“O’ZBEKISTON TEMIR YO’LLARI“ JSC

Tashkent institute of railway engineering

Protection

Permitted

Head of the department

“___”___________2018 y

Department “Electrical communication and radio”

GRADUATION QUALIFICATION WORK

Theme: Frequency division multiple access (FDMA)

Student. Muzaffarov M.M ____________________

Tutor: Ortiqov M.S _________________________

Consultant of language :

Achilov O.R.________________________________

Consultant of labor protection:

Krivoruchko B.V. ___________________________

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Tashkent – 2018 y
“O’ZBEKISTON TEMIR YO’LLARI“ JSC

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Approve

Head of the department “EC and R”

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“___”___________2018 y

THE TASK RECEIVED ON GRADUATION QUALIFICATION WORK
Student:____
full name

1. The theme of graduation work: Frequency division multiple access (FDMA) «____» ___________ 2018 at the meeting session of department.
2. Deadline _______________________________
3. Initially information and basic resources for qualified graduation work: course job and several literatures
4. Should be solved in the following problems
   1. Introduction
   2. Fixed-Assignment Access for Voice-Oriented Networks
   3. Comparison of FDMA, TDMA, and CDMA
   4. Performance of Fixed-Assignment Access Methods
   5. Devices
   6. Labor protection
   7. Conclusion
   8. Literatures

Section consultants of graduation work

SECTIONS	CONSULTANTS	Sign,date	
		Task given	Task received
Labor protection.	KrivoruchkoB. V.		
Main part	Ortiqov M.S		
Language consultant	Achilov O.R.		

The plan of graduation work

№	Sections of graduation work	Expiration date	Sign
1.	Introduction		
2.	Fixed-Assignment Access for Voice-Oriented Networks		
3.	Comparison of FDMA, TDMA, and CDMA		
4.	Performance of Fixed-Assignment Access Methods		
5.	Devices		
6.	Labor protection		
7.	Literature		

Tutor: ass Ortiqov M. S. ________________
(full name) sign

Accepted to accomplishing: Muzaffarov M.M.______________
(full name) (sign)

Task given date ____________________ 2018 year

Contents

ANNOTATION

This graduation dedicated for Frequency Division Multiple Access (FDMA) and in the first part was considered general information about FDMA, TDMA and CDMA standards. Also in the first part we can see several examples, which are related to upon standards.

In the second part we can know what kind of contrasts FDMA, TDMA and CDMA standards. Furthermore we capable of see several checking methods, which are these methods can differentiate those standards.

In the third part was encapsulated fixed-assignment access methods, drawbacks and advantages also discrepancies between two main standards FDMA and TDMA. Moreover in the third part illustrated schemes of FDMA and TDMA standards.

Fourth section is about of devices. There are two types devices and first one is employing wireless connection and this technology is not using our railroad. However second one is employing on our railroad in wire connection.

Finally, in the last part which is related to labor protection. In this part was considered the life safety and technical problems.

This graduation work has been actualized on a technical task, text size 70 pgs., 25 pictures, 6 tables, 7 literatures.

Introduction

President of Uzbekistan Shavkat Mirziyoyev signed a decree "On Uzbekistan's Development Strategy ".

The document has approved Uzbekistan's Five-Area Development Strategy for 2017-2021 which was developed following comprehensive study of topical issues, analysis of the current legislation, law enforcement practices, the best international practices.

The Decree establishes that the timely and effective implementation of the Development Strategy shall be the top priority of all government bodies and their officials.

In the decree, the state bodies and entities, responsible for the implementation of measures envisaged in a state program, have been instructed to pay particular attention to:

- improving the system of handling of letters from natural and legal persons, introducing of new effective tools and methods to establish an open dialogue with ordinary people, putting in practice of a system of reporting to the public, strengthening among the public the trust in authorities;
- inadmissibility of bureaucratic barriers and obstacles when handling of letters from natural and legal persons and inadmissibility of transforming the process of dialogue with ordinary people into "window-dressing";

The Strategy is to be implemented in five stages, each of which provides for approval of a separate annual State program in accordance with a declared name of the year.

This chapter presents an overview of the access methods commonly used in wireless networks. Access methods form a part of Layer 2 of the OSI protocol stack and Layer 3 of the IEEE 802 standard for LANs that is responsible for interacting with the medium to coordinate the successful operation of multiple terminals over the wireless channel. Most multiple access methods were originally developed for wired networks and later on adopted to the wireless medium. However, requirements on the wired and wireless media are different, thereby demanding modifications in the original protocols to make them suitable for the wireless medium. Today the main differences between wireless and wired channels are availability of bandwidth and reliability of transmissions. The wired medium is moving toward optical media with enormous bandwidth and very reliable transmission. Bandwidth in wireless systems is always limited because the medium (air) cannot be duplicated, and the medium is shared between all wireless systems, including multichannel broadcast television and a number of other bandwidth demanding applications and services. In the case of wired operation, we can always lay additional cable to increase the capacity as needed even if it is an expensive proposition. In a wireless environment, we can reduce the size of *cells* to increase capacity. With the reduction of the size of the cells, the number of cells increases, and the need for improvements in the wired infrastructure to connect these cells increases. Also the complexity of the network for handling additional handoffs and mobility management increases posing a practical limitation upon the maximum capacity of the network. As far as transmission reliability is concerned, the wireless medium always suffers from multipath and fading, which causes a serious threat to reliable data transmission over the communication link. Because the wireless channel is so unreliable, as discussed in Chapter 3, people have developed a number of signal processing techniques to improve transmission reliability over the wireless channel. In spite of these techniques, the reliability of the wireless medium is far below that of the wired medium used as the backbone of the wireless networks.

Although in practice we prefer to have the same access method and the same frame structure for wired backbone and the wireless access, wireless networks often use different packet sizes and a modified access method to optimize the performance to the specifics of the unreliable wireless medium.

Example: TF.F.F. 802.3 and IEEE 802.11

The IEEE 802.3 standard (based on Ethernet) is the successful and dominant standard for wired LANs. Consequently, the IEEE 802.11 WLAN standard, in ideal situations, desired using the same access method as previously established with IEEE 802.3. Carrier sense multiple-access with collision detection (CSMA/CD) is the protocol used in the Ethernet. However, collision detection in wireless channels is extremely challenging, and IEEE 802.11 had to resort to carrier sense multiple-access with collision avoidance (CSMA/CA) that can be viewed as a wireless adaptation of IEEE 802.3

Example: ATM and Wireless ATM

In the 1.990s, ATM was perceived to be the transmission scheme for all future networking. In the mid 1990s when wireless solutions were considered, a wireless ATM working group was formed to extend the ATM short packet solution with QoS for wireless access. The group had to make significant compromises as discussed in Chapter 12 because ATM was designed for broadband and reliable transmission over optical channels.

To avoid substantial overlap with existing literature, we describe access methods used in wireless networks with justification of why and how they are employed in different wireless networks.

As we explained in previous chapters, wireless networks have evolved around voice or data applications, and as a result we can divide them into voice- and data-oriented networks. The access methods adopted by voice- and data-oriented networks are quite different. Voice-oriented networks are designed for relatively long telephone conversations as the main application. Each communication session for this application exchanges several megabytes of information in both directions. A signaling channel that exchanges short messages between two calling components sets up the call by obtaining resources (such as the link, switches, etc.) in the telephone network at the beginning of the conversation and terminates these arrangements by releasing the resources at the end of the call. The wireless access methods evolved for interaction with these networks assigns a slot of time, a portion of frequency, or a specific code to a user preferably for the entire length of the conversation. We refer to these techniques as fixed- assignment channel access methods or channel partitioning techniques. Data networks were originally designed for bursts of data for which the supporting network does not have a separate signaling channel. In packet communications each packet carries some “signaling information” related to the address of the destination and the source. We refer to the access methods used in these networks as random-access methods accommodating randomly arriving packets of data. Certain local area data networks also *take turns* in accessing the medium as in the case of token passing and polling schemes. In some other cases, the random access mechanisms are used to temporarily *reserve* the medium for transmitting the packet. In the next two sections of this chapter, we provide a short description of the fixed-assignment and random access methods used in voice- and data-oriented wireless networks, respectively.

1. Fixed-Assignment Access for Voice-Oriented Networks

AII existing voice-oriented wireless networks such as cellular telephony or PCS services use fixed-assignment channel access or channel partitioning techniques. In the fixed-assignment access method, a fixed allocation of channel resources, frequency, time, or a spread spectrum code are made available on a predetermined

basis to a single user for the duration of the communication session. The three basic fixed-assignment multiple-access methods are FDMA, TDMA, and CDMA The choice of an access method will have a great impact on the capacity and QoS provided by a network. The impact of multiple access schemes is so important that we commonly refer to various voice-oriented wireless systems by their channel access method, which is only a part of the layer two specification of the air interface of the network.

Example: Common Terminology for Digital Cellular Systems

The GSM and the North American IS-136 digital cellular standards are commonly referred to as digital TDMA cellular systems and the IS-95/IMT-2000 standards are called digital CDMA cellular systems.

In reality, different systems use different modulation techniques as well. However, as we will see in the rest of this book, the impact of the choice of access method on the capacity and overall performance of the network is much more profound. Consequently, the system is really distinguished by its access method. As we will see in our examples of cellular networks, a network that is identified with an access technique often uses other random or fixed assignment techniques as a part of its overall operation. However, it is identified by the access techniques employed for transferring the main information source for which the network is designed to carry.

Example : Random Access Techniques in Cellular Networks

GSM uses slotted ALOHA (a random access method) to establish a link between the mobile terminal and the base station. It also has an optional frequency- hopping pattern that improves the system performance when there is fading of the radio signal. However, the GSM network is built for voice communications, and each session uses TDMA as the access method.

Another important design parameter related to the access method is the differentiation between the carrier frequencies of the forward (downlink—communication between the base station and mobile terminals) and reverse (uplink—communication between the mobile terminal and the base station) channels. If both forward and reverse channels use the same frequency band for communications, but the forward and reverse channels employ alternating time slots, the system is referred to as employing TDD. If the forward and reverse channels use different carrier frequencies that are sufficiently separated, the duplexing scheme is referred to as FDD. With TDD, because only one frequency carrier is needed for a duplex operation, we can share more of the RF circuitry between the forward and the reverse channels. The reciprocity of the channel in

TDD allows for exact open-loop power control and simultaneous synchronization of the forward and reverse channels. TDD techniques are used in systems intended for low-power local area communications where interference must be carefully controlled and where low complexity and low-power consumption are very important. Thus TDD systems are often used in local area pico- or microcellular systems deployed by PCS networks. FDD is mostly used in macro cellular systems designed for coverage of several tens of kilometers where implementation of TDD is more challenging (see Fig. 1.1).

1.1. Frequency Division Multiple Access (FDMA)

In an FDMA environment, all users can transmit signals simultaneously, and they are separated from one another by their frequency of operation. The FDMA technique is built upon FDM. FDM is the oldest and still a commonly used multiplexing technique in the trunks connecting switches in the PSTN. It is also the choice of radio and TV broadcast, as well as cable TV distribution. FDM is more suitable for analog technology because it is easier to implement. When FDM is used for channel access, it is referred to as FDMA,

Example 4.5: FDMA in AMPS with FDD

Figure 1.1(a) shows the FDMA/FDD system commonly used in 1G analog cellular systems such as AMPS and a number of early cordless telephones. In FDMA/FDD systems, forward and reverse channels use different carrier frequencies, and a fixed subchannel pair is assigned to a user terminal during the communication session. At the receiving end, the mobile terminal filters the designated channel out of the composite signal. As shown in Figure , the AMPS system allocates 30 kHz of bandwidth for each forward and reverse channel. There are a total of 421 channels in 25 MHz of spectrum assigned to each direction; 395 of these channels are used for the voice traffic and the rest for signaling.

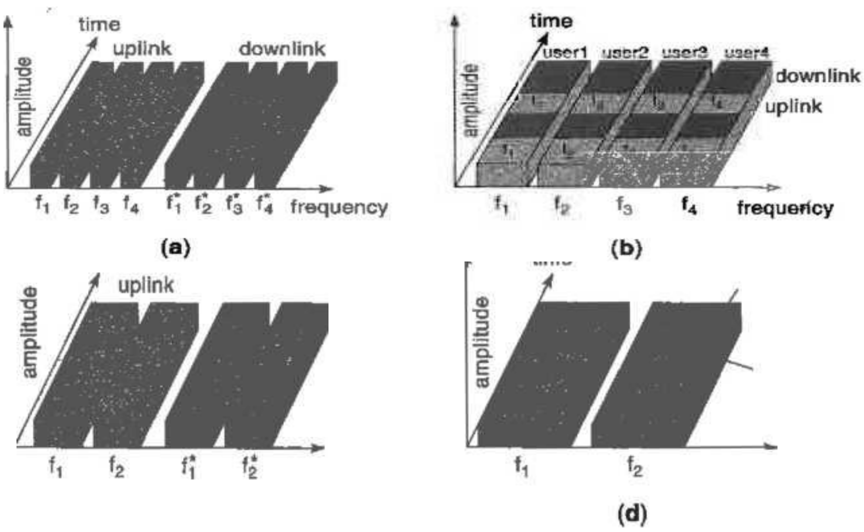


Figure 1.1 (a) FDMA/FDD, (b) FDMA/TDD, (c) TDMA/FDD with multiple carriers, (d) TDMA/TDD with multiple carriers.

Example: FDMA in CT-2 with TDD

Figure 4.1(b) shows an FDMA/TDD system used in the CT-2 digital cordless telephony standard. Each user employs a single carrier frequency for all communications. The forward and reverse transmissions take turns via alternating time slots. This system was designed for distances of up to 100 meters, and a voice conversation is based on 32 kbps ADPCM voice coding. As shown in Figure 4.2(b) the total allocated bandwidth for CT-2 is 4 MHz, supporting 40 carriers each using 100 kHz of bandwidth.

