CHAPTER – II

Literature Survey on Frequency Shift Keying for Nakagami and Rician Fading Channel in Communication System

2.1 Communication

A communication enters our daily lives in so many different ways. The telephone in our homes and offices make it possible for us to communicate with other, no matter how far away. The radio and television sets in our living rooms bring us entertainment from near as well as far-away places. Communication by radio or satellite provides the means for ships on the high seas, aircraft in flight, and rockets and exploratory probes in space to maintain contact with their home bases. Communication keeps the weather forecaster informed of atmospheric conditions that are measured by signals of sensors. Communications make possible for computers to interact [168]. The list of applications involving the use of communications in one way or another goes on.

In the most fundamental sense, communication involves implicitly the transmission of information from one point to another through a succession of processes, as described here:

1. The generation of a thought pattern or image in the mind of an originator;

2. The description of that thought pattern or image, with a certain measure of precision, by a set of aural or visual symbols;

3. The encoding of these symbols in a form that is suitable for transmission over a physical medium of interest; and
4. The transmission of the encoded symbols to the desired destination and the decoding and reproduction of the initial symbols.

### 2.2 Communication System

In telecommunication, a communication system is a collection of individual communication networks, transmission systems, relay stations, tributary stations, and data terminal equipment (DTE) usually capable of interconnection and interoperation to form an integrated one. The components of a communication system serve a common purpose, and are technically compatible, use common procedures to respond to controls and to operate in unison. Telecommunication is a method of communication for sports broadcasting, mass media, journalism, etc.

### 2.3 History and Applications

Data (mainly but not exclusively informational) has been sent via non-electronic (e.g. optical, acoustic, mechanical) means since the advent of communication. Analog signal data has been sent electronically since the advent of the telephone. However, the first data electromagnetic transmission applications in modern time were telegraphy (1809) and teletypewriters (1906), which are both digital signals. The fundamental theoretical work in data transmission and information theory by Harry Nyquist, Ralph Hartley, Claude Shannon and others during the early 20th century, was done with these applications in mind.

Data transmission is utilized in computers in computer buses and for communication with peripheral equipment via parallel ports and serial ports such as Rs-232 (1969), Fire wire (1995) and USB (1996). Data transmission is utilized in computer networking equipment such as, modems (1940), local area networks (LAN) adapters (1964), repeaters, hubs, microwave links, wireless networks access points (1997), etc. The principle of data transmission is also utilized in storage media for Error detection and correction since 1951[168].

In telephone networks, digital communication is utilized for transferring many phone calls over the same copper cable or fiber cable by means of Pulse code modulation (PCM), i.e., sampling and digitization, in combination with Time division multiplexing (TDM) (1962). Telephone exchanges have become digital and software
controlled, facilitating many value added services. For example, the first telephone exchange was presented in 1976. Since late 1980s, digital communication to the end user has been possible using integrated services Digital Network (ISDN) services. Since the end of 1990s, broadband access techniques such as, cable modems, fiber-to-the-building (FTTB) and fiber-to-the-home (FTTH) have become widespread to small offices and homes. The current tendency is to replace traditional telecommunication services by packet mode communication such as, IP telephony and IPTV [169].

 Transmitting analog signals digitally allows for greater signal processing capability. The ability to process a communication signal means that errors caused by random processes can be detected and corrected. Digital signals can also be sampled instead of being continuously monitored. The multiplexing of multiple digital signals is much simpler to the multiplexing of analog signals.

Because of all these advantages and recent advances in wideband, communication channels and solid-state electronics have allowed scientists to fully realize these advantages and hence, digital communication has grown rapidly. Digital communication is quickly edging out analog communication because of the vast demand to transmit computer data, and the ability of digital communication to do so.

The digital revolution has also resulted in many digital telecommunicating applications where the principles of data transmissions are applied. Examples are second generation (1991) and later, cellular telephone, video conferencing, digital TV (1998), digital radio (1999), telemetry, etc.

Advantages:

1. It is fast and easier to operate.

2. The messages can be stored in the device for longer times, without being damaged, unlike paper files that easily get damaged or attacked by insects;

3. Digital communication can be done over large distances through internet.

4. It is comparatively cheaper and the work which requires a lot of people can be done simply by one person as folders and other such facilities can be maintained;
5. It removes semantic barriers because the written data can be easily changed to different languages using software; and

6. It provides facilities like video conferencing which save a lot of time, money and effort.

**Disadvantages:**

1. It is unreliable as the messages cannot be recognized by signatures. Though software can be developed for this, yet the software can be easily hacked;

2. Sometimes, the quickness of digital communication is harmful as messages can be sent with the click of a mouse. The person does not think and sends the message at an impulse;

3. Digital communications have completely ignored the human touch. A personal touch cannot be established because all the computers will have the same font;

4. The establishment of digital communication causes degradation of the environment in some cases. "Electronic waste" is an example. The vibes given out by the telephone and cell phone towers are so strong that they can kill small birds. In fact, the common sparrow has vanished due to so many towers coming up as the vibrations hit them on the head;

5. Digital Communication has made the whole world to be an "office". The people carry their work to places where they are supposed to relax. The whole world has been made into an office. Even in the office, digital communication causes problems because personal messages can come on your cell phone, internet, etc. and

6. Many people misuse the efficiency of digital communications. The sending of hoax messages, the usage by people to harm the society, etc. cause harm to the society on the whole.

**2.4 Type of Communication System**

Communication systems have changed with the advancement of technology. The need for communication has been around since the beginning of humanity. In today's
fast-paced world, there is a variety of communication system available around us. The first primary use of video communication occurred when the television was introduced and became widely used. Video was sent from television stations to satellites which redirected the signal to local antennas and was then broadcast to the antennas located on houses or on top of television sets. The communication system may be broadly classified as:

I. **Voice Communication System**

Voice communication systems require individuals to speak into a receiver/transmitter which transmits the voice as a signal to another receiver/transmitter. This type of communication system has been available since the invention of the telephone, and is still widely used today. Over time, this communication type has been experienced many developments. Analog signals were originally used to transmit the voice over phone lines. In the 1960s, Bell systems were able to produce the T-1 carrier system, which allowed the digital transmission of a voice communication system [168]. Today the carrier system is used in the transmission of Internet signals.

II. **Data Communication System**

The world of Internet and email created a new type of communication system which relies on the use of data transmissions. Data is entered into a computer and is then sent from one IP address to another IP address. Data communication was originally sent over analog phone lines. As computers become faster and technology improved, digital methods of delivery soon appeared. The synchronous optical network was established and is the foundation of the broadband integrated services digital network (ISDN). The ISDN is the method used for most Internet connections today is able to transfer data at the rate of 9.953 gigabits per second [170].

III. **Video Communication System**

Video communication is a mixed form of data and voice communication. A moving picture is created as a part of the data file and can be used to transfer information provided by a form of voice communication. Video communication can be sent over the same communication connections as data, but it often requires more time to send/receive transmission because the files are larger. The first primary use of video
communication occurred when the television was introduced and became widely used. Video was sent from television stations to satellites which redirected the signal to local antennas and was then broadcast to the antennas located on houses or on top of television sets.

In accordance with signal based communication system may be classified as analog communication and digital communication

**Analog Communication:** Analog signals are signals with continuous values. Analog signals are continuous in both time and value. Analog signals are used in many systems, although the use of analog signals has declined with the advent of cheap digital signals. All natural signals are Analog in nature. Analog systems are less tolerant to noise, make good use of bandwidth, and are easy to manipulate mathematically. However, analog signals require hardware receivers and transmitters that are designed to perfectly fit the particular transmission. If you are working on a new system, and you decide to change your analog signal, you need to completely change your transmitters and receivers.

**Digital Communication:** Digital signals are discrete in time and value. Digital signals are signals that are represented by binary numbers, "1" or "0". The 1 and 0 values can correspond to different discrete voltage values, and any signal that *doesn't quite fit* into the scheme just gets rounded off. Digital signals are sampled, quantized & encoded version of continuous time signals which they represent. In addition, some techniques also make the signal undergo encryption to make the system more tolerant to the channel.

**Elements of Communications System**

The purpose of a communication system is to transmit information bearing signals from a source, located at one point, to a user destination, located at another point some distance away. When the message produced by the source is not electrical in nature, which is often the case, an input transducer is used to convert it into a time-varying electrical signal called the message signal. By using another transducer connected to the output end of the system, a "distorted" version of the message is re-created in its original form so that it is suitable for delivery to the user destination. The distortion mentioned here is due to inherent limitations on the communication system. Figure 2.1 is a block diagram of a communication system consisting of three
basic components: transmitter, channel, and receiver. The transmitter has the function of processing the message signal into a form suitable for transmission over the channel (such an operation is called modulation.) The function of the channel is to provide a physical connection between the transmitter output and the receiver input.

![Block Diagram of Communication System](image)

Figure 2.1, Generalized Block Diagram of Communication System

The function of the receiver is to process the received signal so as to produce an "estimate" of the original message signal; this second operation is called detection or demodulation.

There are two types of channels, namely, point-to-point channels and broadcast channels. Examples of point-to-point channels include wire lines, microwave links, and optical fibers. Wire lines operate by guided electro-magnetic waves; they are used to local telephone transmission. In micro-wave links, the transmitted signal is radiated as an electromagnetic wave in free space; microwave links are used in long-distance telephone transmission. An optical fiber is a low-loss; well controlled guided optical medium, Optical fibers are used in optical communications. Although these three channels operate differently, they all provide a physical medium for the transmission of signals from one point to another point and hence, called the term "point-to-point channels" and Broadcast channels, on the other hand, provide capability where many receiving stations may be reached simultaneously from a single transmitter [171]. An example of a broadcast channel is satellite in geostationary orbit, which covers about one third of the earth's surface. Thus, three such satellites provide a complete coverage of the earth surface, except for the Polar Regions.
An optical communication system is any form of telecommunication that uses light as the transmission medium. Equipment consists of a transmitter, which encodes a message into an optical signal, a channel, which carries the signal to its destination, and a receiver, which reproduces the message from the received optical signal. Fiber optical communication systems transmit information from one place to another by sending light through an optical fiber. The light forms an electromagnetic carrier wave that is modulated to carry information.

