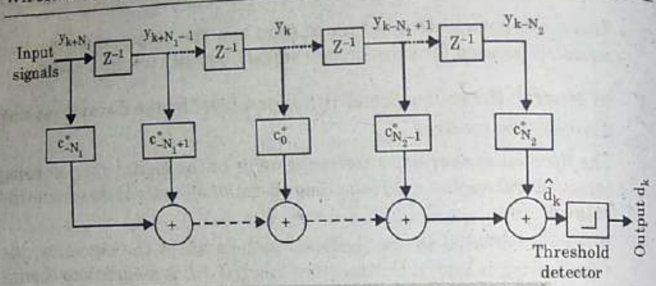

Fig. 3.28.1. Structure of linear transversal equalizer.

3. In such an equalizer, the current and past values of the received signal are linearly weighted by the filter coefficients and summed to produce the output as shown in Fig. 3.28.1.
4. These equalizers are simplest to construct and have current and past values of the received signals.
5. The output of the transversal filter before a decision is made is,

$$\hat{d}_k = \sum_{n=-N_1}^{N_2} (c_n^*)\, y_{k-n} \qquad ...(3.28.1)$$

where, c_n^* = Complex filter coefficients or tap weights

$\hat{d}_k$ = Output of equalizer at time index k

N_1 = Number of taps used in the forward portions of the equalizer

N_2 = Number of taps used in the reverse portions of the equalizer

6. The minimum mean squared error that a linear transversal equalizer can achieve

$$E[|e(n)|^2] = \frac{T}{2\pi}\int_{-\pi/T}^{\pi/T} \frac{N_0}{|F(e^{j\omega T})|^2 + N_0}\, d\omega \qquad ...(3.28.2)$$

where, $F(e^{j\omega T})$ = Frequency response of the channel

and N_0 = Noise power spectral density

Que 3.29. **Discuss in brief adaptive equalizer.**

Answer

1. An adaptive equalizer is a time-varying filter which must constantly be retuned. The basic structure of an adaptive equalizer is shown in Fig. 3.29.1, where the subscript k is used to denote a discrete time index.
2. There is a single input y_k into the equalizer at any time instant. The value of y_k depends upon the instantaneous state of the radio channel and the specific value of the noise. As such y_k is a random process.

3. The adaptive equalizer structure shown in Fig. 3.29.1 is called a transversal filter, and in the case has N delay elements, $N + 1$ taps, and $N + 1$ tunable complex multipliers, called weights.
4. The adaptive algorithm is controlled by the error signal e_k. The error signal is derived by comparing the output of the equalizer, $\hat{d}_k$ with some signal d_k which is either an exact scaled replica of the transmitted signal x_k or which represents a known property of the transmitted signal.
5. The adaptive algorithm uses e_k to minimize a cost function and updates the equalizer weights in a manner that iteratively reduces the cost function.

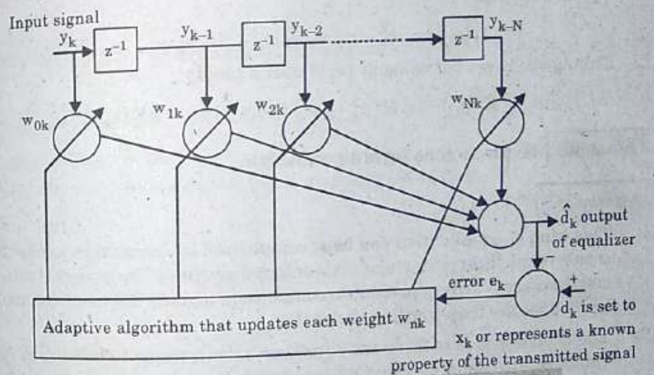

Fig. 3.29.1. A basic linear equalizer during training.

6. The most common cost function is the mean square error (MSE) between the desired signal and the output of the equalizer.
7. The MSE is denoted by $E[e(k)\, e^*(k)]$, and a known training sequence must be periodically transmitted when a replica of the transmitted signal is required at the output of the equalizer.
8. By detecting the training sequence, the adaptive algorithm in the receiver is able to compute and minimize the cost function by driving the tap weights until the next training sequence is sent.
9. Define the input signal to the equalizer as a vector y_k where

$$y_k = [y_k\, y_{k-1}\; y_{k-2} \cdots y_{k-N}]^T \qquad ...(3.29.1)$$

The output of the adaptive equalizer is a scalar given by

$$\hat{d}_k = \sum_{n=0}^{N} w_{nk}\, y_{k-n} \qquad ...(3.29.2)$$

a weight vector can be written as

$$w_k = [w_{0k}\quad w_{1k}\quad w_{2k}\quad \ldots\quad w_{NK}]^T \qquad ...(3.29.3)$$

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Using eq. (3.29.1) and (3.29.3), eq. (3.29.2) may be written in vector notation as

$$\hat{d}_k = y_k^T w_k = w_k^T y_k \quad ...(3.29.4)$$

10. It follows that when the desired equalizer output is known (*i.e.*, $d_k = x_k$), the error signal e_k is given by

$$e_k = d_k - \hat{d}_k = x_k - \hat{d}_k \quad ...(3.29.5)$$

and from eq. (3.29.4)

$$e_k = x_k - y_k^T w_k = x_k - w_k^T y_k \quad ...(3.29.6)$$

11. To compute the mean square error $|e_k|^2$ at time instant k, eq. (3.29.6) is squared to obtain

$$|e_k|^2 = x_k^2 + w_k^T y_k y_k^T w_k - 2x_k y_k^T w_k$$

12. Taking the expected value of $|e_k|^2$ over k yields

$$E\left[|e_k|^2\right] = E\left[x_k^2\right] + w_k^T E[y_k y_k^T] w_k - 2E[x_k y_k^T] w_k$$

Que 3.30. **Explain zero forcing equalizer.**

Answer

1. This kind of equalization can be accomplished by equalizers using the transversal filter structure encountered earlier. Transversal filter equalizers are easily adjustable to compensate against different channels or even slowly time-varying channels.
2. The design goal is to force the equalizer output pulse to have zero ISI values at the sampling (decision-making) instant.
3. In other words, the equalizer output pulses satisfy the Nyquist or the controlled ISI criterion.
4. The time delay T between successive taps is chosen to be T_b, the interval between pulses.
5. For a single pulse p_r at the input of the transversal filter with the tap setting the filter output $p_o(t)$ will be exactly $p_r(t - NT_b)$, that is, $p_r(t)$ delayed by NT_b.
6. For the Nyquist criterion, the output pulse $p_o(t)$ must have zero values at all the multiples of T_b.
7. The output $p_o(t)$ is the sum of pulses of the form $C_k p_r(t - kT_o)$. Thus,

$$p_o(t) = \sum_{n=-N}^{N} C_n p_r(t - nT_b)$$

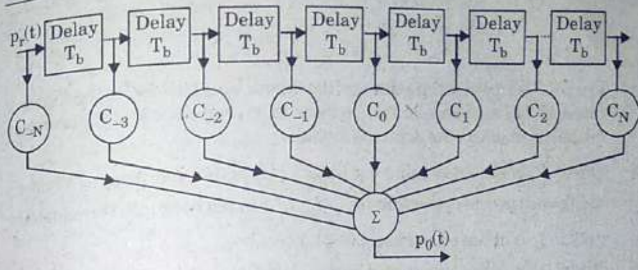

Fig. 3.30.1.

Que 3.31. **Explain adaptive equalization and decision feedback equalizer.**

AKTU 2018-19, Marks 10

OR

Explain the different types of equalization techniques used in wireless communication with support of mathematics and block diagram.

AKTU 2014-15, Marks 05

OR

Explain the different type of equalization techniques used in wireless communication.

AKTU 2016-17, Marks 15

Answer

Types of equalization techniques : Refer Q. 3.27, Page 3–33A, Unit-3.

i. **Linear (transversal) equalization :** Refer Q. 3.28, Page 3–34A, Unit-3.

ii. **Non-linear (Decision feedback) equalization :**

1. The basic idea behind decision feedback (DFE) equalization is that once an information symbol has been detected and decided upon, the ISI that it induces on future symbol can be estimated and subtracted out before detection of subsequent symbols.
2. The DFE can be realized in either the direct transversal form or as a lattice filter. The direct form is shown in Fig. 3.31.1.
3. It consists of a feed forward 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.
4. The equalizer has $N_1 + N_2 + 1$ taps in the feed forward filter and N_3 taps in the feedback filter, and its output can be expressed as

$$\hat{d}_k = \sum_{n=-N_1}^{N_2} c_n^* y_{k-n} + \sum_{i=1}^{N_3} F_i d_{k-i} \quad ...(3.31.1)$$

where c_n^* and y_n are tap gains and the inputs, respectively, to the forward filter, F_i^* are tap gains for the feedback filter, and $d_i (i < k)$ is the previous decisions made on the detected signal.

5. Once $\hat{d}_k$ is obtained using eq. (3.31.1), d_k is decided from it. Then d_k along with previous decisions $d_{k-1}, d_{k-2}, ...$ are fed back into the equalizer and $\hat{d}_{k+1}$ is obtained using eq. (3.31.1).
6. The minimum mean squared error a DFE can achieve, is

$$E[|e(n)|^2]_{min} = \exp\left\{\frac{T}{2\pi}\int_{-\pi/T}^{\pi/T} \ln\left[\frac{N_0}{|F(e^{j\omega T})|^2 + N_0}\right] d\omega\right\} \quad ...(3.33.2)$$

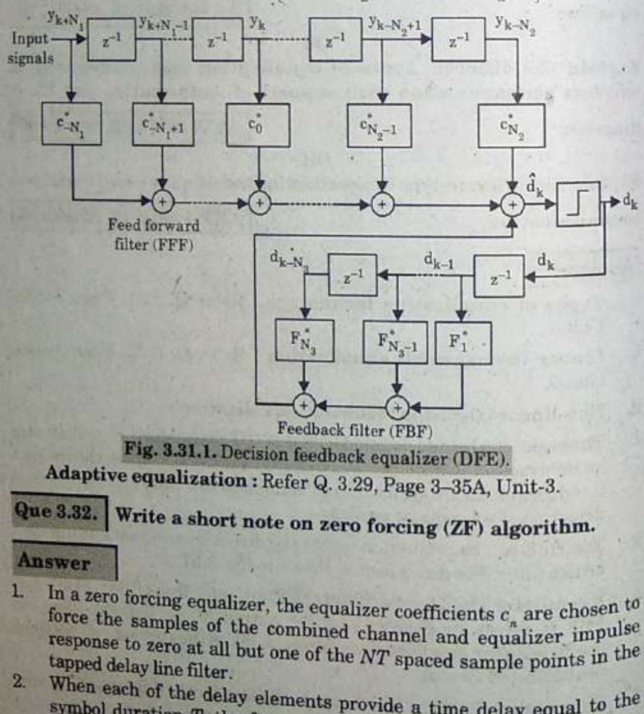

Fig. 3.31.1. Decision feedback equalizer (DFE).

Adaptive equalization : Refer Q. 3.29, Page 3–35A, Unit-3.

Que 3.32. **Write a short note on zero forcing (ZF) algorithm.**

Answer

1. In a zero forcing equalizer, the equalizer coefficients c_n are chosen to force the samples of the combined channel and equalizer impulse response to zero at all but one of the NT spaced sample points in the tapped delay line filter.
2. When each of the delay elements provide a time delay equal to the symbol duration T, the frequency response $H_{eq}(f)$ of the equalizer is periodic with a period equal to the symbol rate $1/T$.

3. According to Nyquist criterion, the combined response of the channel with the equalizer must be

$$H_{ch}(f)\, H_{eq}(f) = 1 \text{ as } |f| < 1/2T \quad ...(3.32.1)$$

Where, $H_{ch}(f)$ is the folded frequency response of the channel.

4. So an infinite length, zero ISI equalizer is simply an inverse filter which inverts the folded frequency of the channel.

Que 3.33. **Write short notes on :**

A. Least squares (LS) algorithm.
B. Least mean square (LMS) algorithm.

Answer

A. Least squares algorithms :

1. Least squares (LS) means that the overall solution minimizes the sum of the squares of the errors made in the results of every single equation.
2. The method of LS channel estimation is a standard approach to the approximate solution of over determined systems, that is, sets of equations in which there are more equations than unknowns.
3. The LS regression process can be understood with the help of the following mathematics.
4. Consider a noise-corrupted communication system through a fading multipath channel h, after which the signal has memory of L symbols.
5. Moreover, n is the white Gaussian noise, which is sampled at the symbol rate.
6. The demodulation problem here is to detect the transmitted bits x from the received signal y, and hence; the estimation procedure is required. For a general linear equation of the received signal,

$$y = hx + n$$

7. The complex CIR h during L training bits can be expressed as

$$h = [h_0\, h_1 \ldots h_L]^T$$

8. Within each transmission burst, the transmitter sends a unique training sequence, which is divided into a reference length of P and guard period of L bits and is denoted by

$$m = [m_0\, m_1 \ldots m_{p+L-1}]^T$$

having bipolar elements $m_i \in \{-1, +1\}$.

9. For estimation during the training period, the equation for the received signal can be rewritten as $y = hm + n$. The circulant training sequence matrix can be formed as

$$M = \begin{bmatrix} m_L & \cdots & m_1 & m_0 \\ m_{L+1} & \cdots & m_2 & m_1 \\ \vdots & \cdots & \vdots & \vdots \\ m_{L+P+1} & \cdots & m_P & m_{P-1} \end{bmatrix}$$

10. The vertical deviation can be calculated using the following formula :

$$d_i = y_i - y = y_i - (hm_i + n)$$

11. If the square of the deviations is minimized, the best line can be calculated as

$$d_i^2 = (y_i - y)^2 = (y_i - hm_i + n)^2$$

In other way,

$$\hat{H} = \arg_h \min \|y - Mh\|^2$$

B. Least mean squares algorithms :

1. Least mean squares (LMS) algorithms are a class of adaptive filters used to mimic a desired filter by finding the filter coefficients that relate to producing the LMS of the error signal (difference between the desired and actual signals).
2. It is a stochastic gradient descent method in which the filter is adapted based on the error at the current time.
3. A simplified diagram explaining the algorithm is shown in Fig. 3.33.1.

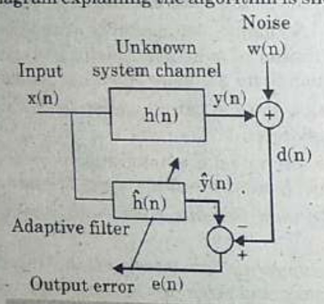

Fig. 3.33.1. LSM formulation diagram.

4. The following equations represent formulation for the algorithm :

$$h^H(n) = [h_0^*(n), h_1^*(n), ..., h_{p-1}^*(n)]$$

(Hermitian transpose or conjugate transpose)

$$y(n) = h^H(n) \cdot x(n)$$
$$d(n) = y(n) + w(n)$$
$$e(n) = d(n) - \hat{y}(n) = d(n) - \hat{h}^H(x) \cdot x(n)$$

5. The basic idea behind the LMS filter is to approach the optimum filter weights by updating the filter weights in a manner to converge to the optimum value.
6. The algorithm starts by assuming a small weight (zero in most cases). At each step, the weights are updated by finding the gradient of the mean square error (MSE).
7. That is, if the MSE gradient is positive, it implies that the error will keep increasing positively if the same weight is used for further iterations; hence, we need to reduce the weights.

8. In the same way, if the gradient is negative, we need to increase the weights. Therefore, the basic weight update equation is

$$W_n + 1 = W_n - \mu\Delta\varepsilon(n)$$

where ε represents the MSE and μ is the step size or the gradient, W_n.

9. A negative value indicates that we need to change the weights in a direction opposite to that of the gradient slope.
10. The MSE as a function of filter weights is a quadratic function; this means that it has only one extreme value that minimizes the MSE, which is the optimum weight. Finally, we may write

$$\hat{h}(n+1) = \hat{h}(n) + \mu e^*(n) \cdot x(n)$$

VERY IMPORTANT QUESTIONS

Following questions are very important. These questions may be asked in your SESSIONALS as well as UNIVERSITY EXAMINATION.

Q. 1. What is the basic mechanism of vocoder and explain any two type of vocoders ?

Ans. Refer Q. 3.2.

Q. 2. What is PN sequence ? Draw suitable PN sequence generator and prove the properties of PN sequence.

Ans. Refer Q. 3.5.

Q. 3. With the help of block diagram and suitable expressions explain the generation and reception of direct sequence spread spectrum (DSSS) signal using BPSK modulation.

Ans. Refer Q. 3.7.

Q. 4. Explain various pulse shaping techniques for zero intersymbol interference.

Ans. Refer Q. 3.17.

Q. 5. Explain the different type of diversity techniques used in wireless communication system.

Ans. Refer Q. 3.19.

Q. 6. Derive an expression for selection diversity improvement in terms of probability of receiving signal using single branch or using *M* branches.