The designer of an FDMA system must pay special attention to adjacent channel interference, in particular in the reverse channel. In both forward and reverse channels, the signal transmitted must be kept confined within its assigned band, at least to the extent that the out-of-band energy causes negligible interference to the users employing adjacent channels. Operation of the forward channel in wireless FDMA networks is very similar to wired FDM networks. In forward wireless channels, in a manner similar to that of wired FDM systems, the signal received by all mobile terminals has the same received power, and interference is controlled by adjusting the sharpness of the transmitter and receiver filters for the separate carrier frequencies. The problem of adjacent channel interference is much more challenging on the reverse channel. On the reverse channel, mobile terminals will be operating at different distances from the BS. The RSS at the BS, of a signal transmitted by a mobile terminal close to the BS, and the RSS at the BS of a transmission by a mobile terminal at edges of the cell are often substantially different, causing problems in detecting the weaker signal. This problem is usually referred to as the near-far problem. If the out-of-band emissions are large, they may swamp the actual information-carrying signal.

Problem 1: Near-Far Problem

a) What is the difference between the received signal strength of two terminals located in 10 m and 1 km from a base station in an open area?

b) Explain the effects of shadow fading on the difference in the RSS.

c) What would be the impact if the two terminals were operating in two adjacent channels? Assume out-of-band radiation that is 40 dB below the main lobe.

Solution:

1. As we saw in Chapter 2, the received signal strength falls by around 40 dB per decade of distance in open areas. Therefore, the received powers from a mobile terminal that is 10 meters from a BS and another, that is at a distance of 1 km, are 80 dB apart.

2. In addition to the fall of the RSS with distance, we also discussed the issues of multi- path and shadow fading in radio channels that cause power fluctuations on the order of several tens of dBs. Therefore, the difference in the received powers due to the near-far problem may exceed even 100 dB.

3. If the out-of-band emission is only *40* dB below that of the transmitted power, it may exceed the strength of the information-bearing signal by almost 60 dB.

To handle the near-far problem, FDMA cellular systems adopt two different measures. First, when frequencies are assigned to a cell, they are grouped such that the frequencies in each cell are as far apart as possible. The second measure employed is power control that is discussed in Chapter 6. In addition, whenever

FDMA is employed, *guard bands* are included in the frequency channel to further reduce adjacent channel interference. This, however, has the effect of reducing the overall spectrum efficiency.

1.2. Time Division Multiple Access (TDMA)

In TDMA systems, a number of users share the same frequency band by taking assigned turns in using the channel. The TDMA technique is built upon the TDM schemes commonly used in the trunks for the telephones systems. The major advantage of the TDMA over FDMA is its format flexibility. Because of the fully digital format and the flexibility of buffering and multiplexing functions, time-slot assignments among multiple users are readily adjustable to provide different access rates for different users. This feature is particularly adopted in the PSTN, and the TDM scheme forms the backbone of all digital connections in the heart of the PSTN. The hierarchy of digital transmission trunks used in North America is the so-called T-carrier system that has an equivalent European system (the E-carriers) approved by the ITU. In the hierarchy of digital transmission rates standardized throughout North America, the basic building block is the 1.544 Mbps link known as T-l carrier. A T-l transmission frame is formed by TDD 24 PCM-encoded voice channels, each carrying 64 kbps of users data. Service providers often lease T-carriers to interconnect their own switches and routers and for forming their own networks.

Example : The Use of T-carriers in Cellular Networks

Cellular networks often lease T-carriers from the long-haul telephone companies to interconnect their own switches referred to as mobile switching centers(MSCs), The difference between the MSC and a regular switch in the PSTN is that the MSC can support mobility of the terminal. The details of these differences are discussed in later chapters when we provide examples of cellular networks. The end-user subscribes to the cellular service provider.

Example: The Use of T-l Lines in the Internet

The routers in the Internet are sometimes connected through leased T-carrier telephone lines to form part of the Internet. The difference between a router and a PSTN switch is that the router can handle packet switching whereas the PSTN switch uses circuit switching. The end-user subscribes to an Internet service provider (ISP) m this case.

With TDMA, a transmit controller assigns time slots to users, and an assigned time slot is held by a user until the user releases it. At the receiving end, a receiver station synchronizes to the TDMA signal frame, and extracts the time slot designated for that user. The heart of this operation is synchronization that was not needed for FDMA systems. The TDMA concept was developed in the 1960s for use in digital satellite communication systems and first became operational commercially in telephone networks in the mid-1970s.

In cellular and cordless systems, the migration to TDMA from FDMA took place in the 2G systems. The first cellular standard adopting TDMA was GSM. The GSM standard was initiated to support international roaming among Scandinavian countries in particular and the rest of Europe in general. The digital voice adoption in TDMA format facilitated the network implementation, resulted in improve- . ments in the quality of the voice, and provided a flexible format to integrate data services in the cellular network. The FDMA systems in the United States very quickly observed a capacity crunch in major cities, and among the options for increasing capacity, TDMA was adopted initially through the IS-54 system that was later on replaced by IS-136. TDMA was adopted in 2G cordless telephones such as DECT to provide format flexibility and to allow more compact and low-power terminals.

Example: TDMA in GSM

Figure 4.1(c) shows an FDMA/TDMA/FDD channel used in 2G digital cellular networks. Figure 4.3 shows a particular example of the 8-slot TDMA scheme used in the GSM system. Forward and reverse channels use separate carrier frequencies (FDD). Each carrier can support up to eight simultaneous users via TDMA, each using a 13 kbps encoded digital speech, within a 200

MHz carrier bandwidth. A total of 124 frequency carriers (FDMA) are available in the 25 MHz allocated band in each direction. One hundred kHz of band is allocated as a guard band at each edge of the overall allocated band.

Example : TDMA in DECT

Figure 4.1(d) shows an FDM A/TDM A/TDD system used in the Pan-European digital PCS standard DECT. Because distances are short, a TDD format allows

Example : TDMA in IS-136

Figure 4.5 shows the frame format for the TDMA/FDD with six slots considered for IS-136 both for the forward (base to mobile) and reverse (mobile to base) channels. In IS-136 each 30 kHz digital channel has a channel transmission rate of 48.6 kbps. The 48.6 kbps stream is divided into six TDMA channels of 8.1 kbps each. The IS-136 slot and frame format, shown in Figure 4.5, is much simpler than that of the GSM standard. The 40-ms frame is composed of six 6.67-ms time slots. Each slot contains 324 bits, including 260 bits of user data, and 12 bits of system control information in a slow associated control channel (SACCH). There is also a 28-bit synchronization sequence, and a 12-bit digital verification color code (DVCC) used to identify the frequency channel to which the mobile terminal is tuned. In the mobile-to-base direction, the slot also contains a guard time interval of a six-bit duration when no signal is transmitted, and a six-bit ramp interval to allow the transmitter to reach its full output power level.

Due to the near-far problem, the received signal on the reverse channel from a user occupying a time slot can be much larger than the received power from the terminal using the adjacent time slot. In such a case, the receiver will have difficulties in distinguishing the weaker signal from the background noise. In a manner similar to FDMA systems, TDMA systems also use power control to handle this near-far problem.

1.3. Code-Division Multiple Access (CDMA)

With the growing interest in the integration of voice, data, and video traffic in telecommunication networks, CDMA appears increasingly attractive as the wireless access method of choice. Fundamentally, integration of various types of traffic is readily accomplished in a CDMA environment as coexistence in such an environment does not require any specific coordination among user terminals. In principle, CDMA can accommodate various wireless users with different bandwidth requirements, switching methods and technical characteristics without any need for coordination. Of course, because each user signal contributes to the interference seen by other users, power control techniques are essential in the efficient operation of a CDMA system.

To illustrate CDMA and how it is related to FDMA and TDMA, it is useful to think of the available band and time as resources we use to share among multiple users. In FDMA, the frequency band is divided into slots, and each user occupies that frequency throughout the communication session. In TDMA, a larger frequency band is shared among the terminals, and each user uses a slot of time during the communication session. As shown in Figure 4.6, in a CDMA environment multiple users use the same band at the same time, and the user is differentiated by a code that acts as the key to identify that user. These codes are selected so that when they are used at the same time in the same band a receiver knowing the code of a particular user can detect that user among all the received signals. In the CDMA/FDD [Figure 4.7(a)] that is used in IS-95 and IMT-2000, the forward and reverse channels use different carrier frequencies. If both transmitter and receiver use the same carrier frequency [Figure 4.7(b)], the system is CDMA/TDD.

In CDMA, each user is a source of noise to the receiver of other users, and if we increase the number of users beyond a certain value, the entire system collapses because the signal received in each specific receiver will be buried under the noise caused by many other users. An important question is, how many users can simultaneously use a CDMA system before the system collapses?

Capacity of CDMA

CDMA systems are implemented based on the spread spectrum technology that is presented in Chapter 3. In its most simplified form, a spread spectrum transmitter spreads the signal power over a spectrum *N* times wider than the spectrum of the message signal. In other words, an information bandwidth of *R* occupies a transmission bandwidth of *W*, where:

$$W = NR \qquad (1)$$

The spread spectrum receiver processes the received signal with a *processing gain* of *N*. This means that during the processing at the receiver, the power of the received signal having the code of that particular receiver will be increased *N* times beyond the value before processing.

Let us consider the situation of a single cell in a cellular system employing CDMA. Assume that we have *M* simultaneous users on the reverse channel of a CDMA network. Further let us assume that we have an ideal power control enforced on the channel so that the received power of signals from all terminals has the same value *P*, Then, the received power from the target user after processing at the receiver is *NP*, and the received interference from *M* - 1 other terminals is *(M* - 1)P. If we also assume that a cellular system is interference limited and the background noise is dominated by the interference noise from other users, the received signal-to-interference ratio for the target receiver will be:

$$S_r=\frac{NP}{(M-1)P} = \frac{N}{M-1} \qquad (2)$$

All users always have a requirement for the acceptable error rate of the received data stream. For a given modulation and coding specification of the system, that error rate requirement will be supported by a minimum S_r

requirement that can be used in Eq. (4.2) to solve for the number of simultaneous users. Then, solving Eqs. (4.1) and (4.2) for *M*, we will have:

$$M = \frac{W}{R}\frac{1}{\mathrm{Sr}} + 1 \cong \frac{W}{R}\frac{1}{\mathrm{Sr}} \qquad (3)$$

CDMA/TDD

Problem 2: Capacity of One Carrier in a Single-Cell CDMA System

Using QPSK modulation and convolutional coding, the IS-95 digital cellular systems require 3 dB < S'_r < 9 dB. The bandwidth of the channel is 1.25 MHz, and the transmission rate is R = 9600 bps. Find the capacity of a single IS-95 cell.

Solution:

Using Equation (3) we can support from up to

$$M = \frac{1.25\,MHz}{9600\,bps}\,\frac{1}{8} \cong 16 \quad to\ M = \frac{1.25\,MHz}{9600\,bps}\,\frac{1}{2} \cong 65\ \ users$$

Practical Considerations

In the practical design of digital cellular systems, three other parameters affect the number of users that can be supported by the system, as well as the bandwidth efficiency of the system. These are the number of sectors in each base station antenna, the voice activity factor, and the interference increase factor. These parameters are quantified as factors used in the calculation of the number of simultaneous users that the CDMA system can support. The use of sectored antennas is an important factor in maximizing bandwidth efficiency. Cell sectorization using directional antennas reduces the overall interference, increasing the allowable number of simultaneous users by a *sectorization gain factor*, which we denote by G_A. With ideal sectorization the users in one sector of a base station antenna do not interfere with the users operating in other sectors, and $G_A = N_{sec}$ where N_{sec} is the number of sectors in the cell. In practice antenna patterns cannot be designed to have ideal characteristics, and due to multipath reflections, users in general communicate with more than one sector. Three-sector base station antennas are commonly used in cellular systems, and a typical value of the sectorization gain factor is assumed to be $G_A = 2.5$ (4 dB). The voice activity interference reduction factor G_v is the ratio of the total connection time to the active talk spurt time. On the average, in a two-way conversation, each user talks roughly 50 percent of the time. The short pauses in the flow of natural speech reduce the activity factor further to about 40 percent of the connection time in each direction. As a result, the typical number used for G_v is 2.5 (4 dB). The interference increase factor H_0 accounts for users in other cells in the CDMA system. Because all neighboring cells in a CDMA cellular network operate at the same frequency, they will cause additional interference. This interference is relatively small due to the processing gain of the system and the distances involved; a value of $H_0 = 1.6$ (2 dB) is commonly used in the industry.

Incorporating these three factors as a correction to Equation (3), the number of simultaneous users that can be supported in a CDMA cell can be approximated by

$$M = \frac{W}{R}\,\frac{1}{\mathrm{Sr}} + 1 \cong \frac{W}{R}\,\frac{1}{\mathrm{Sr}}\,\frac{GaGr}{Ho} \qquad (4)$$

If we define the performance improvement factor in a digital cellular system as

$$K = \frac{GaGr}{Ho} \qquad (5)$$

assuming the typical parameter values given earlier, the performance improvement factor is $K = 4$ (6 dB).

Problem 3: Capacity of One Carrier in a Multi-Cell CDMA System with Correction Factors

Determine the multicell IS-95 CDMA capacity with correction for sectorization and voice activity. Use the numbers from Problem 2.

Solution:

If we continue the previous example with the new correction factor included, the range for the number of simultaneous users becomes $64 < M < 260$.