A radio communication system is composed of several communication subsystems that give exterior communication capabilities. A radio communication system comprises a transmitting conductor in which oscillatory electrical signals are produced and which is arranged to cause such oscillation electrical signal to be propagated through the free space medium from one point to another remote point and a receiving device at such distant point adapted to be excited by the oscillatory electrical signals propagated from the transmitter. Line communication systems operate by impressing a modulated carrier signal on power wires. Different types of power line communications use different frequency bands, depending on the signal transmission characteristics of the power wiring used. Since the power wiring system was originally intended for transmission of AC power, the power wire circuits have only a limited ability to carry higher frequencies. The propagation problem is a limiting factor for each type of power line communication. A duplex communication system is a system composed of two connected parties or devices which can communicate with one another in both directions. The term duplex is used while describing communications between two parties or devices. Duplex systems are employed in nearly all communication networks, either to allow for a communications’ "two-way street" between two connected parties or to provide a "reverse path" for monitoring and remote adjustment of equipment in the field. [96]

A tactical communication system is a communication system that (a) is used within, or in direct support of tactical forces, (b) is designed to meet the requirements of changing tactical situations and varying environmental conditions, (c) is designed to provide secure communications such as, voice, data and video, among mobile users to facilitate command and control within, and in support of tactical forces, and (d) usually requires extremely short installation times, usually on the order of hours, in order to meet the requirements of frequent relocation.
2.5 Digital Communication System

Digital transmission or digital communication is the physical transfer of data (a digital bit stream) over a point-to-point or point-to-multipoint communication channel. Examples of such channels are copper wires, optical fibers, wireless communication channels, and storage media. The data are represented as an electromagnetic signal such as, an electrical voltage, radio-wave, microwave, or infrared signal.

While analog transmission is the transfer of a continuously varying analog signal, digital communication is the transfer of discrete messages. The messages are either represented by a sequence of pulses by means of a line code (baseband transmission), or by a limited set of continuously varying wave forms (pass band transmission); using a digital modulation (also known as detection) which is carried out by modem equipment. According to the most common definition of digital signal, both baseband and pass band signals representing bit streams are considered as digital transmission, while an alternative definition of digital signal considers the baseband signal as digital, and pass band transmission of digital data as a form of digital-to-analog conversion.

Transmitted messages may be digital messages originating from a data source, for example a computer or a keyboard. It may also be an Analog signal such as a phone call or a video signal, digitized into a bit stream, for example using pulse code modulation (PCM) or more advanced source coding (analog-to-digital conversion and data compression) schemes. This source coding and decoding is carried out by the equipment designed for the purpose.

2.5.1 Multichannel Digital Communications in Adaptive White Gaussian Noise (AWGN) Channels

In this section, we confine our attention to multichannel signaling over fixed channels that differ only in attenuation and phase shift. The specific model for the multichannel digital signaling system may be described as follows. The signal waveforms, in general, are expressed as
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\[ s_{m}^{(n)}(t) = \text{Re}\left[ s_{m}^{(n)}(t) e^{j2\pi f_{c}t} \right], \quad 0 \leq t \leq T \]

\[ n = 1, 2, \ldots L, \quad m = 1, 2, \ldots M \]  \hspace{1cm} (2.1)

Where, \( L \) is the total number of channels and \( M \) is the total number of waveforms. The waveform is assumed to have equal energy and to be equally probable a priori. The signal waveform \( \{ s_{m}^{(n)}(t) \} \) transmitted over the \( L \) channels are scaled by the attenuation factors \( \{ \alpha_{n} \} \) phase-shifted by \( \{ \phi_{n} \} \) and corrupted by additive noise. The equivalent low-pass signals received from the \( L \) channels may be expressed as:

\[ r_{i}^{(n)}(t) = \alpha_{n} e^{-j\phi_{n}} s_{m}^{(n)}(t) + z_{n}(t), \quad 0 \leq t \leq T \]

\[ n = 1, 2, \ldots L \quad m = 1, 2, \ldots M \] \hspace{1cm} (2.2)

where, \( \{ s_{m}^{(n)}(t) \} \) is the equivalent low-pass transmitted waveform and \( \{ z_{n}(t) \} \) represent the additive noise processes on the \( L \) channels. It is assumed that \( \{ z_{n}(t) \} \) are mutually statistically independent and identically distributed Gaussian noise random processes.

We consider two types of processing at the receiver is considered namely coherent detection and non coherent detection. The receiver for coherent detection estimates the channel parameters \( \{ \alpha_{n} \} \) and \( \{ \phi_{n} \} \) and uses the estimate in computing the decision variables. Suppose we define \( g_{n} = \alpha_{n} e^{-j\phi_{n}} \) and let \( \hat{g}_{n} \) be the estimated of \( g_{n} \) the multichannel receiver correlates each of the \( L \) receiver signals with a replica of the corresponding estimates \( \{ \hat{g}_{n}^{*} \} \) and sums the resulting signals. Thus, the decision variables for coherent detection are the correlation metrics:

\[ CM_{m} = \sum_{n=1}^{L} \text{Re}\left[ \int_{0}^{T} r_{i}^{(n)}(t) s_{m}^{(n)^{*}}(t) dt \right], \quad m = 1, 2, \ldots, M \] \hspace{1cm} (2.3)

In non-coherent detection, no attempt is made to estimate the channel parameters. The demodulator may base its decision either on the sum of the envelopes (envelope detection) or the sum of the squared envelopes (square-Law detection) of the matched filter outputs. In general, the performance obtained with envelope detection differs little from the performance obtained with square-law detection in
AWGN. However, square-law detection of multichannel signaling in AWGN channels is considerably easier to analyze than envelope detection. Therefore, we confine our attention to square-law detection of the received signals of the L channels, which produces the decision variables:

\[ CM_m = \sum_{n=1}^{L} \left| \int_{0}^{T} r_{1}^{(n)}(t)s_{1}^{(n)*}(t)\,dt \right|^2, \quad m = 1,2,\ldots,M \]  

(2.4)

Let us consider binary signaling first, and assume that \( s_{11}^{(n)}, n=1,2,\ldots,L \) are the L transmitted waveform. Then an error is committed if \( CM_2 > CM_1 \) or, equivalently, if the difference \( D = CM_1 - CM_2 < 0 \) for non coherent detection; this difference may be expressed as:

\[ D = \sum_{n=1}^{L} \left( |X_n|^2 - |Y_n|^2 \right) \]  

(2.5)

Where, the variables \( \{X_n\} \) and \( \{Y_n\} \) are defined as

\[ X_n = \int_{0}^{T} r_{1}^{(n)}(t)s_{11}^{(n)*}(t)\,dt, \quad n = 1,2,\ldots,L \]  

(2.6)

\[ Y_n = \int_{0}^{T} r_{1}^{(n)}(t)s_{12}^{(n)*}(t)\,dt, \quad n = 1,2,\ldots,L \]  

(2.7)

Here, \( \{X_n\} \) are mutually independent and identically distributed Gaussian random variables. The same statement applies to the variables \( \{Y_n\} \). However, for any n, \( X_n \) and \( Y_n \) may be correlated. For coherent detection, the difference \( D = CM_1 - CM_2 \) may be expressed as

\[ D = \frac{1}{2} \sum_{n=1}^{L} \left( X_n^* Y_n + X_n Y_n^* \right) \]  

(2.8)

where, by definition, \( Y_n = \hat{s}_n \), \( n = 1,2,\ldots,L \)

(2.9)

\[ X_n = \int_{0}^{T} r_{1}^{(n)}(t)\left[s_{11}^{(n)*}(t) - s_{12}^{(n)*}(t)\right]\,dt \]  

(2.10)
2.6 Analog Communication System

Analog communication is a data transmitting technique in a format that utilizes continuous signals to transmit data including voice, image, video, electrons, etc. An analog signal is a variable signal continuous in both time and amplitude which is generally carried by use of modulation. Analog circuits do not involve quantization of information unlike the digital circuits and consequently have a primary disadvantage of random variation and signal degradation, particularly resulting in adding noise to the audio or video quality over a distance.

Data is represented by physical quantities that are added or removed to alter data. Analog transmission is inexpensive and enables information to be transmitted from point-to-point, or from one point to many. Once the data has arrived at the receiving end, it is converted back into digital form so that it can be processed by the receiving computer. The advantage and disadvantage analog communication system is given blow

**Advantages:**

1. More easy to generate; and
2. Easy way of communication.

**Disadvantages:**

1. Very difficult to transmit as it is;
2. Devices used are expensive;
3. Lots and lots of noise interruptions;
4. Accuracy is less; and
5. Transmission and reception are not very easy.
2.7 Fading Channel Characterization and Modeling

Radio-wave propagation through wireless channels is a complicated phenomenon characterized by various effects such as, multipath and shadowing. A precise mathematical description of this phenomenon is either unknown or too complex for tractable communication systems' analyses [172]. However, considerable efforts have been devoted to the statistical modeling and characterization of these different effects. The result is a range of relatively simple and accurate statistical models for fading channels which depend on the particular propagation environment and the underlying communication scenario.

The primary purpose of this section is to review briefly the principal characteristics and models for fading channels. A brief qualitative description and main characteristics of fading channels is presented. Models for frequency flat fading channels correspond to narrowband transmission are presented. Models for frequency selective fading channels that characterize fading in wideband channels are described as following.

2.7.1 Main Characteristics of Fading Channels

I. Envelope and Phase Fluctuations

When a received signal experiences fading during transmission, both its envelope and phase fluctuate over time. For coherent modulations, the fading effects on the phase can severely degrade performance unless measures are taken to compensate for them at the receiver. Most often, analyses of systems employing such modulations assume that the phase effects due to fading are perfectly corrected at the receiver, resulting in what is referred to as ideal coherent demodulation. For non-coherent modulation, phase information is not needed at the receiver and therefore the phase variation due to fading does not affect the performance. Hence performance analyses for both ideal coherent and non-coherent modulations over fading channels require only knowledge of the fading envelope statistics. Furthermore, for slow fading wherein the fading is at least constant over the duration of a symbol time, the fading envelope random process can be represented by a random variable over the symbol time.
II. Slow and Fast Fading

The distinction between slow and fast fading is important for the mathematical modeling of fading channels and for the performance evaluation of communications systems operating over these channels. This notion is related to the coherence time $T_c$ of the channels, which measures the period of time over which the fading process is correlated (or equivalently, the period of time after which the correlation functions of two samples of the channel response taken at the same frequency but different time instants drops below a certain predetermined threshold).

The coherence time is also related to the channel Doppler spread $f_d$ by:

$$T_c \approx \frac{1}{f_d} \quad (2.11)$$

The fading is said to be slow if the symbol time duration $T_s$ is smaller than the channels coherence time $T_c$; otherwise, it is considered to be fast. In slow fading, a particular fade level will affect many successive symbols. This leads to burst errors, whereas in fast fading the fading decor relates from symbol to symbol. In the latter case when the communication receiver decisions are made based on an observation of the received signal over two or more symbol times (such as differentially coherent or coded communication), it becomes necessary to consider the variation of the fading channel from one symbol interval to the next. This is done through a range of correlation models that depend essentially on the particular propagation environment and the underlying communication scenario [172].

II. Frequency-Flat and Frequency Selective Fading

Frequency selectivity is also an important characteristic of fading channels. If all the spectral components of the transmitted signal are affected in a similar manner, the fading is said to be frequency nonselective or, equivalently, frequency flat. This is the case for narrowband systems in which the transmitted signal bandwidth is much smaller than the channel's coherence bandwidth $f_c$. This bandwidth measures the frequency range over which the fading process is correlated and is defined as the frequency bandwidth over which the correlation function of two samples of the channel response taken at the same time but at different frequencies falls below a
suitable value. In addition, the coherence bandwidth is related to the maximum delay spread $\tau_{\text{max}}$ by:

$$f_c \approx 1 / \tau_{\text{max}}$$  \hspace{1cm} (2.12)

On the other hand, if the spectral components of the transmitted signal are affected by different amplitude gains and phase shifts, the fading is said to be frequency selective. This applies to wideband systems in which the transmitted bandwidth is bigger than the channel’s coherence bandwidth.