Ans. Refer Q. 3.21.

Q. 7. What do you mean by equalization ? Explain equalizer as inverse filter.

Ans. Refer Q. 3.25.

Q. 8. Discuss in brief adaptive equalizer.

Ans. Refer Q. 3.29.

Q. 9. Explain adaptive equalization and decision feedback equalizer.

Ans. Refer Q. 3.31.

Q. 10. Write short notes on :

A. Least squares (LS) algorithm.

B. Least mean square (LMS) algorithm.

Ans. Refer Q. 3.33.

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4 UNIT

Multiplexing and Multiple Access

CONTENTS

PART-1

Multiplexing and Multiple Access : FDMA, TDMA, CDMA.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 4.1. **Discuss multiplexing schemes.**

Answer

1. Multiplexing schemes are used when signals from multiple users are to be combined and sent on a single channel as a single input stream.
2. Multiplexing is used to enable several users share a medium with minimum or no interference.
3. There are mainly four schemes for multiplexing :

i. Frequency division multiplexing (FDM) :

In FDM, individual users are provided individual channels, which will in combination make the whole transmission bandwidth.

ii. Time division multiplexing (TDM) :

1. In TDM, each individual user is pre-assign a time slot in which he or she can send the information, and once that slot over, the slot for the next user will start.
2. For n users, the bit rate of the TDM stream will increase n-fold. Time synchronization is a very important issue in TDM. There are two methods of TDM : synchronous and asynchronous.

iii. Code division multiplexing (CDM) :

1. In code division multiplexing, separation is achieved by assigning each user channel its own code.
2. Guard spaces are realized by using codes with the necessary distance in code space.
3. Good protection against unauthorized reception is the main advantage of CDM.

iv. Space division multiplexing (SDM) :

In SDM, signals can be transmitted by different directional antennas, or the signals received by a multidimensional antenna can be combined to get all of them back.

Que 4.2. **Discuss multiple access schemes.**

Answer

1. Multiple access schemes allow many simultaneous users to share the same available channel bandwidth or radio spectrum on an individual basis.

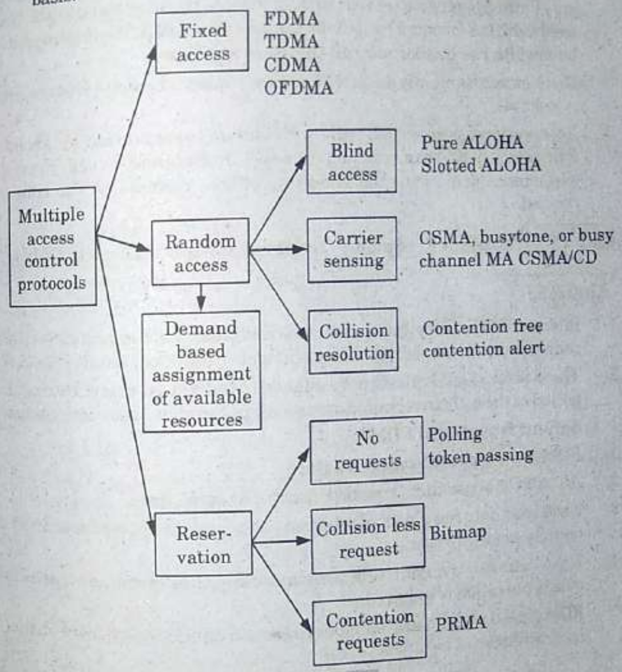


Fig. 4.2.1.

2. Multiple access may be achieved by four different ways as follows :
   a. Fixed assignment of resources in terms of carrier allotment, time slot allocation, code allocation, or area allocation to specific users.
   b. Demand assignment.
   c. Random access, *i.e.*, a dynamic assignment of spectrum resources in time or bandwidth to the users, according to their needs or on the basis of demand.
   d. Reservation-based access, where prior reservations intimate other users about the request of a particular user.
3. A detailed classification of the important multiple access schemes is given in Fig. 4.2.1.

4. Frequency division multiple access (FDMA), time division multiple access (TDMA), code division multiple access (CDMA), and space division multiple access (SDMA) are the four major methods of multiple access be fixed assignment of resources to the users.
5. In FDMA, it is not possible for each user to use the entire bandwidth and only limited bandwidth is allocated. In TDMA, the user can use the total bandwidth but for only a limited slot duration. In CDMA, the total available bandwidth can be shared by all the users at a time.
6. Random access mainly deals with packet radio and mostly storage data is utilized.
7. Reservation-based access deals with channel reservation in advance whenever data transmission is needed. Demand-based channel assignment deals with the allocation of free channel at the time of request.

Que 4.3. **Explain Frequency Division Multiple Access (FDMA).**

Answer

1. In case of frequency division multiple access, the complete available radio spectrum is subdivided into channels (frequency band) 1, 2, ..., N.
2. These individual channels are assigned to individual users. During the period of the call conversation, no two users can share the same channel.

Salient features of FDMA :

1. FDMA systems are analog in nature.
2. All users can use their assigned channel at same time.
3. Each user only has a part of spectrum as complete spectrum is divided equally among N users.
4. Data can be transmitted continuously therefore less need of synchronization is required.
5. FDMA need to use costly bandpass filters to eliminate adjacent channel interference.

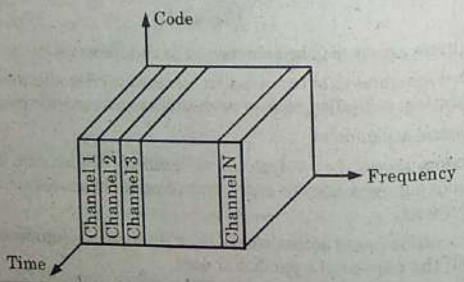


Fig. 4.3.1. Representation of FDMA (N users having N frequency bands).

Que 4.4. **How many numbers of channels can be supported in a FDMA ? Explain efficiency of FDMA.**

Answer

The number of channels that can be simultaneously supported in a FDMA system is given by

$$N = \frac{B_t - 2B_{guard}}{B_c} \quad ...(4.4.1)$$

where, B_t = Total spectrum allocated

B_{guard} = Guard band allocated at the edge of the allocated spectrum band

B_c = Channel bandwidth.

Efficiency of FDMA :

1. The efficiency of FDMA system is a measure of the percentage of transmitted data that contains information as opposed to providing overhead for the access scheme.
2. The frame efficiency η_f is the percentage of bits per frame which contain transmitted data.

$$\eta_f = \frac{\text{Bandwidth available for data transmission}}{\text{System bandwidth}} \quad ...(4.4.2)$$

From eq. (4.4.1), $B_t = NB_c + 2B_{guard}$

3. Let $N_{control}$ be the number of allocated channels and N_{data} be the number of data channels in the system. Then the total number of available channels will be

$$N = N_{data} + N_{control} \quad ...(4.4.3)$$

Substituting eq. (4.4.3) into eq. (4.4.2), we get

$$B_t = N_{data} B_c + N_{control} B_c + 2B_{guard}$$

Then $N_{data} B_c < B_t$

4. $N_{data} B_c$ is the total bandwidth available for data transmission.

So frame efficiency,

$$\eta_f = \frac{N_{data} B_c}{B_t} < 1$$

Que 4.5. **Write short note on TDMA.**

Answer

1. In case of TDMA, the complete radio (wireless) channel is divided into time slots $t_1, t_2, ..., t_N$. Each time slot is assigned to one user and during this period, the user is allowed to use complete radio spectrum.
2. As the radio spectrum is not divided into frequency slots but divided in terms of time slots therefore only one user shares the base station antennas at one time. This reduces the interference.

3. To transmit the data in TDMA scheme, a 'frame' is being formed that consist of a number of slots. Each frame is cyclically repeated over time.
4. Fig. 4.5.1 shows the TDMA frame structure and each slot contents.

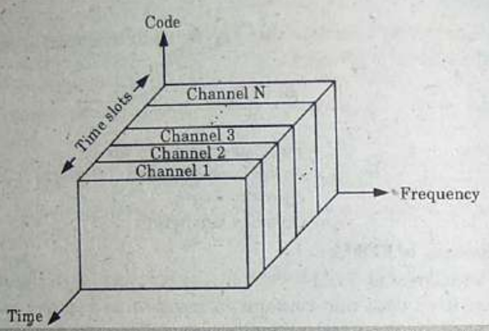


Fig. 4.5.1. Representation of TDMA (N users having different time slots).

5. Here, preamble contains the address of base station and subscriber and synchronization information.
6. Guard time is allocated to make the separation between different frames and time slots.

Salient Features of TDMA :

1. TDMA systems are digital in nature.
2. Many users access whole radio spectrum at different time slots one by one.
3. Due to time division multiplexing the data transmission is not continuous.
4. Duplexers are not required because TDMA uses separate time slots for transmission and reception.
5. Power consumption is low. Since the transmitter can be turned off in idle state.
6. Each user utilizes whole bandwidth.

Que 4.6. **Explain frame efficiency in a TDMA system. How many number of channel slots can be provided in a TDMA mobile system ?**

Answer

1. The efficiency of a TDMA system is a measure of the percentage of transmitted data that contained information as opposed to providing overhead for the access scheme.
2. The frame efficiency, η_f is the percentage of bits per frame which contain transmitted data.

3. The number of overhead bits per frame is

$$b_{OH} = N_r b_r + N_t b_p + N_t b_g + N_r b_g$$

where N_r is the number of reference bursts per frame
N_t is the number of traffic bursts per frame
b_r is the number of overhead bits per reference burst
b_p is the number of overhead bits per preamble in each slot
and b_g is the number of equivalent bits in each guard time interval.

4. The total number of bits per frame b_T is

$$b_T = T_f R$$

where T_f is the frame duration.
and R is the channel bit rate.

5. The frame efficiency η_f is thus given as

$$\eta_f = \left(1 - \frac{b_{OH}}{b_T}\right) \times 100\,\%$$

6. The number of TDMA channels slots that can be provided in a TDMA system is found by multiplying the number of TDMA slots per channel by the number of channels available and is given by

$$N = \frac{m(B_t - 2B_G)}{B_c}$$

where, m is the number of TDMA of users supported on each radio channel
B_t is the total spectrum allocation
B_G is the guard band allocated at the edge of the allocated spectrum band
B_c is the channel bandwidth.

Que 4.7. **Consider a GSM system which is a TDMA/FDD system that uses 20 MHz for forward link which is broken into radio channels and if no guard band is assumed, find the number of simultaneous users that can be accommodated in GSM.**

AKTU 2015-16, Marks 7.5

Answer

The number of simultaneous users that can be accommodated in GSM is given as

$$N = \frac{20\text{ MHz}}{(200\text{ KHz})/8} = 800$$

Thus, GSM can accommodate 800 simultaneous users.

Que 4.8. Write short note on CDMA.

OR

Explain the different type of multiple access schemes (TDMA, CDMA and FDMA). **AKTU 2018-19, Marks 10**

OR

Explain the frequency hopped multiple access and code division multiple access technology. **AKTU 2016-17, Marks 10**

Answer

Types of multiple access scheme :

i. **FDMA :** Refer Q. 4.3, Page 4–4A, Unit-4.

ii. **TDMA :** Refer Q. 4.5, Page 4–6A, Unit-4.

iii. **CDMA :**

1. Code division multiple access (CDMA) is wideband spread spectrum technology. A unique code is assigned to all speech bits (signals).
2. Signals for all calls are spread across a broad frequency spectrum, hence the term 'spread spectrum'.
3. This technique allows numerous users to transmit simultaneously on one radio frequency. As a result, CDMA systems increase the system capacity by 10 to 30 times over conventional cellular systems.
4. In CDMA technology, during transmission process the sound of the user's voice is converted into a digital code.
5. This digital signal is correlated with a code known as pseudorandom noise (PN) code (also called 'Walsh' code).
6. Each user has its own pseudorandom codeword which is approximately orthogonal to all other code words.

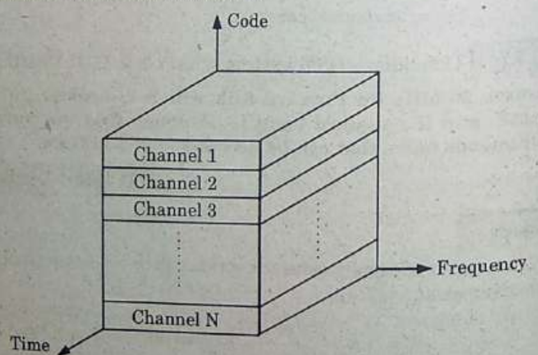


Fig. 4.8.1. Representation of CDMA (*N* users having *N* different codes).

7. The output of the correlator is an encrypted signal which is then spread over a very wide frequency spectrum (1.25 MHz).
8. At the receiving terminal, the spread spectrum signal is demodulated back to a narrow bandwidth using 'decorrelator'.
9. Decorrelator uses the same PN code for detection of the signal that was used in transmission of the signal.
10. A signal correlated with a give PN code and decorrelated with the same PN code returns the original signal.

Salient features of CDMA

1. **Frequency diversity :** Many users can share the same frequency spectrum. This technique reduces multipath fading because the signal is spread over a large spectrum.
2. **Soft hand off :** With soft hand off the wireless call is actually carried by two or more cells simultaneously. It is just like a make-before-break fashion in relation to cell hand off. It may be described as the mobile acquires the new cell before it relinquishes the old.
3. **Near-far problem :** It occurs at a CDMA receiver *i.e.*, signals closer to the receiver are received with less attenuation than signals farther away.
4. **Self-jamming :** It is a big problem in CDMA technique *i.e.*, unless all of the mobile users are perfectly synchronized, the arriving transmission from multiple users will not be perfectly aligned on chip boundaries. Thus the spreading sequence of different users is not orthogonal and there is some level of cross-correlation. This will affect the receiver decision.
5. **Privacy :** CDMA has very high privacy because spread spectrum is obtained by the use of code signals *i.e.*, different for each other.

Frequency hopped multiple access :

1. Frequency hopped multiple access (FHMA) is a digital multiple access system in which the carrier frequencies of the individual users are varied in a pseudorandom fashion within a wideband channel.
2. FHMA allows multiple users to simultaneously occupy the same spectrum at the same time, where each user dwells at a specific narrowband channel at a particular instance of time, based on the particular PN code of the user.
3. The digital data of each user is broken into uniform sized bursts which are transmitted on different channels within the allocated spectrum band.
4. The instantaneous bandwidth of any one transmission burst is much smaller than the total spread bandwidth.
5. The pseudorandom change of the channel frequencies of the user randomizes the occupancy of a specific channel at any given time, thereby allowing for multiple access over a wide range of frequencies.

6. If the rate of change of the carrier frequency is greater than the symbol rate, then the system is referred to as a fast frequency hopping system.
7. If the channel changes at a rate less than or equal to the symbol rate, it is called slow frequency hopping.
8. A frequency hopped system provides a level of security, especially when a large number of channels are used, since an unintended (or an intercepting) receiver that does not know the pseudorandom sequence of frequency slots must retune rapidly to search for the signal it wishes to intercept.

Que 4.9. **Find out an expression for calculating the number of users in a CDMA cellular system.**

Answer

1. For evaluation of CDMA system capacity, let the total number of users are N then the cell base station receives a composite waveform containing the one desired signal of power S and $(N - 1)$ interfering users (each having power, S).
2. Thus, signal to noise ratio (S/N) is given as

$$S/N = \frac{\text{Desired signal power}}{\text{Number of interfering users} \times \text{Desired signal power}}$$

$$= \frac{S}{(N-1).S} = \frac{1}{(N-1)} \quad ...(4.9.1)$$

3. Now, other important parameter (along with SNR) in communication system is bit energy-to-noise ratio (E_b/N_o).

$$\frac{E_b}{N_o} = \frac{\text{Signal power/ baseband information bit rate}}{\text{Interference power/ Total RF bandwidth}}$$

$$\frac{E_b}{N_o} = \frac{S/R}{(N-1)(S/W)} = \frac{W/R}{(N-1)} \quad ...(4.9.2)$$

4. In the spread spectrum bandwidth, the background thermal noise (η) is also accountable.

So,
$$\frac{E_b}{N_o} = \frac{W/R}{(N-1)+(\eta/S)} \quad ...(4.9.3)$$

5. After solving eq. (4.9.3), the number of users able to access the system is given by

$$N = 1 + \frac{(W/R)}{(E_b/N_o)} - (\eta/S) \quad ...(4.9.4)$$

Here, $W/R \rightarrow$ Known as processing gain
$E_b/N_0 \rightarrow$ Bit energy to noise ratio
$\eta \rightarrow$ Thermal noise

6. In order to achieve an increase in capacity, the interference should be minimized. This can be done by antenna sectorization, switching-off the transmitter power during silence (or no voice activity).

