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CONVERGENCE OF TELECOMMUNICATION SERVICES

Manual

For doctoral stud.of the speciality
6D071900-Radio engigeering, electronics and telekommunications

Almaty
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The presented tutorial summarizes, systematizes and presents information about a convergence of telecommunications services.
The manual is intended for doctoral students of the specialty "Radio engineering, electronics and telecommunications".

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Introduction

The current stage is characterized by a variety of technologies with intensive use of the TCP / IP stack in all areas of telecommunications (information, entertainment and consumer electronics). Using IP broadband access, both wired and wireless, including mobile has prepared a base that provides seamless access to any network resources, anytime, anywhere, using any devices. In these conditions, convergence becomes a reality.

The technological convergence brings the common denominator of the development trend of different technologies, telecommunications systems and various user terminals in solutions to common problems. Previously separately functioning communication networks can now share resources and interact with each other. It is appropriate to talk about convergence in infocommunications, convergence of services, convergence of networks, convergence of terminal devices.

Convergence in infocommunications is designed to make the transition from the business model of the service provider further the model of providing connections to the service model, as a result of which both parties optimizing costs will increase their revenues. Services based on the integration of information environments (electronic media, telecommunications network, broadcasting, digital television, telephony and the Internet) based on common standard joints and single terminals will become common.

Convergence of networks gives users the choice of network operators with the best access to services and applications.

Convergence of services - providing the user with new advanced services.

Convergence of terminals will allow using any set of applications from any terminal.

As a result of using these technologies, combining fixed and mobile networks with digital cameras, MP3 players, video cameras, voice recording and playback devices and other devices, which is called FMC (Fixed-Mobile Convergence), an environment is created where voice, data and video are harmonized and are routed through the IP network. Television closely integrates with the mobile phone industry. Today, telephone calls are made using personal computers. Mobile phones except voice transmission, carry out high-speed text and image exchange, video, music, receive television. High-speed resources of the IP network also routing new services with increasing mobility, which means that information technologies everywhere and infocommunication services constantly surround us.

1 Introduction convergence in infocommunications

1.1 Evolution of computer networks

In the late 1960s, computer networks appeared, as a result of which the process of convergence became the result of their influence on other types of telecommunication networks [1]. The prerequisites for rapprochement began when the telephone networks began to transmit voice in digital form. In turn, computer networks began to develop new services - IP-telephony services.

Table 1.1 shows the main stages of the evolution of the development of computer networks.

Table 1.1 - The main stages of the evolution of the development of computer networks

Stage	Time
The first global communications of computers, the first experiments with packet networks	The end of the 1960s.
Beginning of transmission over the telephone networks of voice in digital form	The end of the 1960s.
The emergence of BIS, the first mini-computers, the first non-standard local area networks	Early 1970s.
Creating an IBM SNA Network Architecture	1974
Standardization of X.25 technology	1974
The emergence of a PC, the creation of the Internet in a modern form, the installation on all nodes of the TCP / IP stack	Early 1980s.
The emergence of standard technologies for local networks (Ethernet - 1980, Token Ring, FDDI - 1985)	The middle of the 1980s.
Start of commercial use of the Internet	The end of the 80's.
The invention of the Web	1991

At the end of the 1980s, global and local networks were very different, but over time these differences became smoother, and today this converged network provides a multitude of diverse services to a huge number of users.

Currently, the main moving forces that have determined the development of the telecommunications network have developed:

- progress in the field of microelectronics;
- digitalization;
- development of technologies (Internet, wired, wireless and mobile technologies, digital television, etc.).

The research materials of IBBT analysts today note the following:

- traffic growth;

- number of users;
- methods of transmitted information;
- collected content;
- a variety of user devices;
- various applications for users;
- alternative power supplies.

The main goal of the NGN (Next Generation Networks) network under construction has many characteristics that relate to convergence:

- the core packet network carries information;
- Supports a wide range of services (real and unreal time, streaming and multimedia);
- Users have unrestricted access to different service providers;
- the existence of different identification procedures;
- integration of fixed and mobile services;
- Support for a variety of access technologies;
- providing convergent services based on the NGN concept and IMS architecture (IP Multimedia Subsystem).

The main trends and trends in the development of telecommunications are presented in figure 1.1, which shows that three directions can be distinguished:

- evolution of communication networks;
- globalization and personalization of the provision of services;
- Universalization of terminal equipment.

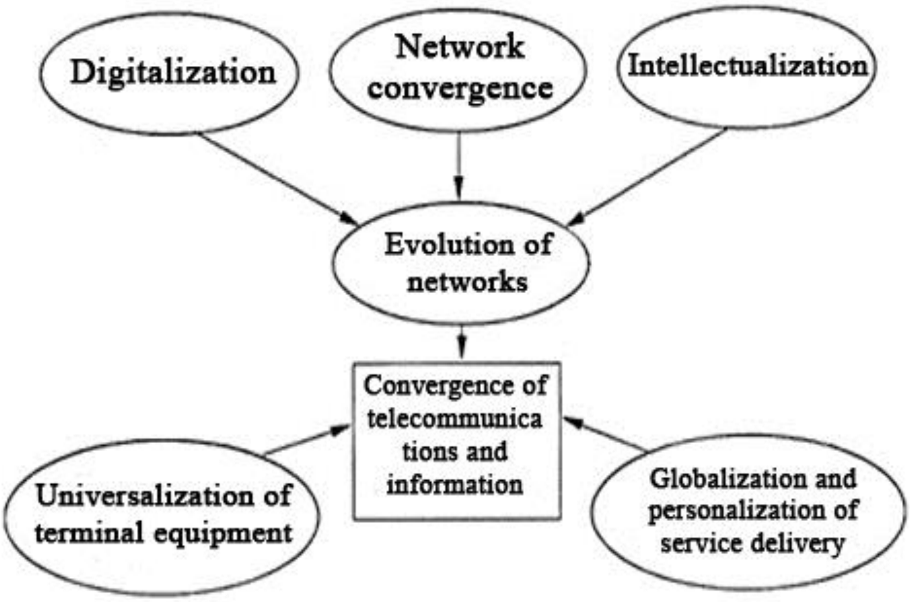


Figure 1.1 - Basic directions of development of telecommunications

Digitalization of networks opens up wide opportunities for providing new infocommunication services, in particular telematic services (e-mail, videoconferencing, database access services, etc.).

The emergence of intelligent networks is also caused by the provision of new types of services.

The need for convergence is caused by the desire to unify existing networks.

The convergence of information and communication technologies facilitates the unification of information flows, which leads to a reduction in the differences between the categories of telecommunications networks. New cloud structures change the usual ways of using information technology and undermine the old operating models. The goal of convergence in infocommunications is the transformation of the business model of the service provider from the connection model to the service provision model, which will increase the revenues of all market participants when optimizing costs. Most infocommunication services are "applications", that is, their functionality is distributed between the equipment of the service provider and the terminal equipment of the user, whose functions also relate to the composition of the infocommunication service. Infocommunication services assume the transmission of multimedia information, which is characterized by high transmission rates and the asymmetry of incoming and outgoing information flows. Infocommunication services are characterized by a variety of application protocols and user service management capabilities [2].

The NGN network is being further developed into an FMC network that will provide convergence of fixed and mobile networks, and then to the IMS network. Then this network will become one of the main components as a result of its upgrading to the level of MOC (Machine Oriented Communications) inter-machine communication support of the all-pervasive USN (Ubiquitous Sensor Networks) sensor networks of the Internet of Things. USN networks include [3]:

- various automated control systems (ACS);
- logistics;
- transport cars, air, sea and railway networks;
- control of environmental data;
- data on the status and location of each person;
- control over animal populations;
- control of plants in nature and in agriculture;
- control and management of various mechanisms, etc.

On the eve of the mobile-cloud era of the IT industry, typical features are ubiquitous access to data and their deep analysis in data centers, the virtualization of infrastructure, workplaces and applications, as well as the explosive growth in the number of devices connected by "man-man" schemes, "man-machine" and "machine-machine."

1.2 General concepts of convergence. Convergence of services

The word "convergence" in Latin means "bring together". In telecommunications, convergence refers to the process of convergence of telecommunication technologies and services that are different in their purpose for the purpose of unifying equipment and expanding functional capabilities.

In the case of the convergence of telecommunications and informatics, it is a new industry about infocommunications. The process of convergence became possible as a result, on the one hand, of technological progress and, on the other hand, new requirements imposed by consumers of services. [4, 5].

Convergence is the ability of various network platforms to provide virtually the same set of services, regardless of the type of receiving terminal device.

Aspects of convergence in infocommunication technologies are:

- Service convergence provides new enhanced functionality for users;
- Network convergence means the convergence of technologies and systems that enable the convergence of services;
- device convergence allows operators and providers to work with devices from different manufacturers and with different technologies and offer new efficient services.

Consequently, service convergence is a package of integrated services and services that combine mobile and fixed communication capabilities based on a single user profile. At the same time, the main purpose of convergence of services is the parallel delivery of all types of media carriers - voice, data and video to an easy-to-use graphical user interface that has mobility. That is, this variety of services (from person to person, from person to application, and from application to person, etc.) can be provided to the same user through different networks. For example, a convergent "single number" solution, which is a single number that combines mobile and fixed communications.

The behavior of the end user is changing rapidly, thanks to globalization, and the value of such concepts as individuality, new structures of social networks and communities, and the "nomadic" way of life are growing. The user expects from the perspective services the convenience of use, reliability and security of services. Therefore, in the near future, convergent service solutions for fixed and mobile networks will be determined by the active development of multimedia applications, as evidenced by an analysis of the development of the following factors in the information community:

- the growing use of consumer electronics - televisions, video tape recorders, video games, CD players, personal computers, etc.;
- development of information, commercial and entertainment services;
- the rapid spread of the Internet.

As a result, convergent services will:

- always available;
- on any device with voice, data, multimedia;
- intuitive;
- simple and personified;
- reliable and safe.

1.3 Convergence of terminals

Convergence of terminals implies the creation of a universal information terminal for the subscriber, which should provide the opportunity to use all types of infocommunication services. And as the emergence and development of new technologies expands the list of services provided by the universal terminal. Universalization (convergence) of terminals is carried out on the way of transformation of subscriber devices for the transmission and reception of specific messages: telephone (voice) stationary and mobile, facsimile, data transmission and others into a single infocommunication terminal (figure 1.2) [6].

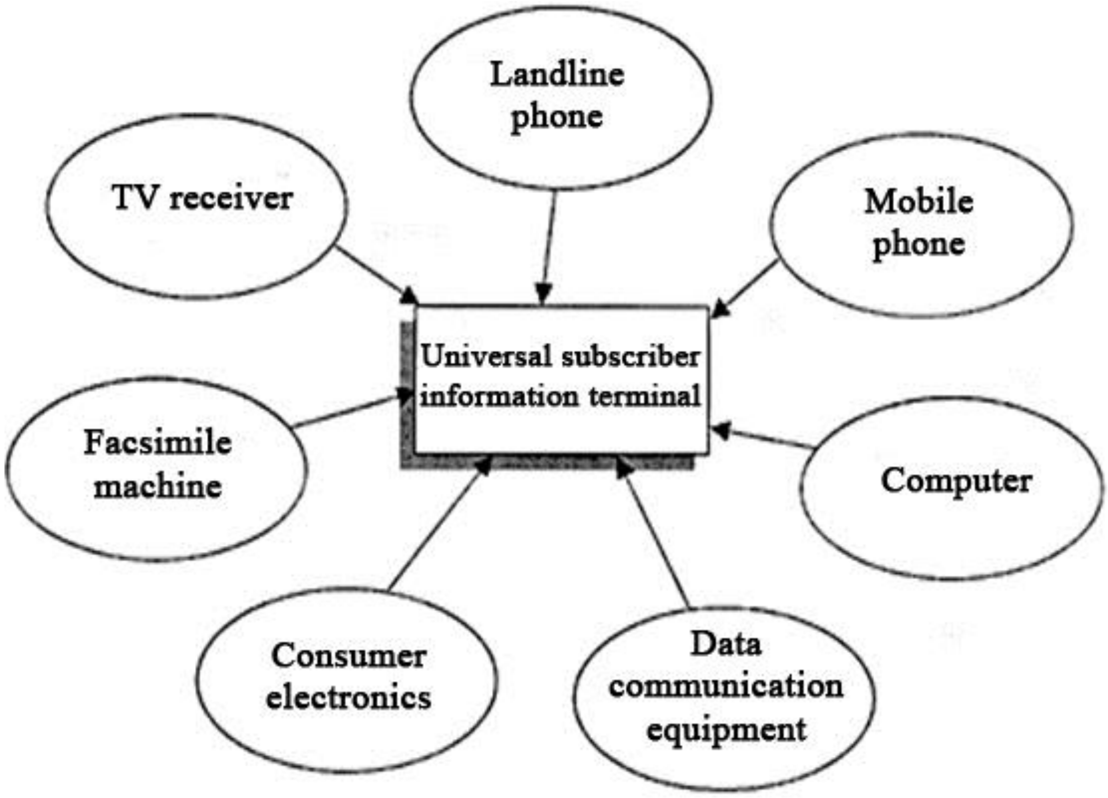


Figure 1.2 - Single infocommunication terminal

Progress in consumer electronics, defined by Moore's law, leads to the creation of increasingly complex micro-processors, an increasing number of functions are combined in one device. Obviously, a converged terminal device will become a device providing personal services, personal data, a personal profile and a personal interface. expected significant improvements in the characteristics of memory devices terminals and battery life, allowing to make convergent services in the terminal real.