### 2.7.2 Modeling of Flat Fading Channels

When fading affects narrowband system, the received carrier amplitude is modulated by the fading amplitude $\alpha$ where, $\alpha$ is a random variable (RV) with mean square value $\Omega = \alpha^2$ and probability density function (PDF) $p_\alpha(\alpha)$, which is dependent on the nature of the radio propagation environment [Lee. ‘Mobile Communications: Design fundamentals’. 1993]. After passing through the fading channel, the signal is perturbed at the receiver by (AWGN), which is typically assumed to be statistically independent of the fading amplitude $\alpha$ and which is characterized by one sided power spectral density $N_0 \alpha^2$. thus, we define the instantaneous signal to noise power ratio (SNR) per symbol by $\gamma = \alpha^2 E / N_0$ and the average SNR per symbol by $\gamma = \Omega E / N_0$ where, E is the energy per symbol in additions, the PDF of $\gamma$ is obtained by introducing a change of variable in the expression for the fading PDF $p_\alpha(\alpha)$ of $\alpha$, yielding

$$p_\gamma(\gamma) = p_\alpha\left(\sqrt{\frac{\Omega \gamma}{\gamma}}\right) \frac{1}{2\sqrt{\frac{\Omega \gamma}{\gamma}}}$$  \hspace{1cm} (2.13)

The moment generating function (MGF) $M_\gamma(s)$, associated with the fading PDF $p_\gamma(\gamma)$ is defined by

$$M_\gamma(s) = \int_0^\infty p_\gamma(\gamma)e^{sy} d\gamma$$  \hspace{1cm} (2.14)
This is another important statistical characteristic of fading channels. In addition, the amount of fading (AF), or "fading figure," associated with the fading PDF is defined as:

\[
AF = \frac{\text{var}(\alpha^2)}{\left(\mathbb{E}[\alpha^2]\right)^2} = \frac{\mathbb{E}\left[(\alpha^2 - \Omega)^2\right]}{\Omega^2} = \frac{\mathbb{E}(\gamma^2) - (\mathbb{E}[\gamma])^2}{(\mathbb{E}[\gamma])^2}
\]  

(2.15)

### 2.7.3 Multipath Fading and models

Multipath fading is due to the constructive and destructive combination of randomly delayed, reflected, scattered, and diffracted signal components. This type of fading is relatively fast and is therefore responsible for the short-term signal variations. Depending on the nature of the radio propagation environment, there are different models describing the statistical behavior of the multipath fading envelope, which are given below.

#### I. Rayleigh Model

The Rayleigh distribution is the frequency used to model multipath fading with no direct line-of-sight (LOS) path. In this case, the channel fading amplitude \( \alpha \) is distributed according to:

\[
p_\alpha(\alpha) = \frac{2\alpha}{\Omega} \exp\left(-\frac{\alpha^2}{\Omega}\right) \quad \alpha \geq 0
\]  

(2.16)

and hence, the following instantaneous SNR per symbol of the channel, \( \gamma \) is distributed according to an exponential distribution given by:

\[
p_\gamma(\gamma) = \frac{1}{\gamma} \exp\left(-\frac{\gamma}{\gamma}\right) \quad \gamma \geq 0
\]  

(2.17)

The MGF corresponding to this fading model is given by:

\[
M_\gamma(s) = \left(1 - s\gamma\right)^{-1}
\]  

(2.18)

In addition, the moments associated with this fading model can be shown by:
where, \( \Gamma(\cdot) \) is the gamma function. The Rayleigh fading model has an AF equal to 1 and typically agrees very well with experimental data for mobile systems where, no LOS path exists between the transmitter and receiver antennas. It also applies to the propagation of reflected and refracted paths through the troposphere, ionosphere and to ship to ship radio.

II. Nakagami-q (Hoyt) Model

The Nakagami-q distribution also referred to as the Hoyt distribution is given in Nakagami by:

\[
p_\alpha(\alpha) = \frac{(1+q^2)^\alpha}{q\Omega} \exp\left[-\frac{(1+q^2)^2}{4q^2\Omega}\right] I_0\left(\frac{(1-q^4)\alpha^2}{4q^2\Omega}\right), \quad \alpha \geq 0
\]

(2.20)

where, \( I_0(.) \) is the zeroth-order modified Bessel function of the first kind, and q is the Nakagami-q fading parameter which ranges from 0 to 1. It can be shown that the SNR per symbol of the channel \( \gamma \) is distributed according to:

\[
p_\gamma(\gamma) = \frac{(1+q^2)^\alpha}{2q^\gamma} \exp\left[-\frac{(1+q^2)^2}{4q^2\gamma}\right] I_0\left(\frac{(1-q^4)\gamma}{4q^2\gamma}\right), \quad \gamma \geq 0
\]

(2.21)

The moments associated with this model are given by

\[
E(\gamma^k) = \Gamma(1+k) _2F_1\left(-\frac{k-1}{2}, -\frac{k}{2}; 1; \frac{(1-q^2)^2}{1+q^2}\right) \gamma^{-k}
\]

(2.22)

Where, \( _2F_1(\cdot; \cdot; \cdot) \) is the Gauss hyper geometric function; and the AF of the Nakagami distribution is therefore given by:
\[ AF_q = \frac{2(1+q^4)}{(1+q^2)^2}, \quad 0 \leq q \leq 1 \] (2.23)

And hence, it ranges between 1 (q=1) and 2 (q=0). The Nakagami-q distribution spans the range from one sided Gaussian fading (q=0) to Rayleigh fading (q=1). It is typically observed on satellite links, subject to strong ionosphere scintillation. It is to be noted here that one sided Gaussian fading corresponds to the worst-case fading or, equivalently, the largest AF for all multipath distributions considered in our analyses.

\section*{III. Nakagami-N (Rice) Model}

The Nakagami-n distribution is also known as the Rician distribution. It is often used to model propagation paths consisting of one strong direct LOS component and many random weaker components. Here, the channel fading amplitude follows the distribution:

\[ p_n(\alpha) = \frac{2(1+n^2)e^{-\frac{\alpha}{n^2}}}{\Omega} \exp \left[ -\frac{(1+n^2)\alpha^2}{\Omega} \right] I_0 \left( 2n\alpha\sqrt{\frac{1+n^2}{\Omega}} \right), \quad \alpha \geq 0 \] (2.24)

where, n is the Nakagami-n fading parameter which ranges from 0 to \( \infty \) and, which is related to the Rician K factor by \( K = n^2 \). The SNR per symbol of the channel \( \gamma \) is distributed according to a non-central chi-square distribution given by:

\[ p_\gamma(\gamma) = \frac{(1+n^2)e^{-\frac{\gamma}{n^2}}}{\gamma} \exp \left[ -\frac{(1+n^2)\gamma}{\gamma} \right] I_0 \left( 2n\sqrt{\frac{(1+n^2)\gamma}{\gamma}} \right), \quad \gamma \geq 0 \] (2.25)

It can also be shown that the MGF associated with this fading model may be by

\[ M_\gamma(s) = \frac{(1+n^2)}{(1+n^2)-s\gamma} \exp \left[ \frac{n^2s\gamma}{(1+n^2)-s\gamma} \right] \] (2.26)
And, that the moments are given by:

\[ E(y^k) = \frac{\Gamma(1+k)}{(1+n^2)^k} \, \Gamma(-k,1-n^2)^{-k} \]  \hspace{1cm} (2.27)

Where, \( \Gamma(\cdot, \cdot) \) is the Kummer confluent hyper geometric function. The AF of the Nakagami-n distribution is given by:

\[ AF_n = \frac{1+2n^2}{(1+n^2)^2} \hspace{1cm} n \geq 0 \]  \hspace{1cm} (2.28)

And hence, it ranges between 0 \((n=\infty)\) and 1 \((n=0)\). The Nakagami-n distribution spans the range from Rayleigh fading \((n=0)\) to no fading (constant amplitude) \((n=\infty)\). This type of fading is typically observed in the first resolvable LOS paths of microcellular urban and suburban land-mobile, Pico cellular indoor and factory environment. It also applies to the dominant LOS path of satellite and ship to ship radio links.

**IV. Nakagami-m Model**

The Nakagami-m PDF is a central chi-square distribution as given by:

\[ p_\alpha(\alpha) = \frac{2^m \alpha^{2m-1}}{\Omega^{m} \Gamma(m)} \exp\left(-\frac{m \alpha^2}{\Omega}\right) \hspace{1cm} \alpha \geq 0 \]  \hspace{1cm} (2.29)

Where, \( m \) is the Nakagami-m fading parameter which ranges from \( \frac{1}{2} \) to \( \infty \).

Figure 2.3 shows the Nakagami-m PDF for \( \Omega=1 \) and various values of the \( m \) parameter. The SNR per symbol \( \gamma \) is distributed according to a gamma distribution given by:
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\[ P_\gamma(\gamma) = \frac{2^m \gamma^{2m-1}}{\gamma^m \Gamma(m)} \exp\left( -\frac{m\gamma}{\gamma} \right) \quad \gamma \geq 0 \]  

(2.30)

It can also be shown that the MGF is given in this case by:

\[ M_\gamma(s) = \left( 1 - \frac{s\gamma}{m} \right)^{-m} \]  

(2.31)

and, that the moments are given by:

\[ E[\gamma^k] = \frac{\Gamma(m+k)}{\Gamma(m) m^k} \gamma^k \]  

(2.32)

which yields an AF as given by:

Figure 2.2 Nakagami PDF for \( \Omega=1 \) and various values of the fading parameter \( m \)
\( AF_m = \frac{1}{m}, \quad m \geq \frac{1}{2} \)  \hspace{1cm} (2.33)

Hence, the Nakagami-m distribution spans via the m parameter the widest range of AF (from 0 to 2) among all the multipath distributions. For instance, it includes the one sided Gaussian distribution \( m = \frac{1}{2} \) and the Rayleigh distribution (m=1) as special cases. In the limit as \( m \to +\infty \), the Nakagami-m fading channels converges to a no fading AWGN channel. Furthermore, when \( m < 1 \), we obtain a one to one mapping between the m parameter and the q parameter, allowing the Nakagami-m distribution to closely approximate the Nakagami-q (Hoyt) distribution, and this mapping is given by:

\[
m = \frac{(1+q^2)^2}{2(1+2q^2)}, \quad m \leq 1
\]

(2.34)

Similarly, when \( m < 1 \), we obtain another one to one mapping between the m parameter and the n parameter (or, equivalently, the Rician K factor), allowing the Nakagami-m distribution to closely approximate the Nakagami-n (Rician) distribution, and this mapping is given by:

\[
m = \frac{(1+n^2)^2}{1+2n^2}, \quad n \geq 0
\]

\[
n = \frac{\sqrt{m^2 - m}}{m - \sqrt{m^2 - m}}, \quad m \geq 1
\]

(2.35)

Finally, the Nakagami-m distribution often gives the best fit to land mobile and indoor-mobile multipath propagation, as well as scintillating ionosphere radio links.