PART-2

OFDMA, SC-FDMA, IDMA, Schemes and Hybrid Method of Multiple Access Schemes, Rake Receiver.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 4.10. **Write a short note on OFDMA.**

Answer

1. Orthogonal frequency division multiple access (OFDMA) is a hybrid multiple access or multiplexing technique with multicarrier modulation, which divides the available spectrum into many carriers, each one being modulated by a low-rate data stream.
2. In OFDMA, information of different users is processed in combination and then allocated to multiple carriers, whereas in OFDMA, out of the total available bandwidth, each narrow channel can be accessed by individual users.
3. Though OFDMA is similar to FDMA, it uses the spectrum much more efficiently by spacing the channels much closer together.
4. This is achieved by making all the carriers orthogonal to one another, thereby preventing interference between the closely spaced carriers.

Que 4.11. **Write short notes on SC-FDMA.**

Answer

1. Single-carrier FDMA (SC-FDMA) is a frequency division multiple access scheme. It is also called linearly precoded OFDMA (LP-OFDMA).
2. Like other multiple access schemes (TDMA, FDMA, CDMA, OFDMA), it deals with the assignment of multiple users to a shared communication resource.
3. SC-FDMA can be interpreted as a linearly precoded OFDMA scheme, in the sense that it has an additional DFT processing step preceding the conventional OFDMA processing.
4. SC-FDMA has drawn great attention as an attractive alternative to OFDMA, especially in the uplink communications where lower peak-to-average power ratio (PAPR) greatly benefits the mobile terminal in

terms of transmit power efficiency and reduced cost of the power amplifier.

5. It has been adopted as the uplink multiple access scheme in 3GPP long term evolution (LTE), or evolved UTRA (E-UTRA).
6. SC-FDMA's advantage of low PAPR makes it desirable for uplink wireless transmission in mobile communication systems, where transmitter power efficiency is of paramount importance.

Transmitter and receiver structure of LP-OFDMA/SC-FDMA :

1. The transmission processing of SC-FDMA is very similar to that of OFDMA. For each user, the sequence of bits transmitted is mapped to a complex constellation of symbols.
2. Then different transmitters (users) are assigned different Fourier coefficients. This assignment is carried out in the mapping and demapping blocks.
3. The receiver side includes one demapping block, one IDFT block, and one detection block for each user signal to be received.
4. Just like in OFDM, guard intervals (called cyclic prefixes) with cyclic repetition are introduced between blocks of symbols in view to efficiently eliminate inter-symbol interference from time spreading (caused by multi-path propagation) among the blocks.
5. In SC-FDMA, multiple access among users is made possible by assigning different users different sets of non-overlapping Fourier coefficients (sub-carriers). This is achieved at the transmitter by inserting (prior to IDFT) silent Fourier coefficients (at positions assigned to other users), and removing them on the receiver side after the DFT.
6. The distinguishing feature of SC-FDMA is that it leads to a single-carrier transmit signal, in contrast to OFDMA which is a multi-carrier transmission scheme. Its transmit signal has a lower peak-to-average power ratio (PAPR), resulting in relaxed design parameters in the transmit path of a subscriber unit.

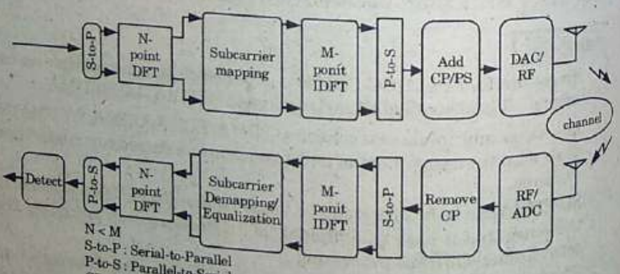


Fig. 4.11.1.

7. Equalization is achieved on the receiver side, after the DFT calculation, by multiplying each Fourier coefficient by a complex number. Thus, frequency-selective fading and phase distortion can be easily counteracted.
8. The advantage is that frequency domain equalization using FFTs requires less computation than conventional time-domain equalization be employed, followed by the IDFT operation.

Que 4.12. **Explain IDMA scheme.**

Answer

1. Interleave-division multiple-access (IDMA) is a recently proposed multi-access scheme, in which users are distinguished by different interleaving patterns.
2. An interleaver is used as a component of a channel encoder to enhance the coding gain, or as a channel interleaver to combat the time/frequency coherent fading by scrambling burst errors into random errors.
3. The block diagram of IDMA scheme is shown in Fig. 4.12.1 for *K* users. The principle of iterative multi user detection (MUD) which is a promising technique for multiple access problems (MAI) is also illustrated in the lower part of Fig. 4.12.1.

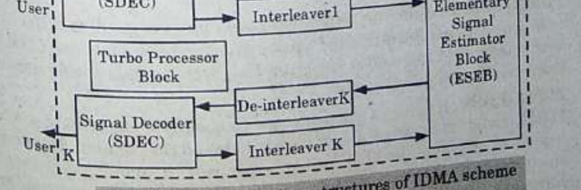


Fig. 4.12.1. Transceiver structures of IDMA scheme with *K* simultaneous users.

4. The turbo processor involves elementary signal estimator block (ESEB) and a bank of K decoders (SDECs). The ESEB partially resolves MAI without considering FEC coding.
5. The outputs of the ESEB are then passed to the SDECs for further refinement using the FEC coding constraint through de-interleaving block.
6. The SDECs outputs are fed back to the ESEB to improve its estimates in the next iteration with proper user specific interleaving. This iterative procedure is repeated a preset number of times (or terminated if a certain stopping criterion is fulfilled).
7. After the final iteration, the SDECs produce hard decisions on the information bits. The complexity involved (mainly for solving a size $K \times K$ correlation matrix) is $0(K^2)$ per user by the well-known iterative minimum mean square error (MMSE) technique in CDMA, while in IDMA, it is independent of user.

Que 4.13. **Discuss hybrid methods of multiple access.**

OR

Explain SC-FDMA, IDMA schemes and hybrid method of multiple access shcemes.

AKTU 2017-18, Marks 10

Answer

A. SC-FDMA : Refer Q. 4.11, Page 4–12A, Unit-4.

B. IDMA : Refer Q. 4.12, Page 4–13A, Unit-4.

C. Hybrid Methods of Multiple Access :

The following are a few combinations of multiple access technique :

1. **FDMA-CDMA :** The spectrum is divided into channels and each channel is a narrowband CDMA system with processing gain lower than the original CDMA system.
2. **DSSS-FHSS :** The direct sequence modulates the signal and hops centre frequency using a pseudo-random hopping pattern. The method avoids near-far effect.
3. **TDMA-CDMA :** Different spreading codes are assigned to different cells. One user per cell is allotted a particular time slot. Only one CDMA user transmits in each cell at any given time. The method avoids severe fades on the channel.
4. **TDMA-FHSS :** It involves a hop to a new frequency at the start of a new TDMA frame. The method avoids severe fades on the channel. Hopping sequences are predefined and unique per cell. It avoids co-channel interference if other base stations transmit on different frequencies at different times.

Que 4.14. **Draw and explain the rake receiver in detail using proper block diagram.**

AKTU 2016-17, Marks 10

OR

Explain the structure of RAKE receiver with the help of neat diagram. What is *M* branch RAKE receiver ?

AKTU 2017-18, Marks 10

OR

Draw and explain RAKE receiver using block diagram.

AKTU 2018-19, Marks 10

Answer

1. There are four RAKE receivers within base transceiver and three RAKE receivers within each mobile phone. These are so called because they resemble a lawn rake.
2. The main function of the RAKE receiver at both ends (mobile and base station transceivers) is to aggregate the diversity received signals.
3. The direct signal at the RAKE receiver is the strongest signal that is combined with multipath reflected signal from the other two or three RAKE receivers to form the composite signals that are used to process the mobile call.
4. The multipath signals are additives to the direct signals to obtain the cleanest, strongest signal possible.
5. Therefore, we can conclude that the signal to noise ratio (SNR) gets improved by using RAKE receivers. The RAKE receiver circuitry is shown in Fig. 4.14.1.

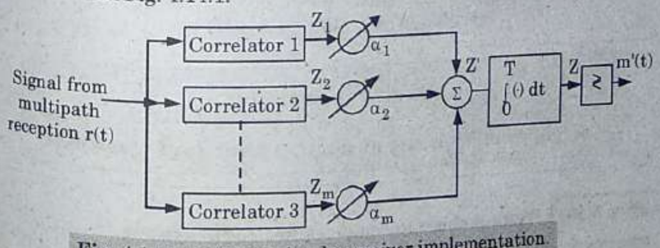


Fig. 4.14.1. A M-branch rake receiver implementation.

6. A RAKE receiver performs its functions in following steps :
   i. It collects the time shifted version of the original signal by providing a separate correlation receiver for each of the multipath signals.
   ii. The range of time delays that a particular correlator can search is called a search window.
   iii. A RAKE receiver having '*M*' correlators separately detect '*M*' strongest multipath components.

iv. The output of each correlator is then weighted to provide a better estimate of the transmitted signals. This is done in weighting network that provide a linear combination of the correlator output.

v. Decisions based on the combination of the 'M' separate decision statistics offered by the RAKE receiver provide a form of diversity which can overcome fading and thereby improve CDMA reception.

Working detail of a *M*-branch RAKE receiver :

1. A 'M branch' RAKE receiver is shown in Fig. 4.14.1. Here $r(t)$ is the input signal after multipath reception which is given to 'M' correlators. The output of 'M' correlators are denoted by $Z_1, Z_2, \ldots, Z_M$. These output are weighted by $\alpha_1, \alpha_2, \ldots \alpha_M$ respectively. The weighting coefficients are based on the power or the SNR from each correlator output.
2. Note that if SNR (or power) is small then it will be assigned a small weighting factor. The overall signal Z' (after combination and weightage) is given by

$$Z' = \sum_{m=1}^{M} \alpha_m Z_m \qquad \ldots(4.14.1)$$

where α_m are weighting coefficients which are normalized to the output signal power of the correlator in such a way that the coefficients sum to unity.

$$\alpha_M = \frac{Z_m^2}{\sum_{m=1}^{M} Z_m^2} \qquad \ldots(4.14.2)$$

PART-3

Multiple Access for RadioPacket System : Pure ALOHA, Slotted ALOHA.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 4.15. **Write a short note on packet radio system.**

Answer

1. In packet radio (PR) access technique, many subscribers attempt to access a single channel in an uncoordinated (or minimally coordinated) manner. Transmission is done by using bursts of data.
2. Collisions from the simultaneous transmissions of multiple transmitters are detected at the base station receiver, in which case an ACK or NACK signal is broadcast by the base station to alert the desired user of received transmission.
3. The ACK signal indicates an acknowledgement of a received burst from a particular user by the base station, and a NACK (negative acknowledgement) indicates that the previous burst was not received correctly by the base station.
4. By using ACK and NACK signals a PR system employs perfect feedback, even though traffic delay due to collisions may be high.
5. Packet radio multiple access is very easy to implement, but has low spectral efficiency and may induce delays. The subscribers use a contention technique to transmit on a common channel.
6. ALOHA protocols are the best examples of contention techniques. ALOHA allows each subscriber to transmit whenever they have data to send.
7. The transmitter subscribers listen to the acknowledgement feedback to determine if transmission has been successful or not.

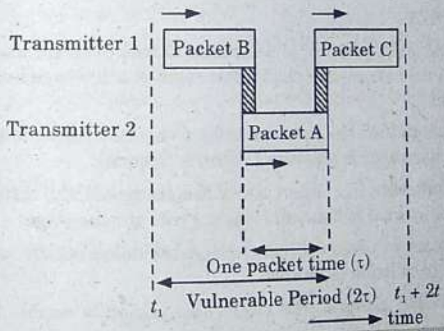


Fig. 4.15.1. Vulnerable period for a packet using the ALOHA protocol.

8. If a collision occurs, the subscriber waits a random amount of time and then retransmits the packet.
9. Packet A will collide with packets B and C because of overlap in transmission time.
10. In order to determine the throughput, it is important to determine the vulnerable period, V_p, which is defined as the time interval during which the packets are susceptible to collisions with transmissions from other users.
11. Fig. 4.15.1 shows the vulnerable period for a packet during the period t_1 to $t_1 + 2\tau$.
12. If τ is the packet duration in seconds, then the traffic occupancy or throughput R of a packet radio network is given by

$$R = \lambda\tau \qquad \ldots(4.15.1)$$

where λ = Mean arrival rate

13. The probability that n packets are generated by the user population during a given packet duration interval is assumed to be Poisson distributed and is given as

$$Pr(n) = \frac{R^n e^{-R}}{n!} \quad ...(4.15.2)$$

14. The probability that zero packets are generated (*i.e.*, no collision) during this interval is given by

$$Pr(0) = e^{-R} \quad ...(4.15.3)$$

Que 4.16. **Explain pure ALOHA and slotted ALOHA.**

OR

Define the term ALOHA. Explain Pure and Slotted ALOHA.

AKTU 2017-18, Marks 10

Answer

Pure ALOHA :

1. The pure ALOHA protocol is a random access protocol used for data transfer. A user accesses a channel as soon as a message is ready to be transmitted.
2. After a transmission the user waits for an acknowledgement on either the same channel or a separate feedback channel.
3. In case of collisions, (*i.e.*, when a NACK is received), the terminal waits for a random period of time and retransmits the message.
4. As the number of users increase, a greater delay occurs because the probability of collision increases.
5. For the ALOHA protocol, the vulnerable period is double the packet duration.
6. Thus, the probability of no collision during the interval of 2τ is found by evaluating $Pr(n)$ given as

$$Pr(n) = \frac{(2R)^n e^{-2R}}{n!} \text{ at } n = 0 \quad ...(4.16.1)$$

7. The probability of no collision is $Pr(0) = e^{-2R}$. The throughput of the ALOHA protocol is given as

$$T = Re^{-2R} \quad ...(4.16.2)$$

8. The maximum through put achieved by using the ALOHA protocol is given by

$$\frac{dT}{dR} = e^{-2R} - 2Re^{-2R} = 0$$

$$R_{max} = \frac{1}{2}$$

$$R_{max} = 1$$

Substituting R_{max} in eq. (3.29.2)

$$T_{max} = \frac{1}{2} e^{-1} = 0.1839 = 0.184$$

Slotted ALOHA :

1. In slotted ALOHA, time is divided into equal time slots of length greater than the packet duration τ.
2. The subscribers each have synchronized clocks and transmit a message only at the beginning of a new time slot, thus resulting in a discrete distribution of packets.
3. This prevents partial collisions, where one packet collides with a portion of another. As the number of user's increases, a greater delay will occur due to complete collisions and the resulting repeated transmissions of those packets originally lost.
4. The number of slots which a transmitter waits prior to retransmitting also determines the delay characteristics of the traffic.
5. The vulnerable period for slotted ALOHA is only one packet duration, since partial collisions are prevented through synchronization.
6. The probability that no other packets will be generated during the vulnerable period is e^{-R}. The throughput for the case of slotted ALOHA is thus given by

$$T = Re^{-R} \quad ...(4.16.3)$$

7. The maximum throughput achieved by using the slotted ALOHA protocol is given by

$$\frac{dT}{dR} = e^{-R} - Re^{-R} = 0$$

Substituting R_{max} in eq. (4.16.3)

$$T_{max} = e^{-1} = 0.3679 = 0.368$$

PART-4

CSMA and Their Versions : Paket and Pooling Reservation Based Multiple Access Schemes.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 4.17. **Give the basic principle of carrier sense multiple access (CSMA) protocols.**

Answer

Basic principle :

1. To minimize the chance of collision, and therefore, increase the performance, the CSMA method was developed.
2. The chance of collision can be reduced if a station senses the medium before trying to use it.
3. Carrier sense multiple access (CSMA) requires that each station first listen to the medium (or check the state of the medium) before sending.
4. In other words, CSMA is based on the principle "sense before transmit" or "listen before talk".
5. CSMA can reduce the possibility of collision, but it cannot eliminate it. The reason is due to possibility of collision still exists because of propagation delay; when a station sends a frame, it still takes time (although very short) for the first bit to reach every station and for every station to sense it.

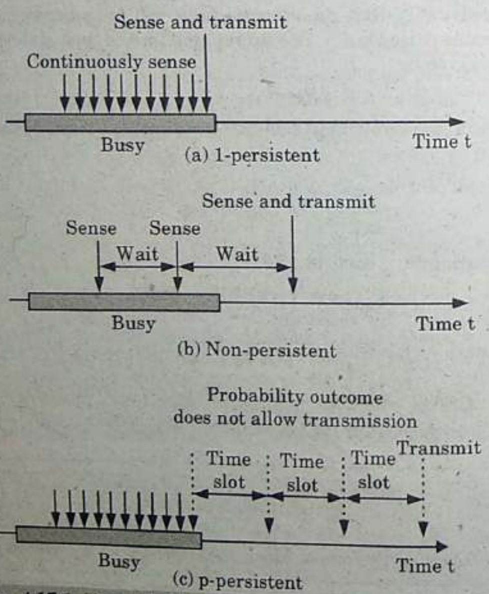


Fig. 4.17.1. Shows the behaviour of three persistence methods.