The main idea is to use an individual communicator - a single universal device that supports various services (telephony, television, personal computer), as well as the transition to mobile devices.

1.4 Networks convergence

The time of networks with channel switching is a thing of the past, replaced with packet switched networks. The first generation of such networks was IP-networks, the second - NGN networks, the third - IMS networks.

The next step is the complete convergence of the FMC (Fixed Mobile Convergence) telecommunications networks with the provision of all telecommunications services over the IP network.

Convergence of networks means convergence or integration of various network technologies to create opportunities to provide users with heterogeneous services. As a result, there is no difference between telephone networks and data networks, or between public networks and corporate networks. At present, a multiservice network is deployed in the RK, which provides voice, data, and multimedia services. Multimedia means the integration of several information types of messages, such as text, images, graphics, animation and much more. The creation of multimedia has become the main direction in the development of information technology of the last decade and has led not only to the emergence of new technologies, but also to the emergence of new services.

The current stage is characterized by the following trends in the convergence of networks [7, 8]:

- splicing of telecommunication networks with computer networks;
- Fiber-optic guide systems are laid everywhere on the mains and in the access networks;
- the volume of data traffic is constantly growing;
- the boundaries between the technologies of the local area network (LAN) and the Wide Area Network (WAN) are being blurred.

Figure 1.3 shows the evolution of three networks: public switched telephone networks (PSTN), mobile networks (MN) and documentary telecommunications networks (DTNs) today coexisting within a single NGN network.

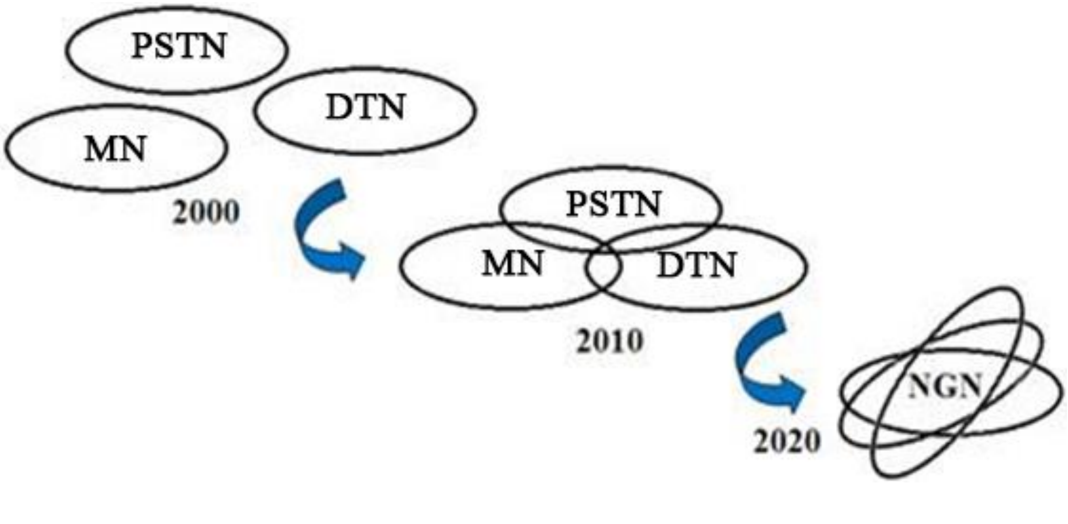


Figure 1.3 - Stages of network evolution

In the simplest form, network convergence involves the consolidation of networks in order to provide the user with various services through several types of access with guaranteed quality of service and cost-effectiveness. In the process of network convergence, backbone networks, access networks and service delivery platforms are involved.

In response to rapidly growing customer needs, network convergence has become the backbone of any digital Internet activity. Web surfing, quality analysis, testing, VoIP (Voice over Internet Protocol), IPTV (Internet Protocol Television), video and audio conferencing and e-commerce are services that use network convergence to interact with the public and business groups. Consequently, the convergence of networks inevitably raises the issue of new models of traffic distribution and transmission, both theoretically and in practice.

Technologies such as VoIP, IPTV, 3rd and 4th generation mobile systems, IMS (IP Multimedia Subsystem), are now seen as driving forces for network convergence.

The NGN network model will include three levels - transport, level of sessions and application level.

At the transport level, based on IP technology, the functions of switching and transmission of signaling information are provided. The main elements of the transport layer are various types of gateways, including Access Gateways through which subscribers are connected, and trunk gateways that connect a single network to separate networks (telephone, data transmission, etc.).

At the session level, the signal flows are controlled by the signaling gateway known as Softswitch.

The application layer provides access to a variety of services and supports a number of service functions related to network management.

1.5 Convergence of various branches of the infocommunication industry

Convergent services deliver personalized services across multiple domains. Service providers need to understand how to move services beyond traditional boundaries and ensure that all aspects of the subscriber's needs are met. In this case, end users will require the same level of services in all domains, at home, at work or on the road. The level of services will be important in the delivery of voice, data and video with the highest level of reliability and quality. The media level will provide content; service providers will need to interact with application developers and network operators. For service providers and operators, the ability to quickly respond to market demands by managing operating costs will be important in relation to the profitability of services. Success in the development of many standards in all sectors of the infocommunication industry - in telecommunications, information and media technology, the entertainment industry and consumer electronics, over the past few years has identified an infrastructure in which all existing types of services, networks and terminals will converge.

1.6 Stages of convergence

Figure 1.4 shows the stages of convergence. Three stages are shown:
- traditional TDM networks;
- Converging networks;
- Converged NGN networks.

Traditional TDM networks are characterized by the following:
- all telephony TDM;
- Over time, aging TDM-PBX, which are replaced by IP-PBX;
- VoIP application for long-distance communication

Converged networks are characterized by the following:
- the first softswitch is replaced by AMTS (automatic long-distance telephone exchange);
- distribution of multiservice subscriber access;
- Telephony - a mixture of TDM and VoIP.

Converged NGN networks are characterized by the following:
- copper in the access network is replaced by fiber and radio;
- Most of the voice traffic goes from local PBX to Softswitch;
- sets of new infocommunication services on the basis of Softswitch;
- all VoIP telephony.

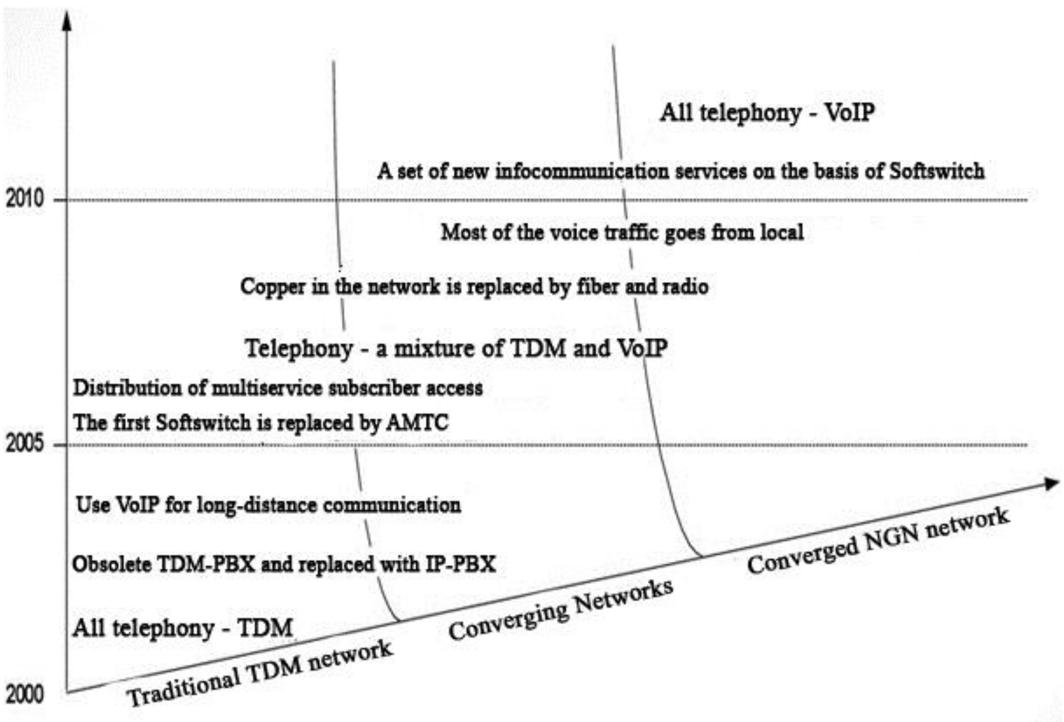


Figure 1.4 – Stages of convergence

1.7 Examples of convergence in infocommunications

The first example of convergence is related to the structure of the transport network. Transport networks of rural telephone network (RTN) and city telephone network (CTN) provide interconnections between nodes on a "everyone with each" basis. The topology of long-distance and p] rural transport networks have structures like "tree" and "star". Transport networks based on digital transmission systems (DTS) of the synchronous hierarchy are built on the basis of ring topology, and the collection of rings can be considered as a cellular topology. That is, the structures of fixed and mobile transport networks in the course of their evolution acquired the maximum similarity.

The second example of convergence - automatic telephone exchanges (ATS) were used earlier in the GTS, which differed from the automatic telephone exchanges developed for RTN. Leading manufacturers of switching equipment have created a set of hardware and software that allow the production of international, intercity, urban and rural stations, which after a certain adaptation are also suitable for servicing mobile traffic.

The third example of convergence - in recent years fixed telephone networks have actively used cordless terminals that expand the boundaries of the territory within which subscribers can move. In addition, there were telephone sets that support the exchange of SMS. The emergence of GPRS, EDGE and other technologies has made it possible to significantly bring together the functionality of fixed and mobile networks.

Approximation in the methods of data transmission occurs on the platform of digital data transmission over fiber-optic communication lines. This transmission medium is used in almost all technologies of local networks for high-speed information exchange at distances over 100 meters, modern main lines of SDH and DWDM primary data networks are also built on it, which provide digital channels for combining the equipment of global computer networks.

Great contribution to the convergence of local and global networks brought the dominance of the IP protocol. This protocol now works on top of any technologies of local and global networks (Ethernet, Token Ring, ATM, Frame Relay), combining different subnets into a single composite network.

The next example is the Internet, in which there were 3 stations in 1969, and today it is a global network with more than two billion users. Initially, it was used for communication between computers and for quick access to information in universities and other educational institutions, and today users watch video films, TV shows, listen to music, transfer files, talk on the IP network. Internet technology is the basis on which convergent services and convergent networks will be created.

Recently, along with the term Mobility, the term Nomadity has appeared, meaning the services of communication with limited mobility for users of personal computers. Here you can observe both the convergence of fixed and mobile networks, and the convergence of telecommunications and informatics networks.

The convergence process in telecommunications required the creation of integrated billing systems.

Convergence in telecommunications has a significant impact on the development of telecommunications network management technologies, and, in particular, leads to the creation of integrated management systems. Integrated management should ensure the provision of all possible resources of converged networks for the provision of any infocommunication services.

The convergence of telecommunications, information and media services leads to the emergence of a single information and communication technology (IST).