### 2.7.4 Log-Normal Shadowing

In terrestrial and satellite land mobile systems, the link quality is also affected by slow variation of the mean signal level due to the shadowing from terrain, buildings and trees. Communication system performance will depend on shadowing only if the radio receiver is able to average out the fast multipath fading or if an efficient micro diversity system is used to eliminate the effects of multipath [173].
Based on empirical measurements, there is a general consensus that shadowing can be modeled by a log normal distribution for various outdoor and indoor environment, in which case the path SNR per symbol $\gamma$ has a PDF given by the standard log normal expression as

$$p_\gamma(\gamma) = \frac{\xi}{\sqrt{2\pi} \sigma \gamma} \exp\left[-\frac{(10 \log_{10} \gamma - \mu)^2}{2\sigma^2}\right]$$  \hspace{1cm} (2.36)

Where, $\xi = \frac{10}{\ln 10} \approx 4.3429$, and $\mu$ (dB) and $\sigma$ (dB) are the mean and standard deviation of $10\log_{10} \gamma$ respectively.

The MGF associated with this slow-fading effect is given by:

$$M_\gamma(s) \equiv \frac{1}{\sqrt{\pi}} \sum_{n=1}^{N_p} H_{x_n} \exp\left\{10^{\frac{\mu + n\theta}{10}}\right\}$$  \hspace{1cm} (2.37)

Where, $x_n$ are the zeros of the $N_p$ order Hermite polynomial and $H_{x_n}$ are the weight factors of the $N_p$ order Hermite polynomial addition, the moments of are given by:

$$E[\gamma^k] = \exp\left[\frac{k}{\xi} \mu + \frac{1}{2} \left(\frac{k}{\xi}\right)^2 \sigma^2\right]$$  \hspace{1cm} (2.38)

yielding an AF of:

$$AF_\sigma = \exp\left(\frac{\sigma^2}{\xi^2}\right) - 1$$  \hspace{1cm} (2.39)

The fading follows Rice (Nakagami-n) PDF. On the other hand, when shadowing is present, it is assumed that no direct LOS path exists and the received signal power (or, equivalently SNR per bit) is assumed to be an exponential/log normal (Hansen-Meno) PDF. The combination is characterized by the shadowing time share factor, which is denoted by $A, 0 \leq A \leq 1$; hence, the resulting combined PDF is given by:

$$p_\gamma(\gamma) = (1-A) \left(\frac{1+K}{\gamma}\right)^e \exp\left[-\frac{(1+K)\gamma}{\gamma}\right] I_0\left[2\sqrt{\frac{K(1+K)}{\gamma}}\right]$$
\[
+ A \int_{0}^{\infty} \frac{1}{w} \exp \left( -\frac{\gamma}{w} \right) \sqrt{\frac{2\pi \sigma^w}{2}} \exp \left[ \frac{(10 \log_{10} w - \mu^w)^2}{2(\sigma^w)^2} \right] dw
\]

(2.40)

where, \( \bar{\gamma}^w \) is the average SNR per symbol during the un-shadowed fraction of time, and \( \mu^w \) and \( \sigma^w \) are the average and standard deviation of \( 10 \log_{10} \gamma \) during the shadowed fraction of time, respectively. The overall average SNR per symbol \( \bar{\gamma} \) is then given by:

\[
\bar{\gamma} = (1 - A) \bar{\gamma}^w + A 10^{\frac{\mu^w + (\ln 10) (\sigma^w)^2}{200}}
\]

(2.41)

Finally, the MGF can be shown to be given by:

\[
M_{\bar{\gamma}}(s) \equiv (1 - A) \frac{(1 + K)}{1 + K - s \bar{\gamma}^w} \exp \left[ - \frac{K s \bar{\gamma}^w}{(1 + K - s \bar{\gamma}^w)} \right] + A \frac{1}{\sqrt{\pi}} \sum_{n=1}^{N_s} H_{s_n} \left( 1 - 10^{\frac{\sqrt{\sigma^w s_n + \mu^w}}{10}} \right)^{-1}
\]

(2.42)

2.7.5 Modeling of Frequency Selective Fading Channels:

When wideband signals propagate through a frequency selective channel, their spectrum is affected by the channel transfer function, resulting in a time dispersion of the waveform. This type of fading can be modeled as a linear filter characterized by the following complex valued low pass equivalent impulse response:

\[
h(t) = \sum_{l=1}^{L_p} \alpha_l e^{-j\theta_l} \delta(t - \tau_l)
\]

(2.43)

Where, \( \delta(\cdot) \) is the Dirac delta function and \( l \) is the channel index. And, the \( \{ \alpha_l \}_{l=1}^{L_p}, \{ \theta_l \}_{l=1}^{L_p}, \) and \( \{ \tau_l \}_{l=1}^{L_p} \), their random channel amplitudes, phases, and delays respectively.

In \( L_p \) is the number of resolvable paths (the first path being the reference path whose delay \( \tau_1 = 0 \) ) and is related to the ratio of the maximum delay spread to the
symbol time. Under the slow fading assumption, \( L_p \) is assumed to be constant over a certain period of time, and \( \{\alpha_l\}_{i=1}^{L_p} \), \( \{\theta_l\}_{i=1}^{L_p} \) and \( \{\tau_l\}_{i=1}^{L_p} \) are all constant over a symbol interval. If the various paths of a given impulse response are generated by different scatters, they tend to exhibit negligible correlations and it is reasonable in that case to assume that the \( \{\alpha_l\}_{i=1}^{L_p} \) are statistically independent random variable (RV). Otherwise, the \( \{\alpha_l\}_{i=1}^{L_p} \) have to be considered as correlated RV's.

Expanding the flat fading notations, the fading amplitude, \( \alpha_l \) of the \( l \)th resolved path is assumed to be a RV whose mean square value \( \overline{\alpha_l^2} \) is denoted by \( \Omega_l \) and whose PDF can be any one of the PDFs presented above. Also as in the flat fading case, after passing through the fading channel, a wideband signal is perturbed by AWGN with a one sided power spectral density \( N_0(W/H) \). The AWGN is assumed to be independent of the fading amplitude \( \{\alpha_l\}_{i=1}^{L_p} \). Hence the instantaneous SNR per symbol is given by \( \gamma_l = \alpha_l^2 E_s / N_0 \), and the average SNR per symbol of is given by \( \overline{\gamma_l} = \Omega_l E_s / N_0 \).

The first arriving path in the impulse response typically exhibits a lower amount of fading than subsequent paths since it may contain the LOS path. Furthermore, since the secular power component typically decreases with respect to delay, the last arriving paths exhibit higher amounts of fading. They are related to the channels power delay profile (PDP), which is also referred to as the multipath intensity profile (MIP) and which is typically a decreasing function of the delay. The PDP model can assume various forms, depending on whether the model is for indoor or outdoor environment and for each environment, the general propagation conditions apply to PDPs for indoor partitioned office building, indoor factory buildings with heavy machinery, and high-density office building in urban area. For example, experimental measurements indicate that the mobile radio channel is well characterized by an exponentially decaying PDP for indoor office building and congested urban areas:

\[
\Omega_l = \Omega_0 e^{-\tau_l / t_{\text{max}}} , \quad l = 1, 2, ..., L_p
\]  
(2.44)
Where, $\Omega_1$ is the average fading power corresponding to the first (reference) propagation path and $\tau_{\text{max}}$ is the channel maximum delay spread. In the literature the delays are often assumed to be equally spaced $(\tau_{i+1} - \tau_i)$ is constant and equal to the symbol time $T_s$ and with this assumption, we get the equally spaced exponential profile given by:

$$\Omega_i = \Omega_1 e^{-t(\tau_{i-1})}\delta, \quad \delta \geq 0 \quad \text{and} \quad l = 1, 2, ..., L_p$$

(2.45)

where, the parameter is the power decay factor, which reflects the rate at which the average fading power decays. Other idealized PDP profile reported or used in the literature include the constant (flat) [42], the flat exponential. The double spike, the Gaussian the power function (polynomial) and other more complicated composite profiles.

2.8 Diversity Techniques:

Various techniques are possible for mitigating the effects of multipath fading. All possible techniques include equalization, coding and diversity. The diversity technique improves the quality of a wireless communication link without altering the common air interface, transmitted power or bandwidth. Along with the choice of the diversity system, a proper choice of the combining method is necessary to get the most of the gain promised by diversity. Diversity combining as a whole system consists of receiving redundantly the same information bearing signal over two or more fading channels and then combining these multiple replicas at the receiver in order to increase the overall received SNR. The intuition behind diversity combining is to take advantage of the low probability of concurrence of deep fades in all the diversity branches. There are various means of realizing diversity. These include Space Diversity, Frequency Diversity and Time Diversity.

I. Space Diversity

Space diversity is the diversity technique that is most popular in wireless communication systems, because of its low added complexity with respect to gained performance improvement. It is also called antenna diversity. Antenna diversity requires multiple antennas at the receiver and is therefore usually bulkier than other
diversity systems. However, operating at high frequency bands allows for the size reduction of antennas elements and it is feasible to have multiple antennas not only at the base stations but also on the mobile handset.

Antenna diversity can be achieved through spatial, polarization, or pattern configurations. The spatial configuration is the most common of the three and requires two or more antennas to be separated in space at the terminal. Two antennas that are physically separated in space experience different propagation environments, multipath components and different summation at each antenna. Ideally, for a spatial diversity system, the antennas are spaced far enough in distance such that the branch signals have a higher probability of fading independently and this separation could be in the order of tens of radio frequency (RF) carrier wavelengths.

For the subsequent models employed in the thesis, two kinds of antenna configurations will be considered due to their relative ease of installation. They are uniform linear and triangular arrays. The statistical correlation between the antenna diversity branches is a major hurdle for the antenna system performance. The correlation matrix of an antenna array depends not only on the array configuration but also on the incident angle of the incoming signal. Different parameters affect the correlation differently. The increase of antenna separation between two antennas decreases their correlations. Similarly, the increase of height of antenna is followed by lowering the correlation. The correlation matrix of a given antenna array varies also with the signal incident angle.

For the polarization configuration, the polarization of the transmitted signal is often altered as it travels along the channel. The signal arrives at the terminal with a polarization which differs from the transmitted one. In addition, the mobile transmitter or receiver rarely transmits or receives with a fixed polarization as a result of random antenna movement by the user. Antenna with orthogonal polarization is used as part of polarization diversity systems to benefit from the times when there might be a larger signal received on one polarization than at another. It is not uncommon for one polarization to fade while the orthogonal one has adequate signals strength. One of the advantages of polarization diversity is that it permits the antennas to be co-located. Pattern diversity, on the other hand, uses antennas creating unalike interference patterns of the signal at each branch.
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As a result each channel receives the transmitted signal with different strengths depending on the branch pattern and the propagation characteristics at that moment in time. As with polarization diversity, pattern diversity may allow for the co-location of the antenna elements. The signal-to-noise ratio can be successfully increased in flat fading channels by using one or a combination of the above mentioned diversity systems.