6. In another words, a station may sense the medium and find it idle, only because the first bit sent by another station has not yet been received.
7. Then the two signals will collide and both frames are destroyed.

Persistence Methods :

1. If there are any channel found to be busy or idle, the station control the channel by the following methods : the 1-persistent method, the non-persistent method, and the p-persistent method.
2. Fig. 4.17.1 shows, the behavior of three persistence methods when a station finds a channel busy.

There exist several variations of the CSMA strategy.

i. **1-persistent CSMA :** The terminal listens to the channel and waits for transmission until it finds the channel idle. As soon as the channel is idle, the terminal transmits its message with probability one.

ii. **Non-persistent CSMA :** In this type of CSMA strategy, after receiving a negative, acknowledgement the terminal waits a random time before retransmission of the packet. This is popular for wireless LAN applications, where the packet transmission interval is much greater than the propagation delay to the farthermost user.

iii. **p-persistent CSMA :** p-persistent CSMA is applied to slotted channels. When a channel is found to be idle, the packet is transmitted in the first available slot with probability p or in the next slot with probability $1-p$.

iv. **CSMA/CD :** In CSMA with collision detection (CD), a user monitors its transmission for collisions. If two or more terminals start a transmission at the same time, collision is detected, and the transmission is immediately aborted in midstream. This is handled by a user having both a transmitter and receiver which is able to support listen-while talk operation. For a single radio channel, this is done by interrupting the transmission in order to sense the channel.

v. **CSMA/CA :** In CSMA with collision avoidance (CSMA/CA), an attempt is made to improve the performance of CSMA. CSMA/CA by default uses the carrier sensing mechanism with exponential back-off.

Que 4.18. **Discuss in detail reservation based multiple access schemes.**

Answer

In multiple access schemes based on reservation, there is a provision for users to reserve their slot or resources in advance and then the transmission will be followed in that order. Such reservation times are separately provided.

i. **Packet reservation multiple access :**

1. Packet reservation multiple access (PRMA) is an implicit reservation scheme. Here, slots can be reserved implicitly as follows.

2. A certain number of slots form a frame, which is repeated in time just like a TDM pattern.
3. A base station broadcasts the status of each slot to all mobile stations. All stations receiving this vector will then know which slot is occupied and which slot is currently free.
4. If all the slots in a vector are reserved except one, then more than one station attempts to access this free slot.
5. Hence, a collision occurs, and a new status will be generated with the same free slot, indicating that one slot is still available for reservation.
6. Again, stations can compete for this slot and the procedure continues. The actual transmission of packets in the network will follow the order in the final reservation status.

ii. Polling and token passing :

1. Polling is a strictly centralized scheme with one master station and several slave stations.
2. The master can poll the slaves according to many schemes: round robin randomly according to the reservations, and so on.
3. The master can also establish a list of stations wishing to transmit during a contention phase. After this phase, the station polls each station on the list.
4. Token passing is mainly suitable for wired networks. IEEE 802.4 is a token bus protocol on the LAN whereas IEEE 802.5 is a token ring protocol. Here, the key part is the token, which is a small bit pattern.
5. The station that is transmitting captures the token during the transmission; after reception of the complete data, it releases the token, again circulating it among various users.
6. Thus, a logical or physical ring configuration is necessary for the token passing type of scheme.

VERY IMPORTANT QUESTIONS

Following questions are very important. These questions may be asked in your SESSIONALS as well as UNIVERSITY EXAMINATION.

Q. 1. Discuss multiplexing schemes.

Ans. Refer Q. 4.1.

Q. 2. Consider a GSM system which is a TDMA/FDD system that uses 20 MHz for forward link which is broken into radio

channels and if no guard band is assumed, find the number of simultaneous users that can be accommodated in GSM.

Ans. Refer Q. 4.7.

Q. 3. Explain the different type of multiple access schemes (TDMA, CDMA and FDMA).

Ans. Refer Q. 4.8.

Q. 4. Explain SC-FDMA, IDMA schemes and hybrid method of multiple access shcemes.

Ans. Refer Q. 4.13.

Q. 5. Explain the structure of RAKE receiver with the help of neat diagram. What is M branch RAKE receiver ?

Ans. Refer Q. 4.14.

Q. 6. Define the term ALOHA. Explain Pure and Slotted ALOHA.

Ans. Refer Q. 4.16.

Q. 7. Give the basic principle of carrier sense multiple access (CSMA) protocols.

Ans. Refer Q. 4.17.

☺☺☺

5

UNIT

GSM System For Mobile Telecommunication

CONTENTS

PART-1

GSM System for Mobile Telecommunication, General Packet Radio Service, Edge Technology.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 5.1. **What is GSM ? Mention the GSM services and features.**

Answer

GSM :

1. Global system for mobile (GSM) is a second generation cellular system standard that was developed to solve the fragmentation problems of the first generation cellular systems in Europe.
2. GSM was the world's first cellular system to specify digital modulation and network level architectures and services, and is the world's most popular 2G technology.
3. GSM was originally developed to serve as the pan-European cellular service and promised a wide range of network services through the use of ISDN.

GSM services and features :

1. GSM services follow ISDN guidelines and are classified as either teleservices or data services.
2. Teleservices include standard mobile telephony and mobile originated or base originated traffic.
3. Data services include computer-to-computer communication and packet switched traffic. User services may be divided into three major categories:
   i. Telephonic services.
   ii. Bearer service or data services.
   iii. Supplementary ISDN services.
4. From the user's point of view, one of the most remarkable features of GSM is the subscriber identity module (SIM), which is a memory device that stores information such as the subscriber's identification number, the networks and countries where the subscriber is entitled to service, privacy keys and other user specific information.
5. A second remarkable feature of GSM is the on-the-air privacy which is provided by the system.

Que 5.2. Draw the GSM architecture and also explain radio subsystem in mobile radio communication.

AKTU 2018-19, Marks 10

OR

Explain the term GSM in detail in mobile radio communication using system architecture.

AKTU 2016-17, Marks 10

Answer

GSM services : Refer Q. 5.1, Page 5–2A, Unit-5.

GSM architecture :

1. The GSM system architecture consists of three major interconnected subsystems :
   i. The base station subsystem (BSS)
   ii. Network and switching subsystem (NSS) and the
   iii. Operation support subsystem (OSS).
2. The mobile station (MS) is also a subsystem, but is usually considered to be part of the BSS for architecture purposes.
3. Fig. 5.2.1 shows the block diagram of the GSM system architecture. The mobile stations (MSs) communicate with the base station subsystem (BSS) over the radio air interface.

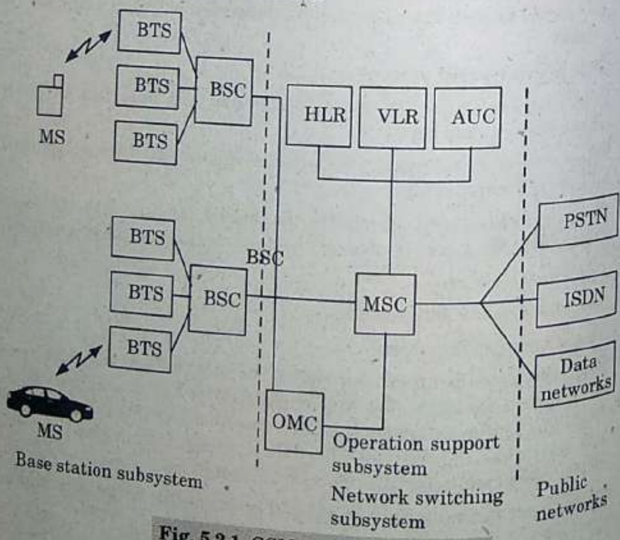


Fig. 5.2.1. GSM system architecture.

Base station subsystem (BSS) :

1. The BSS, also known as the radio subsystem, provides and manages radio transmission paths between the mobile stations and the mobile switching center (MSC).
2. The BSS also manages the radio interface between the mobile stations and all other subsystems of GSM.
3. Each BSS consists of many base station controllers (BSCs) which connect the MS to the NSS via the MSCs.
4. The BSS consists of many BSCs which connect to a single MSC, and each BSC typically controls upto several hundred base transceiver stations (BTSs).
5. Mobile handoffs between two BTSs under the control of the same BSC are handled by the BSC and not the MSC. This greatly reduces the switching burden of the MSC.

Network and switching subsystem (NSS) :

1. The NSS manages the switching functions of the system and allows the MSCs to communicate with other networks such as the PSTN and ISDN.
2. The NSS handles the switching of GSM calls between external networks and the BSCs in the radio subsystem and is also responsible for managing and providing external access to several customer databases.
3. In the NSS, there are three different databases :

i. **HLR (Home Location Register) :**
1. The HLR is a database which contains subscriber information and location information for each user who resides in the same city as the MSC.
2. Each subscriber in a particular GSM market is assigned a unique International Mobile Subscriber Identity (IMSI) and this number is used to identify each home user.

ii. **VLR (Visitor Location Register) :**
1. The VLR is a database which temporarily stores the IMSI and customer information for each roaming subscriber who is visiting the coverage area of a particular MSC.
2. Once a roaming mobile is logged in the VLR, the MSC sends the necessary information to the visiting subscribers HLR so that calls to the roaming mobile can be appropriately routed over the PSTN by the roaming user's HLR.

iii. **AUC :**
1. The authentication center (AUC) is a strongly protected database which handles the authentication and encryption keys for every single subscriber in the HLR and VLR.
2. The authentication center contains a register called the Equipment Identity Register (EIR) which identifies stolen or fraudulently altered phones that transmit identity data that does not match with information contained in either the HLR or VLR.

Operation support subsystem (OSS) :

1. The OSS supports the operation and maintenance of GSM and allows system engineers to monitor, diagnose and troubleshoot all aspects of GSM system.
2. The OSS supports one or several Operation Maintenance Centers (OMC) which are used to monitor and maintain the performance of each MS, BS, BSC and MSC within a GSM system.
3. The OSS has three main functions, which are :
   i. To maintain all telecommunications hardware and network operations with a particular market.
   ii. Manage all charging and billing procedures, and
   iii. Manage all mobile equipment in the system.

Que 5.3. **Explain the GSM architecture and frame structure in mobile radio communication system in detail.**

AKTU 2014-15, Marks 10

Answer

GSM architecture : Refer Q. 5.2, Page 5–3A, Unit-5.

GSM frame structure :

1. A simplest GSM frame structure is shown in Fig. 5.3.1. Each frame consists of 8 time slots and 156.25 bits/slot. Therefore, a frame contains $8 \times 156.25 = 1250$ bits/frame.
2. Each frame has a period of 4.615 ms.

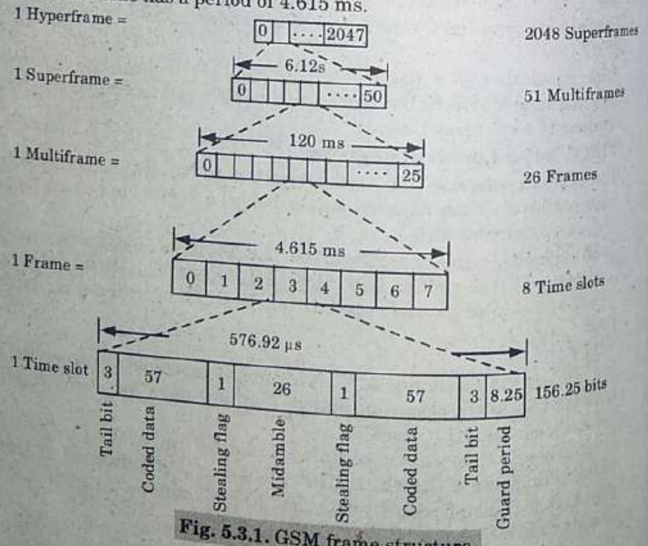


Fig. 5.3.1. GSM frame structure.

3. Each time slot contains tail bits, coded bits, stealing bits, midamble and guard period which are used to provide the information like whether the time slot (TS) contains voice data or control data , to analyze the radio channel characteristics before decoding the user data, to measure the signal strength, to bear (carry) the information etc.
4. Each of the normal speech frames is grouped into larger structures called multiframes, superframes and hyper frames.
5. One multiframe consists of 26 frames and has a period of 120 ms.
6. One superframe consists of 51 multiframes. The overall period of a superframe is 6.12 sec.
7. The 13^{th} and 26^{th} frame are not used for traffic but, for control purposes.
8. A hyperframe contains 2048 superframes (or 2,715, 648 frames). It is sent about every 3 hours, 28 minutes and 54 seconds.

Que 5.4. **Explain signal processing and GSM operations from speech input to speech output with diagram.**

AKTU 2015-16, Marks 7.5

OR

Explain signal processing and GSM operations from speech input to speech output with diagram. Calculate the total available channels for a cellular system having a total bandwidth of 60 MHz which uses two 50 kHz simplex channel to provide full duplex voice and control channels assume that the system uses nine cell reuse pattern and 1 MHz of the total bandwidth is allocated for control channels. Also calculate the number of control channels and voice channels per cell.

AKTU 2017-18, Marks 10

Answer

Fig. 5.4.1 illustrates all of the GSM operations from transmitter to receiver.

1. **Speech coding :** The GSM speech coder is based on the residually excited linear predictive coder (RELP), which is enhanced by including a long-term predictor (LTP). The coder provides 260 bits for each 20 ms blocks of speech, which yields a bit rate of 13 kbps.
2. **Channel coding :** The output bits of the speech coder are ordered into groups for error protection, based upon their significance in contributing to speech quality.
3. **Interleaving :** In order to minimize the effect of sudden fades on the received data, the total of 456 encoded bits within each 20 ms speech frame or control message frame are broken into eight 57 bit sub-blocks. These eight sub-blocks which make up a single speech frame are spread over eight consecutive TCH time slots.
4. **Burst formatting :** Burst formatting adds binary data to the ciphered blocks, in order to help synchronization and equalization of the received signal.

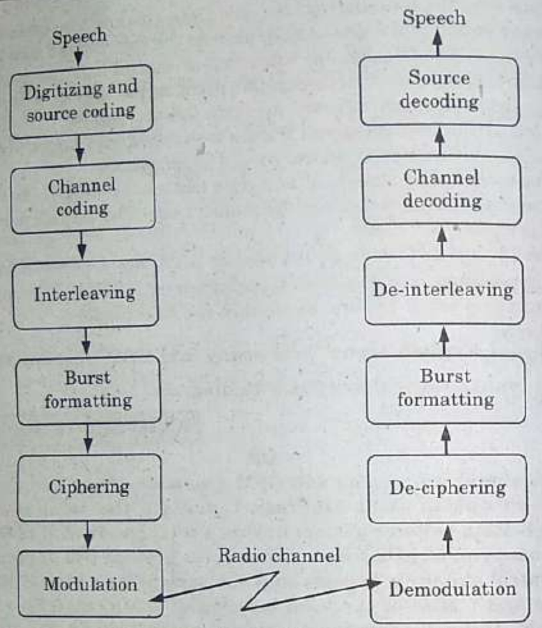


Fig. 5.4.1. GSM operations from speech input to speech output.

5. **Ciphering :** Ciphering modifies the contents of the eight interleaved blocks through the use of encryption techniques known only to the particular mobile station and base transceiver station. Two types of ciphering algorithms called *A*3 and *A*5 are used in GSM.
6. **Modulation :** The modulation scheme used by GSM is 0.3 GMSK, where 0.3 describes the 3 dB bandwidth of the Gaussian pulse shaping filter with relation to the bit rate (*e.g.*, BT = 0.3).
7. **Frequency hopping :** Under normal conditions, each data burst belonging to a particular physical channel is transmitted using the same carrier frequency. However, if users in a particular cell have severe multipath problems, the cell may be defined as a hopping cell by the network operator, in which case slow frequency hopping may be implemented to combat the multipath or interference effects in that cell.
8. **Equalization :** Equalization is performed at the receiver with the help of the training sequences transmitted in the midamble of every time slot. The type of equalizer for GSM is not specified and is left up to the manufacturer.

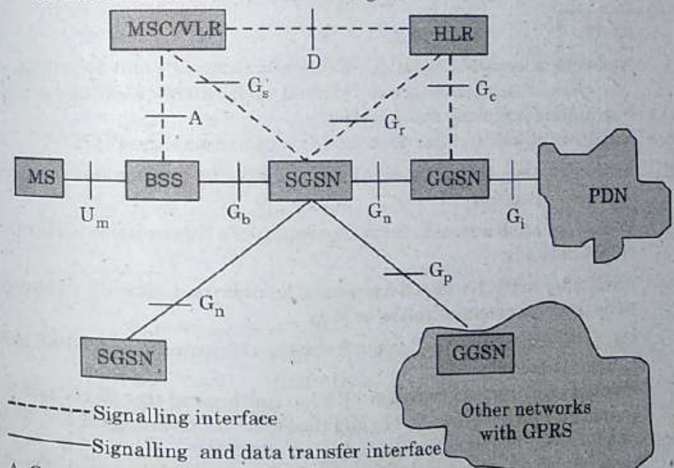
9. **Demodulation :** The portion of the transmitted forward channel signal which is of interest to a particular user is determined by the assigned TS and ARFCN. The appropriate TS is demodulated with the aid of synchronization data provided by the burst formatting. After demodulation, the binary information is deciphered, de-interleaved, channel decoded and speech decoded.