2. Comparison of FDMA, TDMA, and CDMA

With the success of IS-95 CDMA systems in its challenge to conventional IS-136 TDMA systems in the United States and the adoption of W-CDMA as the primary choice for the 3G cellular networks, one wonders why CDMA has become the favorite choice for wireless access in voice-oriented networks. Spread spectrum technology became the favorite technology for military applications because of its capability to provide a low probability of interception and strong resistance to interference from jamming. In the cellular industry, CDMA was introduced as an alternative to TDMA to improve the capacity of 2G cellular systems in the United States. As a result, much of the early debates in this area were focused on calculation of the capacity of CDMA as it is compared with TDMA. However, capacity is not the only reason for the success of the CDMA technology. As a matter of fact, calculation of the capacity of CDMA using the simple approach provided earlier is *not* very conclusive and is subject to a number of assumptions such as perfect power control that cannot be practically met. The first CDMA service providers in the ’ United States were using slogans such as “you cannot believe your ears!” to address j the superior quality of voice for the CDMA. However, the superiority of voice is partially dependent on the speech coder, and it is not a CDMA versus TDMA issue. In order to provide a good explanation for the success of a complex and multidisciplinary technology, such as a cellular network, addressing consumer market issues | has always been very important. Those of us involved in this debate for the past | decade have seen the discussion of the ups and downs of CDMA in variety of forums. One of the most interesting events that the principal author remembers was in | 1997 in a major wireless conference in Taipei where one the most famous figures in this debate in his keynote speech at the opening of the conference declared that “we have seen in the past that the VHS which was not a better technology defeated BETA.” In his perception, at that time, CDMA was similar to BETA. In less than a year or so after that, CDMA was selected by a number of different communities around the world as the technology of choice for 3G and ШТ-2000.

In the rest of this section, we bring out a number of issues that may enlighten the reader toward a deeper understanding of the technical aspects of CDMA systems as they are compared with TDMA and FDMA networks. We hope that this may lead the reader to her/his own conclusion about the success of CDMA.

2.1 Format Flexibility.

Telephone voice was the dominant source of income for the telecommunication industry up to the end of the past century. In the new millennium, the strong emergence of Internet and cable TV industries has created a case for other popular multimedia applications. The cellular phones that were designed for telephony applications are now being used for other applications and need support for multimedia applications. To support a variety of data rates with different requirements, a network needs format flexibility. As we discussed earlier, one of the reasons for migrating from analog FDMA to digital TDMA was that TDMA provides a more flexible environment for integration of voice and data. The time slots of a TDMA network designed for voice transmission can be used individually or in a group format to transmit data from users and to support different data rates. However, all these users should be time synchronized and the quality of the transmission channel is the same for all of them. The chief advantage of CDMA relative to TDMA is its flexibility in timing and the quality of transmission. In CDMA users are separated by their codes, unaffected by the transmission time relative to other users. The power of the user can also be adjusted with respect to others to support a certain quality of transmission. In CDMA each user is far more liberated from the other users, allowing a fertile setting to accommodate different service requirements to support a variety of transmission rates with different qualities of transmission to support multimedia or any other emerging application.

2.2 Performance in Multipath Fading.

As we saw in Chapter 2, multipath in wireless channels causes frequency selective fading. In frequency selective fading, when the transmission band of a narrowband system coincides with the location of the fade, no useful signal is received. As we increase the transmission bandwidth, fading will occupy only a portion of the transmission band, providing an opportunity for a wideband receiver to take advantage of the portion of the transmission band not under fade and a more reliable communication link. In Chapter 3 we introduced DFE, OFDM, sectored antennas, and spread spectrum as technologies that can be employed in wideband systems to handle frequency selective fading. The wider the bandwidth, the better is the opportunity for averaging out the faded frequency.

These technologies are not used in the 1G analog cellular FDMA systems because they were analog systems and these techniques are digital. The Pan European GSM digital cellular system uses 200 kHz of band, and the standard recommends using DFE. The North American digital cellular system, IS-136, uses digital transmission over the same analog band of 30 kHz of the North American AMPS system and does not recommend equalization because the bandwidth is not very large. An equalizer needs additional circuitry, and some power budget at the receiver that was one of the drawbacks considered in IS-

136. The bandwidth of the IS-95 CDMA system is 1.25MHz and W-CDMA systems for 3G networks use bandwidths that are as high 10 MHz. RAKE receivers are used to increase the benefits of wideband transmission by taking advantage of the so-called in-band or time diversity of the wideband signal. This is one of the reasons for having a better quality of voice in CDMA systems. As we mentioned earlier, quality of voice is also effected by the robustness of the speech-coding algorithm, coverage of service, methods to handle interference, handoffs, and power control as well.

2.3 System Capacity.

Comparison of the capacity depends on a number of issues, including the frequency reuse factor, speech coding rate, and the type of antenna. Therefore a fair comparison would be difficult unless we go to practical systems. The following simple example compares the capacity of FDMA (AMPS), TDMA (IS-136), and CDMA (IS-95) used in debates to evaluate alternatives for the 2G North American digital cellular systems to replace the 1G analog.

Problem 4: Comparison of the Capacity of Different 2G Systems

Compare the capacity of IS-95 CDMA with AMPS FDMA and IS-136 TDMA systems. For the CDMA system, assume an acceptable signal to interference ratio of 6 dB, data rate of 9600 bps, voice duty cycle of 50 percent, effective antenna separation factor of 2.75 (close to ideal 3-sector antenna), and neighboring cell interference factor of 1.67.

Solution:

For the IS-95 CDMA using Equation (4.4) for each carrier with W = 1.25 MHz, R = 9600 bps, $S_r = 4$ (6dB), G_v = 2 (50 percent voice activity), G_A • 2.75, and H_0 = 1.67 we have:

$$M = \frac{W}{R}\frac{1}{\mathrm{Sr}} + 1 \cong \frac{W}{R}\frac{1}{\mathrm{Sr}}\frac{GaGr}{Ho} = 108 \text{ users per cell}$$

For the IS-136 with a carrier bandwidth of W_c = 30 kHz, the number of users per carrier of N_u = 3, and frequency reuse factor of K = 4 (commonly used in these systems), each W = 1.25 MHz of bandwidth provides for

$$M = \frac{W}{Wc}\frac{Nm}{K} = 31.25 \text{ users per cell}$$

For the AMPS analog system with earner bandwidth of W_c= 30 kHz, and frequency reuse factor of K — 7 (commonly used in these systems), each W = 1.25 MHz of bandwidth provides for

$$M = \frac{W}{Wc}\frac{1}{K} = 6 \text{ users per channel}$$

Another example of this form is instructive to compare these systems with the GSM.

Problem 5: Comparison of NA Systems with GSM

Determine the capacity of GSM for K = 3.

Solution:

For the GSM system with a carrier bandwidth of W_c= 200 kHz, the number of users per carrier of N_u = 8, and frequency reuse factor of K = 3 (commonly used in these systems), each W = 1.25MHz of bandwidth provides for

$$M = \frac{W}{Wc}\frac{Nm}{K} = 16{,}7 \text{ users per cell}$$

2.4 Handoff.

As we discuss in Chapter 6, handoff occurs when a received signal in an MS becomes weak and another BS can provide a stronger signal to the MS. The 1G FDMA cellular systems often used the so-called hard-decision handoff in which the base station controller monitors the received signal from the BS and at the appropriate time switches the connection from one BS to another. TDMA systems use the so-called *mobile-assisted handoff* in which the mobile station monitors the received signal from available BSs and reports it to the base station controller which then makes a decision on the handoff. Because adjacent cells in both FDMA and TDMA use different frequencies, the MS has to disconnect from and reconnect to the network that will appear as a click to the user. Handoffs occur at the edge of the cells when the received signals from both BSs are weak. The signals also fluctuate anyway because they are arriving over radio channels. As a result, decision making for the handoff time is often complex, and the user experiences a period of poor signal quality and possibly several clicks during the completion of the handoff process. Because adjacent cells in a CDMA network use the same frequency, a mobile moving from one cell to another can make “seamless” handoff by the use of signal combining. When the mobile station approaches the boundary between cells, it communicates with both cells, A controller combines the signals from both links to form a better communication link. When a reliable link has been established with the new base station, the mobile stops communicating with the previous base station, and communication is fully established with the new base station. This technique is referred to as soft handoff. Soft handoff provides a dual diversity for the received signal from two links which improves the quality of reception and eliminates clicking as well as the Ping-Pong problem. Handoff is an important issue that has many more details.

2.5 Power Control.

As we discussed earlier in this chapter, power control is necessary for FDMA and TDMA systems to control adjacent channel interference and mitigate the unexpected interference caused by the near-far problem. In FDMA and TDMA systems, some sort of power control is needed to improve the quality of the voice delivered to the user. In CDMA, however, the capacity of the system depends *directly* on the power control, and an accurate power control mechanism is needed for proper operation of the network. With CDMA, power control is the key ingredient in maximizing the number of users that can operate simultaneously in the system. As a result, CDMA systems adjust the transmitted power more often and with smaller adjustment steps to support a more refined control of power. Better power control also saves on the transmission power of the MS, which increases the life of the battery. The more refined power control in CDMA systems also helps in power management of the MS, which is an extremely important practical issue for users of the mobile terminals..

Implementation Complexity**.** Spread spectrum is a two-layer modulation technique requiring greater circuit complexity than conventional modulation schemes.This in turn will lead to higher electronic power consumption and larger weight and cost for mobile terminals. Gradual improvements in battery and integrated circuit technologies, however, have made this issue transparent to the user.

3. Performance of Fixed-Assignment Access Methods

Fixed assignment access methods are used with circuit switched cellular and PCS telephone networks. In these networks, in a manner similar to the wired multichannel environments, the performance of the network is measured by the blockage rate of an initiated call. A call does not go through for two reasons: (1) when the calling number is not available, and (2) when the telephone company is out of resources to provide a line for the communication session. In POTS, for both cases the user hears a busy tone signal and cannot distinguish between the two types of blockage. In most cellular systems, however, type (1) blockage results in a response that is a busy tone and type (2) with a message such as “All the circuits are busy at this time please try your call later.” In the rest of this book, we refer to blockage rate only as a type (2) blockage rate. The statistical properties of the traffic offered to the network are also a function of time. The telephone service providers often design their networks so that the blockage rate at peak traffic is always below a certain percentage. Cellular operators often try to keep this average blockage rate below 2 percent.

The blockage rate is a function of the number of subscribers, number of initiated calls, and the length of the conversations. In telephone networks, the Erlang equations are used to relate the probability of blockage to the average rate of the arriving calls and the average length of a call. In wired networks, the number of lines or subscribers that can connect to a multichannel switch is a fixed number. The telephone company monitors the statistics of the calls over a long period of time and upgrades the switches with the growth of subscribers so that the blockage rate during peak traffic times remains below the objective value. In cellular telephony and PCS networks, the number of subscribers operating in a cell is also a function of time. In the downtown areas, everyone uses their cellular telephones during the day, and in the evenings they use them in their residential area which is covered by a different cell. Therefore, traffic fluctuations in cellular telephone networks are much more than the traffic fluctuations in POTS. In addition, telephone companies can easily increase the capacity of their networks by increasing their in-. vestment on the number of transmission lines and quality of switches supporting network connections. In wireless networks, the overall number of available channels for communications is ultimately limited by the availability of the frequency bands assigned for network operation. To respond to the fluctuations of the traffic and cope with the bandwidth limitations, cellular operators use complex frequency assignment strategies to share the available resources in an optimal manner.

3.1 Traffic Engineering Using the Erlang Equations

The Erlang equations are the core of the traffic engineering for telephony applications. The two basic equations used for traffic engineering are Erlang B and Erlang C equations. The Erlang B equation relates the probability of blockage

B(N,p) to the number of channels *N* and the normalized call density in units of channels p. The Erlang B formula is:

$$B(N, \rho) = \frac{\rho^N / N!}{\sum_i^N (\rho^i, i!)} \qquad (6)$$

where p = λ/μ , λ is the call arrival rate and μ is the service rate of the calls.

Problem 6: Call Blocking Using Erlang B Formula

We want to provide a wireless public phone service with five lines to a ferry crossing between Helsinki and Stockholm carrying 100 passengers where on the average each passenger makes a three-minute telephone call every two hours. What is the probability of a passenger approaching the telephones and none of the four lines are available?

Solution;

In practice, often the probability of call blockage is given, and we need to calculate the number of subscribers. Here we need an inverse function for the Erlang equation that is not available. As a result, a number of tables and graphs arc available for this inverse mapping. Figure 4.8 shows a graph relating the probability of blockage *B(N,p)* to the number of channels *N* and the normalized traffic per available channels p. From this graph, we can estimate the blocking probability. The traffic load is 100 users X 1 call/user X 3 minutes/call per 120 minutes = 2.5 Erlangs. Because there are five lines available and the traffic is 2.5 Erlangs, the blocking probability is roughly 0.07.

Problem 7: Capacity Using Erlang B Formula

An IS-136 cellular phone provider owns 50 cell sites and 19 traffic carriers per cell each with a bandwidth of 30 kHz. Assuming each user makes three calls per hour and the average holding time per call of five minutes, determine the total number of subscribers that the service provider can support with a blocking rate of less than 2 percent.

Solution:

The total number of channels is *N* = 19 X 3 = 57 per cell. For *B(N*,p) = 0,02 and N = 57 Figure 4.8 shows that the p = 45 Erlangs. With an average of five calls per minute, the service rate is ц, = 1/5 minutes, and the acceptable arrival rate of the calls is X = p X |x = 1/5 (min^{-1}) X 45 (Erlang) = 9 (Erlang/min). With an average of 3 calls per hour, the system can accept 9 (Erlang/min) / 3 (Erlang) / 60 (min) = 180 subscribers per cell. Therefore the total number of subscribers are 180 (subscribers/cell) X 50 (cells) = 8,000 subscribers.

The Erlang C formula relates the waiting time in a queue if a call does not go through, but it is buffered until a channel is available.