Antenna diversity could also be used at the transmitter side termed as transmit antenna diversity. Transmit antenna diversity is part of the proposed standards of Wideband code Division Multiple Access (WCDMA), but WCDMA system offers more path diversity than does a narrowband one. Thus, the needs for multiple transmit antenna to provide spatial diversity in WCDMA is less than that in narrowband NCDMA.

II. Frequency Diversity

When two frequencies separated by a bandwidth $B_c$ are such that two received fading signals on two different frequencies are uncorrelated, the value of $B_c$ can be determined by:

$$B_c = \frac{\Delta \omega}{2\pi} = \frac{1}{2\pi \Delta}$$

(2.46)

where, $\Delta$ is time delay spread and $\Delta \omega_c$ is frequency separation.

Above equation shows that a value of $B_c$ larger than 50 kHz should be used for urban areas, and larger than 300 kHz for suburban areas. In open area the value of $B_c$ should be greater than 0.8 MHz. In open areas no fading is observed; so no diversity is needed. To use frequency diversity in both urban and suburban area, $B_c$ has to be 300 kHz or greater.

III. Time Diversity

Time diversity means transmitting identical messages in different time slots. This yields two uncorrelated fading signals at the receiving end. Time diversity is a good scheme for reducing intermediation at a multichannel site. But in a mobile radio environment a mobile unit may be at a standstill at any location that has a weak local
mean or is caught in a deep fade. In either situation, time diversity would not reduce the fades.

2.9 Combining Techniques on Diversity Schemes

The principle diversity combining techniques are selection combining (SC), Maximal ratio combining (MRC), Equal gain combining (EGC) and Generalized selection combining (GSC).

I. Selection Combiners

Selection diversity combining is the simplest among all methods. From a collection of diversity branches, the branch that receives the signal with the largest SNR at any time is selected and connected to the demodulator. In SC, the receiver monitors the SNR of all channels and connects the branch with the largest SNR at any instant in time to the demodulator. When one signal falls below the other the receiver switches between branches and the phase discontinuities occurs. In order to prevent phase discontinuities both channels are co-phased constantly.

A variation of selection diversity is switched diversity combining (SWC). Under this combining scheme, the receiver does not monitor all branches simultaneously as in selection diversity. The receiver connected to a branch and monitors its signal to noise ratio. If the SNR is above a certain threshold, it stays connected to that branch. As soon as the SNR drops below the acceptable limit, the receiver picks the next available branch and measures its received SNR. If the SNR is acceptable, the receiver stays connected to the branch otherwise it moves to a new branch and the process is repeated. The advantage of using the selection combining scheme is that it requires a single received only.

II. Maximal-Ratio Combining:

The term maximal ratio indicated maximal signal to noise ratio. This is the best combining technique. At a baseband the combined signal is the sum of the instantaneous SNRs \( \gamma_i \) on the individual branches.

\[
\gamma = \sum_{i=1}^{M} \gamma_i
\]  
(2.47)
This combining technique for two branch diversity needs two receivers. Maximal ratio combining (MRC) takes better advantage of all the diversity branches in the system. All branches are weighed by their respective instantaneous SNR's. The branches are then co-phased prior to summing in order to insure that all branches are added in phase for maximum diversity gain. The summed signals are then used as the received signal and connected to the demodulator. MRC will always perform better than either selection diversity or equal gain combining because it is an optimum combiner. The information on all channels is used with this technique to get more reliable received signals. The disadvantage of MRC is that it is complicated and requires accurate estimates of the instantaneous signal level and average noise power to achieve optimum performance. The advantage is that improvements can be achieved with this configuration even when branches are completely correlated. [172]

III. Equal-Gain Combining

Equal-gain combining (EGC) is a co-phase combining that brings all phases to a common point and combines them. The combined signal is the sum of the instantaneous fading envelopes of the individual branches.

\[ r = \sum_{i=1}^{M} r_i \] (2.48)

Equal-gain combining has only one dB degradation as compared to maximal-ratio combining. Equal-gain combining is usually used at base stations due to simplicity of circuit, it can be considered as a variant of MRC in which the branches are equally weighted and then summed. EGC is better implemented as Post detection combining. MRC is best implemented as pre detection combining. The major problem with EGC is the combining loss where no coherent post detection combining is often used.

V. General Selection Combiners

Generalized selection combining (GSC) is a hybrid technique that builds on the two extremes of SC and MRC, where it can be better understood through the pitfalls and merits of either of these extremes in different applications. MRC is sensitive to channel estimation errors and these errors tend to be more important when
the instantaneous SNR is low. On the other hand, SC and SWC use only one path out of the available multipath and hence do not fully exploit the amount of diversity offered by the channel. So, GSC was proposed to bridge the gap between these two extremes MRC/EGC and SC which adaptively combines the strongest resolvable paths among the available ones and is denoted as GSC/MRC and GSC/EGC accordingly. GSC/MRC receivers are expected to be more robust toward channel estimation errors because the weakest SNR paths are excluded from the combining process. Finally, GSC/MRC was shown to approach the performance of MRC. GSC/EGC was shown to approach the performance of conventional post detection EGC because it is less sensitive to the no coherent and differentially coherent post detection system.

2.10 M-ARY Frequency Shift Keying (FSK)

The M-ary of FSK, for which the transmitted signals are defined by:

\[ s_i(t) = \sqrt{\frac{2E}{T}} \cos \left( \frac{\pi}{T} (n_c + i) t \right), \quad 0 \leq t \leq T \]

(2.49)

where, \( i = 1, 2, \ldots, M \), and the carrier frequency \( f_c = n_c/2T \) for some fixed integer \( n_c \). The transmitted symbols are of equal duration \( T \) and have equal energy \( E \). Since the individual signal frequencies are separated by \( 1/2T \) Hz, the signals in above equation are orthogonal, that is:

\[ \int_0^T s_i(t) s_j(t) \, dt = 0, \quad i \neq j \]

(2.50)

This property of M-ary FSK suggests that we may use the transmitted signal \( s_i(t) \) themselves except for energy normalization as a complete orthogonal set of basis function, which is shown by:

\[ \phi(t) = \frac{1}{\sqrt{E}} s_i(t), \quad 0 \leq t \leq T \]

(2.51)

Where \( i = 1, 2, \ldots, M \)
The minimum distance $d_{\text{min}}$ in M-ary FSK is $\sqrt{2E}$.

The probability of error for MFSK is:

$$P_e \leq \frac{1}{2} (M - 1) \text{erf} \left( \frac{E}{\sqrt{2N_0}} \right)$$

When the orthogonal signals of an M-ary FSK signal are detected coherently, the adjacent signals need only to be separated from one another by a frequency $1/2T$ so as to maintain orthogonally. Hence, we may define the channel bandwidth required to transmit M-ary FSK signals as:

$$B = \frac{M}{2T} \quad (2.52)$$

For multilevel with frequency assignments that make the frequency spacing uniform and equal to $1/2T$, the bandwidth $B$ contains a large fraction of the signal power. The symbol period $T$ is equal to $T_b \log_2 M$. Hence, using $R_b = 1/T_b$, we may redefine the channel bandwidth $B$ for M-ary FSK signals as:

$$B = \frac{R_b M}{2 \log_2 M} \quad (2.53)$$

And, the bandwidth efficiency is:

$$\rho = \frac{R_b}{B} = \frac{2 \log_2 M}{M} \quad (2.54)$$

### 2.10.1 Non-Coherent Communication in AWGN

In non-coherent decoding, the carrier phase of the incoming signal is not known. An M-ary non-coherent receiver can be implemented as a bank of M matched filters, with the output of each being passed through an envelope detector. In the non-coherent case, the probability density function for the decision variables is not Gaussian.
There are two types of decision variables. The output of a filter, when subjected to an orthogonal waveform to which it is not matched, contains only noise. When the matched waveform is present at the filter input, the filter output contains signal plus noise. If the noise is additive, white, and Gaussian, then the \( M \) matched-filter outputs consist of \((M-1)\) values for noise only and one value for signal-plus-noise. As these are passed through envelope detectors, we must then calculate the probability density functions for the envelopes of Gaussian noise and for a constant signal plus Gaussian noise. The Gaussian noise process to have zero mean and variance \( \sigma^2_0 \) at the matched filter output.

In an \( M \)-ary orthogonal alphabet of equiprobable equal energy waveforms a symbol is properly decoded only if its corresponding decision variable is larger than all of the other \((M-1)\) decision variables. The decision variable \( U_1 \) corresponding to the correct symbol is Rician-distributed with probability density \( p_1(r) \). All other \((M-1)\) variables \( U_n, n=2...M \), are Rayleigh distributed with probability density \( P_0(r) \). The probability of correct decoding is therefore, given by:

\[
p_c(M, \gamma) = \int_0^{\infty} \left[ \prod_{i=2}^{M} \Pr \{U_i < U_1, U_i < U_1, \ldots, U_i < U_1 \} p_1(U_1) \right] dU_1
\]

(2.55)

\[
= \int_0^{\infty} \left( \prod_{n=2}^{M-1} \Pr \{U_n < U_1 | U_1 \} \right)^{M-1} p_1(U_1) dU_1
\]

(2.56)

The probability is given by:

\[
\Pr \{ U_n < U_1 | U_1 \} = \int_0^{U_1} P_0(r) dr
\]

\[= 1 - e^{-U_1^2/2\sigma^2_0} \]

(2.57)

The probability of correct decoding can be written as:

\[
p_c(M, \gamma) = \frac{1}{\sigma^2_0} \int_0^{\infty} \left[ 1 - e^{-x^2/2\sigma^2_0} \right]^{M-1} e^{-\left(\gamma r^2 + A^2/2\sigma^2_0\right)} I_0 \left( \frac{rA}{\sigma^2_0} \right) rdr
\]

(2.58)

Letting \( x = r^2/2\sigma^2_0 \) and \( \gamma = A^2/2\sigma^2_0 \) this expression can be simplified to:

\[
p_c(M, \gamma) = e^{-\gamma} \int_0^{\infty} \left( 1 - e^{-x^2} \right)^{M-1} e^{-x} \left[ 2(\gamma x)^{1/2} \right] dx
\]
\[ = \sum_{m=0}^{M-1} \left( -1 \right)^m \left( \frac{M - 1}{m} \right) e^{-m \gamma / (m+1)} \]  

(2.59)

The equivalent bit error probability can be found as:

\[ \text{p}_{be} (M, \gamma) = P_e (M, k \gamma_b) 2^{k-1} / (2^k - 1) \]  

(2.60)

where, \( P_e (M, k \gamma_b) = 1 - P_c (M, \gamma) \), \( K = \log_2 M \), and \( \gamma = k \gamma_b \). The decoding performance of no coherent M-ary orthogonal signaling is shown in figure 2.4

---

**Figure 2.3 Non-coherent M-ary Orthogonal Signaling Bit**

The performance behavior of non-coherent orthogonal signaling as M becomes large; we present the following derivation due to

\[ P_c (M, \gamma) = \int_0^\infty g (x) \exp \left[ - (x + y) \right] I_0 \left[ 2 (\gamma x)^{1/2} \right] dx \]  

(2.61)

whereas, M becomes large, function \( g (x) \) can be approximated by:
\[ g(x) \approx \begin{cases} 0 & \text{if } x < \ln(M-1) \\ 1 & \text{if } x > \ln(M-1) \end{cases} \]

Substituting this in the expression for \( P_c(M, \gamma) \) gives:

\[ P_c(M, \gamma) \approx \int_{\ln(M-1)}^{\infty} g(x) e^{-(x+\gamma)} I_0 \left[ 2(\gamma x)^{1/2} \right] dx \]

\[ = \int_{\ln(M-1)}^{\infty} e^{-(x+\gamma)} I_0 \left[ 2(\gamma x)^{1/2} \right] dx \]

This integral is the Marcum Q function of detection theory.