Numerical : Refer Q. 1.8, Page 1–9A, Unit-1.

Que 5.5. **Explain GPRS architecture.**

Answer

1. The GSM technology was developed for voice services, but it did not have the capability to provide data services.
2. To develop a higher data rate capability and to enhance the services the GPRS protocol was developed on the GSM platform. The scenario and GPRS architecture are shown in Fig. 5.5.1.



A, G_s, G_r, G_c, D are interface for signalling.
U_m, G_b, G_n, G_p, and G_i are interfaces for signalling and transmission.

Fig. 5.5.1. GPRS architecture with modifications in the GSM, including additional GPRS components.

3. GPRS could offer data rates upto 115 kbps, which allowed web browsing and other services requiring data transfer. GGSN-gateway GPRS support node; SGSN-serving GPRS support node; PDN-packet data network.
4. GPRS uses packet-switched data rather than circuit-switched data, which is much more efficient for using the available capacity. This is because data transfer takes place in a bursty manner.

5. The transfer occurs in short peaks, followed by breaks when there is little or no activity. Packet switching permits sharing the overall capacity among several users.
6. The data is split into packets and then tags are inserted into each packet to mark its destination address. Packets from several sources can then be transmitted over the link. It is unlikely that the data burst for all the users will occur at the same time, so sharing the overall resource in this fashion makes the system efficient.
7. For GPRS, the data from the BSC is routed through the serving GPRS support node (SGSN).
8. This forms the gateway to the services within the network and then a gateway GPRS support node (GGSN), which forms the gateway to the outside world.

Que 5.6. **Write a short note on GPRS functional groups.**

Answer

1. **Network access:** Network access supports standard point-to-point data transfer and anonymous access (without authentication and ciphering). The functions include the following :
   i. Registration, which associates the MS identity with the PDPs.
   ii. Authentication and authorization are for security purposes to avoid anonymous access.
   iii. Packet terminal adaptation, which adapts data transmission across the GPRS network.
   iv. Admission control, which determines the radio and network resources to be used for communication of MSs.
   v. Charging information collection for packet transmission in GPRS and external networks.
2. **Packet routing and transfer :** Packet routing and transfer is used to route the data between the MS and the destination through the SGSNs and GGSNs. The functions include the following :
   i. Relay function, which is used by the BSS to forward packets between the MS and the SGSN and is also used by the SGSN to forward packets between the BSS and the SGSN or GGSN.
   ii. Routing which determines the destinations of packets.
   iii. Address translation and mapping, which converts a GPRS network address to an external data network address and vice-versa.
   iv. Encapsulation and tunnelling, which encapsulate packets at the source of a tunnel, deliver the packets through the tunnel, and decapsulate them at the destination.
   v. Compression and ciphering, which reduce and protect the database.

3. **Mobility management :** Mobility management keeps track of the current location of an MS which includes the following :
   i. Cell update.
   ii. Routing area update.
   iii. Combined routing area and location area update.
4. **Logical link management :** Logical link management maintains the communication channel between the MS and the GSM network across the radio interface, which includes the following :
   i. Logical link establishment.
   ii. Logical link maintenance.
   iii. Logical link release.
5. **Radio resource management :** Radio resource management allocates and maintains radio communication paths, which includes the following:
   i. Cell selection, which enables the MS to select the optimal cell for radio communication.
   ii. Path management, which maintains the communication paths between the BSSs and SGSNs.
6. **Network management:** Network management provides the mechanisms to support operations, authentication and maintenance (OAM) functions related to GPRS.

Que 5.7. **Write a short note on GPRS device categories and its modes.**

Answer

A. GPRS device categories :

Class A : Mobile phones in this class can be connected to both GPRS and GSM services at the same time.

Class B : Mobile phones in this class can be attached to both GPRS and GSM services, but they can be used on only one service at a time.

Class C : Mobile phones in this class can be attached to either GPRS or GSM services, but the user needs to switch manually between the two different types.

B. Modes :

i. Initialization or idle :

1. When a mobile is turned ON, it must register with a network and update the location register. It is referred to as location update.
2. When the mobile performs its location update, the network also performs an authentication to ensure that it is allowed to access the network. Like GSM, it accesses the HLR and VLR as necessary for the location update and the AUC for authentication.

3. The SGSN also maintains a record of the location of the mobile so that data can be sent there if required.

ii. Stand by :

1. The mobile then enters a standby mode, periodically updating its position as required.
2. It monitors to ensure that it has not changed BSs and also looks for stronger BSCCHs. The mobile will also monitor the PPCH in case of an incoming alert indicating that data is ready to be sent.

iii. Ready :

1. In the ready mode, the mobile is attached to the system and virtual connection is made with the SGSN and GGSN.
2. This connection enables the network to know where to route the packets when they are sent and received.

Que 5.8. **Write a short note on EDGE technology.**

Answer

1. Enhanced data for GSM evolution (EDGE) is an enhancement to the GSM mobile cellular phone system.
2. It enables data to be sent over a GSM TDMA system at speeds up to 384 kbps.
3. It is regarded as a 2.5 G system and also known as EGPRS (enhanced GPRS).
4. Operators who have not been able to secure full 3G licenses use EDGE to provide data services. It is highly spectrally efficient and high-speed system.

EDGE system requirements :

1. The EDGE technology is applied to GSM networks where the enhancements provided by GPRS have already been added.
2. GSM provides voice services based on circuit switching whereas GPRS provides data services based on packet switching. Hence, the infrastructure for both is to be adopted by EDGE.
3. In terms of implementation, EDGE is intended to build on the enhancements provided by the addition of GPRS where packet switching is applied to a network.
4. It then enables a threefold increase in the speed at which data can be transferred by adopting a different form of modulation.
5. EDGE changes the modulation to 8PSK, which is a form of phase shift keying (PSK) where eight phase states are used.
6. The advantage is that it can transmit high data rates, although it is not as immune to interference and noise. By using 8PSK, data can be transferred at 48 kbps per channel rather than 9.6 kbps, the rate that is possible using GMSK. By allowing the use of multiple channels, the technology allows the transfer of data at rates up to 384 kbps.

Specification : Edge services have the following specifications :

i. They offer data services upto 400 kbps.
ii. They are used on voice over packet-switched network.
iii. They support IP-based applications.
iv. Typical applications are internet on mobile, mobile LAN, videophone, wireless real-time applications and file transfer.

PART-2

CDMA Based Standards : IS95 to CDMA 2000, Wireless Local Loop, IMT-2000 and UMTS.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 5.9. **Explain forward link and reverse link in IS-95 system.**

Answer

A. Forward link (BS – MS) :

1. The IS-95 system specifies 869-894 MHz band for the forward link.
2. It consists of upto 64 logical CDMA channels each occupying the same 1228 KHz bandwidth. The forward link supports four types of channels :

i. **Pilot (Channel 0) :** This channel allows the mobile unit to acquire timing information, provide phase reference for the demodulation, and provides a means of signal strength comparison for the purpose of handoff determination. The pilot channel consists of all zeros transmitted at higher power then user channels.

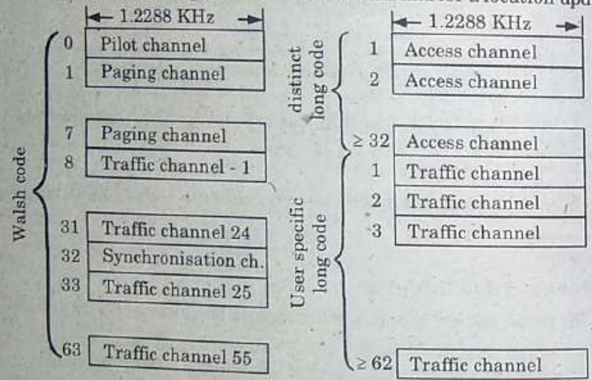
ii. **Synchronization (Channel 32) :** A 1200 bps channel used by the MS to obtain identification information about the cellular system (system time, long code state, protocol revision etc)

iii. **Paging (Channel 1 - 7) :** These channels contain messages for one or more mobile stations.

iv. **Traffic channel (Channel 8 - 31 and 33 - 63) :** The forward channel supports 55 traffic channels. The original specification supported data rates upto 9.6 kbps. All these channel use the same bandwidth. The chipping code is used to distinguish among the different channels. For the forward channel, the chipping codes are the 64 orthogonal 64 bit codes derived from a 64 × 64 walsh matrix.

B. Reverse link (MS – BS) :

1. The IS-95 system specifies 824-849 MHz band for the reverse channel.
2. The reverse link consists of upto 94 logical CDMA channels each occupying the same 1228 KHz bandwidth.
3. The reverse link supports upto 32 access channels and upto 62 traffic channels. The traffic channels in the reverse link are mobile unique.
4. Each station has a unique long code mask based on its electronic serial number. The long code mask is a 42 bit number. There are $2^{42} - 1$ different masks.
5. The access channel is used by a mobile to initiate a call to respond to a paging channel message from the base station and for a location update.



(a) Forward channel (b) Reverse channels.

Fig. 5.9.1.

Que 5.10. **Describe the forward CDMA channel and reverse CDMA channel using proper block diagram.** **AKTU 2014-15, Marks 05**

Answer

IS-95 forward CDMA channel :

1. For voice traffic the speech is encoded at a data rate of 9.6 kbps including error detection bits. During quiet periods the data rate is lowered to 1200 bps.
2. The 2400 bps is used to transmit transient in the background noise. 4800 bps is used to mix digitized speech signalling data.
3. The data or digitized speech is transmitted in 20 ms blocks will forward error correction provided by convolutional encoder with rate 1/2, thus doubling the effective data rate to 19.2 kbps rate.

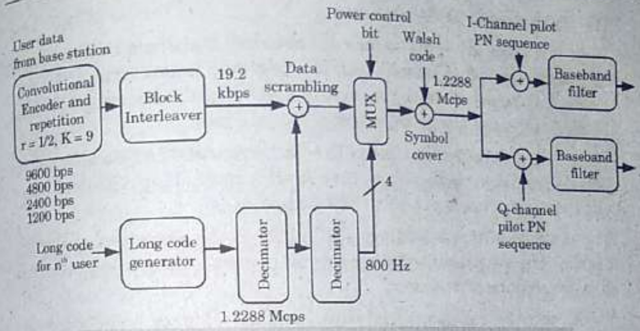


Fig. 5.10.1. Forward CDMA channel modulation process.

4. For lower data rates, the encoder O/P bits are replicated to yield 19.2 kbps rate.
5. The data are then interleaved in blocks to reduce the effects of errors by spreading them out.
6. Following the interleaver, the data bits are scrambled. The purpose of this is to serve as privacy mask and also to prevent sending of repetitive patterns.
7. The scrambling is accomplished by means of a long code that is generated as a pseudo random number from a 42-bit long shift register.
8. The shift register is initialised with the user's electronic serial number. The output of long code generator is at a rate of 1.2288 Mcps.
9. The decimator keeps only the first chip out of every 64 consecutive PN chips. The O/P of decimator is at 19.2 kbps.
10. The data scrambling is performed by modulo-2 addition of the interleaver O/P with the decimator O/P.
11. The next step in the processing inserts power control bits at a rate of 800 bps by replacing some of the traffic channel bits, using the long code generator to encode the bits.
12. The next step in the DSSS function spreads the 19.2 kbps to a rate of 1.2288 Mbps (Mcps) using one row of 64 × 64 Walsh matrix. One row of the matrix is assigned to a MS during call set up.
13. After this bits are spread in quadrature. Two separate bit stream (I and Q), data are split in I and Q channels.
14. Data in each channel XORed with a unique short-code. The short-binary spreading sequence, period $2^{15} - 1$, chips is generated from a 15 bit long shift register. Thus, data is transmitted using QPSK modulation.
15. A pilot code on the forward link is also transmitted simultaneously and at a higher power level thereby allowing all mobiles to use coherent detection while estimating the channel conditions.

IS - 95 reverse CDMA channel :

1. First few steps are same as forward channel. Data rate tripled here to 28.8 kbps, then block interleaved. The next step is spreading of the data using Walsh matrix. In the reverse channel, the data coming out of the block interleaver are grouped into units of 6 bits.
2. Each 6 bit unit serves as a index to select a row of 64 × 64 walsh matrix ($2^6 = 64$) and that row is substituted for the input. Thus, the data row is expanded by a factor of 64/6 to 307.2 kbps.
3. The purpose of this encoding is to improve the reception at the BS, because the 64 possible coding are orthogonal, the coding enhances decision making at receiver.
4. We can view this Walsh modulation as a form of block error correcting code with $(n, k) = (64, 6)$ and $d_{min} = 32$ (all distances are 32).
5. The data burst randomizer is used to help to reduce interference from other mobile stations. The operation involves using the long code mask to smooth the data out over each 20 ms frame.
6. The next step in the process is DSSS function in case of reverse channel, the long code unique to mobile is XORed with the output of randomizer to produce 1.2288 Mcps final data stream.
7. The digital bit stream is then modulated on to a corner using an offset QPSK modulation.
8. The reason the modulators are different is that in forward channel, the spreading codes are orthogonal, all coming from Walsh matrix, whereas in reverse channel, orthgonality of the spreading code is not guaranteed.

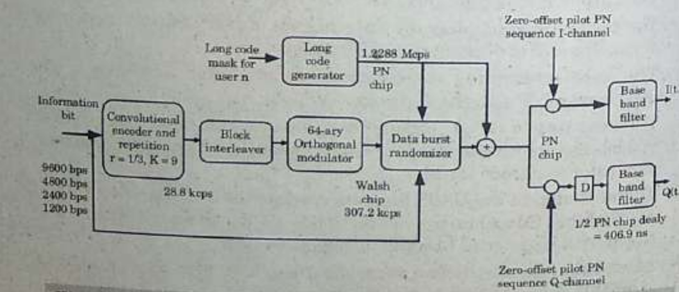


Fig. 5.10.2. Reverse IS-95 channel modulation process for a single user.

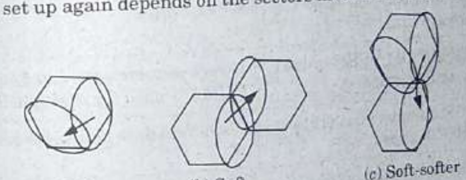
Que 5.11. **Discuss in detail the features of IS-95.**

Answer

The most important features of IS-95 are :

i. **Soft handoff :**

1. Generally, soft handoff refers to 'make before break' connection. In the IS-95 standard, three types of soft handoffs are defined that are shown in Fig. 5.11.1.
2. In the softer handoff case, shown in Fig. 5.11.1(*a*), the handoff is between two sectors of the same cell.
3. In the soft handoff case, as shown in Fig. 5.11.1(*b*), the handoff is between two sectors of different cells.
4. In the soft-softer handoff case, as shown in Fig. 5.11.1(*c*), the candidates for handoff includes two sectors from the same cell and a third sector from a different cell.
5. Weather the connection in the infrastructure needs to be turn down and set up again depends on the sectors involved in the final handoff.



(*a*) Softer (*b*) Soft (*c*) Soft-softer

Fig. 5.11.1. Types of soft handoff.

ii. **Power control :**

1. Like all cellular telephony systems, CDMA is also interference limited. However co-channel and adjacent channel interference are not the major problems here.
2. Instead the interference is from other users transmitting in the same frequency band at the same time.
3. In order to avoid the near-far effect, it is important to implement good power control. Also, in order to maintain a good link quality, effects such as fading and shadowing need to be countered by increasing the transmit power.
4. In IS-95, power control is very important especially on the reverse link where non-coherent detection is employed.
5. Two types of power control are implemented—an open loop and a closed loop. A slow mobile assisted power control is employed on the forward link.

iii. Provide extensive path diversity.

iv. Improve performance in difficult propagation environment.

v. Improvement in system capacity and coverage.

Que 5.12. **How does CDMA technology work in principle ? Give detailed features of GSM and CDMA mobile standards.**

AKTU 2017-18, Marks 10

Answer

CDMA technology : Refer Q. 5.10, Page 5–13A, Unit-5.