Prospects for further deepening of convergence in infocommunications many researchers associate with the creation of the Global Information Infrastructure (GII). The Telecommunication Standardization Sector (TSS) in the Y-series recommendations defines the GII as "a set of networks, end-user equipment, information and human resources that can be used to access useful information, connect users with each other, work, get entertainment at any time and from any place at an affordable price. " Thus, for users of GII is, in fact, some sort of universal network in which convergence of all possible types of infocommunication services is carried out. The creation of the GII will be carried out evolutionarily through continuous convergence, both existing technologies and existing and emerging technologies.

The next expected stage of convergence, according to some experts, will be the convergence of nano-, bio- and information technologies. The first steps in this direction have already been made. There were, for example, cell phones with built-in fingerprint sensors or blood sugar meters.

2 Evolution of telecommunication networks in the direction of building NGN

2.1 Levers of the telecommunications sector development

Today, the moving forces that determine the development of the world telecommunications sector are in fact that telecommunications around the world are at the stage of intensive development and in this sector of the economy there are significant changes at both the macro and micro levels.

Global changes at the macro level: the formation of a new legislative and regulatory environment; evolution of networks, services and terminal equipment in the direction of convergence, determined, on the one hand, by progress in key technologies and, on the other, new requirements and growing expectations of users.

The transition from monopolies to a competitive environment in the telecommunications sector began in the 1990s with the formation of a new regulatory and legislative environment - liberalization and privatization, which means competition between operators for each user, spreading the activity of operators not only in their own country, but also for their outside. This led to changes in the global telecommunications market.

Figure 2.1 shows that it is possible to implement industry reforms in three ways:

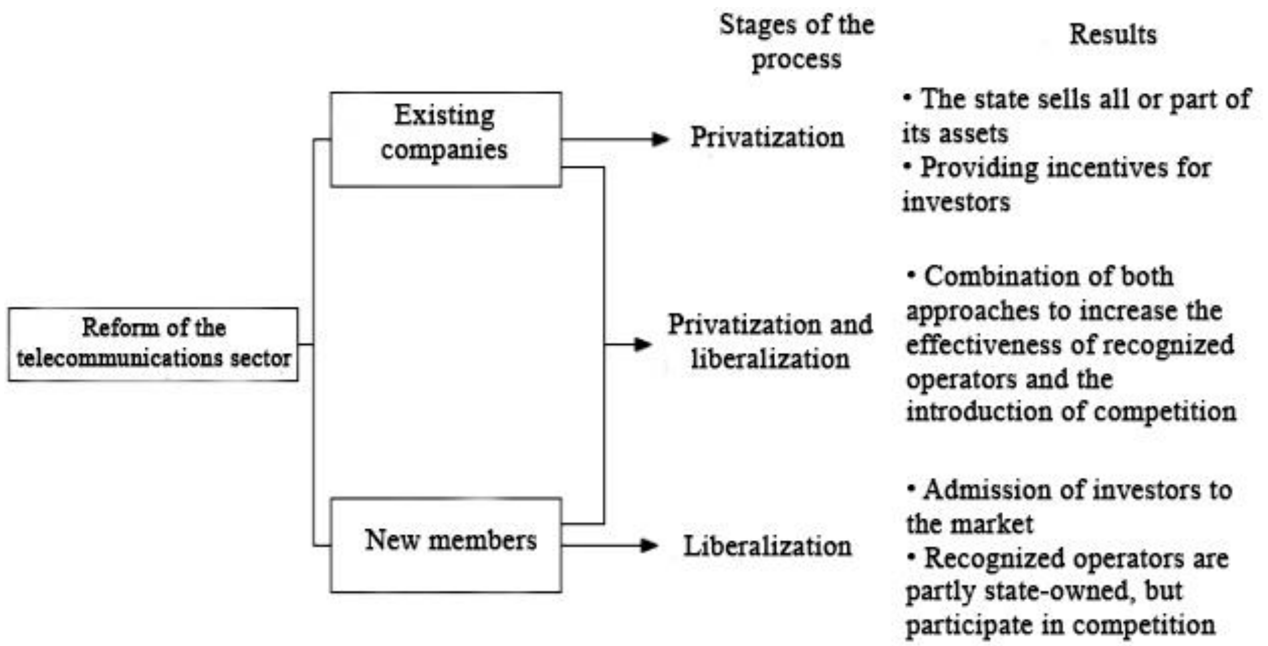


Figure 2.1 - Stages of the telecommunications sector reform process

- through privatization;

- through liberalization;

- a combination of the above methods, which leads to a complete reform of the sector [9].

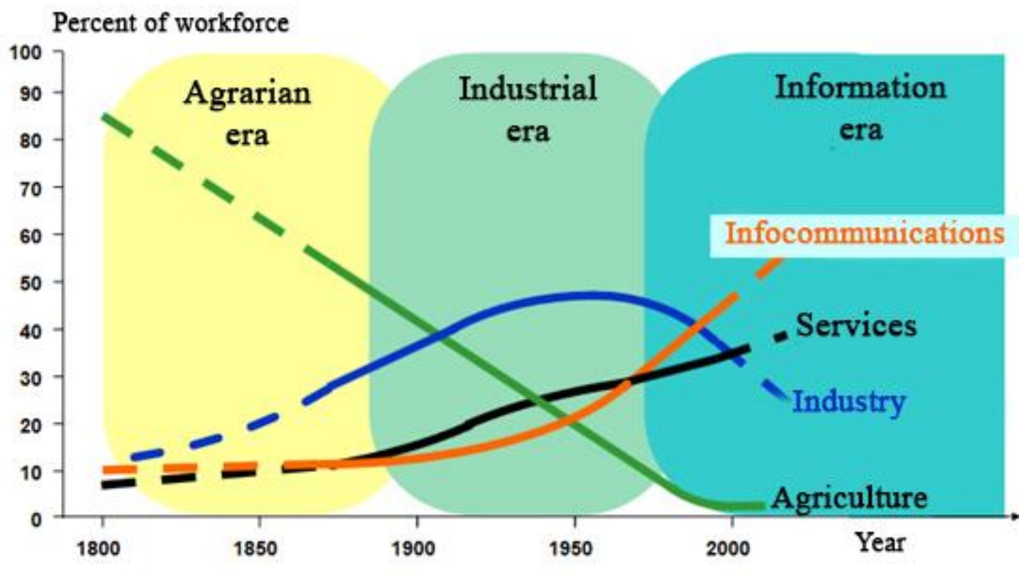
On the other hand, the key factors in the development of technological progress are the following:

- the growing performance of microprocessors;
- progress in the field of fiber-optic communication systems;
- the emergence of powerful digital signal processors;
- creation of highly effective methods of compression and information transport;
- the emergence of networks with very high throughputs;
- dynamics of the number of users of the telecommunication network;
- increase in the number and decrease in the cost of communication services.

Success in solving problems depended on the following main factors:

- access to services;
- level of competitiveness of prices;
- choice of services;
- connection rules for Internet and data transmission;
- percentage of fixed and mobile access lines, allowing them to be used for Internet access.

The employment of labor in these or those branches of production is evidence of the transition to a new stage in the development of human society. Figure 2.2 shows that, depending on the share of employment of labor, taking into account the development of agriculture, industry, services and infocommunications, the era is called agrarian, industrial and informational.



Источник: Leo A. Nefiodow, "Der fünfte Kondratieff", Gabler, Wiesbaden 1990

Figure 2.2 - Trends in labor force employment in various sectors

In accordance with the generally accepted in the world model of Kondratieff, the dynamics of economic processes, beginning with the end of the 18th century, is determined by large waves of growth and a fall in the values of economic indicators. The growing front of each wave is characterized by shorter industrial innovation cycles (two to three decades) at the beginning of the growing phase of each new wave. These short innovative cycles are characterized by a large number of new technical discoveries and inventions leading to the reorganization of existing production relations. But every large wave of Kondratieff begins with a significant invention or development of a fundamentally new technology, which determines the profile of each next large wave (figure 2.3).

Today is the beginning of the innovation cycle, which is connected with the discoveries in the field of information processing and transmission.

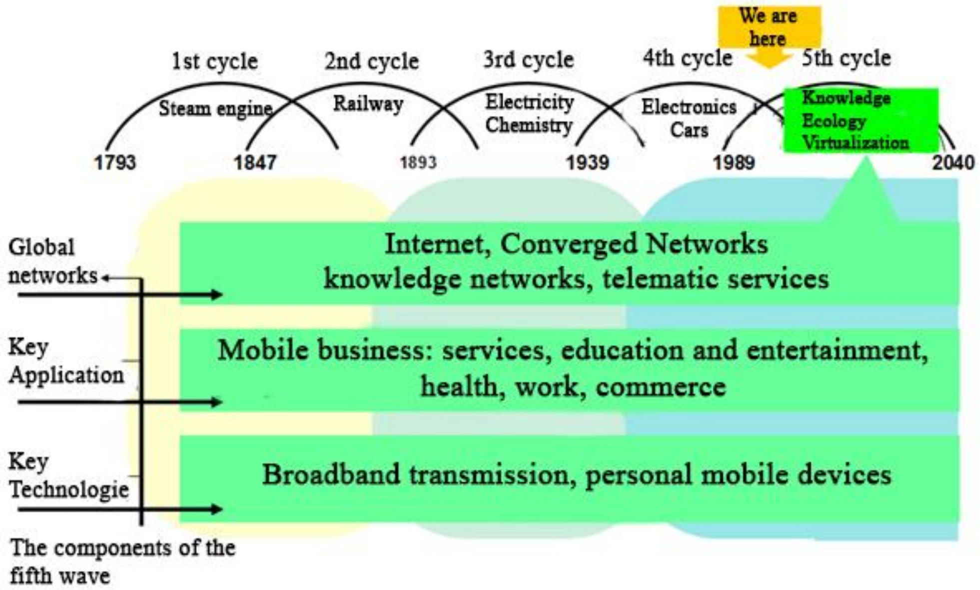


Figure 2.3 - Innovative cycles by Kondratieff

The key components of the innovation cycle that forms the profile of the new era are broadband geographically distributed / global IP-based communications networks (the World Wide Web), applications related to mobile networks and devices, and a huge number of available services in the areas of commerce, health, education and entertainment. The main priorities of the new era of human development are the problems of ensuring health and safety and personal well-being.

2.2 Basic technologies that form evolutionary processes in infocommunications

The current stage is characterized by the development of an information society based on advances in microelectronics (figure 2.4) and photon technologies (figure 2.5).