As \( M \) tends to infinity in \( P_c(M, \gamma) \),

\[ \lim_{M \to \infty} P_c(M, \gamma) \]

\[ \lim_{M \to \infty} \exp \left( -\frac{\gamma_b}{\ln 2} \ln M - \ln(M-1) \right) \sum_{n=0}^{\infty} \left( \frac{\gamma_b}{\ln 2} \right)^n I_n \left[ 2\ln M \left( \gamma_b / \ln 2 \right)^{1/2} \right]. \]

This expression converges only if \( \gamma_b < \ln 2 \) in which case it converges to zero:

\[ \lim_{M \to \infty} P_c(M, \gamma) \]

\[ = 1 - \lim_{M \to \infty} \exp \left( -\frac{\gamma_b}{\ln 2} \ln M - \ln(M-1) \right) \sum_{n=0}^{\infty} \left( \frac{\gamma_b}{\ln 2} \right)^n I_n \left[ 2\ln M \left( \gamma_b / \ln 2 \right)^{1/2} \right] \]

(2.63)

which converges to 1 provides \( \gamma_b > \ln 2 \).

Summarizing this result and using \( P_e(M, \gamma) = 1 - P_c(M, \gamma) \), we have

\[ \lim_{M \to \infty} P_e(M, \gamma) = \begin{cases} 0 & \text{if } \gamma_b > \ln 2 \\ 1 & \text{if } \gamma_b < \ln 2 \end{cases} \]

Thus, we may conclude that as the signaling complexity increases without bound, the performance of no coherent orthogonal signaling approaches ideal performance in which an arbitrarily small decoding error probability can be enjoyed as \( M \) is made sufficiently large, provided \( \gamma_b > \ln 2 \).
2.10.2 Coherent Binary Frequency-Shift keying (BFSK)

M-ary PSK and M-ary QAM share a common property; both are examples of linear modulation. In this section we study a nonlinear method of pass band data transmission, namely, coherent frequency-shift keying (FSK) [168]. We begin the study by considering the simple case of binary FSK.

In a binary FSK system, symbols 1 to 0 are distinguished from each other by transmitting one of two sinusoidal waves that differ in frequency by a fixed amount. A typical pair of sinusoidal waves is described by:

\[
\phi_i(t) = \begin{cases} 
\sqrt{2E_b/T_b} \cos(2\pi f_i t), & 0 \leq t \leq T_b \\
0, & \text{otherwise}
\end{cases}
\]

where, \( i = 1, 2 \) and \( E_b \) is the transmitted signal energy per bit; the transmitted frequency is:

\[
f_i = \frac{n_c + i}{T_b}
\]

Thus, symbol 1 is represented by \( s_1(t) \) and symbol 0 by \( s_0(t) \). The FSK signal described here is known as Sunde’s FSK. It is a continuous phase signal in the sense that phase continuity is always maintained, including the inter-bit switching times. This form of digital modulation is an example of continuous phase frequency shift keying (CPFSK).

We observe directly that the signals \( s_1(t) \) and \( s_2(t) \) are orthogonal, but not normalized to have unit energy. We therefore deduce that most useful form for the set of orthogonal basis functions is:

\[
\phi_i(t) = \begin{cases} 
\sqrt{2E_b/T_b} \cos(2\pi f_i t), & 0 \leq t \leq T_b \\
0, & \text{otherwise}
\end{cases}
\]

where, \( i = 1, 2 \). Correspondingly, the coefficient \( s_j \) for \( i = 1, 2 \), and \( j = 1, 2 \) is defined by:
\[
S_{ij} = \int_0^{T_b} s_i(t)\phi_j(t)\,dt
\]
\[
= \int_0^{T_b} \sqrt{2E_{b_i} \over T_b} \cos(2\pi f_i t) \sqrt{2E_{b_i} \over T_b} \cos(2\pi f_i t)\,dt
\]
\[
= \begin{cases} 
\sqrt{E_{b_i}}, & i = j \\
0, & i \neq j 
\end{cases}
\]

(2.66)

**Error Probability of Binary FSK**

The observation vector \(x\) has two elements \(x_1\) and \(x_2\) that are defined by respectively:

\[
x_1 = \int_0^{T_b} x(t)\phi_1(t)\,dt
\]

(2.67)

and,

\[
x_2 = \int_0^{T_b} x(t)\phi_2(t)\,dt
\]

(2.68)

Where, \(x(t)\) is the received signal, the form of which depends on which symbol was transmitted. Given that symbol will be 1 was transmitted, \(x(t)\) equals \(s_1(t) + w(t)\), where \(w(t)\) is the sample function of a white Gaussian noise process of zero mean and power spectral density \(N_0/2\). If, on the other hand, symbol 0 was transmitted, \(x(t)\) equal \(s_2(t)+w(t)\).

We here find that the observation space is partitioned into two decision regions, labeled \(Z_1\) and \(Z_2\). The decision boundary, separation region \(Z_1\) from \(Z_2\) is the perpendicular bisectors of the line joining the two message points. The receiver decides in favor of symbol 1 if the received signal point represented by the observation vector \(x\) falls inside region \(Z_1\). This occurs when \(x_1 > x_2\) if, on the other hand, we have \(x_1 > x_2\) the received signal points falls inside region \(Z_2\) and the receiver decides in favor of symbol, 0. On the decision boundary, we have \(x_1 = x_2\) in which case the receiver makes a random guess in favor of symbol 1 or 0.

Let us define a new Gaussian random variable \(Y\) whose sample value \(y\) is equal to the difference between \(x_1\) and \(x_2\), that is,

\[
y = x_1 - x_2
\]

(2.69)
The mean value of the random variable $Y$ depends on which binary symbol was transmitted. Given that symbol 1 was transmitted, the Gaussian random variables $x_1$ and $x_2$ whose sample values are denoted by $x_1$ and $x_2$ have mean values equal to and zero, respectively. Correspondingly, the conditional mean of the random variable $Y$, given that symbol 1 was transmitted is:

$$E[Y|1] = E[X_1|1] - E[X_2|1]$$

$$= + \sqrt{E_b}$$

(2.70)

On the other hand given that symbol 0 was transmitted, the random variables $x_1$ and $x_2$ have mean values equal to zero and $\sqrt{E_b}$ respectively, correspondingly, the conditional mean of the random variable $Y$, given that symbol 0 was transmitted, is:

$$E[Y|0] = E[X_1|0] - E[X_2|0]$$

$$= - \sqrt{E_b}$$

(2.71)

The variance of the random variable $Y$ is independent of which binary symbol was transmitted. Since the random variable $x_1$ and $x_2$ are statistically independent, each with a variance equal to $N_0/2$, it following that:

$$Var[Y] = Var[X_1] + Var[X_2]$$

$$= N_0$$

Supposing that symbol 0 was transmitted. The conditional probability density function of the random variable $Y$ is then given by:

$$f_y(y|0) = \frac{1}{\sqrt{2\pi N_0}} \exp \left[ - \frac{(y + \sqrt{E_b})^2}{2N_0} \right]$$

(2.72)

Since the conditions $x_1 > x_2$ or equivalently $y > 0$, corresponds to the receiver making a decision in favor of symbol 1. That the conditional probability of error, given that symbol 0 was transmitted is:
\[ p_{01} = P(y > 0| \text{symbol 0 was sent}) \]
\[ = \int_0^\infty f_y(y|0)dy \]
\[ = \frac{1}{\sqrt{2\pi N_0}} \int_0^\infty \exp \left[ -\frac{(y + \sqrt{E_b})^2}{2N_0} \right] dy \]
\[ \frac{y + \sqrt{E_b}}{\sqrt{2N_0}} = z \]

Then, changing the variable of integration from \( y \) to \( z \):
\[ p_{01} = \frac{1}{\sqrt{\pi}} \int_{\sqrt{E_b/2N_0}}^{\infty} \exp \left( -z^2 \right) dz \]
\[ = \frac{1}{2} \text{erfc} \left( \sqrt{\frac{E_b}{2N_0}} \right) \]

Similarly, we may show the \( p_{01} \) the conditional probability of error given that symbol 1 was transmitted. Accordingly, averaging \( p_{10} \) and \( p_{01} \) we find that the average probability of bit error or, equivalently, the bit error rate for coherent binary FSK is (assuming equiprobable symbols)[86]
\[ P_e = \frac{1}{2} \text{erfc} \left( \frac{E_b}{2N_0} \right) \]

### 2.10.3 Non-Coherent Binary Frequency-Shift Keying (NFBSK)

In the binary FSK case, the transmitted signal is defined by:
\[ s_i(t) = \begin{cases} \sqrt{\frac{E_b}{T_b}} \cos(2\pi f_i t), & 0 \leq t \leq T_b \\ 0, & \text{otherwise} \end{cases} \]

where, the carrier frequency \( f_i \) equals one of two possible values, \( f_1 \) and \( f_2 \) to ensure that the signals representing these two frequencies are orthogonal, we choose \( f_i = n_i / T_b \)
The transmission of frequency $f_1$ represents symbol 1, and the transmission of frequency $f_2$ represents symbol 0. For the non-coherent detection of this frequency modulated wave, the receiver consists of a pair of matched filters followed by envelope detectors. The filter in the upper path of the receiver is matched to $\cos(2\pi f_1 t)$ and the filter in the lower path is matched to $\cos(2\pi f_2 t)$ and in both cases $0 \leq t \leq T_b$. The resulting envelope detector outputs are sampled at $t = T_b$ and their values are compared. The envelope samples of the upper and lower paths are $l_1$ and $l_2$ respectively. Then, if $l_1 > l_2$ the receiver decides in favor of symbol 1, and if $l_1 < l_2$; it decides in favor of symbol 0 if the receiver simply makes a guess in favor of symbol 1 or 0.