Comaprision :

S. No.	GSM standard	CDMA standard
1.	It uses 890-915 MHz frequency band for reverse link and 935-960 MHz for forward link.	It uses 824-849 frequency band for reverse link and 869-894 for forward link.
2.	TDMA/FDMA/FDD is used for multiple access.	Direct sequence spread spectrum CDMA is used for multiple access.
3.	GSM channel's data rate is 270.833 kbps.	Maximum data rate is 9.6 kbps.
4.	Carrier separation is 200 kHz.	A forward and reverse channel pair is separated by 45 MHz for cellular band operation.
5.	The modulation technique used is Gaussian minimum shift keying (GMSK).	All data transmitted on the reverse channel are convolutionally encoded and modulated by a 64-ary orthogonal modulation.

Que 5.13. **Write a short note on wireless local loop (WLL) system.**

Answer

1. A WLL is defined in the last-mile system category. WLL services may be defined as fixed wireless services intended to provide access to the telephone network.
2. In general local loop means exchange-to-exchange closed loop. Conventionally, telephone local loops are unshielded twisted pair (UTP) or shielded twisted pair (STP) based cables.
3. If we remove the wires used for communication, that is, establish an RF link, it becomes a wireless local loop, of course with supporting hardware.

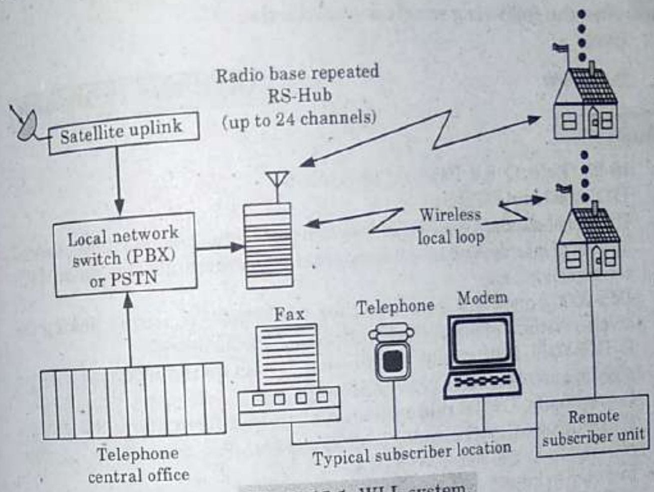


Fig. 5.13.1. WLL system.

4. Wireless local loop systems will generally divide a geographical region into many similar sized cells. Each cell will be serviced by a BS (wireless access network unit or WANU), which will communicate with all the WLL customers (wireless access subscriber units or WASUs) within the cell, as shown in Fig. 5.13.1 and 5.13.2.

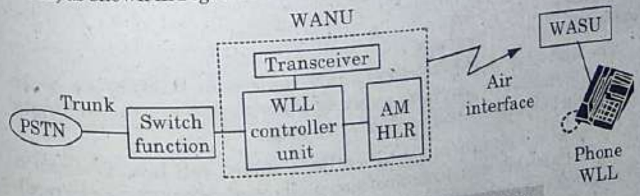


Fig. 5.13.2. WLL access unit and their functions.

5. The BS may be as simple as a small omnidirectional antenna and control box hanging from the overhead electrical lines. Each customer will be equipped with a transceiver and a small patch antenna.
6. The WANU, the interface between the underlying telephone network and the wireless link, consists of the following :
   i. BTS
   ii. Radio port control unit (RPCU).

Que 5.14. **Write a short note on IMT-2000 and UMTS.**

OR

Describe the following wireless standards :

i. **IS-95**

ii. **IMT-2000**

AKTU 2018-19, Marks 10

Answer

A. **IS-95 :** Refer Q. 5.9, Page 5–12A, Unit-5.

B. **IMT-2000 and UTMS :**

1. The global standard for 3G wireless communications is IMT-2000, defined by a set of interdependent international telecommunication union (ITU) recommendations.
2. IMT-2000 provides a framework for worldwide access by linking the diverse systems of terrestrial and/or satellite networks.
3. In IMT-2000, 2000 stands for the year it was introduced and also the spectrum used (around 2000 MHz).
4. Conceptually, UMTS represents an evolution from 2G to 3G. It uses WCDMA to carry the radio transmissions, and the system is often referred to by the name WCDMA.
5. In order to create and manage a system as complicated as UMTS or WCDMA, it is necessary to develop and maintain a large number of documents and specifications.
6. For UMTS, these are now managed by a group known as the 3GPP—third-generation partnership programme.

Que 5.15. **Describe the network architecture of UMTS with suitable diagram.**

Answer

1. The universal mobile telecommunication system (UMTS) is a system that is capable of providing a variety of mobile services to a wide range of global mobile communication standards.
2. To handle a mixed range of traffic, a mixed cell layout (shown in Fig. 5.15.1), that would consist of macrocells overlaid on micro and picocells is one of the architecture plans being considered.
3. This type of network distributes the traffic with the local traffic operating on the micro and pico cells, while the highly mobile traffic is operated on the macrocells, thus reducing the number of handoffs required for the fast moving traffic.
4. Macrocells will also be able to avoid the failures of the overlapped cells. However the major disadvantage of the overlaid architecture is the reduced spectrum efficiency.
5. The UMTS architecture will provide radio coverage with a network of base stations interconnected to each other and to a fixed network exchange.

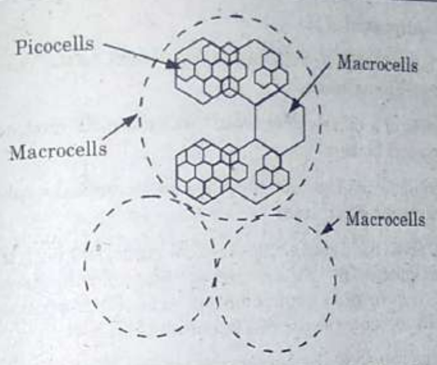


Fig. 5.15.1. Network architecture for UMTS.

Que 5.16. **Describe the elements of UMTS or WCDMA system.**

Answer

1. The network for UMTS can be split into three main constituents as shown in Fig. 5.16.1.

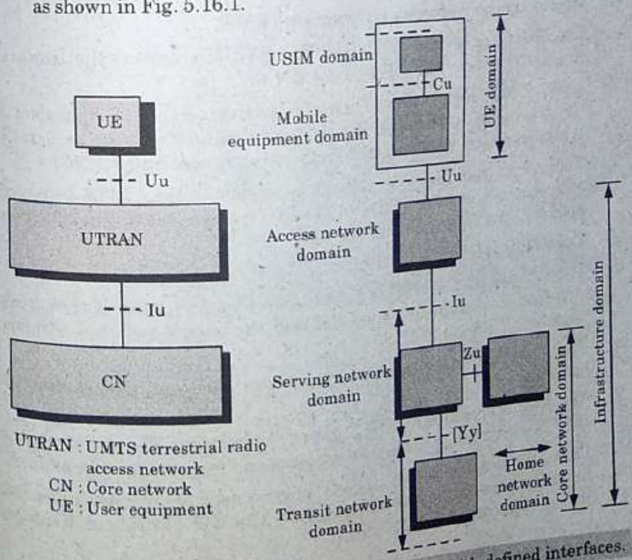


Fig. 5.16.1. UMTS architecture and domain along with defined interfaces.

i. User equipment (UE) :

1. The UE for UMTS or WCDMA is equivalent to the mobile equipment used on GSM networks.
2. It consists of a variety of elements including RF circuitry, processing, antenna and battery.
3. The circuitry used within the UE can be broadly split into RF and baseband processing areas.

a. The RF areas handle all elements of the signal, for both the receiver and the transmitter. One of the major challenges for the RF power amplifier was to reduce power consumption. The form of modulation used, WCDMA requires the use of a linear amplifier.

b. Baseband signal processing consists mainly of digital circuitry.

4. The UE contains a SIM card, although in the case of UMTS, it is termed as the universal identity module (USIM). It contains the IMSI and mobile station international subscriber directory number (MSISDN).
5. It also holds information regarding the preferred language to enable the correct language information to be displayed, especially when roaming and a list of preferred and prohibited public land mobile networks (PLMN).

ii. Radio access network subsystem (RNS) :

1. The RNS is the section of the UMTS or WCDMA network that acts as an interface between the UE and the CN.
2. Under UMTS terminology, the radio transceiver or BTS equivalent (in GSM) is known as the node B. This communicates with the various UE. It also communicates with the radio network controller (RNC).
3. The overall radio access network is known as the UMTS terrestrial radio access network (UTRAN). The RNC component of the radio access network (RAN) connects to the CN.

iii. Core Network (CN) :

1. The CN used for UMTS is based upon a combination of the circuit-switched elements used for GSM and the packet-switched elements used for GPRS and EDGE.
2. Thus, the CN is divided into circuit switched and packet switched domains.
3. The ATM is specified for UMTS core transmission. The architecture of the CN may change when new services and features are introduced.
4. Gateway Location Register (GLR) may be used to optimize subscriber handling between network boundaries. The MSC, VLR, and SGSN can merge to become a UMTS MSC.

Que 5.17. **Explain the term long term evolution in wireless communication.**

AKTU 2018-19, Marks 10

Answer

1. Long-term evolution transits from the existing UMTS circuit and packet switching combined network to an all-IP flat architecture system. Fig. 5.17.2 shows the simplicity of the LTE architecture.
2. The network side of the evolved UMTS terrestrial radio access network (E-UTRAN) is composed only of evolved Node Bs (eNode Bs or eNBs), hence the simplified architecture.
3. LTE supports cell sizes from tens of meters radius to 100 km radius macrocells.

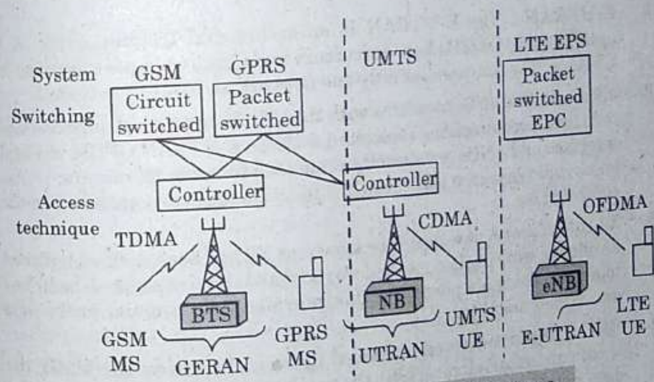


Fig. 5.17.1. Major differences in the architectures of GSM, GPRS, UMTS and LTE.

5. The new blocks specific to evolved UMTS evolution, LTE also known as the evolved packet system (EPS) are the E-UTRAN and the evolved packet core (EPC). The EPS is purely IP-based.
6. The LTE access network E-UTRAN is simply a network of BSs and eNBs, generating a flat architecture. There is no centralized intelligent controller and the eNBs are normally interconnected by the X2 interface and towards the CN by the S1 interface.

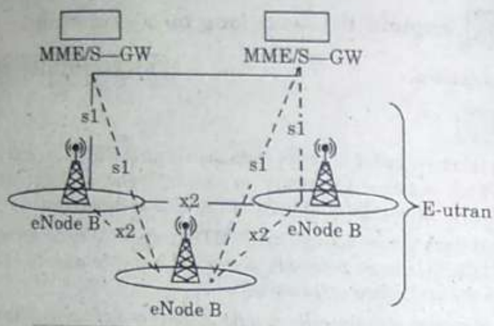


Fig. 5.17.2. Simplified LTE architecture - EPS.

Elements of LTE-EPS :

1. **E-UTRAN :** The E-UTRAN is an orthogonal frequency division multiplexing (OFDM)-based structure and is quite simple compared to UTRAN. It is composed of only one network element the eNode B.
2. **eNode B:** The eNB interfaces with the UE. It supports all physical and data link layer functions associated with the E-UTRAN OFDM physical interface and eNBs are directly connected to network routers. It also hosts radio resource control (RRC) functionality corresponding to the control plane.
3. **X2 interface:** A new interface known as X2 has been defined between eNode Bs, working in a meshed way to make them possible to be linked together. The main purpose of this interface is to minimize packet loss due to user mobility.
4. **EPC :** This is composed of several functional entities: the MME, the home subscriber server (HSS), the serving gateway (S-GW), the packet data network gateway (PDN-GW), and the policy and charging rules function (PCRF) server.
5. **MME :** The mobility management entity (MME) is charge of all the control plane functions related to subscriber and session management. The MME supports the following :

i. Security procedures.

ii. Terminal to network session handling.

iii. Idle terminal location management.

6. **HSS :** The HSS is the concatenation of the HLR and the AUC-two functions that are already present in GSM and UMTS networks. The HLR part of the HSS is in charge of storing and updating when necessary. The database containing all the user subscription information, including the following :

i. User identification and addressing which corresponds to the IMSI and MSISDN or mobile telephone number.

ii. User profile information which includes service subscription states and user-subscribed QoS information.

7. **S-GW :** The S-GW is the termination point of the packet data interface towards the E-UTRAN. The S-GW routes and forwards user data packets.
8. **PDN-GW :** The PDN-GW is the termination point of the packet data interface towards the packet data network (PDN). The PDN-GW provides connectivity from the UE to external PDNs by being the point of exit and entry of traffic for the UE.
9. **PCRF server :** It manages the service policy and sends the QoS setting information for each user session and the accounting rule information.

Que 5.18. **Explain the working of channels used in LTE.**

Answer

Long-term evolution channels are of three types :

1. **Physical channels :** The LTE physical channels are again grouped into uplink (SC-FDMA based) and downlink (OFDMA based) channels as each has different requirements and operates in a different manner.

Downlink channels : There are four downlink physical channels for correspondence from eNode B to UE.

i. **Physical broadcast channel (PBCH) :** This channel carries system information for the UE requiring access to the network. It carries only the master information block (MIB) messages. The modulation scheme is always QPSK, and the information bits are coded and rate matched.

ii. **Physical control format indicator channel (PCFICH) :** This channel informs the UE about the format of the signal being received. It indicates the number of OFDM symbols used for the physical downlink control channels (PDCCHs).

iii. **Physical downlink control channel (PDCCH) :** The main purpose of this channel is to carry mainly scheduling information of different types, downlink resource scheduling uplink power control instructions uplink resource grant, the indication for paging or system information.

iv. **Physical hybrid automatic repeat request indicator channel (PHICH) :** This channel is used to report the HARQ status. It carries the HAEQ ACK/NACK signal indicating whether a transport block has been correctly received.

Uplink channels: The following three different channels are specified as uplink physical channels :

i. **Physical uplink control channel (PUCCH) :** This channel provides the various control signalling requirements.

ii. **Physical uplink shared channel (PUSCH) :** This physical channel found on the LTE uplink is the uplink counterpart of the PDSCH.

iii. **Physical random access channel (PRACH) :** The channel is used for random access functions. This is the only non synchronized transmission that the UE can make within LTE.

2. **Logical channels :** The logical channels cover the data carried over the radio interface. Logical channels are again grouped into TCHs and CCHs, like in the GSM system.

i. **Traffic channels :** The LTE TCHs carry the user plane data.

ii. **Dedicated traffic channel :** This channel is used for the transmission of user data.

iii. **Multicast traffic channel (MTCH) :** This channel is used for the transmission of multicast data.

iv. **Control channels :** The LTE CCHs carry the control plane information.

v. **Broadcast control channel (BCCH) :** The channel provides system information to all mobile terminals connected to the eNode B.

vi. **Common control channel (CCCH) :** This channel is used for random access information needed for multicast reception.

vii. **Multicast control channel (MCCH) :** This channel is used for information needed for multicast reception.

viii. **Dedicated control channel (DCCH) :** This channel is used for carrying user-specific control information, for example, for controlling actions such as power control and handover.

3. **LTE transport channels :** The LTE transport channels are also defined for the uplink and the downlink, as each has different requirements and operates in a different manner.

Downlink : Four downlink transport channels are specified as follows:

i. **Broadcast channel :** This channel maps to the BCCH.

ii. **Downlink shared channel (DL-SCH) :** This channel is the main channel for downlink data transfer. It is used by many logical channels.

iii. **Paging channel :** This channel is used to convey the PCCH information.

iv. **Multicast channel (MCH) :** This channel is used to transmit MCCH information to set up multicast transmissions.

Uplink : The following two transport channels are specified for uplink :

i. **Uplink shared channel (UL-SCH) :** This channel is the main channel for uplink data transfer. It is used by many logical channels.

ii. **Random access channel (RACH) :** This channel is used for random access requirements.

Que 5.19. **Write a short note on mobile satellite communication.**

Answer

1. In mobile satellite communication, the satellite and mobile systems are combined.
2. Antenna spot beams can be used for frequency reuse. The beam projection areas are treated as cells.
3. Small satellite earth stations are used to send and receive the signals from the satellite. These signals can be further routed to the MSC, which will then take care of further routing of the signals.
4. The system offers communication services to mobile users operating within a predefined service area. The users communicate with other mobiles or with fixed users through one of the visible satellites.
5. Users in fixed network are accessed through large fixed stations called gateways, which carry a large amount of traffic, whereas the mobiles are small portable units that can support only a few channels.
6. There may be one or more space segments, which may consist of one satellite or a group of interlinked satellites. Depending on the service area and application, the space segments can be utilized.
7. Telemetry and control ground stations, used for monitoring and controlling satellites, constitute a part of the space segment.