Advantages and drawbacks FDMA

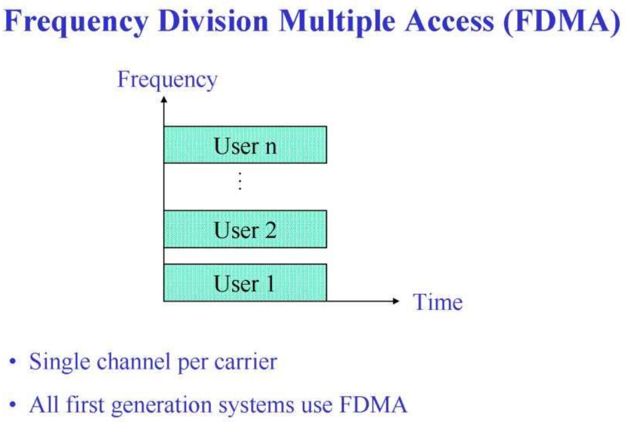
In a wide variety of communication systems, modulation as a fundamental technique plays a very important role in data transmission through air within a specified spectral bandwidth. A modulation signal, which is usually represented by a low-frequency baseband signal and is commonly referred to as an information- bearing signal, varies or modulates one of three parameters: *amplitude, phase,* and *frequency* of the radio frequency (RF) carrier signal such that the baseband signal is carried by a varied parameter of the carrier signal through atmosphere propagation to the destination. Why does the information-bearing signal need to modulate a high-frequency carrier signal for this transmission? This is necessary because the size of the antenna used to radiate the signal to free space depends on the wavelength X of the transmitted signal. The wavelength X is equal to *c/f,* where c is the speed of light and equals 3 x 10^8 m/s, and f is the frequency of the transmitted signal. For cellular communication systems, antennas are typically X/4 in size [1]. If a baseband signal with a frequency of 15 kHz were to be transmitted through an antenna without modulating a carrier signal, the size of the antenna would be X/4 = 5,000 m. However, it is only 8 cm if a carrier signal with a frequency of 900 MHz is modulated by such a baseband signal. For this reason, a high-frequency signal usually called a carrier signal is needed for all wireless communication systems to carry the modulation baseband signal. Thus, modulation is a necessary process in all wireless communication systems. Through this book, we mainly consider either the phase modulation scheme or a combination of an amplitude and phase modulation scheme such as M-QAM for its simple implementation and robust performance. In the phase modulation scheme, the information-bearing baseband signal is used to change the phase of a sinusoidal carrier signal whenever the polarity of the baseband signals changes. The phase change of the carrier signal, indicating either 1 or 0, is carried by the carrier signal.

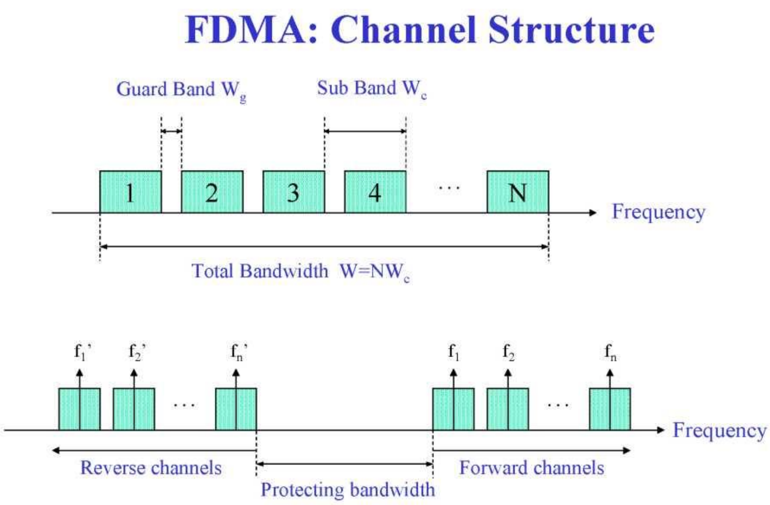
After the carrier signal is modulated by the baseband signal, the RF-modulated signal needs to access the desired frequency channel for the transmission so that the receiver can reliably detect the received signal and extract the original information bits from it. Several different techniques allow access to the RF channel:

- Frequency division multiple access (FDMA)
- Time division multiple access (TDMA)
- Code division multiple access (CDMA)
- Time and frequency division multiple access (TDMA/FDMA)

FDMA is the earliest multiple-access technique mentioned above and is widely used in both satellite/earth station communications and first-generation cellular systems. In this technique a user is assigned a pair of frequencies for sending or receiving a call. One frequency is used for downlink (base station to mobile in cellular systems or satellite to earth station in satellite systems) and one for uplink (mobile to base station or earth station to satellite). This process is called "frequency division multiplexing." Even though the user may not be talking, a pair of frequencies cannot be reassigned as long as a call is in place. Second-generation cellular systems, such as the global system for mobile communications (GSM), use an FDMA/TDMA technique for voice and data transmissions. The available spectrum band is divided into frequency slots, each with a 200-kHz sub-band. In addition, each frequency slot is also divided into time slots. Each user is assigned a pair of frequencies for uplink and downlink and a time slot during a frame. Therefore, the FDMA technique still plays an important role in modern digital communication systems.

In the early stages (during the late 1980s) satellite digital communication systems adopted an SCPC/FDMA (single channel per carrier) technique, where a total of 800 data/voice channels occupied a 36-MHz transponder bandwidth of the satellite. In this system each user can transmit and receive either data at a rate of 64 kbps with a Quadrature Phase Shift Keying (QPSK) format or voice at a rate of 32 kbps with a Binary Phase Shift Keying (BPSK) format. The channel spacing is 36 MHz/800 = 45 kHz. In the 2G GSM systems with a Gaussian Filtered Minimum Shift Keying (GMSK) modulation format, every symbol represents one bit, which means that symbol rate and bit rate are equal. In the 2.5G GSM Evolution Enhanced Data for Global Evolution (EDGE) systems with an 8PSK modulation format, every symbol represents three consecutive bits, which indicates that bit rate is three times the symbol rate. With the same symbol rate of 270.833 kbps for GMSK modulation, a bit rate up to 812.5 kbps for the EDGE system with an 8PSK modulation can be achieved within the same transmission bandwidth (200 kHz) as GSM systems.





FDMA systems have some common features: the relative low data transmission rate, narrow channel spacing, and restrictive transmission bandwidth. These features dictate that spectrally efficient modulation schemes are the best approach to achieving bandwidth-efficient transmission without significantly causing interference in adjacent channels. Meanwhile, the modulated signals in these applications prefer to have constant envelope or small envelope fluctuation in order to achieve energy efficiency when the power amplifiers operate in the saturation region or close to it. The energy-efficient operation of the power amplifier can extend given bandwidth in a specific communication system to achieve reliable performance. Spectral efficiency is viewed as bits per second per hertz (bits/s/Hz) and defined as

$$\eta_S = \frac{R}{B_W} \quad (7)$$

where R is the information rate or transmission rate in bit/s, and B_w is the *passband* (or double-side) transmission bandwidth in Hz. For a certain information rate, the spectral efficiency can be maximized by minimizing the transmission bandwidth. The minimum transmission bandwidth for intersymbol interference (ISI) free is determined by the Nyquist bandwidth (B_N) criterion, which states that the theoretical minimum bandwidth B_N of an ideal and linear phase brick-wall channel lowpass filter used for impulse transmission at a transmission rate of f_s symbols per second without ISI is equal to the Nyquist frequency f_N, or $B_N = f_N = f_s/2$. For a passband transmission system, the minimum *passband* bandwidth of B_w for ISI free is twice the Nyquist bandwidth, or $B_w = 2B_N = 2f_N$. If the transmission bandwidth of B_w is less than twice the Nyquist bandwidth B_N, the responses to these impulses at the output of the channel lowpass filter have ISI at the optimal sampling instants.

If rectangular pulses rather than impulses are used in the transmission channel, an inverse SINC function, or x/sin(x), shaped amplitude equalizer should be added before the ideal brick-wall channel filter so that the rectangular pulses can be transferred to the impulses before the ideal brick-wall channel filter.

Spectrum efficiency or bandwidth efficiency in (2.1) describes how efficiently the allocated bandwidth is utilized to accommodate the higher data transmission rate. For a given information rate *R*, the symbol rate f_s is associated with the information rate *R* or bit rate f_b through a modulation format such as M-order QAM. Therefore, the relationship between f_b and f_s is determined by the modulation format.

For a signaling alphabet with *M* alternative symbols, each symbol contains $N = \log_2 M$ bits, or N-bit represents the number of $M = 2^N$ symbols. The relationship between the bit rate f_b and symbol rate f_s is expressed as

$$f_b = N f_x \qquad (8)$$

Or

$$f_x = = \frac{f_b}{log_2 M} \qquad (9)$$

In the case of M-QAM, *M* represents *M* alternative symbols in (3). The theoretical spectral efficiency for the pass band BPSK transmission, where the bit rate is equal to the symbol rate $f_b = f_s$), is given by

$$\eta_s = \frac{f_b}{B_w} = \frac{f_s}{2 * f_N} = \frac{f_s}{2(f_s/2)} = 1 bit/s/Hz \qquad (10)$$

Table 2.1 illustrates the theoretical bandwidth efficiency limits for some main modulation formats. It should be noted that these figures cannot actually be achieved in practical radios since the ideal brick-wall channel filter is required, which is impossible to practically design.

In practice, a raised cosine filter with a roll-off factor of *a* is widely used to approximate the ideal brick-wall filter. The double-side bandwidth of the raised cosine filter is given by

$$B_W=2(1+\alpha)f_N\ , \quad 0 \leq \alpha \leq 1 \qquad (11)$$

Energy Efficiency

In highly integrated wireless systems, such as wireless system on chip (SoC) devices, the RF power amplifier is the subsystem that consumes most DC power in the whole transmission system. For example, it may consume up to more than 70% of DC power in the transmit path of certain RF mobile transceivers. For this reason, the “energy efficiency” of the RF power amplifier, which is often described as “power efficiency” in the literature (and this is incorrect), is directly proportional to and mainly represents the efficiency of the overall system, especially in the transmission path. In addition to power consumption, it is now apparent that energy consumption is an important metric for transmitter circuits. Energy consumption more accurately predicts the battery life, especially when a portable system operates with a wide range of the output power. Hence, energy efficiency tends to be a better metric of the performance than power efficiency in terms of the battery life. Actually, energy consumption depends on the power consumption and time spent in the power consumption duration.

Usually, in the published literature, the power efficiency of the power amplifier refers to how efficiently the input power to the power amplifier, including the input AC and DC powers, is converted to the output AC power to a load or an antenna, regardless of time. Energy efficiency of the power amplifier, however, is identical to its power efficiency as long as all powers, including the input AC power to the PA, power supply DC power, dissipated power as heat, and the output AC power from the PA are measured equivalently in time . Therefore, we use the term *energy efficiency* instead of *power efficiency* in this book even though definitions are the same [3].

All power amplifiers can be represented by a four-port network: an input DC port, a RF input signal port, a RF output signal port, and a ground. The measured input DC power PDC includes the power associated with all the bias lines of the power amplifier. It is assumed here that the PA is perfectly matched and has infinite reverse isolation. Therefore, the measured power PIN at the input RF port corresponds to the input signal power at the fundamental frequency only. The measured power POUT at the output RF port corresponds to the output power at the fundamental frequency and all the spurious frequencies, which are generated by the PA itself.

A power amplifier is evaluated for efficiency using a conservation of power calculation based on the flows identified in

$$P_{IN} + P_{DC} = P_{OUT} + P_{D} \qquad (12)$$

The energy efficiency of the PA is a measure of its ability to convert the DC power of the power supply into the RF signal power delivered to the load. There are two definitions of power amplifier efficiency. One is basic PA efficiency and the other is power-added efficiency.

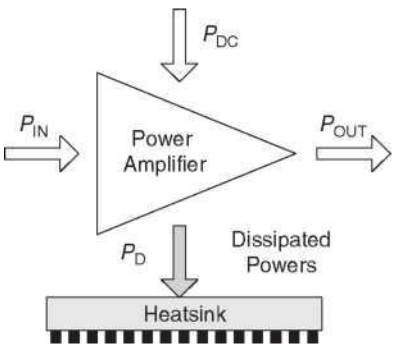


Fig. 2.1 Power flows in a power amplifier. Redrawn from

Basic PA Efficiency: The basic PA efficiency is a ratio of the RF output power to the DC power and is derived from

More precisely, PA efficiency is also called the overall efficiency . If the power amplifier has relatively high power gain, the direct contribution of the RF input signal power to the RF output signal power is insignificant, and therefore it can be neglected. The basic efficiency can be approximated by

$$\eta_{B=}1\text{-}P_D / P_{DC} \qquad (13)$$

Power-Added Efficiency: When the gain of the power amplifier is not significantly high, the RF input power needs to be subtracted from the RF output power in the efficiency expression, and then the power efficiency is referred to as the power- added efficiency (PAE):

$$\eta_{pae}=(Pout\text{-}Pin)/ P_{DC} \qquad (14)$$

If the PA has a relatively large power gain, then, $n_B \sim n_{PAE}$, as expressed in (13) and (14). The power-added efficiency can be interpreted as the efficiency of the network to convert the input DC power into the amount of the RF net output power. PAE definition in (2.13) is widely used as a useful metric for evaluating the efficiency of the RF power amplifier. PAE becomes zero when the power gain of G_{PA} is unit or $P_{OUT} = G_{PA} \times P_{IN} = P_{IN}$. This means the power amplifier does not convert any DC power to the RF output power.