The non-coherent binary FSK described herein is a special case of non-coherent orthogonal modulation with $T = T_b$ and $E = E_b$. We find that the bit error rate for non-coherent binary FSK is:

$$P_e = \frac{1}{2} \exp\left(-\frac{E_b}{2N_0}\right)$$

The above expression is derived as a special case of no coherent orthogonal modulation. [46]

### 2.11 Communication System Reliability Evaluation

In recent years, communication system has envisaged a tremendous rise in popularity. The continued minimization of communication devices, decreasing trend of cost and the extraordinary rise of processing power of processors combine to put more and better communication applications into the hands of a growing segment of the population. Mobile communication has enjoyed a tremendous rise in popularity during the last decade. This trend is mainly forced by recent advances and revolutions in microelectronics, telecommunications and wireless communications, which leads to develop a new generation of communication devices. Depending on the application
scenario, different requirements have to be considered for message communication. The communication system reliability is particularly important for mission-critical applications such as monitoring of disaster-struck regions, remotely located civil works management, under construction dam site management for multi purpose large hydro power plants, remotely located patient monitoring, battlefield monitoring, home automation, tracking of chemical and explosive agents, etc. These applications are loss-and-delay sensitive and therefore, reliable and timely delivery of information is critical [174].

A communication network is a group of hardware devices that purposefully interconnected to provide communication services. It has the transportation ability to support applications for users. It is concerned with the interconnectivity of various elements in the form of network, or graph, as exemplified by telecommunication, distribution and computer networks. In a computer communication network, the nodes might represent the physical computers (servers, switches, routers, etc.) and the edges of such a network might represent existing communication links between these nodes. A network service is a function that a network provides for users or a group of users with some demanded performance requirements. Usually, a function is supported by software or a group of compactable operating software. Application profile represents the information of an application, including information of the users involved and a set of operation capabilities of the application. Usage profile contains a set of application profiles and their occurrence probabilities.

There are numerous types of communication networks, of which the two most common are the circuit switched and packet switched networks. The circuit switched is mostly used in telephone networks dominated by fiber optical networking, while the packet switched is employed for the overwhelming majority of computer networks. In circuit switched communication network, a specific circuit/link is switched on for dissemination of information/message. In packet switched networks, the nodes in such a network exchange discrete blocks of data (i.e. some piece of application data as a file, a piece of e-mail, an image) called packets to each other. Internetworking is the process of moving a packet of data from source to destination.
The process of determining systematically how to forward messages toward the destination node based on its address is called routing. Routing is a key feature of the Internet because it enables messages to pass from one computer to another and eventually reach the target machine. Each intermediary computer performs routing by passing along the message to the next computer. Part of this process involves analyzing a routing table to determine the best path/route. Routing is usually performed by a dedicated device called a router.

A cognitive network is a network with a cognitive process that can perceive current network conditions, and then plan, decide, and act on those conditions. The network can learn from these adaptations and use them to make future decisions, all while taking into account end-to-end goals.

**Communication System Reliability, \( R (t) \):** Communication system reliability is the capability that the network will satisfy user’s normal requirements under specified operation conditions for a stated period of time. The specified operation conditions include not only network traffic, node mobility, terrain, and weather, but also equipment/device stochastic failures in the process of network operation (except failures caused by human factors). In other words, this is a network’s ability to perform a designated set of functions under certain conditions for specified operational time period.

**The Reliability Function:** The Reliability function represents the expression of the system reliability in terms of its hazard rate. This may be derived from the classical definition of probability. The probability of occurrence of an event \( A \) can be expressed as

\[
Pr(A) = \frac{N_s}{N} = \frac{N_s}{N_s + N_f}
\]

(2.73)

Where, \( N = \) Total number of trials = \((N_s + N_f)\),

\( N_s = \) Number of favorable or successful outcomes

\( N_f = \) Number of unfavorable outcomes
Now, let us consider $N$ units are taken on life test. They are operated under exactly the same environmental conditions and ensuring that failures of the all units are independent and do not influence one other. It is considering catastrophic failure only. At some predetermined intervals of time, the number of the survivors and failed units are recorded. Let us assume that $N_s(t)$ units are survived after the time $t$ out of total $N$ number of units is taken for life test and remaining units, $N_f(t)$ are failed over the time $t$. Then, Reliability of such units at time $t$ can be expressed as

$$R(t) = \frac{N_s(t)}{N} = \frac{N_s(t)}{N_s(t) + N_f(t)} \quad (2.74)$$

Similarly unreliability, $Q(t)$ which is also called cumulative failure distribution function (CDF) represented as $F(t)$ can be written as

$$Q(t) = F(t) = \frac{N_f(t)}{N} \quad (2.75)$$

At the beginning i.e. $t \to 0$, $N_s = N$, hence, $R(t) \to 1$ and $Q(t) \to 0$

$t \to \infty$, $N_s = 0$, then, $R(t) \to 0$ and $Q(t) \to 1$

or $R(t) = 1 - Q(t)$

$$= 1 - F(t) = 1 - \frac{N_f(t)}{N} \quad (2.76)$$

Differentiating the eqn. (v) both side with respect to $t$, we get

$$\frac{dR(t)}{dt} = -\frac{dN_f(t)}{dt} \frac{1}{dt.N}$$

Or, $\frac{dN_f(t)}{dt} = -N \frac{dR(t)}{dt} \quad (2.77)$

Where, $\frac{dN_f(t)}{dt}$ is failure rate of the units. This is the rate at which devices or items are failing at time $t$. 
**Hazard rate:** This is known as the instantaneous probability of failure is defined as

\[
h(t) = \frac{(\text{Number of failures per unit time})}{(\text{Number of survivals at time } t)}
\]

\[
= \frac{1}{N_s(t)} \cdot \frac{dN_f(t)}{dt}
\]

\[
= \lim_{\Delta t \to 0} \frac{N_f(t + \Delta t) - N_f(\Delta t)}{\Delta t}
\]

Hence, “The Hazard rate may be defined as the rate of failing of units normalized to the number of survivors at time \(t\)”. Hazard rate actually says that how hazardous the situation is at the time, \(t\).

Actually the rate of falling doesn’t provide the clear picture of how hazardous the situation is. For instance, rate of falling at two instant of time may be same but the hazard rate may not be same. For example,

\[
\frac{dN_f(t_1)}{dt} = \frac{dN_f(t_2)}{dt} \quad \text{for} \quad t_2 > t_1
\]

But as the population is decaying, the number of survivors would go on decreasing as the time increases.

\(N_s(t_2) < N_s(t_1)\) for \(t_2 > t_1\)

Now, Hazard rate at time \(t_2\)

\[
h(t_2) \quad \text{i.e.} \quad \frac{1}{N_s(t_2)} \cdot \frac{dN_f(t_2)}{dt} \neq h(t_1) \quad \text{i.e.} \quad \frac{1}{N_s(t_1)} \cdot \frac{dN_f(t_1)}{dt}
\]

Obviously, at time \(t_2\) the situation is more hazardous than at the time \(t_1\).

This Hazard rate, \(h(t)\) may be constant, i.e. not changing with time called Constant hazard rate. This may vary with time.

Substituting the value of \(\frac{d}{dt}N_f(t)\) in the above equations, we get
Chapter II

\[ h(t) = \frac{1}{Ns(t)} dNf(t) \frac{dt}{dt} \]

\[ = \frac{1}{Ns(t)} \cdot \frac{-N \cdot dR(t)}{dt} \]

\[ = \frac{-N}{Ns(t)} \cdot \frac{dR(t)}{dt} \]

\[ = \frac{-1}{R(t)} \cdot \frac{dR(t)}{dt} \]

The above equation is the first order differential eqn. Its solution may be found by integrating both sides as given below:

\[ \int_{0}^{t} h(u) \, du = - \int_{0}^{t} \frac{1}{R(u)} \cdot dR \] Where u is a dummy variable.

\[ = - \log R(t) \]

or, \(- \log R(t) = h(u)\int_{0}^{t} = h(t) \cdot t = \lambda t \) where \( h(t) = \text{Constant} = \lambda \)

or, \( R(t) = \exp[ - \int_{0}^{t} h(u) \, du ] = e^{-\int_{0}^{t} h(u) \, du} \) (2.78)

\[ = e^{-H}, \text{ where, } H = \int_{0}^{t} h(u) \, du \]

Above equation is called the reliability function of the system with its hazard rate \( h(u) \).

Also, \( R(t) = \exp[ -\lambda t ] = e^{-\lambda t} \) for \( h(u) = \lambda \), Constant

Similarly, Unreliability;

\[ Q(t) = F(t) = 1 - \exp( - \int_{0}^{t} h(u) \, du ) \] (2.79)
In engineering systems, there may be different hazard models followed by systems during their operation. They may follow one type of hazard model or a mixture of two or more of the hazard models. The hazard models may be of Constant, Linearly increasing or decreasing, Weibull, etc hazard models. The useful life of a piece of electronic equipment is the period of relatively constant failure rate. In this hazard model, the hazard rate is constant throughout the operation of the system. Most of the system has constant hazard rate. Mathematically,

Hazard rate, \( h(t) = \lambda \)  Where \( \lambda \)  is a constant.

We know,
\[
R(t) = \exp(- \int_{u}^{t} h(u) \, du) = e^{-H}, \quad \text{where } H = \int_{u}^{t} h(u) \, du
\]

Thus, \( R(t) = e^{-H} = e^{-\lambda t} \)

Failure Density Function \( f(t) , f(t) = \lambda \cdot e^{-\lambda t} \)

Unreliability, \( F(t) = 1 - e^{-\lambda t} \)

Reliability of constant hazard model is decaying exponentially. The failure density function is exponential distribution function.

**System Reliability Evaluation:** A system consists of several subsystems. Each subsystem may have several elements or components. These elements or components may be connected in different way to form the proper function of the subsystem. Similarly all subsystems of the systems are connected properly for the function of the system. For reliability evaluation, a system may be classified as series, parallel, mixed and non series- parallel systems.

Reliability of non series- parallel systems can be obtained by using any of the following methods i.e. Factoring Theorem or Decomposition method, Path Set method, Tie Set and Minimal Tie Set method, Cut Set and Minimal Cut Set method, Possibility Status method, Event Tree and Reduced event Tree method and Delta – Star conversion method.
2.12 A General Approach for Reliability Evaluation of Communication Network

A communication network may include a number of mobile nodes that should have high responsiveness to deal with the mobility. The mobility effect on responsiveness will compound the reliability challenge. This is very suitable for disaster management during disaster. Many other applications for this type of network require immediate and guaranteed action such as medical emergency alarm, fire alarm detection; monitoring and management of civil works in remote locations particularly dam site, ware monitoring and management, etc. In these situations, information packets have to be communicated in a reliable way and in time through the nodes of the network. Here, besides the energy consumption and delay, data reliability becomes more prominent for the proper functioning of the network. Direct communication between any node and destination could be subject to just a small delay, if the distance between the source and the destination is short. It suffers energy wastage when the distance increases. Therefore, often multi hop short range communications through other sensor nodes, acting as intermediate relay, are preferred in order to reduce the energy consumption in the network. In such a scenario, it is necessary to define efficient technique that can ensure reliable communication with very tight delay constraint. The stochastic factors that affect the reliability of ad hoc networks can be divided into external factors (include network traffic, node mobility, terrain, weather, etc.) and internal factors (include the reliability of network components/equipment, network topology, network protocol, Quality of Service (QoS) assurance mechanism, etc.). Many of these stochastic factors are difficult to incorporate and ascertain when evaluating the reliability of the network [174].