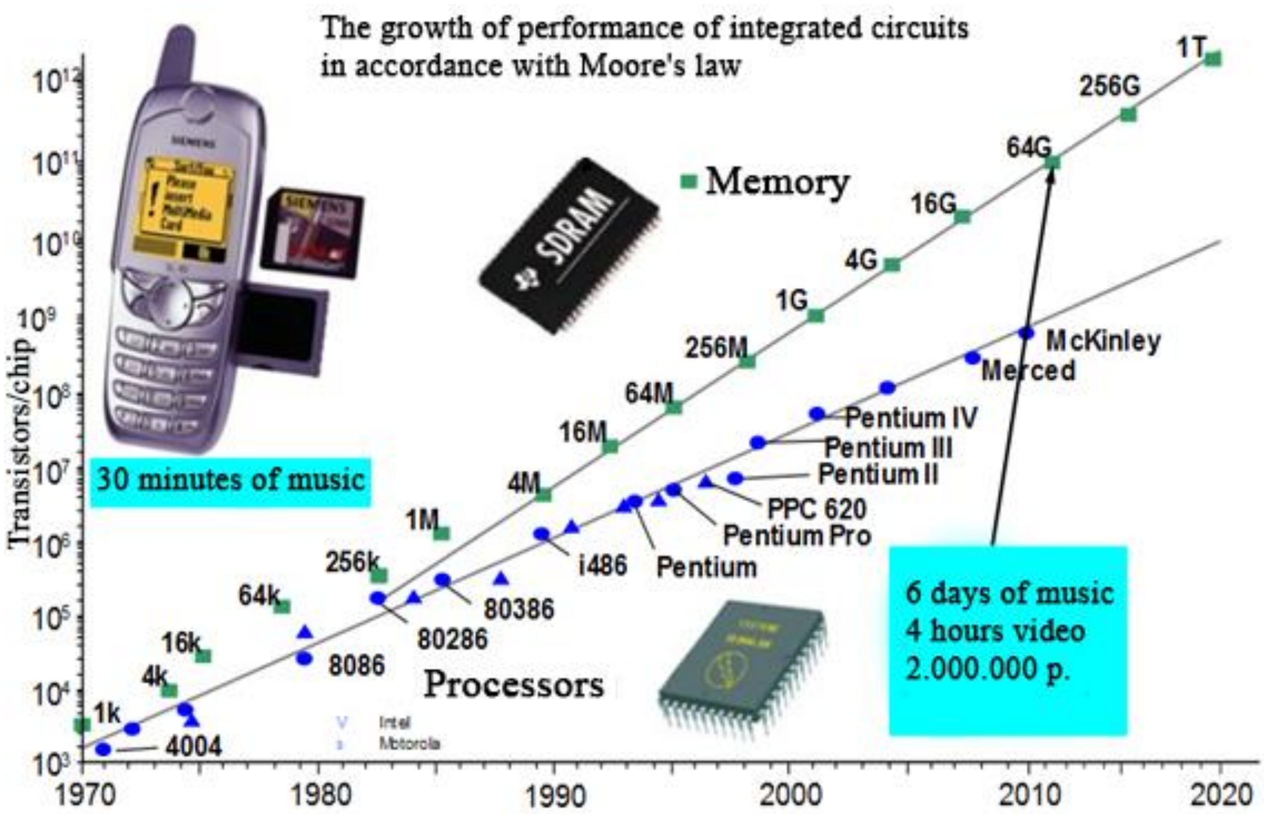


Figure 2.4 - Changing the number of transistors per chip by year

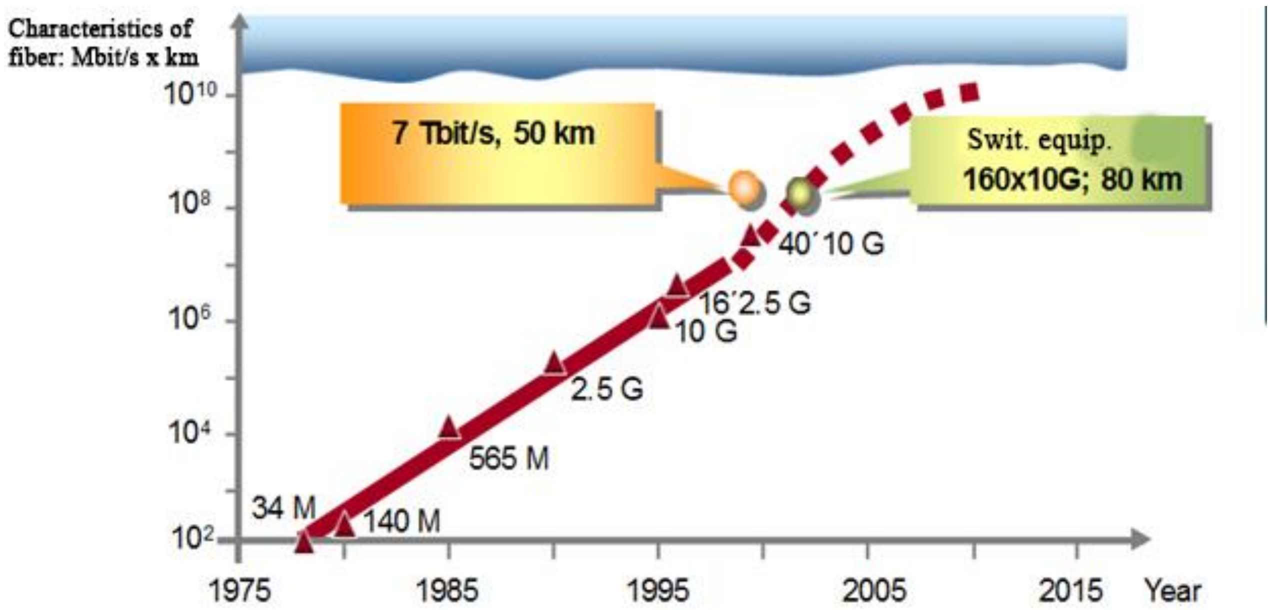


Figure 2.5 - Progress in the field of photonic systems

Microelectronics is the main base for the development of infocommunication systems. Today, microprocessors and memory chips are used in almost every product available on the telecommunications market. For example, a modern cell phone equipped with a multimedia card allows you to store and play high-quality video for several hours.

Given the continuous increase in the number of transistors per chip, we can expect in the near future the emergence of terminal devices (personal computers (PCs), mobile phones, etc.) in the form of single chips.

At the same time, semiconductor-based transistors, which are universally used today, can be miniaturized to 2015 or 2020. That is, the developers of silicon devices have come to a certain physical limit and it is necessary to search for new physical media for the implementation of microprocessor devices.

As predicted by experts, due to physical and technological reasons, time will require new devices. At present, some types of devices that will replace traditional transistors are already known. Among the new elements of circuit technology, they note the development of the following new devices:

- carbon nanotubes;
- organic transistors;
- molecular devices;
- Quantum processors.

Progress in network technologies in the future is related not only to advances in semiconductor technology, but also to advances in fiber optic technologies, which are the second key driving force in the development of infocommunications. The use of fiber-optic systems on communications networks began in the mid-1970s.

The main trends in the development of fiber-optic transmission systems are as follows:

- transition from multimode fiber to single-mode fiber;
- change in the wavelength of the used spectral window with □□ = 0.85 μm to □□ = 1.33 / 1.55 μm;
- decrease in attenuation in the fiber from several tens of dB / km to values of the order of 0.2 dB / km;
- increase in transmission rates, accompanied by a decrease in the cost of systems.

Progress in the use of fiber-optic systems on communication networks will make it possible in the near future to build photonic networks in which all the processes of transmission and processing of information are carried out on the basis of optical signals alone. In this case, the transformation of electrical signals into optical signals and vice versa is carried out in the signal source and in the optical receiver.

The bandwidth of photonic networks grows faster than systems based on semiconductor technologies. The capacity of fiber-optic transmission systems doubled every 10 to 12 months in the interval of the last thirty years of the 20th century.

At present, DWDM systems allow the transmission of information at a rate of tens of Tbit / s over the fiber.

Today, the bandwidth required to transfer the entire amount of international and inter-city traffic does not exceed 1 Tbit / c, that is, this total traffic can be transmitted over a single fiber.

A system with a capacity of 10 Tbit / s can transmit information stored in the human brain across the Atlantic Ocean for a time interval of 1 to 10 s (the required throughput is in the range of 10 to 100 Tbit / s).

2.3 Trends in the development of infocommunication

The number of global trends that determine infocommunications in the future include:

- all-pervasive digitalization;
- growth of mobile communication networks;
- growth in the scale of Internet networks;
- convergence of communication networks, terminals, services and industries of the information and communication industry.

In the 60-ies. the transition began from the analog form of information of all types to the digital format, which made it easy to process, store and transport information.

Communication networks are now becoming fully digital, in which all-pervasive computing is widely used.

Analysts predict to 2020 the main factors for the further growth of mobile networks:

- further rapid growth of mobile telecommunications services;
- Increasing requirements for intellectual services;
- Increasing security requirements;
- increase in the share of mobile payments;
- growth of mobile and fixed Internet traffic.

As for the Internet, today it is determined by the huge growth in traffic volumes, due to the growth in the number of consumers and the increase in the number and variety of applications.

The volume of traffic to the Internet is growing due to the following main applications:

- WWW, e-mail, file transfer;
- peer-to-peer applications P2P (Peer-to-Peer);
- files (Freenet, Gnutella, Morpheus, Kazaa, etc.);
- video applications, including video telephony, YouTube, IPTV;
- Video on demand;
- Data processing applications in which the software is provided to the user as an Internet service;
- games distributed over the Internet;
- applications (tele-participation - tele-education, telemedicine).

2.4 Growth and change in traffic patterns

For many years, the telecommunication network served narrowband voice traffic, which was formed in fixed (fixed) telephone networks and in mobile networks of the first and second generations. In the 90-ies. the volume of voice traffic increased by about 5-7% per year. Traffic data in 1970-1980 accounted for only a fraction of the percentage of total voice traffic that formed the system of e-mail and local networks. Since the mid-1990s. traffic data has become a growth.

Figure 2.6 shows the growth in data traffic, according to Cisco analysts [10]. At the same time, an 18-fold increase in data traffic is expected due to the connection to the network of myriad fitness trackers, smart watches and glasses, sports accessories and medical monitoring devices.

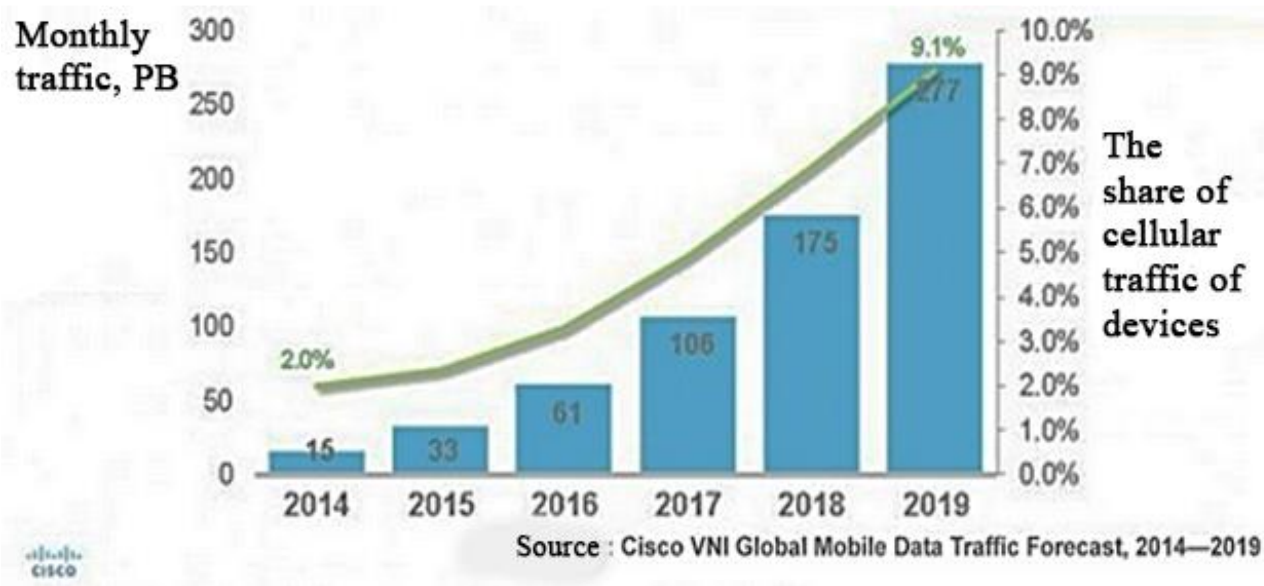


Figure 2.6 - Data traffic growth according to Cisco analysts

In [11], the key points of the report of Cisco VNI 2017 are described. According to the forecast of analysts of Cisco, which will triple the global volume of IP traffic in 2021 by three times. and will reach 278 exabytes per month (in 2016, the same figure was 96 exabytes). The annual volume of IP traffic by 2021 can reach 3.3 zettabytes. Internet traffic is 4.6 times (average annual increase of 35%) and by 2021 will reach 4.3 Pb / s. Only devices connected to Wi-Fi and mobile networks will generate 73% of Internet traffic by 2021. The number of public Wi-Fi access points, including domestic ones, for the period 2016-2021. will grow sixfold, from 94 to 541.6 million. The number of home access points of Wi-Fi in the world for this period will grow from 85 to 526.2 million.

By 2021, 56% of connected flat-panel TVs will support 4K resolution (in 2016 - 15%).

The rejection of traditional TV services (Cord cutting) implies an increasing substitution for viewing traditional on-air and subscription television by other services, such as mobile and online video, which are available to users via fixed and mobile Internet.

The average annual increase in traffic of software-defined global networks (SD-WAN) will be 44%, of traditional WAN networks - 5%.

2.5 Evolution of network technologies

At the beginning of the XXI century, a change in transmission and switching technologies took place. Modern telecommunication systems and networks are a complex set of various technical means that ensure the transmission of various messages to any distance with given quality parameters.

Now, in telecommunications, a situation has arisen where customers do not want to simply receive communication services, and operators can not provide them with new services based on existing networks, which in turn underwent their development evolution (figures 2.7 and 2.8).