It can be seen from (2.11) and (2.13) that the PA efficiency increases as DC

Table 2.2 Classical modes of power amplifier operation

Classical mode	Conduction angle (°)	Operation range
A	360	Linear
AB	180-360	Either linear or nonlinear[a]
B	180	Nonlinear
C	0-180	Nonlinear

[a]Class-AB is not a complete linear amplifier; a RF signal with non-constant envelope will be distorted significantly at its peak power level

power of P_{DC} is reduced. The DC power P_{DC} is given by

$$P_{DC}=V_{DC}I_{DC} \qquad (15)$$

The DC current I_{DC} can decrease monotonically as the conduction angle of the power amplifier is reduced [5]. The conduction angle of the power amplifier determines the classical modes of the power amplifier operation as listed in Table 2.2. In 3G and 4G cellular and 802.11 WLAN communication systems, power amplifiers usually operate in a class AB mode to achieve high efficiency by reducing the DC current I_{DC} and to avoid severe nonlinear distortion on the RF output signal as well. In general, DC supply voltage of V_{DC} is fixed during the power amplifier operation except the envelope tracking technique based power amplifier [6, 7], where the supply voltage dynamically and synchronously tracks or follows the envelope of the RF input signal. Furthermore, even in the class AB mode, the efficiency can be further increased due to the reduction of DC current $\mathrm{I_{DC}}$ as the output back-off of the power amplifier from its P1dB compression point is reduced. Therefore, in terms of achieving high efficiency, power amplifiers are preferred to operating in the small back-off condition as long as the transmitted power spectral density (PSD) and error vector magnitude (EVM) meet the standard specifications with enough margins.

The expressions in (14) and (2.15) also demonstrate that the consideration of power amplifier efficiency is the same as the consideration of either amplifier output power or amplifier power dissipation. The amount of the total input power to the power amplifier in Fig. 2.1, either DC power or RF input power that is not converted into the RF output power, is dissipated as heat. Higher power output corresponds to higher energy efficiency. From (2.10), lower power dissipation leads to higher power output, which in turn results in higher power amplifier efficiency. From a design point of view, the design objective of maximum energy efficiency is identical to the design objective of either minimum power dissipation P_D or DC power P_{DC}. Therefore, the energy efficiency of the power amplifier has become a challenging requirement for most PA designers. Cell phone handset PAs have to operate efficiently to conserve battery power and base station PAs are also need to be efficient as possible due to cooling limitations.

The energy consumption of a system is highly related to the power consumption through a time interval *T*. There are many different definitions of the energy consumption in the literature. Considering that the energy consumption covered in this book only focus on the transmitter, especially for the PA, we define the energy consumption of the PA at the transmitter as shown in Fig. 2.1 during the time interval *T* as

$$E = T(\mathrm{Pdc} + \mathrm{Pd})\ [\mathrm{Joule}] \qquad (16)$$

where P_{DC} is the DC consumption power of the PA and P_D represents the static power dissipated in the PA as heat. Substituting $P_{DC} = P_{OUT}/n_B$ in into (16), the energy consumption in (17) can be rewritten as:

$$E=T(\mu P_{out} + P_D)\ [\mathrm{Joule}] \qquad (17)$$

where $\mu = 1/\eta_B$ is a factor, with η_B the efficiency of the transmit power amplifier. In the case that the PA has a relatively large power gain, we have shown the relationship between η_B and η_{PAE}, or $\eta = \eta_B \sim \eta_{PAE}$-

If P_D includes the power dissipated in all other circuit blocks of the transmitter and receiver, the energy consumption expression in (16) represents the energy consumption of a system, which is the same as one used in [9, 10]. From the PA standpoint of view, expressions of (17) and (17) precisely represent the energy consumption of the PA.

As the energy consumption of the PA has been defined, the energy efficiency of the PA needs to be defined next. The energy efficiency of the PA is the ratio of the benefit obtained after sustaining the energy cost and the amount of energy consumption in (16) and (17). The benefit is usually related to the amount of data reliably transmitted in the time interval *T*. Several performance functions have been used in the literature to evaluate this quantity, such as *system capacity*, and *system throughput* in [9, 10]. Throughput is the amount of data reliably transmitted over a communication channel in the certain time period T. The throughput metric is measured in bits/s and also depends on the signal-to-noise ratio (SNR) and the transmission channel condition. compared to capacity, throughput is more practically used to evaluate the system performance.

Energy Efficiency **:** With a general function ƚ/(SNR) that represents throughput as the system benefit, the energy efficiency of the PA is defined as

$$\mathrm{EE}=\frac{T*\dagger(\mathrm{SNR})}{T*(\mu\mathrm{Pout}+\mathrm{PD})}=\frac{\dagger(SNR)}{\mu\mathrm{Pout}+\mathrm{PD}} \qquad (17)$$

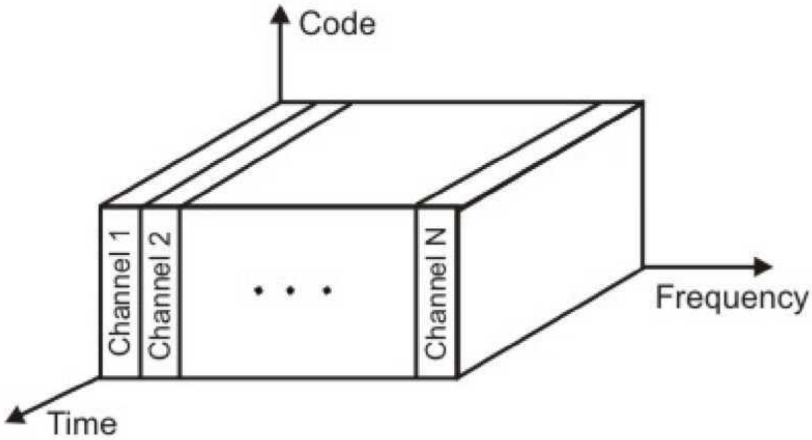
EE can be improved by either increasing the numerator or decreasing the denominator in (2.17). In this book, however, we focus on increasing the energy efficiency by decreasing the denominator through the increase of the efficiency n of the PA. It is clear that EE can be improved by decreasing the factor« = 1/n through the increase of the efficiency n of the PA, which in turn requests either to increase the output power P_{OUT} of the PA or to decrease the DC power P_{dc}. Even though EE can be improved by increasing the throughput in (2.17), the increase of the throughput takes the entire system involved, including a transmitter and receiver, multi-antennas, a channel condition and so on. Therefore, energy efficiency improvements through increasing the throughput are very complicated and are beyond the scope of this book.

Medium Access Control Techniques

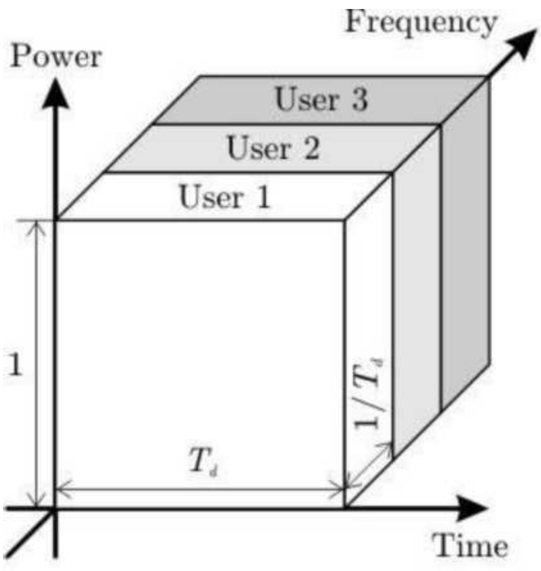
Channelization is a multiple access method in which the available bandwidth of a link is shared in time, frequency or using code by a number of stations. Basic idea of these approaches can be explained in simple terms using the *cocktail party theory.* In a cocktail party people talk to each other using one of the following modes:

FDMA: When all the people group in widely separated areas and talk within each group. *TDMA:* When all the people are in the middle of the room, but they take turn in speaking. *CDMA:* When all the people are in the middle of the

room, but different pairs speak in different languages.
Basic principle of these approaches are briefly explained below:
FDMA: The bandwidth is divided into separate frequency bands. In case of busty traffic, the efficiency can be improved in FDMA by using a dynamic sharing technique to access a particular frequency band; channels are assigned on demand as shown in Fig 5



FDMA divides available bandwidth into a number of orthogonal channels of smaller bandwidths.
A channel is used continuously over the duration of the message. FDMA is limited to narrowband applications due to its limited transmission rate. If the same channel is reused at another physically separate location, an increase in transmit power will negatively affect the carrier-to-interference ratio at that location.



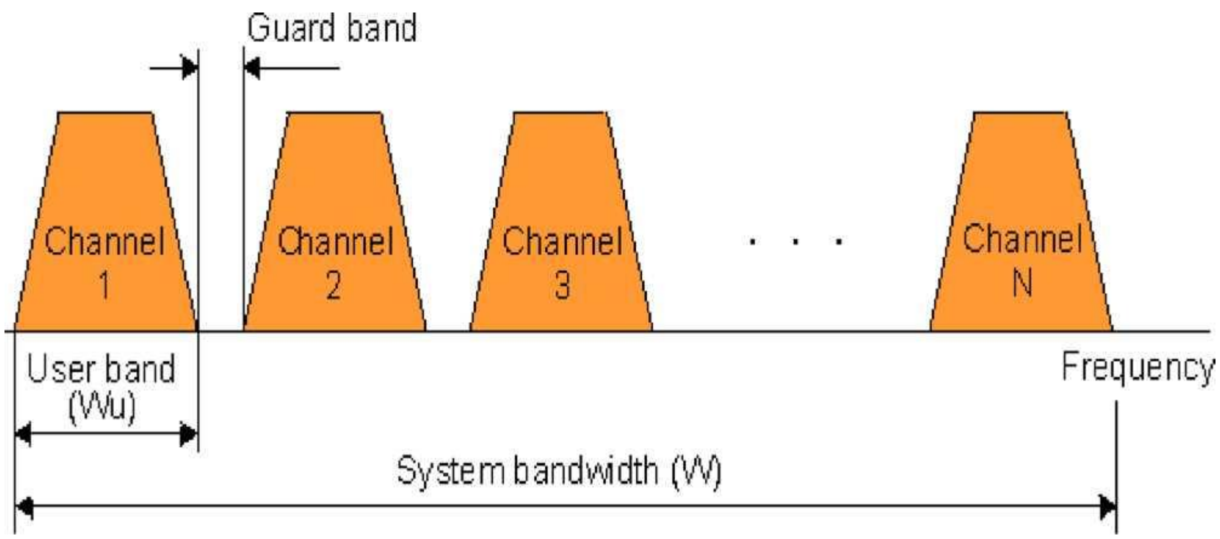
FDMA

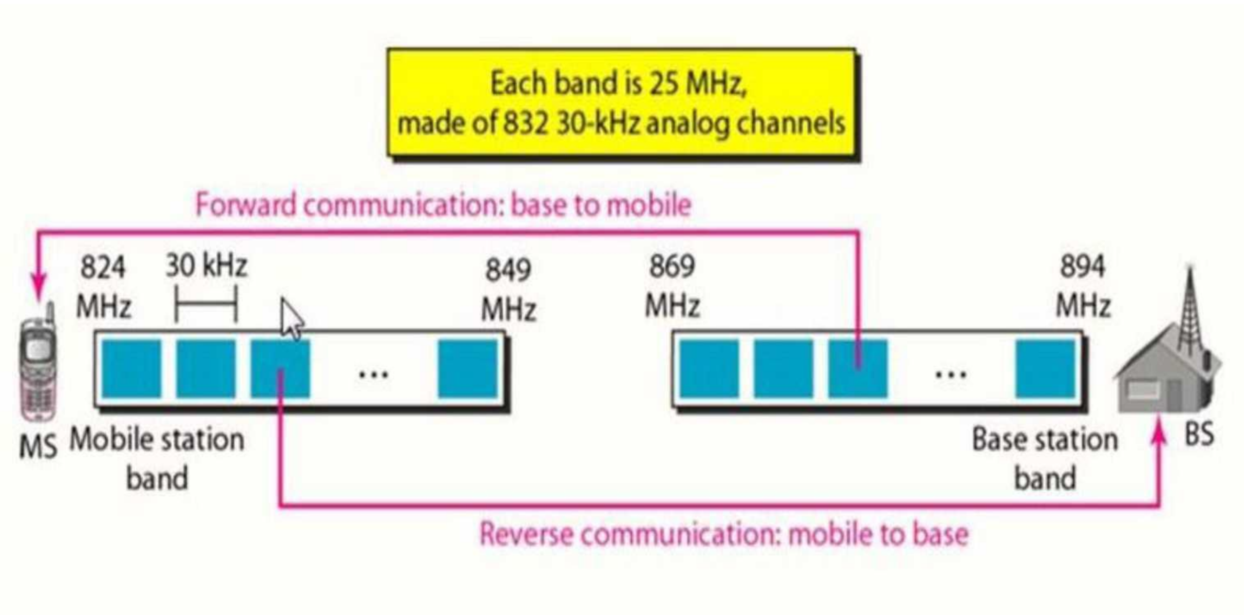
FDMA is employed in first generation cellular technology advanced Mobile Phone Systems (AMPS). A total bandwidth of 50MHz is divided equally into two:

25 MHz for forward link;

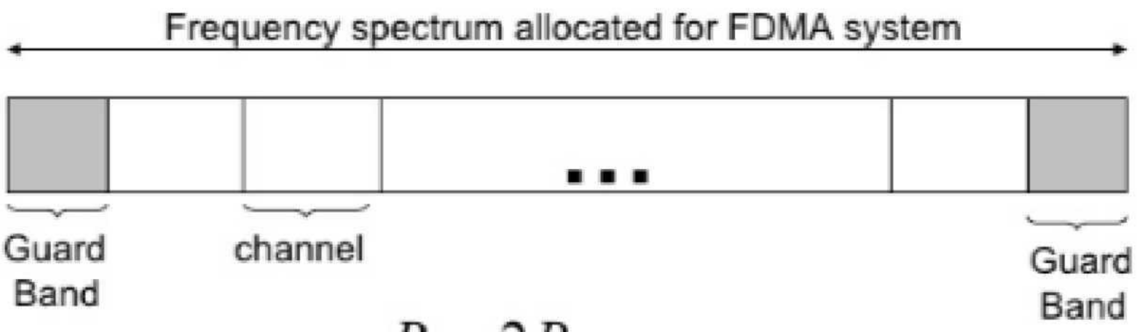
25MHz for reverse link.