PART-3

Introduction to Mobile Adhoc Networks, Li-Fi Communication, Ultra-Wideband Communication.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 5.20. **What is basic concept of adhoc network ? Why and how proper route is required to discuss in adhoc network ?**

AKTU 2015-16, Marks 10

OR

Explain mobile adhoc network in wireless communication.

AKTU 2016-17, Marks 10

Answer

1. Mobile adhoc networks (MANETs) are collections of mobile nodes dynamically establishing short-lived networks in the absence of fixed infrastructure.

2. Each mobile node is equipped with a wireless transmitter and a receiver with an appropriate antenna.
3. These mobile nodes are connected by wireless links and act as routers for all other mobile nodes in the network.
4. Nodes in mobile adhoc networks are free to move and organize themselves in an arbitrary manner.
5. These features make MANETs very practical and easy to deploy in places where existing infrastructure is not capable enough to allow communication, for instance in disaster zones or infeasible to deploy locations.

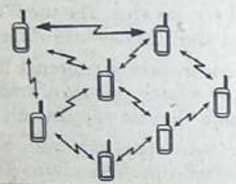


Fig. 5.20.1. Mobile adhoc netwrork.

6. MANETs are the short term temporary spontaneously wireless networks of mobile nodes communicating with each other without the intervention of any fixed infrastructure or central control.
7. It is an autonomous system of mobile nodes, mobile terminals, or mobile stations serving as routers interconnected by wireless links.
8. Depending on the locations, antenna coverage pattern, transmit power levels and co-channel interference levels, a wireless connectivity exists among participating mobile nodes at a given time, either in the form of random multihop transmissions or adhoc network.
9. Network communication and management tasks are typically performed in a distributed manner.
10. As the nodes moves or adjust their transmission and reception parameters, MANET topology may change from time to time.
11. An adhoc network can be of the following types :
   i. Wireless personal area network (WPAN).
   ii. Wireless local area network (WLAN).

Que 5.21. Describe challenges and issues in Mobile AD-HOC Networks (MANETs) in wireless communication.

AKTU 2018-19, Marks 10

Answer

1. Mobility in ad-hoc networks causes frequent link failure, which in turn causes packet losses.

2. Transmission control protocol treats loss or delay of a package acknowledgement as traffic congestion. It is considered as a complex task in mobile ad-hoc networks.
3. Some of the most challenging issues are :
   i. Limited wireless transmission range.
   ii. Broadcast nature of the wireless medium.
   iii. MAC related issues such as hidden terminal and exposed terminal problems.
   iv. Routing problem due to change in route because of node mobility.
   v. Packet losses due to transmission errors and mobility.
   vi. Battery constraints.
   vii. Security issues leading to ease of snooping.
   viii. Design of the MAC protocols which define how the wireless medium is shared by all nodes.

Que 5.22. What are the applications of MANET ? Explain any one application in brief.

OR

What do you understand by Mobile Data Network ? Explain important features of mobile Ad-Hoc networks.

AKTU 2017-18, Marks 10

Answer

A. MANET : Refer Q. 5.20, Page 5–26A, Unit-5.

B. Applications of MANET are as follows :

i. Data networks,
ii. Home networks,
iii. Device networks,
iv. Sensor networks and
v. Distributed control systems.

Data networks :

1. Mobile adhoc data networks primarily support data exchange between laptops, palmtops, personal digital assistants (PDAs), and other information devices.
2. These data networks generally fall into three categories based on their coverage area : LANs, MANs, and WANs, Infrastructure based wireless.
3. However, mobile adhoc data networks have some advantages over these infrastructure based networks.
4. First, only one access point is needed to connect to the backbone wired infrastructure: this reduces cost and installation requirements.
5. In addition, it can be inefficient for nodes to go through an access point of base station. For example, PDAs that are next to each other can exchange information directly rather than routing through an intermediate node.

6. Wireless MANs typically require multi-hop routing since they cover a large area. The challenge in these networks is to support high data rates, in a cost-effective manner, over multiple hops, where the link quality of each hop is different and changes with time.
7. The lack of centralized network control and potential for high-mobility users further complicates this objective.
8. Military programs such as DARPA's GLOMO (Global mobile information systems) have invested much time and money in building high speed adhoc wireless MANs that support multimedia, with limited success.
9. Wireless WANs are needed for applications where network infrastructure to cover a wide area is too costly or impractical to deploy. For example, sensor networks may be dropped into remote areas where network infrastructure cannot be developed.
10. In addition, networks that must be build up and torn down quickly, *e.g.*, for military applications or disaster relief, are infeasible without an adhoc approach.

C. Features of MANET :

1. Mobile ad-hoc networks are formed dynamically by an autonomous system of mobile nodes that are connected via wireless links.
2. No existing fixed infrastructure or centralized administration and no base station.
3. Mobile nodes are free to move randomly.
4. Network topology changes frequently.
5. Each node work as router.
6. Distributed operation and multi hop routing.

Que 5.23. **Explain mobile adhoc network in wireless communication and discuss any two applications.**

AKTU 2014-15, Marks 10

Answer

Mobile adhoc network : Refer Q. 5.20, Page 5–26A, Unit-5.

Applications of MANET :

Data networks : Refer Q. 5.22, Page 5– 28A, Unit-5.

Application of mobile adhoc network :

1. Mobile adhoc networks also enable distributed control applications, with remote plants, sensors and actuators linked together via wireless communication channels.
2. Such networks allow coordination of unmanned mobile units, and greatly reduce maintenance and reconfiguration costs over distributed control systems with wired communication links.

3. Mobile adhoc networks can be used to support coordinated control of multiple vehicles in an automated highway system, remote control of manufacturing and other industrial processes, and coordination of unmanned airborne vehicles for military applications.

Que 5.24. **Write a short note on Li-Fi communication.**

Answer

1. Light Fidelity (Li-Fi) is a bidirectional, high-speed and fully networked wireless communication technology similar to Wi-Fi. The term was coined by Harald Haas and is a form of visible light communication and a subset of optical wireless communications (OWC) and could be a complement to RF communication (Wi-Fi or cellular networks), or even a replacement in contexts of data broadcasting.
2. It is wire and UV visible-light communication or infrared and near-ultraviolet instead of radio-frequency spectrum, part of optical wireless communications technology, which carries much more information and has been proposed as a solution to the RF-bandwidth limitations.
3. Li-Fi has the advantage of being useful in electromagnetic sensitive areas such as in aircraft cabins, hospitals and nuclear power plants without causing electromagnetic interference. Both Wi-Fi and Li-Fi transmit data over the electromagnetic spectrum, but whereas Wi-Fi utilizes radio waves, Li-Fi uses visible light.
4. Li-Fi is expected to be ten times cheaper than Wi-Fi. Short range, low reliability and high installation costs are the potential downsides.
5. Like Wi-Fi, Li-Fi is wireless and uses similar 802.11 protocols; but it uses visible light communication (instead of radio frequency waves).
6. Standard defines three PHY layers with different rates :
   i. The PHY I was established for outdoor application and works from 11.67 kbit/s to 267.6 kbit/s.
   ii. The PHY II layer permits reaching data rates from 1.25 Mbit/s to 96 Mbit/s.
   iii. The PHY III is used for many emissions sources with a particular modulation method called colour shift keying (CSK). PHY III can deliver rates from 12 Mbit/s to 96 Mbit/s.
7. The modulation formats recognized for PHY I and PHY II are on-off keying (OOK) and variable pulse position modulation (VPPM).
8. The Manchester coding used for the PHY I and PHY II layers includes the clock inside the transmitted data by representing a logic 0 with an OOK symbol "01" and a logic 1 with an OOK symbol "10", all with a DC component. The DC component avoids light extinction in case of an extended run of logic 0's.

Que 5.25. **Write a short note on ultra-wideband communication.**

Answer

1. Ultra-wideband (UWB) is for indoor and short-range outdoor communication.
2. It is a rapidly emerging wireless technology that promises data rates well beyond those possible with currently deployed technologies such as 802.11*a*, *b*, *g* and WiMAX.
3. UWB can be used in both commercial and military applications.
4. The method of transmission employed by UWB is totally different to that used by most other wireless technologies in use today.
5. Rather than using a specified frequency with a carrier, the technique that is used by traditional transmissions, UWB uses what can be termed time domain electromagnetic.

PART-4

Mobile Data Networks, Wireless Standard IMT 2000.

Questions-Answers
Long Answer Type and Medium Answer Type Questions

Que 5.26. **What is mobile data networks ? Give classification of mobile data networks.**

Answer

1. By mobile data networks we refer to those services, technologies, and standards that are related to data services over wide area coverage, areas spanning more than the local area or campus.
2. In addition to traditional mobile data services, SMS services can also be considered as a part of these systems.
3. As shown in Fig. 5.26.1 we can classify mobile data networks into three categories : independent, shared and overlay networks based on the way they relate to the cellular infrastructure.

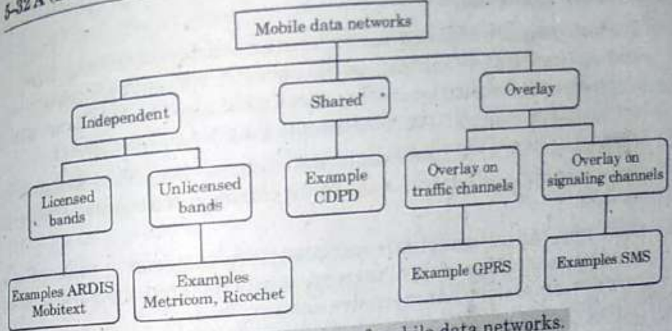


Fig. 5.26.1. Classification of mobile data networks.

i. **Independent mobile data :**

1. Independent networks have their own spectrum that is not coordinated with any other service and their own infrastructure that is not shared with any other service.
2. These networks are divided into two groups according to the status of their operating frequency band.
3. The first group uses independent spectrum in licensed bands. Examples of such networks are ARDIS and Mobitex and historically they were the first mobile data services that were introduced.
4. The second group of independent mobile data networks makes use of unlicensed spectrum that is shared among a variety of applications and users. Metricom's was an example of this service.
5. This service was deployed in airports and some metropolitan areas for wireless internet access.

ii. **Shared mobile data :**

1. These networks share the spectrum and part of the infrastructure with an already existing voice-oriented analog service. The service operate in the same radio channels used for analog voice, but they have their own air-interface and MAC protocols.
2. In addition to dedicated channels for data, these mobile data services can also use the available unused voice channels.
3. These systems share an existing system infrastructure, therefore the initial investment is not huge and it could be made as gradually as possible.
4. Initial deployment could be made in areas where there is subscriber demand and subsequent penetration into other areas is considered as the customer base enlarges.
5. The CDPD service, which shares spectrum and part of the infrastructure with AMPS, is an example of such networks.
6. It does have an independent air-interface and MAC layer along with additional infrastructure required for operation of data services.

iii. Overlay mobile data :

1. The last group of mobile data networks in an overlay on existing networks and services. This means that the data service will not only make use of the spectrum allocated for another service but also the MAC frames and air-interface of an existing voice-oriented digital cellular system.
2. GPRS and GSM's SMS are examples of such overlays. They make use of free time slots available within the traffic channels and signaling channels in GSM.
3. This way, the amount of new infrastructure required is reduced to a bare minimum. Most of the extra components required and implemented in software, making it inexpensive and easy to deploy.
4. GPRS type of service uses computer keyboards to communicate longer messages and SMS use the cellular phone dialing keypad to communicate short messages.

Que 5.27. **Write the short notes on following :**

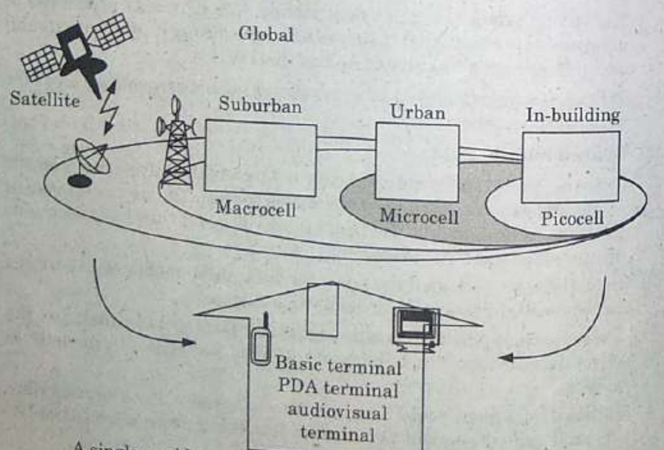
i. Wireless standard IMT 2000.

ii. RAKE Receiver.

AKTU 2014-15, Marks 10

Answer

i. Wireless standard IMT 2000 :



A single world standard to support all wireless environments.

Fig. 5.27.1. Scope of mobile transmission technologies (RTTs) for IMT-2000.

1. The vision for an IMT-2000 system and its capabilities is summarized which illustrates that IMT-2000 will provide capabilities constituting

significant improvements over the current mobile systems, especially in terms of global mobility for the users and support of services like high speed data, multimedia, and internet.

2. Since, however, it is generally accepted that these IMT-2000 capabilities will to a large extent be achieved by evolving existing wireline and wireless networks, IMT-2000 will be a family of systems rather than a single, monolithic network.
3. **IMT-2000 vision :**
   i. Common spectrum worldwide (1.8 - 2.2 GHz band).
   ii. Multiple radio environments (cellular, cordless, satellite, LANs).
   iii. Wide range of telecommunication services (voice, data, multimedia, internet).
   iv. Flexible radio bearers for increased spectrum efficiency.
   v. Data rates up to 2 Mb/s (phase 1) for indoor environments.
   vi. Maximum use of IN capabilities (for service provision and transport).
   vii. Global seamless roaming.
   viii. Enhanced security and performance.
   ix. Integration of satellite and terrestrial system.
4. A global standard to satisfy market demand for mobile services in the 21st century.
5. In scope, IMT-2000 service environments will address the full range of mobile and personal communication applications shown in Fig. 5.27.1 : in building (picocell), urban (microcell), suburban (macrocell), and global (satellite), as well as communications types that include voice, data and image.
6. Support communication needs for developing countries in the form of fixed wireless access (FWA) application is also included in the scope of IMT-2000.

ii. RAKE Receiver : Refer Q. 4.14, Page 4–14A, Unit-2.

Que 5.28. **Explain general requirements for radio transmission technologies (RTTs) for IMT-2000.**

Answer

Radio transmission technologies (RTTs) for IMT-2000 :

1. As opposed to second-generation wireless systems, which are generally optimized for circuit-switched services in a single-radio operating environment, IMT-2000 is intended to provide access, by means of one or more radio links, to a wide range of telecommunication services (from voice to multimedia and internet), operating in a variety of radio environments.

2. The presence of a wide variety of radio operating environments translates into a number of factors that impact the choice of candidate(s) for radio access technologies.
3. Some examples of these factors are as follows :
i. Speed of movement (from zero to very high).
ii. User density (city centers to remote areas).
iii. Physical environment (indoor, urban, suburban, rural, maritime, aeronautical).
iv. Coverage (continuous or islands).
v. Delivery mode (terrestrial or satellite).
4. Another basic requirement is the need for IMT-2000 terminals to be able to roam globally and receive telecommunications services offered by IMT-2000 networks.
5. Key features of the radio access for IMT-2000 therefore include the following :
i. High level of flexibility.
ii. Cost-effectiveness in all operating environments.
iii. Commonality of design worldwide.
iv. Operation within the designated IMT-2000 frequency bands.

Que 5.29. **Draw and explain the component of mobile network structure of IMT-2000.**

AKTU 2015-16, Marks 7.5

AKTU 2017-18, Marks 10

Answer

A. General requirements :

1. IMT-2000 may be implemented as a stand-alone network with gateways and interworking units towards the supporting networks, in particular toward PSTN, ISDN, packet data networks (e.g., Internet), and B-ISDN (broadband ISDN).
2. This is comparable to the current implementations of public land mobile networks, and it is also a solution in cases of fixed and radio networks that are run by different operators. However, IMT-2000 may also be integrated with the fixed networks.
3. One of the key service objectives of IMT-2000 is to enable the provision of multimedia services (in circuit and packet mode operation) and internet services (high speed packet data).
4. Requirements for network functions must therefore take into account the support of multimedia services.

5. IMT-2000 radio resources must be shared among circuit mode as well as packet mode services.
6. In addition, IMT-2000 systems should support global roaming and the virtual home environment concept, that is, the user will be provided with a comprehensive set of services and features that have the same look and feel regardless of whether they are accessed from the home or a visited network.