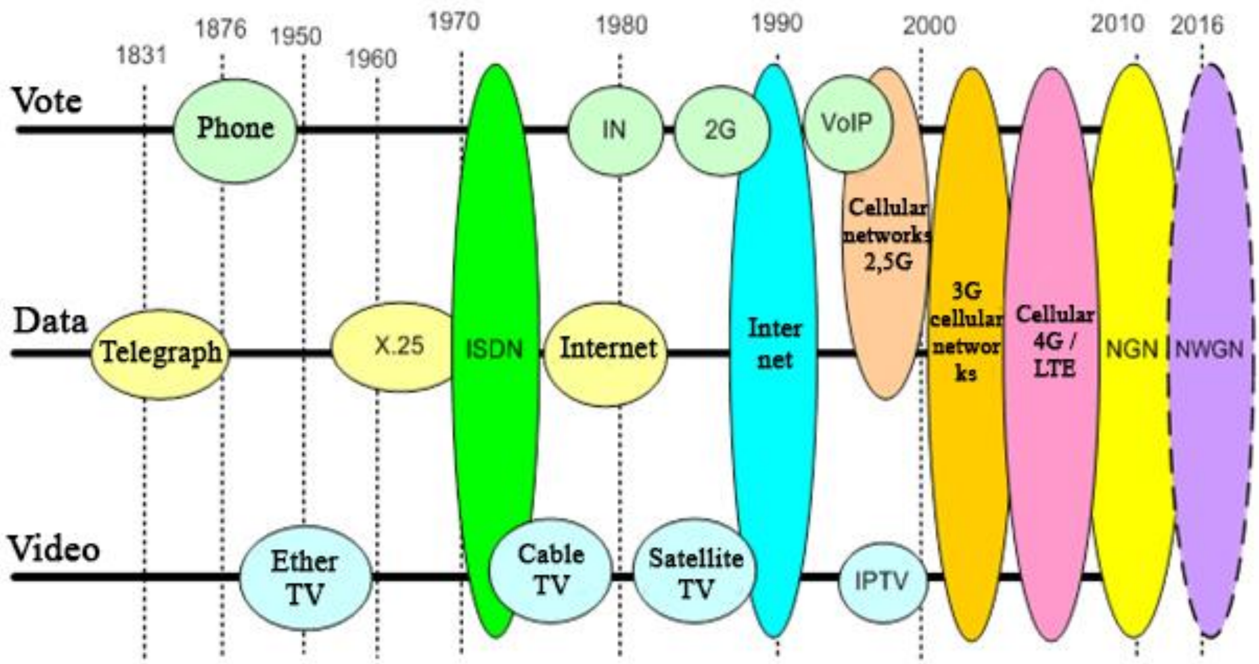


Figure 2.7 - Evolution of communication networks and technologies

In traditional networks, the quality of the service is completely determined by the technology features (PSTN, FR, 2G, etc.).

In multi-service networks, several types of traffic are transmitted over the same network (ISDN, ATM, 3G, TCP / IP).

In the NGN network, various multiservice services are managed independently of the technology (IMS, SIP).

Post-NGN networks manage the habitat, organize the creation of a single information space, interpenetrate ideas and technologies of automation and telecommunications.

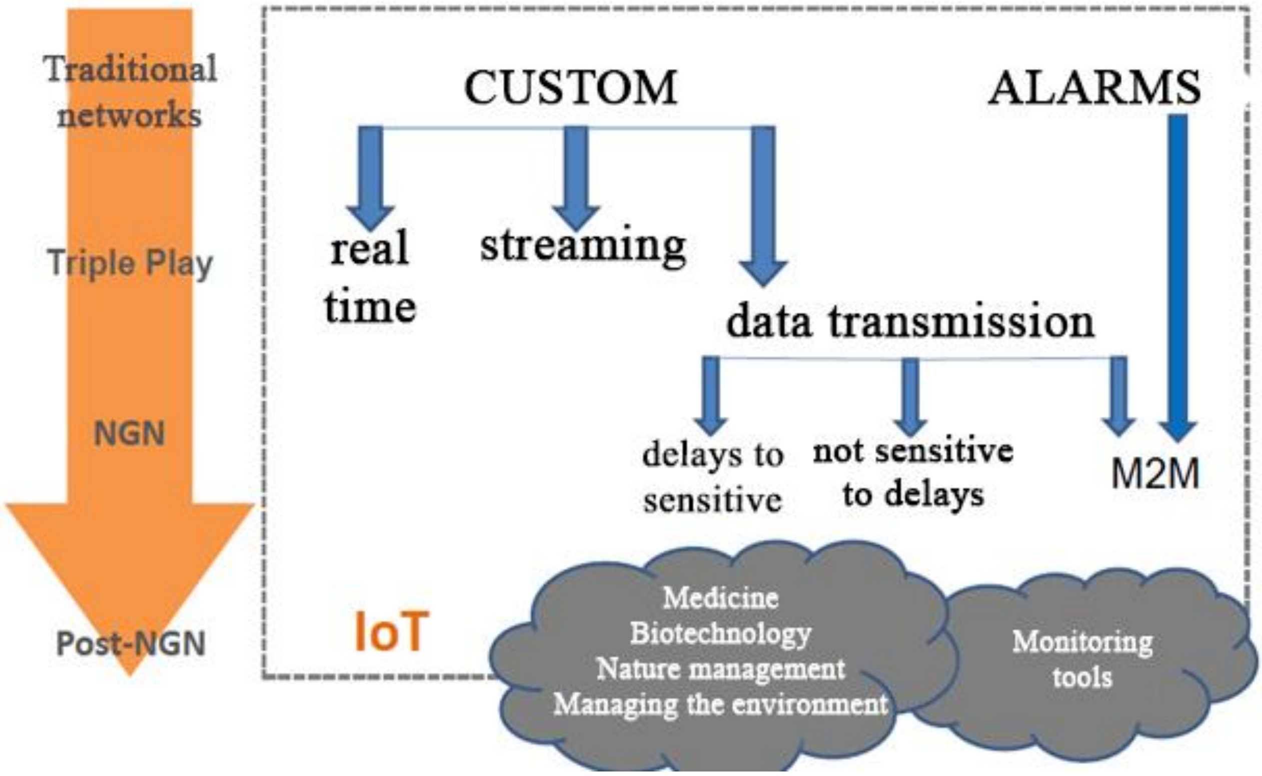


Figure 2.8 - Evolution of telecommunications technologies

The features of the NGN network are:

- independent development of the levels according to the NGN model (separation of service level from transport);
- mobility of the user;
- multiserver;
- splicing of telecommunication and computer technologies:
- development of virtualization technologies.

The prerequisites for the transition to post-NGN:

- saturation of the telecommunications market by the number of users and the range of services;
- stagnation of telecommunication technologies with the transition to all IP;
- increase in the number of mobile users;
- active penetration of web technologies into the infrastructure of settlements and everyday life (smart home, smart city, etc.);
- development of automation and computerization of vehicles, medicine, monitoring tools.

Transition to post-NGN networks [12]:

- the philosophy of controlling remote sites online, increasing interactivity;
- creation of an infocommunication environment in which the user acts as a consumer of the properties of the environment, and not traffic (smart environment);

- changing approaches to managing the network, service, resources, infocommunication environment;

- further active computerization of telecommunication devices;

- change of addressing - transition to IPv6.

The change in the structure of traffic of post-NGN networks is based on the following:

- the number of fixed telephony users decreases, the sharp growth of mobile users;

- new types of traffic appear, with about 70% of traffic coming from video, 10-11% to telephony, 11-12% to web traffic, about 6% to M2M traffic;

- the amount of traffic focused on creating an infocommunication environment is increasing;

- custom applications are increasingly focused on the use of virtualization technologies, including - cloud.

Signs of the post-NGN network are as follows:

- expansion of the range of services through the inclusion of new fields of knowledge: medicine, biotechnology, nature management, automation and monitoring;

- priority use of wireless access (Wi-Fi, ZigBee, LTE);

- the emergence of access networks of a new type - based on ad hoc / mesh architecture, with a cluster organization, swarm structures;

- new quality assurance mechanisms - for many applications the requirements for delays (up to 50 ms) and losses are tightened;

- introduction of mechanisms of self-organization.

The modern network is a complex system with a large number of variable parameters and degrees of freedom. Such systems are prone to self-organization.

The task of post-NGN networks is to artificially apply the principles of self-organization when building and / or developing a network.

Areas of application of self-organization - networks of access levels and aggregation, peer-to-peer and mesh-networks, superimposed networks.

The post-NGN network's problems are as follows:

- Development of self-contained long-acting power supplies;

- development of algorithms for self-organization and routing, taking into account energy saving and tolerance to failures;

- development of procedures for switching and unloading traffic, taking into account the declared requirements for quality;

- development of new mechanisms to ensure the required quality of services and their control;

- development of standards and standard solutions for basic post-NGN services;

- development of new wireless communication technologies, including in other environments.

In the future, it will be necessary to ensure the transmission of ever-increasing volumes of traffic and the connection to the network of an increasing number of devices. From the moment of the civilization's creation and until 2003, humanity has created 5 exabytes of information. In 2016, the volume of global traffic is estimated at 1.4 zetabytes, that is, 30 exabytes a week [13].

With the growth in the number and the quality of services, convergence of the telecommunication (TC) and information (IR) complexes into a single infocommunication complex takes place.

Infocommunication network is a technological system intended for transmission of information through communication lines, access to which is carried out using computer facilities.

TSS describes the Global Information Infrastructure (GII) as a collection of networks, end-user equipments, information and human resources that can be used to access useful information, connect users with each other, work, entertainment at any time and from anywhere at an affordable price .

From the point of view of users, GII infrastructure is a common environment of a set of interconnected networks, penetrating into which through the user terminal (PT) the user is able to satisfy his information and communication needs (figure 2.9) [14].

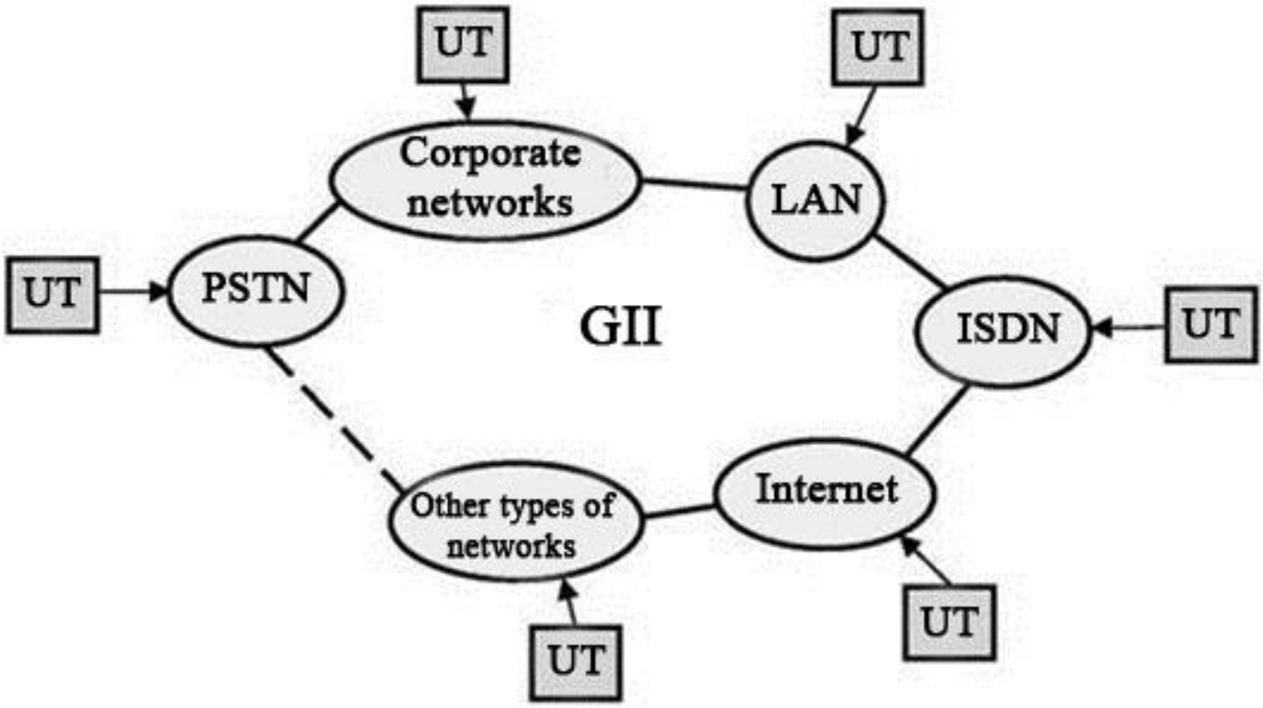


Figure 2.9 - Global Information Infrastructure

The General concept of development of the global information infrastructure – is an evolutionary approach, support existing and future telecommunications, informatization, entertainment, consumer electronics, their integration in a common infrastructure (figure 2.10).