12,5 MHz each is allocated to two competing network operators





In AMPS, a channel bandwidth of 30kHz and a total of 832 channels are available. A guard band of 10kHz is allowed at the edge to reduce inter-system interference.



$$N = \frac{B_t - 2B_{guard}}{B_c}$$

B_t : Total spectrum allocation

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

B_c : Bandwidth of a channel

Advantages of FDMA

Low Inter-symbol Interference (ISI).It has significant channel delay spread relative to the symbol period because the transmission bandwidth is wider than the channel coherence bandwidth.

Lower Overhead. FDMA system uses channels on a continuous basis thereby periodic timing and synchronization controls are not needed. Only fewer bits are required for signaling and control.

Simple Hardware. It does not require adaptive equalization and slot timing adjustment control.

Drawbacks of FDMA

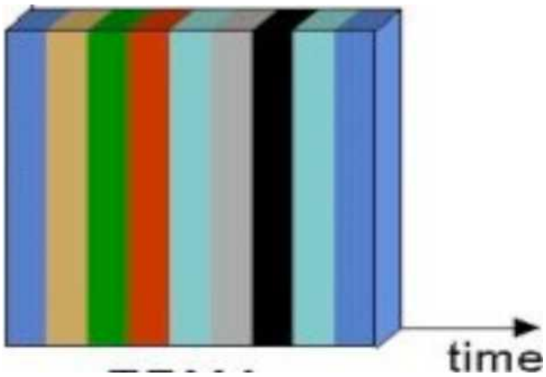
Higher base station cost. The requirement of individual T_x and R_x for each channel augments the BS cost.

Requires a duplexer in the mobile unit. Duplexer is essential for simultaneous reception and transmission in FDMA mobile unit. The cost of a duplexer is 10% of the total cost of mobile unit. Perceptible degradation of link quality during handoffs

TDMA

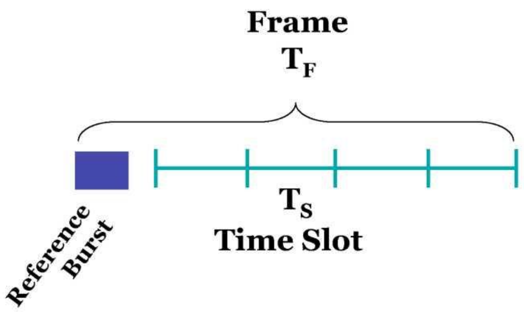
Each user is allowed to transmit only within specified time intervals (Time Slots). Different users transmit in different Time Slots.

When users transmit, they occupy the whole frequency bandwidth (separation among users is performed in the time domain).



TDMA requires a centralized control node, whose primary function is to transmit a periodic reference burst that defines a frame and forces a measure of synchronization of all the users.

The frame so-defined is divided into time slots, and each user is assigned a Time Slot in which to transmit its information.



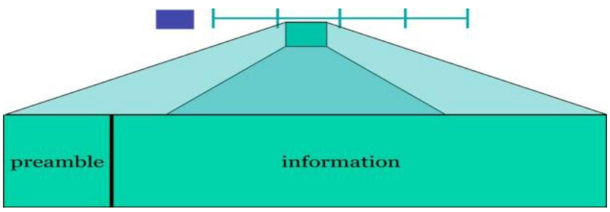
Since there are significant delays between users, each user receives the reference burst with a different phase, and its traffic burst is transmitted with a correspondingly different phase within the time slot.

There is therefore a need for guard times to take account of this uncertainty.

Each Time Slot is therefore longer than the period needed for the actual traffic burst, thereby avoiding the overlap of traffic burst even in the presence of these propagation delays.

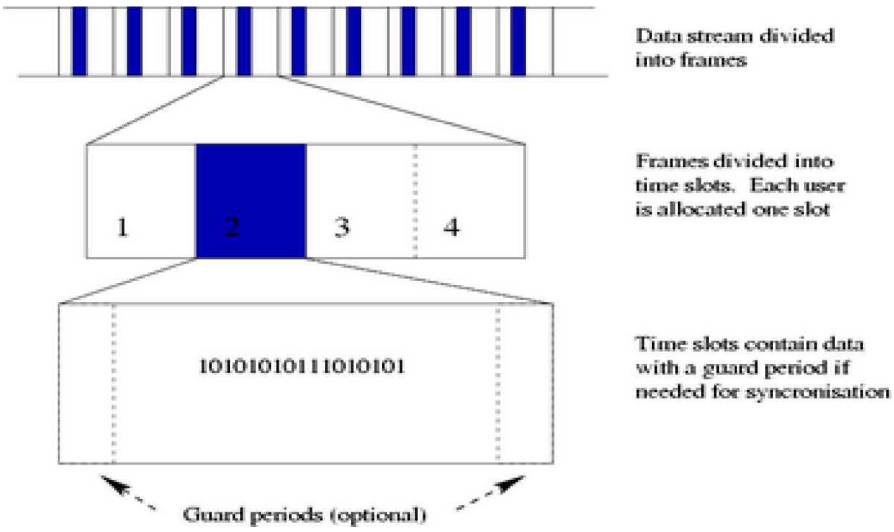
Since each traffic burst is transmitted independently with an uncertain phase relive to the reference burst, there is the need for a preamble at the beginning of each traffic burst.

The preamble allows the receiver to acquire on top of the coarse synchronization provided by the reference burst a fine estimate of timing and carrier phase.



Here the channel time is divided into slots which are arranged into frames. Each active user is allocated a unique slot within a frame, in order to support several channels per carrier. The entire channel bandwidth is used during each slot.

Thus, in contrast to FDMA, the train transmission in TDMA is dis continuous. Users transmit in bursts which are confined to slots specifically assigned to them. A set of time slots are assembled into a frame.



Each slot has:

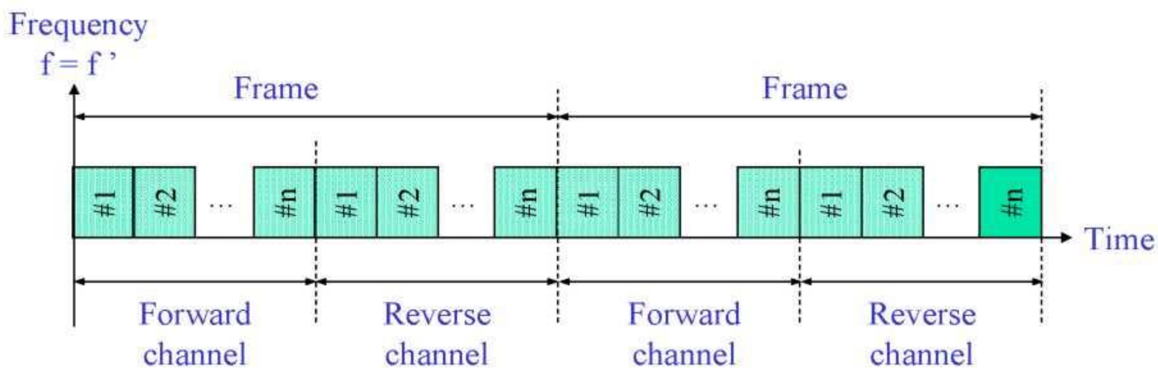
1. Preamble or frame sync
2. Word sync
3. Control and supervisory bits
4. User data

The number of slots per frame depends on:

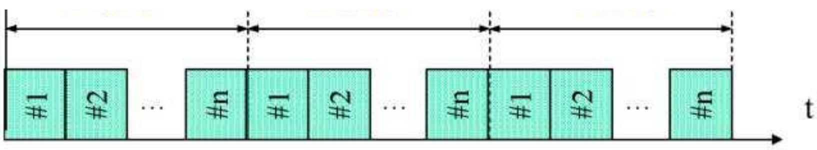
1. RF bandwidth
2. Modulation format
3. Transmission rate

In TDMA, the message to be transmitted is split into time bursts of length equal to the time slot. At the receiver, these bursts are collected to assemble the message. The use of non-overlapping frequencies in FDMA and time slots in TDMA effectively create channels that are orthogonal in one of the dimension of the time-frequency space.

TDMA: Frame Structure (Cont'd)



Channels in Simplex Mode



(b). Reverse channel

Advantages of TDMA

TDMA base stations are relatively smaller in physical dimensions and cheaper. Duplexers are not needed in TDMA since the time between the assigned slots is sufficient for switch over from one frequency to another. TDMA handles inter-channel interference with ease.

Drawbacks of TDMA

The discontinuous mode of transmission and reception in TDMA requires sizable number of overhead bits. Inclusion of guard time between the slots reduces the usable channel time as synchronization, control bit overhead and slot guard time could use up to 30% of the channel time.

Complexity related to synchronization and dynamic slot alignment sub-system. Equalization to mitigate inter-symbol interference (ISI) resulting from channel delay spread

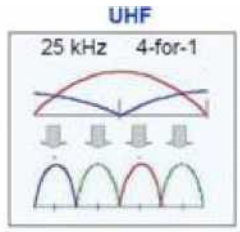
4. Devices

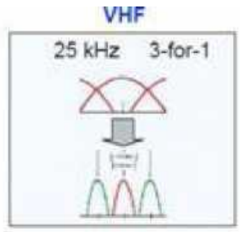
4.1 6.25kHz FDMA Information Center

The LMCC held a meeting recently to address issues about licensing our 6.25

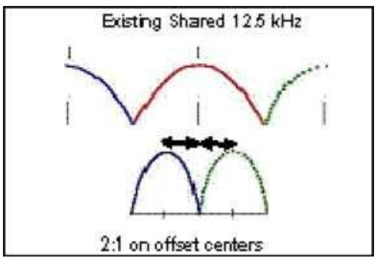
kHz FDMA technology:

Exclusive Channels - Exclusive channels are about 20% of licenses. No one else is on them (uses FB8 classification). The LMCC decided they can do anything they want (“exclusive”) including 2:1, 4:1 splits.



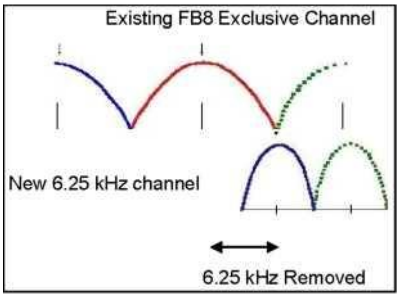


Shared Channels - They agreed to split 12.5 kHz channels 2:1 (previously they would only split “exclusive channels”). This is new because previously they would only coordinate 6.25 kHz licenses on the center frequency of 12.5 kHz channels (no advantage). Now they will license on the 6.25 kHz “offset” frequencies within the 12.5 kHz channel. They agreed that no waiver was necessary under the current rules.



Adjacent to exclusive channels - They protect the rights of exclusive channels (3,300) by preventing adjacent channel interference with a “buffer” between the new license and the exclusive license. New 6.25 kHz channels are “dropped in” between existing 12.5 kHz exclusive channels. Now they will allow as close as

6.25 kHz from the center of the 12.5 kHz channel.



This was a very good meeting as the LMCC is now moving aggressively to license 6.25 kHz channels in these 3 ways.

As existing spectrum becomes increasingly scarce, the demand for more options grows. The FCC essentially created additional spectrum by adding hundreds of new licenses with 6.25kHz bandwidth. To take advantage of this opportunity, Icom and Kenwood entered into a joint agreement to develop 6.25kHz technology. This technology is a new digital communications protocol that provides quality voice and data, and is designed as a non-proprietary protocol. It accomplishes this by using 4LFSK (4- Level Frequency Shift Keying) and FDMA (Frequency Division Multiple Access).

History

This technology was developed in response to an FCC revision of the rules concerning transmitters in the 150MHz to 174MHz and 421MHz to 512MHz range. To receive FCC certification after January 1st, 2005, transmitters must have proved compliant as a multi-mode device. This requirement could be achieved by using 6.25kHz channel bandwidth. In addition to the FCC requirement, Europe and Japan are also moving toward 6.25kHz technologies. Because some in the industry believed that this requirement could not be met by 2005, the FCC suspended this requirement.

Even with the FCC deferring narrowband conversion over a ten year period, Icom set out to meet the requirement without delay. This was impossible using analog technology, so it became necessary to develop a new digital protocol. Other methods were also considered, including ACSB and the proposed APCO Project 25 (P25) Phase II CQPSK. However, both required a more expensive linear amplifier in the transmitter and neither is compatible with existing analog FM hardware.

Instead, 4LFSK modulation was selected using FDMA for transmission. This method has a number of advantages:

- better communication range
- simpler design
- easy to maintain and service
- lower cost for business and industry customers
- compatible with existing FM radio hardware

Icom's first radio with this technology is the F3061/F4061. To enable backwards compatibility, the radio is both analog and digital and also works in 25kHz and 12.5kHz channel bandwidths (conventional and LTR® trunked operation in analog mode).

Backwards compatibility to analog only radios enables a planned migration path to "digital" with existing radios operating analog only and new radios operating analog and digital.

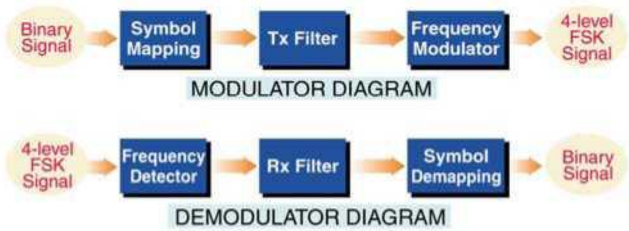
General specifications:

- Access method: FDMA
- Transmission rate: 4800 bps
- Modulation: 4-level FSK
- Vocoder: AMBE+2
- Codec rate: 3600 (voice 2,450 + error correction 1,150 bps)

Modulation with 4LFSK uses a symbol mapping scheme. When the radio receives a binary number, that number is mapped to a symbol, which is interpreted as a 1050Hz frequency deviation.