Since MANET has no regular structure. The network consists of more than one path between source node and destination node. The reliability of the network can be evaluated by using different methods as factoring theorem, minimal cut set, minimal tie set, etc. But as the complexity of the network increases, it is more cumbersome to use these methods to evaluate its reliability. Here, an effort is made to describe a generalized method to evaluate the reliability of the directed network by using an example. The reliability of the network is the probability that there exists an
operational path between source node \((V_s)\) and destination node \((V_d)\), at that time and all other paths are faulty with their given link failure probabilities. For example, let us evaluate the reliability of sending a message in between source node \((V_s)\) and destination node \((V_d)\) through a bridge network as shown in the Figure 2.4.

The above bridge network consists of 4 nodes as \(V_1, V_2, V_3\) and \(V_4\). These nodes are interconnected by five directed edges/links as \(u_{12}, u_{13}, u_{23}, u_{24}, \) and \(u_{34}\). The each of the edge has two states viz. faulty/failure state and operating/good state. The path reliability for sending message from source node \((V_1)\) to destination node \((V_4)\) can be calculated by knowing the path set linking from node \(V_1\) to node \(V_4\). There are three successful paths exist between source node \((V_s)\) and destination node \((V_d)\). These paths are given as follows:

- Path \(X_1 = \{ u_{12}, u_{24} \}\)
- Path \(X_2 = \{ u_{13}, u_{34} \}\)
- Path \(X_3 = \{ u_{12}, u_{23}, u_{34} \}\)

Let \(p_{ij}\) = reliability of fault free operation of path/link \(u_{ij}\) and 
\(q_{ij} = 1 - p_{ij}\) is the unreliability of path/link \(u_{ij}\) where \(i < j\)
It is assumed that all nodes are perfectly reliable i.e. fault free. For successful sending a message from source to destination, at least one of the paths \( X_1, X_2 \) and \( X_3 \) must be operational. Hence, the path reliability for sending a message from source to destination can be calculated by constructing a set of disjoint (i.e. mutually exclusive) events and then add up their probabilities. The disjoint events between node \( V_1 \) and node \( V_4 \) are as follows:

i) \( X_1 \) is up/working

ii) \( X_2 \) is up and \( X_1 \) is down

iii) \( X_3 \) is up but both \( X_1 \) and \( X_2 \) are down

Reliability in each of the above cases is based on conditional probabilities and may be evaluated as

i) **Path \( X_1 \) is working**: Reliabilities of this path = \( p_{12} \cdot p_{24} \)

ii) **Path \( X_2 \) is up and path \( X_1 \) is down**: Reliabilities of this event under given condition may be evaluated as below:

let \( E'_{1/2} \) = Event that path \( X_1 \) is faulty given that \( X_2 \) is operating. This will be evaluated as \( X_{1/2} = \{ X_1 \} - \{ X_2 \} \)

\[
= \{ u_{12}, u_{24} \} - \{ u_{13}, u_{34} \}
\]

\[
= \{ u_{12}, u_{24} \} \text{ which must be faulty/failed, while } X_2 \text{ is working.}
\]

Hence, event reliability under given condition is

\[
R(E'_{1/2}) = R(X_{2/1}) = p_{13} \cdot p_{34} [1 - p_{12} \cdot p_{24}]
\]

iii) **\( X_3 \) is up but both \( X_1 \) and \( X_2 \) are down**: For finding event reliability on this condition, several conditional event sets must be considered. The event sets are given as follows:

let \( E'_{1/3} \) = Event that path \( X_1 \) is faulty given that \( X_3 \) is operating. This will be evaluated as

\[
X_{1/3} = \{ X_1 \} - \{ X_3 \}
\]

\[
= \{ u_{12}, u_{24} \} - \{ u_{12}, u_{23}, u_{34} \}
\]

\[
= \{ u_{24} \}
\]
let $E_{2/3}' = \text{Event that path } X_2 \text{ is faulty given that } X_3 \text{ is operating. This will be evaluated as} \quad X_{2/3}' = \{ X_2 \} - \{ X_3 \}
\quad = \{ u_{13}, u_{34} \} - \{ u_{12}, u_{23}, u_{34} \}
\quad = \{ u_{13} \}

X_{1/3}' \text{ and } X_{2/3}' \text{ must be failed, while } X_3 \text{ is working.}

Now, path reliability on this condition is $p_{12} p_{23} p_{34} [1 - p_{24}] [1 - p_{13}] = p_{12} p_{23} p_{34} [q_{24} q_{13}]

Finally, Network reliability is obtained by adding of reliabilities of these paths under specified conditions.

$R(V_S, V_D) = R(V_1, V_4) = p_{12} p_{24} + p_{13} p_{34} [1 - p_{12} p_{24}] + p_{12} p_{23} p_{34} [q_{13} q_{24}]$

Now, this concept of evaluating network reliability may be extended for a large network having several nodes and several directed edges/links as shown in the Figure 2.5. Any node may act like source and the message may be sent to any destination node other than source node. All the existing nodes between source node and destination node are the intermediate/relay nodes. Assume that the network consists of $n$ paths as $X_1, X_2, X_3, \ldots X_n$ exist in between source node $N_S$ and destination node $N_D$. These paths may be given as follows:

Path $X_1 = \{ u_{12}, u_{24}, u_{47}, u_{78}, \ldots \}$
Path $X_2 = \{ u_{12}, u_{25}, u_{58}, \ldots \}$
Path $X_3 = \{ u_{13}, u_{35}, u_{58}, \ldots \}$

$\ldots \ldots$

Path $X_n = \{ u_{12}, u_{36} \} u_{68}, \ldots \ldots \}$

For successful sending a message from source node $N_S$ and destination node $N_D$, at least one of the paths among $X_1, X_2, X_3, \ldots X_n$ must be operational.

Let the $E_i$ be the event in which path $X_i$ is operational and $E_i' = \text{Event that path } X_i \text{ is faulty i.e. the complement of event } E_i$.

The expression for the network reliability may be written as

$R(V_S, V_D) = \text{Prob}\{E_1 U E_2 U \ldots \ldots U E_n\}$
The events $E_1, E_2, \ldots, E_n$ are not disjoint, but they can be decomposed into a set of disjoint events as given below:

$$E_1 \cup E_2 \cup \ldots \cup E_n = E_1 \cup [E_2 \cap E_1'] \cup [E_3 \cap E_1' \cap E_2'] \cup \ldots \cup [E_n \cap E_1' \cap E_2' \cap E_3' \cap \ldots \cap E_{n-1}']$$

![Diagram of a General Communication Network with k Nodes](image)

**Figure 2.5 A General Communication Network with k Nodes**

Now, the reliability of the network will be obtained by adding reliabilities of their paths.

$$R_{(V_i,V_j)} = \text{Probability}\{E_1 \cup E_2 \cup \ldots \cup E_n\}$$

$$= \text{Prob}\{E_1\} + \text{Prob}\{E_2 \cap E_1'\} + \text{Prob}\{E_3 \cap E_1' \cap E_2'\} + \ldots + \text{Prob}\{E_n \cap E_1' \cap E_2' \cap E_3' \cap \ldots \cap E_{n-1}'\}$$

$$= \text{Prob}\{E_1\} + \text{Prob}\{E_2\} \text{Prob}\{E_1'/E_2\} + \ldots + \text{Prob}\{E_n\} \text{Prob}\{E_1' \cap E_2' \cap E_3' \cap \ldots \cap E_{n-1}'/E_n\}$$

Here, $\text{Prob}\{E_1' \cap E_2' \cap E_3' \cap \ldots \cap E_{n-1}'/E_n\} = \text{Prob}\{E_1' \cap E_2' \cap E_3' \cap \ldots \cap E_{n-1}'/E_n\}$ and $E_{i/j}' = \text{Event that path } X_i \text{ is faulty given that path } X_j \text{ is operating.}$
\[ X_i - X_j = x_{ij} \]
\[ x_i / x_k \in x_i \text{ and } x_k \text{ is not } \in x_j \]

A flowchart based on above algorithm for network reliability evaluation is present in the Figure 2.6.

**Figure 2.6 Flowchart of Network Reliability Evaluation**
2.13 Conclusion

Thus, it can be concluded from the above discussion that communication system is a system or facility for transferring data between persons and equipment. The system usually consists of a collection of individual communication networks, transmission systems, relay stations, tributary stations and terminal equipment capable of interconnection and interoperation so as to form an integrated whole. These individual components must serve a common purpose, be technically compatible, employ common procedures, respond to some form of control and generally operate in unison. The Communication system may be categorized based on their physical infrastructure and the specifications of the signals they transmit. The physical infrastructure pertains to the type of the channel used and the hardware design of the transmitting and receiving equipment. The signal specifications signify the nature and type of the transmitted signal. Thus, the types of communication systems based on their infrastructure and signal specifications.

In wireless communications, fading is deviation of the attenuation affecting a signal over certain propagation media. The fading may vary with time, geographical position or radio frequency, and is often modeled as a random process. A fading channel is a communication channel comprising fading. In wireless systems, fading may happen either be due to multipath propagation, referred to as multipath induced fading, or due to shadowing from obstacles affecting the wave propagation, sometimes referred to as shadow fading. Among the various types of fading models for the distribution of the attenuation is a dispersive fading model, with several echoes, each exposed to different delay, gain and phase shift, often constant. This results in frequency selective fading and inter-symbol interference. The gains may be Rayleigh or Rician distributed. The echoes may also be exposed to Doppler shift, resulting in a time varying channel model. Others are namely Nakagami fading, Log-normal shadow fading, Rayleigh fading, Rician fading, Weibull fading, etc. are discussing.

The markets for wireless telephones and communication devices are experiencing rapid growth. It is required and stressed that the reliability of communication network must be good and ascertained during disaster management and just after the natural disaster. In this chapter, different types of communication
network available are demonstrated and their performance related parameters are discussed. An effort is made, to evaluate the reliability of the mobile ad hoc communication network, which is very suitable for reliable communication during disaster management and just after the natural disaster. This method for evaluation of reliability of communication networks is describes by using directed graph theory by taking an example of a bridge network and finally implemented to a large network consists of several nodes and edges. This method is particularly important when communication network reliability is to be evaluated for mission-critical applications such as monitoring of disaster-struck regions, remotely located civil works management, under construction dam site management for multi purpose large hydro power plants, remotely located patient monitoring, battlefield monitoring, home automation, tracking of chemical and explosive agents, etc. These applications are loss-and-delay sensitive and therefore, reliable and timely delivery of information is critical.