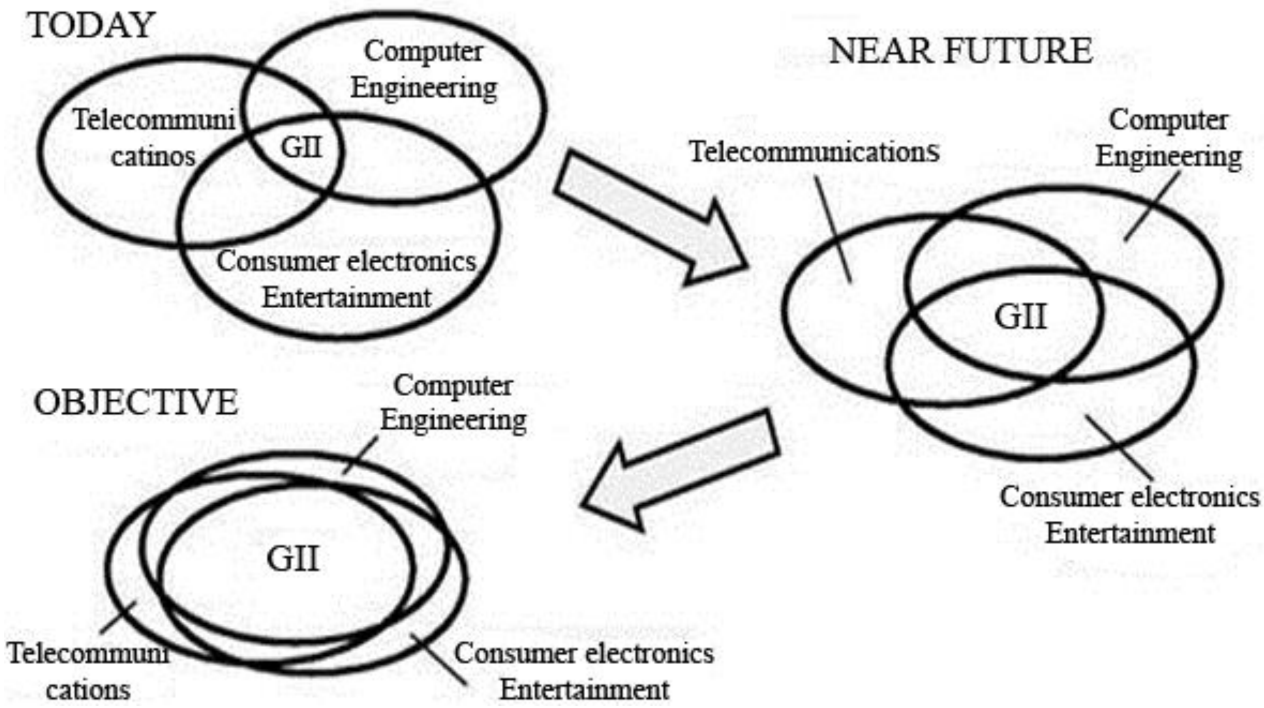


Figure 2.10 – Evolution of the GII

Feature of the global information infrastructure in future will be support a variety of applications, examples of which are: the realization of remote educational technologies and electronic libraries, telemedicine, teleworking, e-Commerce, games, etc.

Areas of Infocommunications development is as follows:

- system (globalization, personalization, intelligence, mobility and convergence of communications and Informatics, e-business);

technology (broadband, multimedia, dynamic telephone traffic, integration of terminals, quality of services);

- structural (de-monopolization of the market, alliances, mergers and acquisitions operators, structuring, and restructuring);
- economic (increase in the number of subscribers and new services, a flexible tariff and financial policy, reducing the cost of equipment and services).

Figure 2.11 shows the structure of information and communication technologies.

The basic components of information and communication technologies are:

- Hardware HW (Hardware), which includes all equipment network or system;
- microelectronics ME (Micro electronics) - semiconductor devices, LICs, SLICs, etc .;
- software SW (Software), which determines the algorithms and programs of operation of the equipment;

- Computers and processors CP (Computers & Processors), integrating elements of computer technology;

- radio technologies RT (Radio technologies), providing the use of radio waves for information transfer;

- fiber optic communication lines FO (Fiber Optics), using optical fibers for transmission of information based on optical radiation;

- Power supply AB (Accumulators & Batteries) is an important element of infocommunication technologies, especially in mobile performance;

- Design of PD (Projects & Design) networks and infocommunication systems, based on the use of computer equipment and databases (DB).

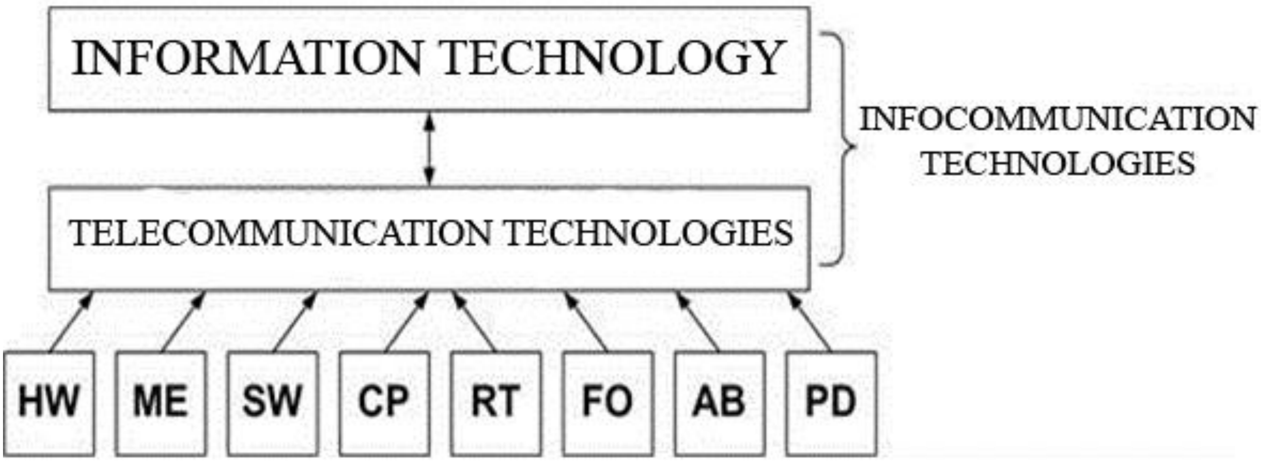


Figure 2.11 - Basic components of infocommunications

2.6 VoIP Systems

The telecommunications industry is in the process of transitioning to all-IP systems, which is due to a fundamental need - to reduce costs, create new services that generate additional revenue and introduce new business models.

Most of the additional services that bring revenue to GSM cellular operators are IP-based services: WAP access, MMS, download of ring tones / pictures / games via GPRS, etc.

The advantages of all-IP networks are the versatility and flexibility of creating new services, integrating technologies and services, and reducing costs.

VoIP or IP telephony is a communication system that provides voice transmission over the Internet or any other IP networks. The signal on the communication channel is transmitted in digital form, before transmission is transformed (compressed) in order to remove redundancy.

Currently, enterprise-class IP-PBXs are designed to replace the traditional PBX (enterprise PBX).

IP-PBX can be IP-telephony systems:

- convergent systems that support both packet solutions and solutions based on circuit switching technology. Thus, these are intermediate, convergent solutions on the way from classical TDM switches to "pure" IP-PBXs. Advantages of convergence application (reduction of network maintenance costs: simpler management and reduction of the total amount of equipment) and communication itself (low tariffs for IP traffic, increase of productivity of each employee (efficiency and quality in decision making due to access to common information resources outside depending on the location (office, home, hotel, public hot spot)), increasing the productivity of the team as a whole (working together in real time with common databases / information outside of depending on the location of each employee), improving the efficiency of client base maintenance (speed and quality of responding to a request) and real protection and a low return on investment (figure 2.12) [15];

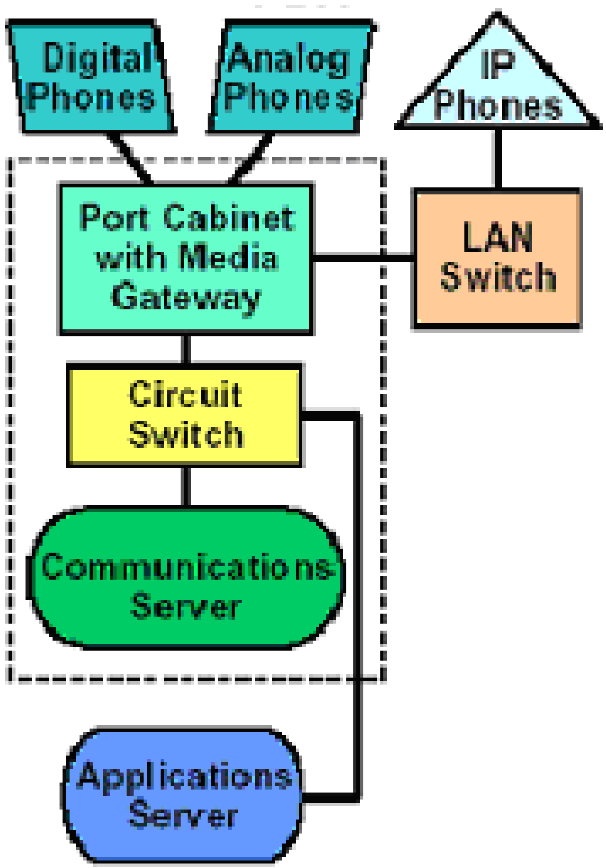


Figure 2.12 - Converged IP Telephony System

- IP-telephony solutions based on a local area network (LAN), in which there is no channel switching (figure 2.13) [16].

LAN telephony is understood as such a telephone communication organization in which only one signal wiring remains (in addition to power) in the building with a local computer network, either computer-network (for example, Ethernet) or telephone. In this case, both computers and telephone sets are connected to the same outlets.

For the organization of telephone communications in these conditions, use alternative PBX (also called pseudo-PBX or un-PBX) - these are PBXs based on personal computers (PCs). Those un-PBXs that use the same signal wiring to transmit telephone signals and data can be classified as PBX for LAN telephony.

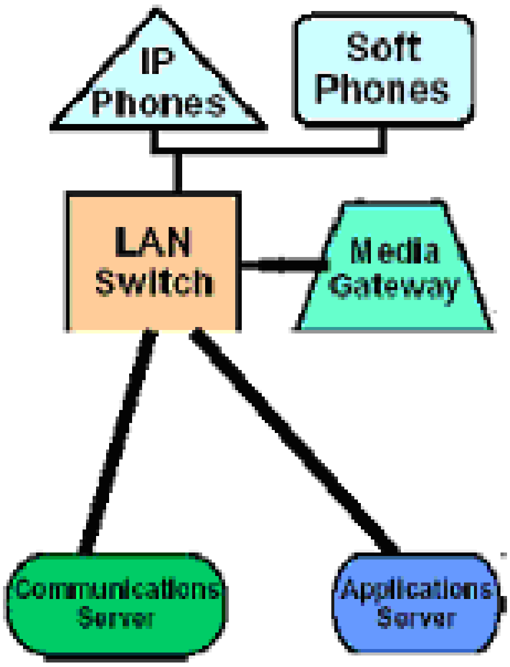


Figure 2.13 - IP-telephony based on the local network

Systems of this class are divided into the following three main categories:

- distributed LAN-PBX. These systems consist of several elements: telephone servers, gateways and telephone switches;

- integrated communication systems (all-in-one-box) - specialized devices with built-in OS and application software;

- PBX with IP gateway - traditional PBX, where IP gateway boards are installed for IP communication.

IP-PBX of the carrier class carries out switching of voice channels, transit of calls, billing based on H.323, SIP and MGCP standards.