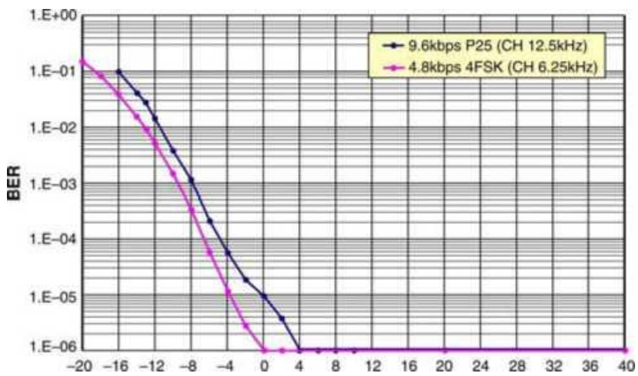
Information	Symbol	Deviation
01	+ 3	+ 1050Hz
00	+ 1	+ 350Hz
10	-1	- 350Hz
11	-3	-1050Hz

During demodulation, that deviation is detected, filtered and "unmapped" as a binary signal for transmission.



Signal Quality

The FDMA signal BER performance exceeds that of P25 phase 1 radios, which have already been accept by the LMR market.



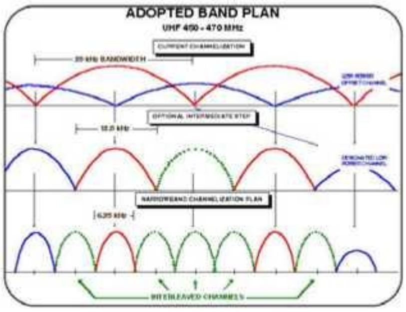
Audio Quality

The 6.25kHz technology also offers superior audio quality compared to P25 audio. Test engineers using a Mean Opinion Sample (MOS) found the audio quality was uniformly better, ranging from clean conditions to 5% BER. Using the AMBE+2 Vocoder in the Icom IC-F3061 makes this possible.

Spectrum Considerations (VHF & UHF)

While most users are operating on 25kHz channels, they will have to migrate to 12.5kHz bandwidth by 2013. Narrowband migration is currently impeded by the lack of incentives for end users to invest in the technology.

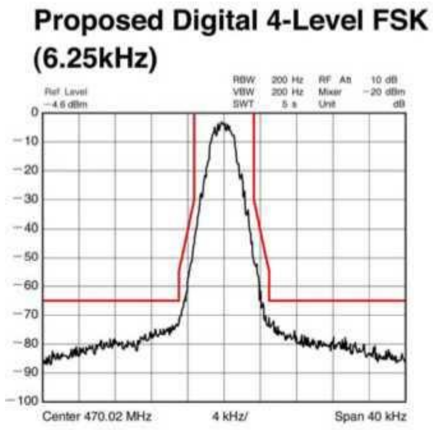
A channel is defined by the deviation either side of the center line frequency. Migrating from a 25kHz channel to a 12.5kHz channel on the same centerline frequency is a 1-for-1 move. There is no increase in the capacity to load radio users.



There are 500 new 6.25kHz frequencies (VHF and UHF) available now. Most are unused because no 6.25kHz radios were available. With Icom's FDMA technology, frequency coordinators have total flexibility to either assign a 6.25kHz channel within an existing 25kHz or 12.5kHz channel or as a stand-alone frequency some where else on the band.

Frequency coordinator will coordinate channels for minimum adjacent channel interference. Because the emission mask is tight, 6.25 kHz channels can be used next to each other without causing interference.

The emission mask above is established by the FCC for 12.5kHz channels. The signal must operate within the mask.



This is the emission mask for a 6.25kHz channel. The Icom FDMA signal clearly operates within the mask. Accordingly, the FCC certified the F3061/F4061 as the first ever 6.25kHz radio.

4.2 CALCULATION OF THE NUMBER OF CHANNELS ON THE SITES PROJECTED NETWORK

The choice of the type of cable line, transmission system and cable brand is determined primarily by the number of channels of the tonal frequency (PM) of the primary network.

The required number of PM channels on each i-th section of the network is defined as the sum of the channels required for the automatically switched ViAC network and for manual maintenance of ViPOs, ViPD data networks, ViTG telegraph communication, and ViOT operational and technological communications.

In the course work, the required number of channels on individual sections of the network for each type of communication is determined from the individual task.

Knowing the number of channels for each type of communication, the total number of channels on the i-th section of the network

$$V_i = 2V_{iАК} + 2V_{iРО} + V_{iПД} + 2V_{iТГ} + V_{iОТ}$$

The type of cable line and the transmission system are chosen according to the number of Vi channels in the main line

CHOICE OF TYPE OF CABLE COMMUNICATION LINE

Depending on the required number of PM channels in the backbone, one of the following types of cable lines can be selected in the course work.

A single-cord line is a line in which all kinds of channels of PM, operational-technological communication circuits and railway automation are organized in one symmetrical cable

A two-cable communication line is a line in which two symmetrical cables are laid and a one-way four-wire K-60p type equipment is used to form PM channels. This equipment for forward and reverse transmission requires the same frequency band 12-252 kHz and two pairs of wires placed in two separate cables. When one, two and three quadruples are sealed in each of the two cables, the K-60p system provides the formation of 120, 240 and 360 channels of PM. Chains of operational-technological communication and railway automation are allocated in the same cables.

A three-cable line is a line in which two symmetric cables of minimum capacity (4 x 4) are laid to form the channels of PM, the circuits of which are condensed. In the case of the K-60p system and the sealing of all the cables, 480 channels of PM are formed. For chains operational-technological communication and railway automation is the third cable.

CHOICE OF CABLES AND TYPE OF TRANSMISSION TYPE

The choice of the type and capacity of the cable depends on the type of cable line, the type of traction of the trains, the linear spectrum of the equipment of transmission systems and the required number of channels of PM. The list of cables most suitable for rail transport is given in Table. 4.1

Cable and system brand cable line	Sealing device	Number of channels of TF and LF circuits and		
		TF	LF	Auto
Single-cord line МКПАБ 7x4x1,05+5x2x0,7+1x0,7 МКБАБ 14x4x1,2+5x0,9	К-12+12 К-12+12	24 48	12 19	5 5
Two-cord line МКСАБП 7x4x1,2 МКПАБ 7x4x1,05+5x2x0,7+1x0,7	К-60п К-60п	24 0 36 0	20 20	– 10
Three-cord line (two main cable К2 и К3) МКПАБ 4x4x1,05+1x2x0,7+1x0,7 МКПАБ 7x4x1,05+5x2x0,7+1x0,7 (third cable К1) ТЗПАПБП 19x4x1,2 МКБАБ 14x4x1,2+5x0,9	К-60п К-60п К-12+12 К-12+12	48 0 48 0 24 48	1 12 22 12	– 10 14 12

The third cable K1 in the three-cable line is intended mainly for the formation of low-frequency circuits and small bundles of TF channels within the

In each of the transmission systems, the equipment of the terminal points (EPs), the serviced amplifying stations (SAS) and the non-serviced amplifying stations (NAS) are distinguished.

In one set of terminal equipment in the K-60p transmission system are the following main racks: individual converters SIP-60, group converters SGP-K-60p, linear amplifiers and correctors SLUK OP K-60p, generator equipment SUGO-1-5. The additional equipment includes: a call-tone stand and differential systems STV-DS-60, installed with manual method of connecting telephone sets of subscribers to channels; transit counter of primary groups and correctors СТПГ-К, placed in points of re-reception and transit of channels; remote power supply rack of NAS amplifiers; the rack of the input-cable equipment SVKO K-60p for input and cable cutting

The SAS equipment in the K-60p transmission system includes: the SVKO stand, linear amplifiers and proof-readers of SLUK SAS-2 or SLUK-3, SDP K-60p.

In the unattended amplifying point, the input-cable equipment VKSh-1 and the stand of the intermediate amplifiers of the non-service stations SPUN K-60p are installed

In a 12-channel single-cable two-way transmission system with a line spectrum of 12-120 kHz, the terminal equipment is placed on one rack. The SAS equipment is also located on one rack. The equipment of unattended amplifying points is made in the form of a metal frame with the amplifying equipment placed on it, and also in the form of a metal container with cassettes of equipment inside.

The K-12 + 12 transmission system allows one or two three-channel groups to be allocated in the frequency spectrum for the AB direction of 12-36 kHz, and for the AB direction of 96-120 kHz. On the section AB can be organized 6 straight and 6 dedicated channels with the help of equipment VPK-12 + 12

The K-60p transmission system allows one or two primary 12-channel groups of channels to be allocated using the SVGG-1 or SVPG-2 rack. The SVGS-1 stand allocates one 12-channel group in the 12-60 kHz frequency band, and the SVGS-2 stand has two 12-channel groups in the frequency band 12-108 kHz.

A brief description of the transmission systems of K-12 + 12 and K-60p is given in Table. 4.2.

For the organization of operational-technological types of telephone communication on two-cable lines, the transmission system K-24T is used.

Basic data of transmission systems over cable communication lines

Characterization	Values of characterization	
	К-60р	К-12+12
System of the connection	Single-lane two-cable	Two-way single-cable
Linear frequency spectrum , kHz	12-252	12,3-57,4 и72,6-119,7
Number of channels of TF generated on the circuit: two-wire four-wire	 - 60	 12 -
The bandwidth of effectively transmitted frequencies of the channels , kHz	0,3-3,4	0,3-3,4
The maximum length of the transitive section at tonal frequencies in the presence of HF transits , km	2500	840
Maximum communication range , km	12500	1680
Attenuation of the amplifying section at the maximum temperature of the ground , dB: nominal maximal	 51,3 54,8	 43,5 47,5
Maximum amplification of amplifiers, dB NAS SAS and EPs	 55,6 61,0	 34,7/47,7* 34,7/47,7*
Minimum amplification of amplifiers, dB : NAS SAS and EPs	 45,0 48,6	

Nominal transmission levels on the channels at the output of the linear amplifier, dB in the absence of pre-emphasis with a distortion along the upper channel	-4,8 -0,9	-4,3 -
The maximum number of NAS between SAS-SAS according to the DP schemes: "Wire - earth" "Wire - wire"	12 6	10 4

Numbers in the numerator for the lower frequency group, in the denominator for the upper frequency group.

It is single-side, four-wire, takes the spectrum of frequencies 12-108 kHz and provides 12 direct and 12 group channels. Direct channels use in the linear path a frequency range of 12-60 kHz and pass from one terminal station to the other without isolation at intermediate stations. Group channels operate in the frequency spectrum of 60-108 kHz and all 12 channels are allocated at each intermediate station, forming a group path. The transmission system includes: a terminal stand SB K-24T, an intermediate SP K-24T for allocation of group channels, intermediate amplifiers a maintenance-free SPUN. The K-24T system is consistent with the K-60p system and their amplifying regions coincide, so the SPUNs are located in the same NASs as the K-60p

PLACEMENT ON THE CABLE LINE CROSSING PMO AND NUP OF K-60p and K-12 + 12 TRANSMISSION SYSTEMS

The projected cable backbone must have re-entry points. Re-reception is the reception of communication signals in one terminal equipment and sending them to other terminal equipment for transmission along the chain further (transit). Re-reception can be carried out at a tonal frequency, by primary 12-channel and by secondary 60-channel groups. There are two sets of terminal equipment in the re-receiving station

Re-reception on the tonal frequency allows allocation of PM channels in the re-access point, as well as branching of the channels and their transfer to other transmission systems. Therefore, the re-receiving points on the railway lines are combined with the locations of the road and gubernatorial control stations, where separation and branching of the channels is necessary

The length of the transitive section at the tonal frequency and the number of transitions in the primary and secondary groups on the site in one or another transmission system are determined by the possibilities of equalizing the amplitude-frequency characteristic of the linear path of the transmission system, the value of the psophometric noise power generated by the equipment of the linear and converter paths, the accuracy of damping of the circuits cable on amplifying sections. The possibility of retransmission of transmission systems K-12 + 12 and K-60p is indicated in Table. 4.2

Transition sections of the cable main are divided into control sections. Under the control section is meant the section located between the NAS, in the apparatus of which the automatic gain control is performed under temperature changes of the attenuation of the cable circuits.

On the cable lines of the railway transport, the placement of the PMU is determined by the location of large and central railway stations, where appropriate equipment for allocating channels with power supplies is installed and their maintenance is required. Other important factors determining the boundaries of the control and positioning sections of the PMU are the adjustment capability of the AGC devices in the amplifying equipment of the PMU and the remote power supply conditions of the NPM.

In the K-60p transmission system, the amplifying equipment installed in the PMU can have a flat-slanted AGC controlled by currents of two reference frequencies or a plane-oblique-curved AGC controlled by currents of three control frequencies. NAS with two-frequency AGC (NAS-2) is placed at distances up to 200-250 km, and PMO with three-frequency AGC (NAS-3) - at distances up to 500-600 km

In the two-way single-cable transmission system K-12 + 12 with the transfer of the lower frequency group 12.3 - 57.4 kHz in the direction AB and the upper frequency group 72.6 - 119.7 kHz in the direction of BA for automatic signal level adjustment in the OP and GPR, a flat inclined AGC is used, controlled by a single reference frequency, transmitted in the AB direction of 60 kHz and in the direction of BA at 72 kHz. The maximum length of the section is 235 km.