B. The ITU three-stage process for interface specification :

1. The starting point for specification of interfaces in terms of detailed protocols is the set of services and service features.
2. These are then translated into signaling requirements in terms of the following :
i. A functional architecture containing required functional entities (FE) and relationships.
ii. Information flows between the functional entities and associated information elements (IE) to support various service features.
iii. System description language (SDL) diagrams that graphically capture the information flows and functional entity actions (FEA).
3. Detailed protocols for the interfaces identified in the last step of a stage 2 process are then developed by protocol experts using the SDL diagrams.

C. IMT-2000 functional architecture :

1. The broad classes of functions that need to be supported by an IMT-2000 system are illustrated in Fig. 5.29.1 which recognizes that separation of call and connection control functions is desirable for supporting multimedia and advanced services in IMT-2000.

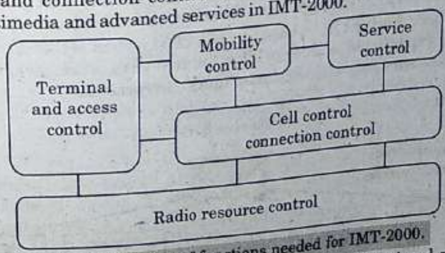


Fig. 5.29.1. Classes of functions needed for IMT-2000.

2. ITU Study Group 11 has developed a detailed functional model for IMT-2000, which contains functional entities and shows the relationships between these FEs.
3. Based on this functional model, detailed information flows have been developed which depict flow of messages between specific FEs to support individual IMT-2000 service features, and also the required information elements that need to be transported as part of these information flows.

4. Together with the SDL diagrams, these form the stage 2 service descriptions for IMT-2000 and will be used by protocol experts to develop detailed protocol definitions across various IMT-2000 interfaces.

Que 5.30. **Describe the key features of IMT-2000.**

Answer

IMT-2000 has the following key features :

1. **Flexibility :**
   i. With the large number of mergers and consolidations occurring in the mobile industry, and the move into foreign markets, operators wanted to avoid having to support a wide range of different interfaces and technologies. This would surely have hindered the growth of 3G worldwide.
   ii. The IMT-2000 standard addresses this problem, by providing a highly flexible system, capable of supporting a wide range of services and applications.
   iii. The IMT-2000 standard accommodates five possible radio interfaces based on three different access technologies (FDMA, TDMA, and CDMA), as shown in Fig. 5.30.1.
2. **Affordability :** There was agreement among industry that 3G systems had to be affordable, in order to encourage their adoption by consumers and operators.
3. **Compatibility with existing systems :** IMT-2000 services have to be compatible with existing systems. 2G systems, such as the GSM standard will continue to exist for some time and compatibility with these systems must be assured through effective and seamless migration paths.
4. **Modular design :** The vision for IMT-2000 system is that they must be easily expandable in order to allow for growth in users, coverage areas, and new services, with minimum initial investment.

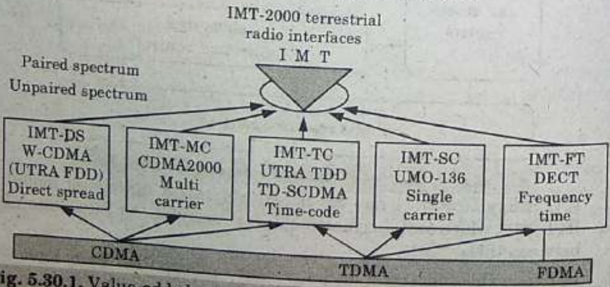


Fig. 5.30.1. Value-added services and worldwide applications development on the basis of one single standard accommodating five possible radio interfaces based on three technologies.

PART-5

Introduction to 4G and Concept of NGN.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 5.31. **Describe the working of 4G wireless network and also explain the principle technologies used by it.**

OR

Explain in brief 4G technologies and also compare with 1G, 2G and 3G technology.

AKTU 2015-16, Marks 05

OR

Explain 4G technology in detail and also compare it with 1G, 2G, and 3G technology.

AKTU 2017-18, Marks 10

Answer

A. **4G :**

1. 4G stands for the fourth generation of cellular wireless standards. It is a successor to 3G and 2G families of standards.
2. Speed requirements for 4G service set the peak download speed at 100 Mbit/s for high mobility devices (such as cellphones) and 1Gbit/s for low mobility devices (such as stand-alone wireless modems).
3. A 4G system is expected to provide a comprehensive and secure all IP based solution where facilities such as ultra-broadband internet access, IP telephony, gaming services, and streamed multimedia may be provided to users.
4. The wireless telecommunications industry as a whole has generally adopted the term 4G as a short hand way to describe those advanced wireless technologies that, among other things, are based on or employ wide channel OFDM technology and all IP based architecture.
5. In 4G the user has freedom and flexibility to select any desired service with reasonable QOS and affordable price, anytime and anywhere.

B. **Principal technologies :**

i. **Physical layer transmission techniques :**

a. **MIMO :** To attain ultra high spectral efficiency by means of spatial processing including multi-antenna and multi-user MIMO.

b. **Frequency domain equalization :** For example, multi-carrier modulation (OFDM) in the downlink or single carrier frequency domain-equalization (SC-FDE) in the uplink. To exploit the frequency selective channel property without complex equalization.

c. **Frequency domain statistical multiplexing :** For example OFDMA or single carrier FDMA in the uplink. Variable bit rate by assigning different sub-channels to different users based on the channel conditions.

d. **Turbo principle error-correcting codes :** To minimize the required SNR at the reception side.

ii. **Channel dependent scheduling :** To utilize the time-varying channel.

iii. **Link adaptation :** Adaptive modulation and error-correcting codes.

C. **Comparison :**

The following can be stated as the major differences in the generations :

1G—Cell structure, analog communication

2G—Cell structure, digital communication, convolution coding, power control

3G—Hierarchical cell structure, turbo coding, hybrid automatic repeat request (HARQ)

4G— Smart antenna, adaptive systems over above secnario.

Que 5.32. **Give the objectives of the 4G wireless communication standard.**

Answer

1. Flexible channel bandwidth, between 5 to 20 MHz, optionally up to 40 MHz.
2. A nominal data rate of 100 Mbit/s while the client physically moves at high speeds relative to the station, and 1Gbit/s while clients and station are in relatively fixed positions as defined by the ITU-R.
3. A data rate of at least 100 Mbit/s between any two points in the world.
4. Peak link spectral efficiency of 15 bit/s/Hz in the downlink and 6.75 bit/s/Hz in the uplink.
5. System spectral efficiency of up to 3 bit/s/Hz/cell in the downlink and 2.25 bits/s/Hz/cell for indoor usage.
6. Smooth handoff across heterogeneous networks.
7. Seamless connectivity and global roaming across multiple networks.
8. High quality of service for next generation multimedia support (real time audio, high speed data, HDTV video content, mobile TV, etc.)
9. Interoperability with existing wireless standards.
10. An all IP, packet switched network.

Que 5.33. **Discuss a complete model of Next Generation Network (NGN) systems for mobile communication. How it is useful for network security.**

AKTU 2017-18, Marks 10

OR

What is 4G system ? And explain the concept of Next Generation Networks.

AKTU 2018-19, Marks 10

Answer

4 G : Refer Q. 5.31, Page 5–38A, Unit-5.

NGN :

1. The IP telecommunication network architecture and software layer architecture are shown in Fig. 5.33.1 in which bearer control layer and logical bearer network perform network control together.
2. NGN layered architecture can be described from five function layers :

i. **Application layer :** It contains the typical middleware for authorization, accounting, directory, search and navigation for millions of users.

ii. **Network control layer :**

1. It aims at overcoming the bottleneck problems at edge nodes or servers and it is composed of a series of control agents for admission control, call set up and end-to-end QoS control through available bandwidth detection, local information control, class priority and intelligent scheduling.
2. Multicast and anycast group managements will be implemented to leverage the load for admission control or service /message distributions.

iii. **Adaptation layer :**

1. It supports different network configurations and network mobility. This layer can provide soft switching between networks on different level such as IPv4, IPv6, ATM, ethernet, WLAN, WMAN and 3G networks.
2. It supports both packet and circuit switching and provides interconnection between the two switching networks.

iv. **Network transmission layer :**

1. It provides the effective end-to-end QoS control for real-time requests and flows through integration of parameterized QoS control and class priority control.
2. This is particularly important to resolve the bottleneck problems such as multipath routing that enables the multiple choices for the path and anycast routing that enables the selection from different (replicated) servers.

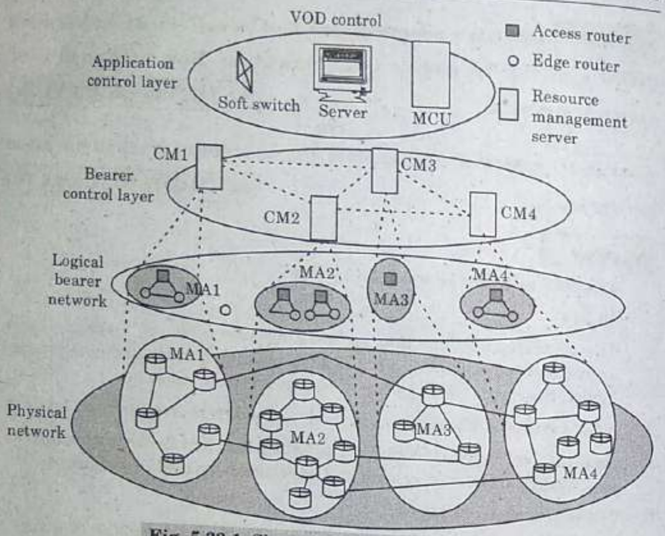


Fig. 5.33.1. Shows NGN network architectures.

v. **Management layer :** It provides web-based GUI browser and wireless connection information such as the data access using XML and web based visualization for data presentation, monitoring, modification and decision making in NGN.

Next generation network (NGN) is useful for network security because it offers the possibility to provide a detailed service control and security within network, so that networks are aware of both the services that they are carrying and the users for whom they are carrying them, and are able to respond in different ways to this information while Internet aims to provide basic transmission.

Que 5.34. **Why next generation network is important ? Explain the next generation network in detail.**

AKTU 2014-15, 2016-17; Marks 10

Answer

A. **NGN layered architecture :** Refer Q. 5.33, Page 5–40A, Unit-5.

B. **Functional requirements of NGN are as follows :**

i. **Very high-speed and high-quality transmission :**

1. Next-generation mobile communication systems should be able to handle a large volume of multimedia information like downloading an e-book or sending a report file.

2. This would be possible by various means like transmitting data at 50-100 Mbps, having asymmetric speeds in up and down links with QoS mechanisms (*i.e.*, efficient encoding, error detection and correction techniques, voice equalizer) at low, affordable and reasonable operating costs.

ii. **Open platform :**

1. Next-generation networks should be open regarding mobile phone platform, service nodes and mobile network mechanisms.
2. This means that the user can freely select protocols, applications and networks.
3. Location and charging information can be shared among networks and applications.

iii. **Flexible and variable service functions :**

1. Next-generation networks should be seamless with regard to the medium, whether it is wireless or optical fiber or satellite link with regard to corresponding hosts or service providers as well as have interconnectivity with other networks like GSM and CDMA.

Que 5.35. **Describe all network elements in network architecture of NGN.**

Answer

NGN network architecture : The detailed network architecture for next-generation networks is shown in Fig. 5.35.1.

Network elements : The network architecture of NGN shown in Fig. 5.35.1 involves many interconnected elements which are described below :

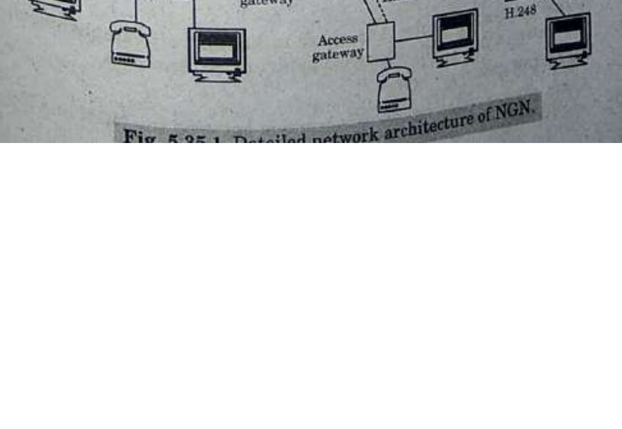


Fig. 5.35.1. Detailed network architecture of NGN.

Packet based networks :

i. Trend is to use IP based networks over various transport possibilities (ATM, SDH, WDM ...).

ii. IP based networks must offer guarantee of quality of service (QoS) regarding the real time characteristic of voice, video and multimedia.

Access gateways :

i. Allows the connection of subscriber lines to the packet network.

ii. Converts the traffic flows of analogue access (Pots) or 2 Mb/s access devices into packets.

iii. Provides subscriber access to NGN network and services.

Trunking gateways :

i. Allows interworking between classical TDM telephony network and packet-based NGN networks.

ii. Converts TDM circuits/trunks (64 kbps) flows into data packets, and vice versa.

Soft switch/MGC :

i. Referred to as the call agent or media gateway controller (MGC).

ii. Provides the "service delivery control" within the network.

iii. In charge of call control and handling of media gateways control (Access and/or trunking) via H.248 protocol.

iv. Performs signaling gateway functionality or uses a signaling gateway for interworking with PSTN N7 signaling network.

v. Provides connection to intelligent network/applications servers to offer the same services as those available to TDM subscribers.

Application server (AS) :

i. A unit that supports service execution, *e.g.*, to control call servers and NGN special resources (*e.g.*, media server, message server).

H.248 Protocol :

i. Known also as MEGACO : standard protocol, defined by ITU-T, for signaling and session management needed during a communication between a media gateway, and the media gateway controller managing it.

ii. H.248/MEGACO allows to set up, keep and terminate calls between multiple endpoints as between telephone subscribers using the TDM.

SIP :

Session initiation protocol in order to handle call establishment, maintenance and termination from packet mode terminals.

Signaling gateway (SG) :

A unit that provides signaling conversion between the NGN and the other networks (*e.g.*, STP in SS7).

ENUM :

Electronic numbering protocol that allows to establish a correspondence between the traditional telephone numbering (E.164) and the network address related to the packet mode networks.

MPLS :

Multiprotocol label switch or protocol that assigns labels to information packets in order to allow the node routers to treat and route flows in the network paths according to established priority for each category.

CAC :

Call acceptance control function in order to accept/reject traffic in the network that allows guarantee of QoS for services with a service level agreement.

BGP :

Border gateway protocol to negotiate flow routing procedures and capacities across different NGN network domains.

Que 5.36. **Explain various applications and service of next generation network (NGN).**

AKTU 2015-16, Marks 7.5

Answer

NGN services :

1. Although we have a feel for the types of service characteristic that will be important in an NGN environment, no one really knows what the "killer applications" will be.
2. A variety of services, some already available, others still at the conceptual stage, have been linked to NGN initiatives and considered likely candidates for NGN implementations.
3. While some of these services can be offered on existing platforms, others benefit from the advanced control, management, and signaling capabilities of NGNs.
5. Most traditional services related to basic access/transport/routing/switching services, basic connectivity/resources and session control services, and various value-added services.

Applications :

1. Specialized resource services (*e.g.*, provision and management of transcoders.
2. Processing and storage services (*e.g.*, file servers.
3. Middleware services (*e.g.*, brokering, security, licensing, etc.)
4. Application-specific services (*e.g.*, business applications, e-commerce applications, interactive video games, etc.).

5. Content provision services that provide or broker information content (*e.g.*, electronic training, information push services etc.).
6. Interworking services for interactions with other type of applications, services, networks, protocols or formats (*e.g.*, EDI translation).
7. Management services to maintain, operate and manage communications/computing networks and services.

VERY IMPORTANT QUESTIONS

Following questions are very important. These questions may be asked in your SESSIONALS as well as UNIVERSITY EXAMINATION.

Q. 1. Draw the GSM architecture and also explain radio subsystem in mobile radio communication.

Ans. Refer Q. 5.2.

Q. 2. Explain signal processing and GSM operations from speech input to speech output with diagram.

Ans. Refer Q. 5.4.

Q. 3. Describe the forward CDMA channel and reverse CDMA channel using proper block diagram.

Ans. Refer Q. 5.10.

Q. 4. How does CDMA technology work in principle ? Give detailed features of GSM and CDMA mobile standards.

Ans. Refer Q. 5.12.

Q. 5. Write a short note on IMT-2000 and UMTS.

Ans. Refer Q. 5.14.

Q. 6. Explain mobile adhoc network in wireless communication.

Ans. Refer Q. 5.20

Q. 7. Draw and explain the component of mobile network structure of IMT-2000.

Ans. Refer Q. 5.29.

Q. 8. Describe the working of 4G wireless network and also explain the principle technologies used by it.

Ans. Refer Q. 5.31.

Q. 9. Discuss a complete model of Next Generation Network (NGN) systems for mobile communication. How it is useful for network security.

Ans. Refer Q. 5.33.

Q. 10. Explain various applications and service of next generation network (NGN).

Ans. Refer Q. 5.36.

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