2.7 H.323 network architecture

Networks based on H.323 standards are focused on integration with digital telephone networks and are considered as ISDN networks superimposed on the data network. Therefore, the procedures for establishing connections in such networks are based on Recommendation Q.931, which is similar to the procedures used in ISDN networks. This option is used by operators of local telephone networks who are interested in using a packet switched network (IP network) to provide long-distance and international telecommunications services. The main components of the network are the following devices (figure 2.14) [17]:

- Terminal (Terminal);

- Gateway;
- gatekeeper;
- MCU Conference Control Unit (Multipoint Control Unit).

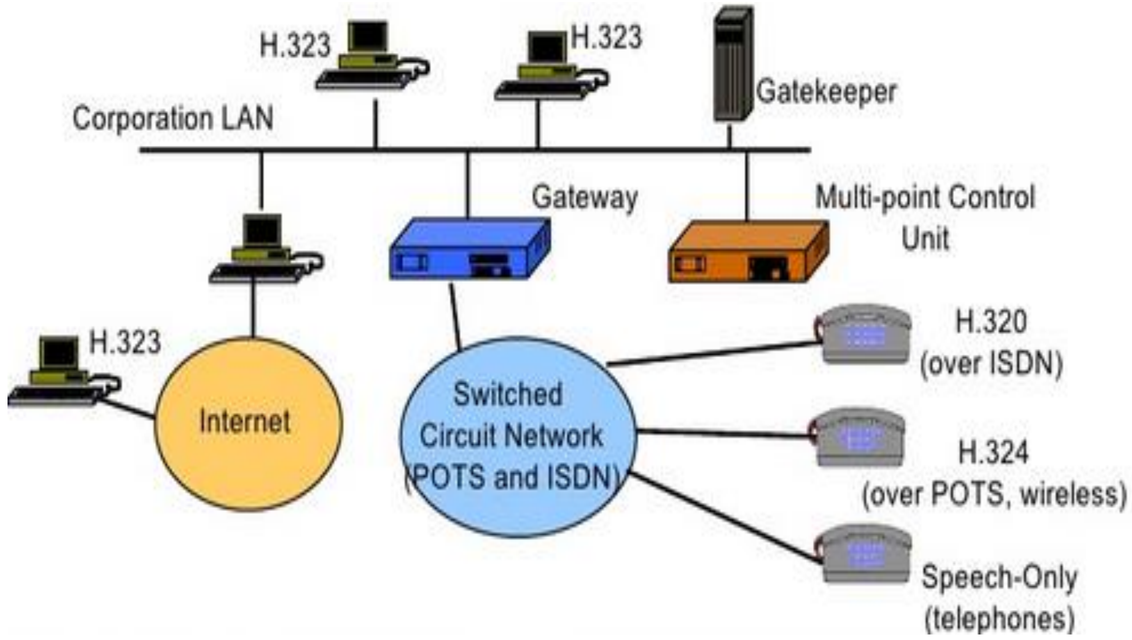


Figure 2.14 - H.323-based network architecture

Terminal H. 323 is an IP telephony network user device that provides two-way voice (multimedia) communication with another H.323 terminal, gateway or MCU.

The IP-telephony gateway implements the transmission of voice traffic and signaling messages over networks with IP packet routing over H.323 protocol, most of the gateway functions are performed in application-layer processes.

The gatekeeper manages the IP network area in which the terminals, gateways and conference control devices that are registered with this gatekeeper are located. The main functions of the gatekeeper are as follows:

- converting alias address (subscriber's name, telephone number, e-mail address, etc.) to the transport address of networks with IP packet routing (IP address and TCP port number);
- control of access of system users to IP telephony services using RAS signaling;
- control, management and redundancy of network capacity;
- routing of signaling messages between terminals located in one zone;
- the gatekeeper is the network administrator.

The MCU consists of a multipoint MC controller and a processor for processing user information in multipoint MP connections (there may be more than one).

Recommendation H.323 provides for three types of conferences:

- centralized conference with the connection of endpoints in the point-to-point mode with the device MCU;

- decentralized conference with connection of terminals in point-to-point mode. In this case, the terminal devices process (switch or mix) the information flows coming from other participants in the conference;

- a mixed conference, i.e. combination of the two previous species.

The H.323 stack consists of seven groups of protocols (figure 2.15):

- control and signaling (H.225.0, H.245);
- processing of audio signals (audio codecs);
- processing of video signals (video codecs);
- conference call;
- transmission of multimedia information (RTP);
- ensuring information security (H.235);
- additional services (H.450).

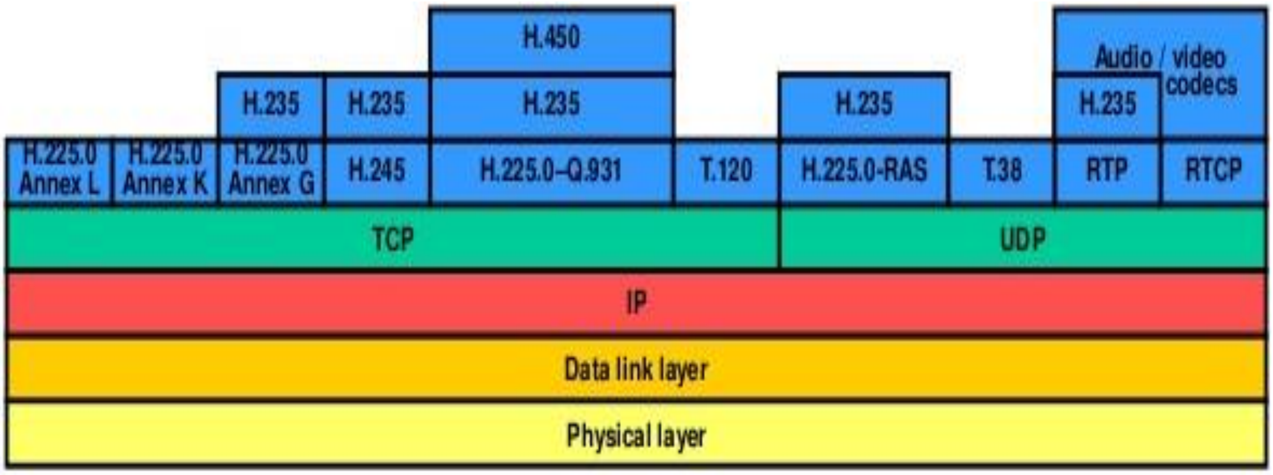


Figure 2.15 - H.323 protocol stack

2.8 SIP network architecture

The architecture of the SIP-network is based on the client-server model. The client sends requests to the server, which, after processing the request, issues a response, in which there may be a notification of the successful execution of the request, an error notice or information requested by the client.

The architecture of the SIP network consists of terminals and servers that differ in the functions performed by the proxy servers, redirection servers, registration servers and location servers (figure 2.16).

A terminal / computer with a software-implemented phone or a hardware-implemented VoIP phone. Since voice transmission requires bidirectional interaction, both client and server agents are implemented in the terminal. They are called the User Agent Client and the User Agent Server.

The proxy server performs the main job of managing the messaging session (SIP itself does not transmit voice, it is a protocol that helps organize it). It accepts requests from clients, processes them, and gives answers to the user. The proxy performs tasks such as locating the user, establishing a connection, and routing calls. He also has to make calls himself if the session needs the participation of a third-party proxy server.

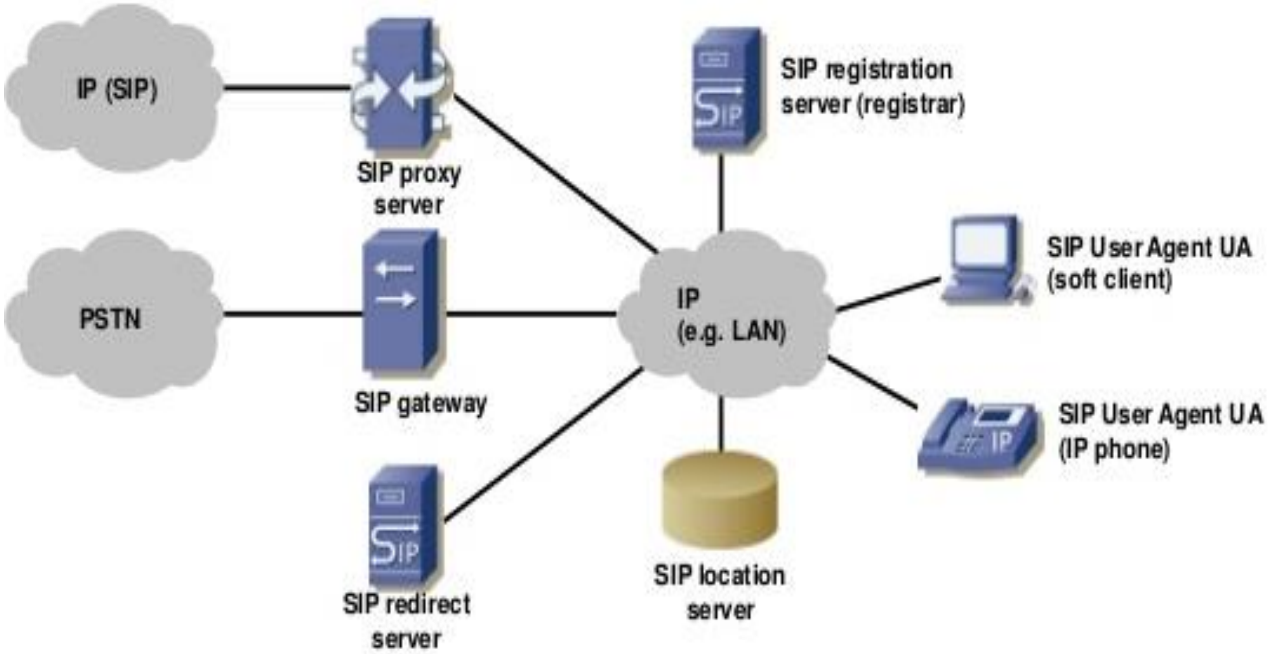


Figure 2.16 - SIP network architecture

Typically, the service provider tells its subscriber the address of the proxy server. In the future, the user equipment will send him a request to connect with any phone. The server itself, or using the location server, finds the addressed user and sends him a connection request. Receives from him the answer and sends to the first terminal the signal about the held or rejected connection [18].

The location server, or the registration server, is engaged in determining the user's location on the network, that is, by determining possible routes to it. This information is obtained as a result of registration of the terminal (the REGISTER command). Registration can be carried out one-time (fixed telephone) or periodically (mobile).

The redirection server handles the transfer of a call from a known address to a real one.

2.9 The architecture of a network based on the MGCP protocol

The network architecture based on the MGCP gateway control protocol developed by the IETF MEGACO group for terminal management contains a gatekeeper (control device) or a SIP server. Next, the MEGACO group created the

MEGACO protocol, which manages the MG transport gateways using the MGC call agent (softswitch), which defines the traffic processing logic (figure 2.17).

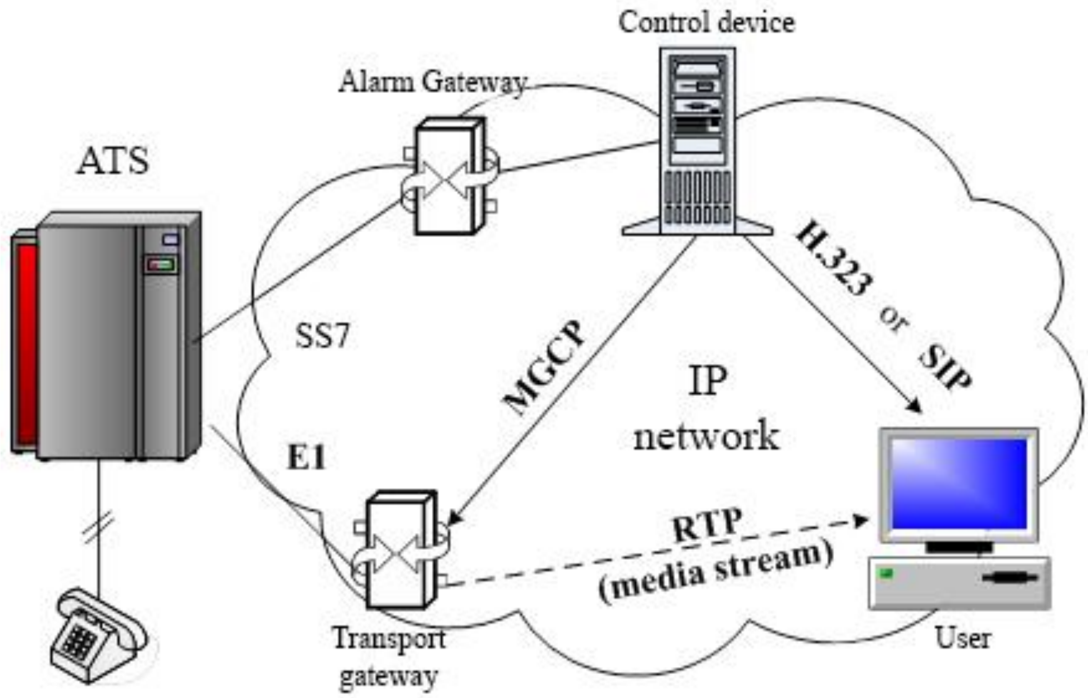


Figure 2.17 - Terminal Management in the MGCP Network

In this case, the gateway is divided into separate functional blocks [19]:

- Media Gateway, which performs the functions of converting voice information coming from the PTN side with a constant speed, into a view suitable for transmission over packet-routed networks IP encoding and voice data packets to RTP / UDP / IR packets and inverse transformation;
- Signaling Gateway, which provides the delivery of signaling information coming from the PTN side to the gateway management device and transfer of the signal information in the reverse direction.