If one starts from the conditions of remote power supply of the NUP, the distance between the PMO depends on the power scheme. Thus, in the K-60p system with remote power supply in the "wire-wire" scheme, there are six remotely supplied NUPs between the two PMUs, and according to the "wire-ground" power scheme -12 NWT, which corresponds to a distance of 130-260 km.

In the K-12 + 12 system, in the power-to-ground power scheme, eight to ten remotely powered NPCs can be located in the control section of the PMO-PMU, and for a wire-to-wire scheme, no more than four.

In the wire-ground remote power supply scheme, an artificial circuit is used as the direct wire formed on a pair of cable strands, and the reverse is the earth.

After determining the location of the NAS and the boundaries of the control sections, the location of the SAS on each section should be selected. It is recommended that you try to ensure that the NUs break the section into sections of the same length as possible. Basic data necessary for calculating the transmission quality are given in Table. 3.2.

The ground AGC of the NAS amplifiers is designed to compensate for the temperature variation of the attenuation of the rated cable. At the same time, it most accurately corrects the frequency distortion of a section of nominal length.

5. Part of labor protection

Calculation and analysis of electric injury in automated system

The results of a continuous analysis of statistical data on electrical injury are the base of planning a set of preventive measures aimed at reducing the number of electrical injuries in electrical safety field.

In the sphere of CIS, "Guidelines and recommendations on the study and analysis of electrical injury" are used. It describes the order of electric injuries, the procedure of accounting and analysis

In addition to the documents that need to be documented for each accident, "In the production process, in accordance with the rules for inspection and accounting of accidents, three copies of the Card are charged for electric injury. One of them will stay in an organization that has been traumatized. The second one sent to the Ministry, the third is transferred to the labor protection service.

The experience with electric injury cards indicates the lack of this method, but only 20-30% of the cards are made for electric injury, which supplements the extra time to complete. In such cases, statistical data obtained as a result of card analysis can not be convincing.

At the same time, electric injury can be seen as a result of the accidental adverse effects in space and time. It is necessary to determine one factor by several factors, as a result of analysis it is necessary to define the level of each index and each factor, as well as the correlation between them.

In order to test the "Automated Accident Detection and Analysis Technique", developed by the ITI Department, "Data on electric trauma in the state energy audit has been analyzed”.

Part of the safety guard

Automated system of accounting and analysis of electric injury

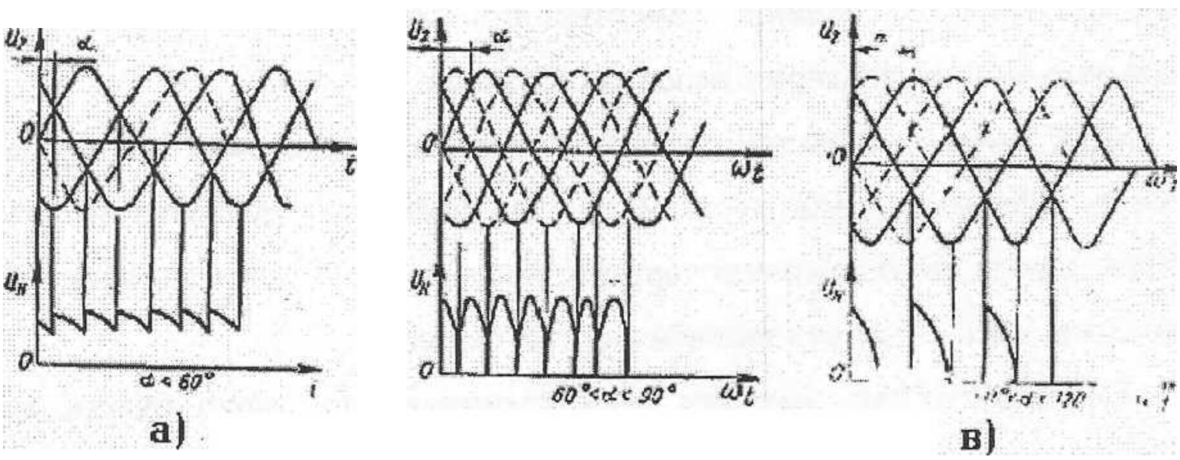
The results of continuous monitoring of statistical data on electricity consumption are the basis of planning of a number of preventive measures aimed at reducing the number of electric accidents and the number of electric injury hazards.

In the framework of the CIS, "Methodological Guidelines and Recommendations on Detection and Control of Electrical Outbreaks of Production" are used. It describes the procedure for the provision of electrical trauma, the procedure for completing accounting records and conducting the analysis.

In order to test the "Automated accounting and control of electric injury", which was developed by the Labor Safety Department, data on electric shock had been obtained at the enterprises of the State Energy Inspection.

5.1 picture). When the voltage is a <60 °, then the $U_{n\alpha} = U_0\cos\alpha$ [U0 - voltage at the output of a three phase non-controlling nozzle], and a> 60 °

$$U_{i\alpha}=U_0[1+\cos(\frac{\pi}{3}+\alpha)].$$



Thus, by changing the starting angle a, you can change the value of the tension in the output of the flare. Controlled hinges are used to control uninterrupted power supplies. You can control the speed of the electric drive by controlling the output voltage of the G-D system by installing a control quicksilver for a generator back. Large tire sticks can be used at quicksilver controllers.

As mentioned above, the voltage at the output of a tire riser is dependent on the angle of the thyratron control. The function of the operate system, based on the alarm management system for the opening of the thyratron, is as follows:

- Ensure that the control pulses are sewn;
- Symmetrical control phases are phases

to ensure that

- wide range of adjustment.

The impassable one is only in the opening angle of the tractor. The closing of the thyratron will cause the anode to be zero if it is zero. Therefore, the control pulses should be shorter, but should be effective over the time that the anode is equal to the working current.

Preliminary data; quality of state energy surveillance enterprises

Annual Reports were used. The data were collected in the 31-item "Electric Wound Cal cane" finished coded. Along with analysis, it is about electric shock

the importance of the information provided. This, on the one hand, illustrates the lack of information in the report and, on the other hand, determines that the accident is scrutinized fairly and in full.

The 9 most common causes of electric shock as a result of the analysis are related to these causes; 5 factors have been identified. About 80% of electrical injuries are caused by defects in work organization. 48.5% indicated that the material was neglected by management in the materials, 20.7% insisted on the safety of the victims, 31.5% were in violation of labor regulations, and 17.2% did not have work permits.

It should be noted that the proposed classifier is made up of traditional classification, from technical, organizational and other factors, to causing electrical injuries. To date, there are some difficulties in which group of reasons to enter.

Additional features such as "dangerous conditions" and "hazardous movements" have been included to describe the occurrence of the accident, which will clarify the case event.

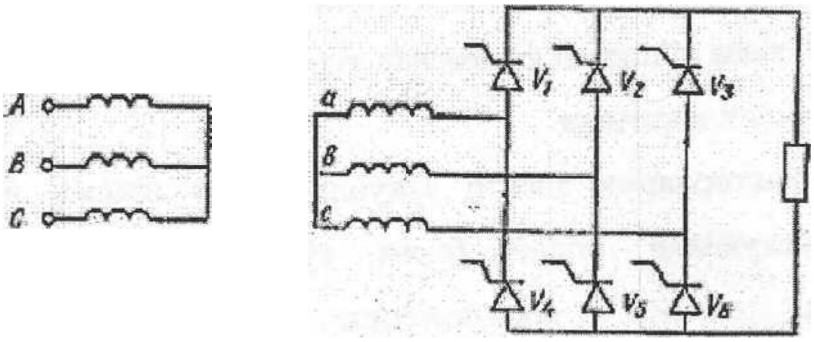
The above factors have been obtained with the equipment for the operation of electrical equipment, as well as the data for improving the design of electrical equipment, protective equipment and technological processes. The "preventive measures" classifier for the automation of the development of the complex of preventive measures has been developed. All classifiers, excluding the Profession Classifier, are built on a two-dimensional scheme. This is the data received from sectors and regions, and electrical injuries

The consumer's volatility is dependent on the consumer's character. If consumers are inductive, Vi and V2 are off the consumer even when the ventilators are closed. Due to the magnetic field energy of the consumer, the current is constant.

Consumer stress is as follows:

The value of the adjusted voltage depends on the dependence of the consumer on the value of a. The curves of consumer and vent voltage lines are given in 5.5, and v. Prior to connecting the ventilator V1, the voltage of the transformer U21 of the secondary transformer is given. When a = ω, Vi is opened and the deceleration in the voltage is equal to zero. When ωt = 180 °, V1 is closed and under voltage of the transformer's secondary coat. The tension in V2 is like the voltage in V1, but the phase is eventually switched to half the time

To control the three-phase currents, a wire diagram and spatial scheme are used. Let's look at the operation of the coupled steering wheel (Fig. 5.2). The scheme uses a three-phase transformer and six valves



In this case, double valves always work, for example, V1 and V4, V2 and V5, V6 and V3 and so on. In order for the circuit to work properly, it is important to put the control gate open to the appropriate ventilator.

When the angle of control a <60 ° (5.1, a), the voltage is constant and a> 60 ° is constant (5.1, b, c)

At present, the current value of the human body and the time it takes to determine the degree of electric injury by the human body. Production electrical equipment In case of disastrous operation, regulatory documents have been developed and approved, which determine the cost of transit passages from the human body. These documents describe the value of currents (depending on their impact time). According to the physiological criterion of the influence of the effect of the effect on the currents of short and long-term exposure

Functional connectivity of the TOE over time (t)

Coordinate is a curve that is described by the analytic expression in the flat system

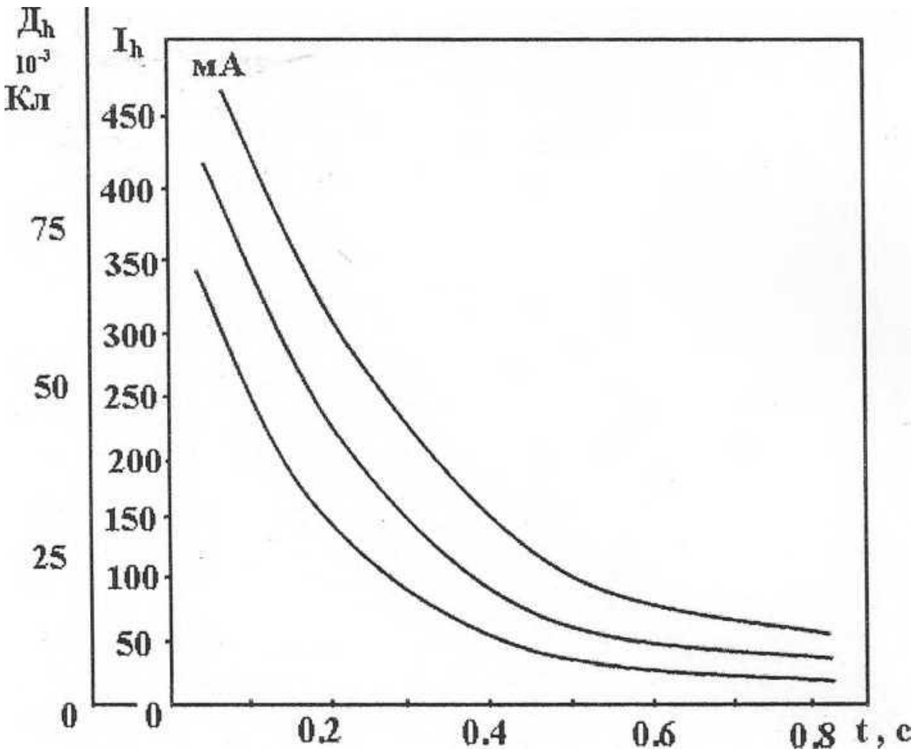


Figure 5.2 shows a curve that has a hyperbolic link:

$$Ih (t) = c / t$$

where S values different values.

Equations with 1-3 curves continues through the interval,

Analyzing these functional connections, one can be sure of the fact that the value of time and time for voluntary values remains unchanged

From the point of view of physics, this magnitude represents the amount of electric charge that affects the body of a human being under the voltage, Qh,

The electron energy that affects the human body is determined by the amount of electric charge available to the body.

It is possible to come to the conclusion that the entire range of electrical current has an amount of electric charge, which exceeds the amount of physiological changes in the human body.

It is proposed to add the integral size (dose of human dose), which is determined by the following formula as the normative size on the basis of the above:

Conclusion

By way of conclusion, I am capable of saying that on current days, FDMA is being employing in Uzbekistan railways, unfortunately this FDMA standard is inadequate system for Uzbekistan railways. Why? From mentioned information illustrated above, we are aware of that channels distributed per the specifically frequency. Admittedly, amplitude of dispatching signal is addictive to other outside signals in during connection time. That is why signal was decreased when receiver get sending signal and in that time, perhaps we might lose very essential part of information. Admittedly, losing information is unforgivable mistake in every sphere in the present time. Also, in the economic side is not affordable too. Practically, lying cables might be rot not to manage expulsive sufficiently.

In briefly, there has been much discussion revolving around the issue of whether FDMA standard is working right now, however subsequently this standard will not expulsive in Uzbekistan railways.

Literatures

1. Uzbekistan's development strategy for 2017-2021
2. Nick J.K. Wireless communication devices. Eng.: Eco – Trend, 2006 y.
3. Jane W.A. Several types of multiple accesses. Germ.: Scientists-w, 2003 y
4. Donna T.U. Wireless medium access alternatives. New York.: Vintage contemporaries. 1996 y.
5. W. Gao, Energy and Bandwidth- Efficient Wireless Transmission Signals and Communication. Switzerland.: Springer International Publishing, 2017 y.
6. Elaine. Sh.W. Houston D.A. Jay M.H. Multiple division techniques. Kentucky.: Boston Globe, 1998 y.
7. Georgie S.M Jessica P.P. Alvarez T.W. Cellular telephone networks. Lon.: Scientists 200y