Thus, the entire intellect of the functionally distributed gateway is located in the control device, whose functions, in turn, can be distributed among several computer platforms. The signaling gateway serves as the STP - transit point of the common channel signaling system - ACS7. Transport gateways perform only the functions of converting voice information. One control device serves several gateways at the same time.

The transfer of MGCP messages provides a protocol of not guaranteed delivery - UDP.

If the distributed gateway is connected to the PTN by signaling via dedicated signal channels (DSC), the signal information along with the user information is first fed to the transport gateway and then transmitted to the control device without the mediation of the signaling gateway.

One of the main requirements for the MGCP protocol is that the devices implementing this protocol should operate in a mode without storing information about the sequence of transactions between the control device and the transport gateway.

The main drawback of this approach is the unfinished standards. The functional blocks of distributed gateways, developed by different telecommunication equipment manufacturers, are almost incompatible. The functions of the gateway management device are not exactly defined. The mechanisms for transferring signal information from the Signaling Gateway to the control device and in the opposite direction are not standardized. The drawbacks include the lack of a standardized protocol of interaction between control devices. This means that a gatekeeper or a SIP server must be present on the MGCP-based network for terminal management.

The IETF MEGACO Working Group, continuing the studies aimed at improving the gateway management protocol, has created a more functional MEGACO protocol that defines the interaction on one side of the gateway between different media environments of the MG (Media Gateway), which converts the voice signal of a circuit switched network to a packet traffic and, on the other hand, a media gateway controller between the MGC (Media Gateway Controller), sometimes called a call agent or Softswitch, which determines the logic for processing traffic.

In other words, MEGACO is designed for intra-domain remote management of devices responsible for establishing a connection or conducting a communication session, such as VoIP gateways, remote access servers, digital Subscriber Line Access Multiplexers (DSLAMs), multiprotocol switching routers using MPLS (Multiprotocol Label Switching), optical cross-connects, PPP session aggregation modules, and others.

In H.323 v.4, the ITU-T introduced the principle of decomposing gateways. An important feature of the ITU-T innovation was the fact that MG transport gateways are managed by the MGC transport gateway controller using the MEGACO protocol adapted for the H.323 network environment. The specifications of this protocol are given in ITU-T Recommendation H.248. For the transfer of MEGACO / H.248 signaling messages, UDP, TCP, SCTP or ATM technology can be used. Support for these purposes of the UDP protocol is one of the mandatory requirements for the gateway controller. The TCP protocol must be supported by both the controller and the transport gateway.

2.10 NGN Network

The prerequisites for the NGN network appearance are the following:

- Mass introduction of modern systems and communication facilities, the characteristic features of which are multiservice and multiprotocol;

- a significant change in network architectures: the abandonment of a rigid hierarchy, characteristic of classical public telephone networks, under the influence of the introduction of new means of communication, the principles of transmission and processing of information;

- functional division of the level of the transport switched network and the level of service formation resulting from the introduction of intelligent networks (IN) and fixed in NGN (thanks to the Internet, the operator does not need to have its own transport network, and the range of services went beyond traditional communication services, -the concept of "telematic services");
- Increasing competition in dynamic market sectors, such as mobile communications, the Internet, services for corporate users;
- the division of the business model of the new services operator into two parts: infrastructure (creation and maintenance of the network) and service (related to marketing);
- the presence of intermediate links - virtual operators that form and implement value-added service packages, as system integrators in IT do;
- changing the status of infocommunication services: the network itself is losing its value, it is acquiring services;
- Reducing the role / share of voice services in modern packages Triple Play (TP) and Quadruple Play (QP);
- use of conditionally free services based on the operation of the Internet (for example, a service provided by Skype);
- reduce the investment attractiveness, competitiveness and profitability of traditional communication systems.

The communication networks built in accordance with the NGN concept ensure the provision of an unlimited set of infocommunication services:

- Voice VoIP services;
- video calls;
- videoconferencing;
- the Internet;
- corporate networks, VPN;
- IPTV, VoD;
- the organization of data transfer for municipal services, housing and communal services, for organizing public order control, and transport traffic.

The main difference between next-generation networks from traditional networks is that all the information circulating in the network is divided into two components. This is the signaling information providing subscriber switching and service provisioning, and directly user data containing the payload intended for the subscriber (voice, video, data). The paths of the signaling messages and the user load may not coincide. To date, the main device for voice services in NGN networks is Softswitch - the so-called softswitch, which manages VoIP sessions. Another important function of the software switch is the connection of next generation NGN networks with existing traditional PSTN networks, through signal and media gateways that can be performed in one device [20].

The problem of switching from traditional circuit-switched networks to packet-switched networks (NGN) is one of the most urgent for telecom operators. Perspective developments in the field of IP communications are associated with the creation of integrated solutions that allow the development of next-generation networks to maintain existing connections and ensure uninterrupted operation in any network:

- on the infrastructure of copper pairs;
- through optical channels;
- Wireless (WiMAX, WiFi);
- a wired (ETTH, PLC, etc.) network.

The main difference between NGN-networks and networks of the old generation is that the division between the transport level and the level of service provision becomes logical rather than physical. Because NGN-network means easy and fast start of new services. This is what operators of communication are interested in - not in building transport for each new service, but in rapidly deploying services based on modern platforms.

The introduction of the NGN network involves the following actions:

- creation of a unified information environment for data transmission;
- formation of distributed transparent and flexible multiservice data transmission networks;
- optimization of IT infrastructure management;
- use of modern call management services;
- provision of multiservice services;
- management of services in real time;
- support for mobile users;
- monitoring the quality of services provided and the operation of network equipment

As a result, all information flows are integrated into a single network.

Components of the NGN network:

- Universal transport network, which includes the IP core of the network and access networks using different technologies;
- service delivery system;
- Subscriber accounting system, billing system;
- Network management systems.

The main properties of NGN are:

- Multiservice, which is understood as the independence of technologies for providing services from transport technologies;
- broadband, i.e., the ability to flexibly and dynamically change the information transfer rate in a wide range, depending on the current needs of the user;
- multimedia - the ability of the network to transmit multi-component information (voice, data, video, audio) with the necessary synchronization of these components in real time;

- Intelligence, which means the ability to manage a service, call and connection from the user or service provider;

- multi-operator - participation of several operators in the process of providing the service and sharing their responsibilities depending on the area of their activities.

ITU-T Recommendation Y.2001 describes that the NGN network has packet switching suitable for providing services in telecommunications networks using several broadband transport technologies with QoS enabled, where the service-related functions do not depend on the underlying technologies responsible for transportation. It allows for unimpeded access of users to networks and competing service providers and / or to the services they select and supports universal mobility that provides constant and ubiquitous service to users (figure 2.18) [21].

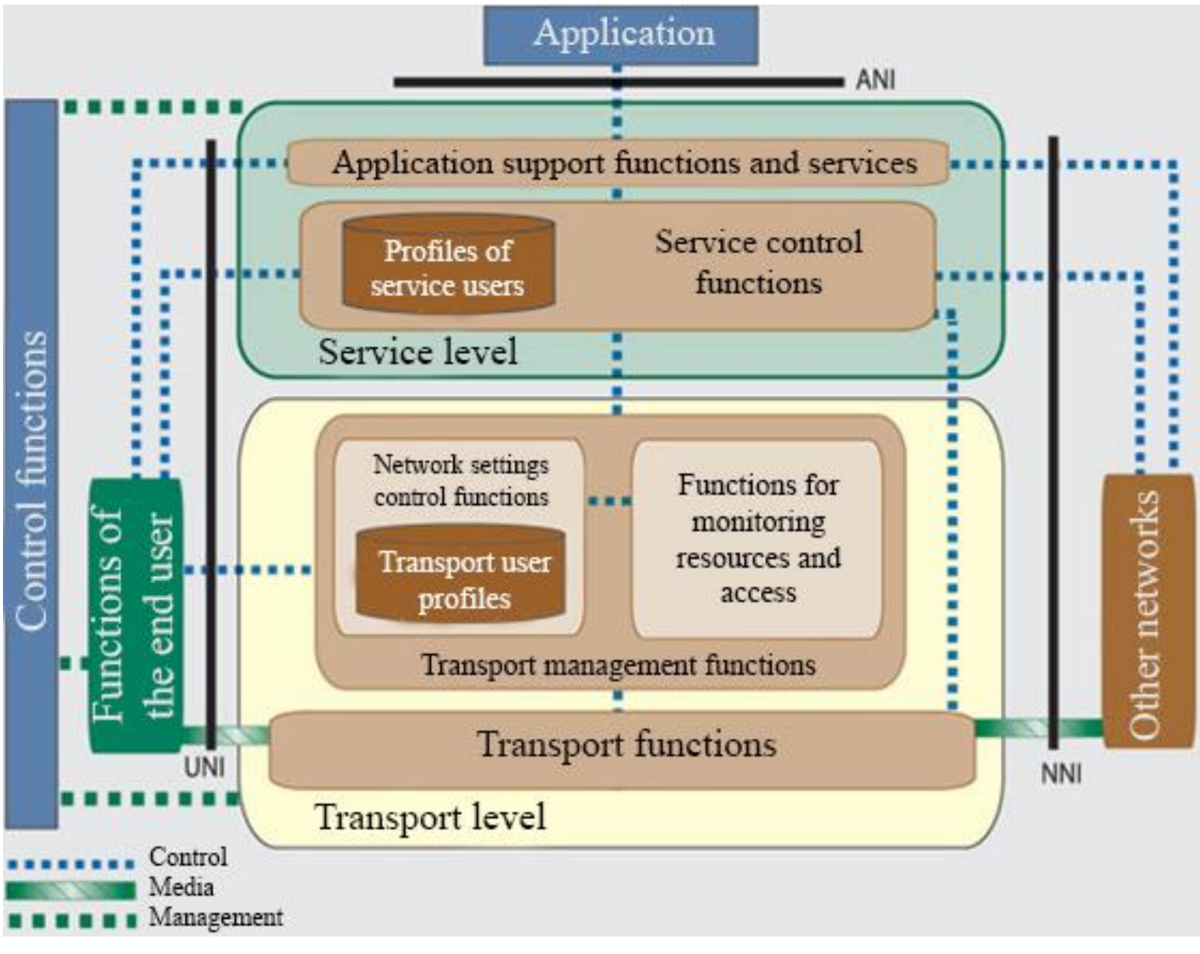


Figure 2.18 - The reference architecture of the NGN network

The difference of the NGN network architecture from previous technologies lies in the inter-model of the logical interaction of the subsystems. The traditional construction of the communication network was based on the nodes of switching at different levels of the hierarchy when the access nodes worked with them. At the same time, the same equipment (communication node) performs several functions simultaneously: broadcasting of user information (data), call service in the process of establishing a dial-up connection, and providing services and additional services. The reference architecture of the NGN network is characterized by the following levels:

- transport level;
- level of services;

application layer.

To solve its tasks, NGN will be implemented in accordance with the architectural solution of IMS, which is standardized within 3GPP for all tasks (roaming, billing and subscriber registration) formulated for NGN.

2.11 SoftSwitch Technology

The structure of the NGN network is shown in figure 2.19.

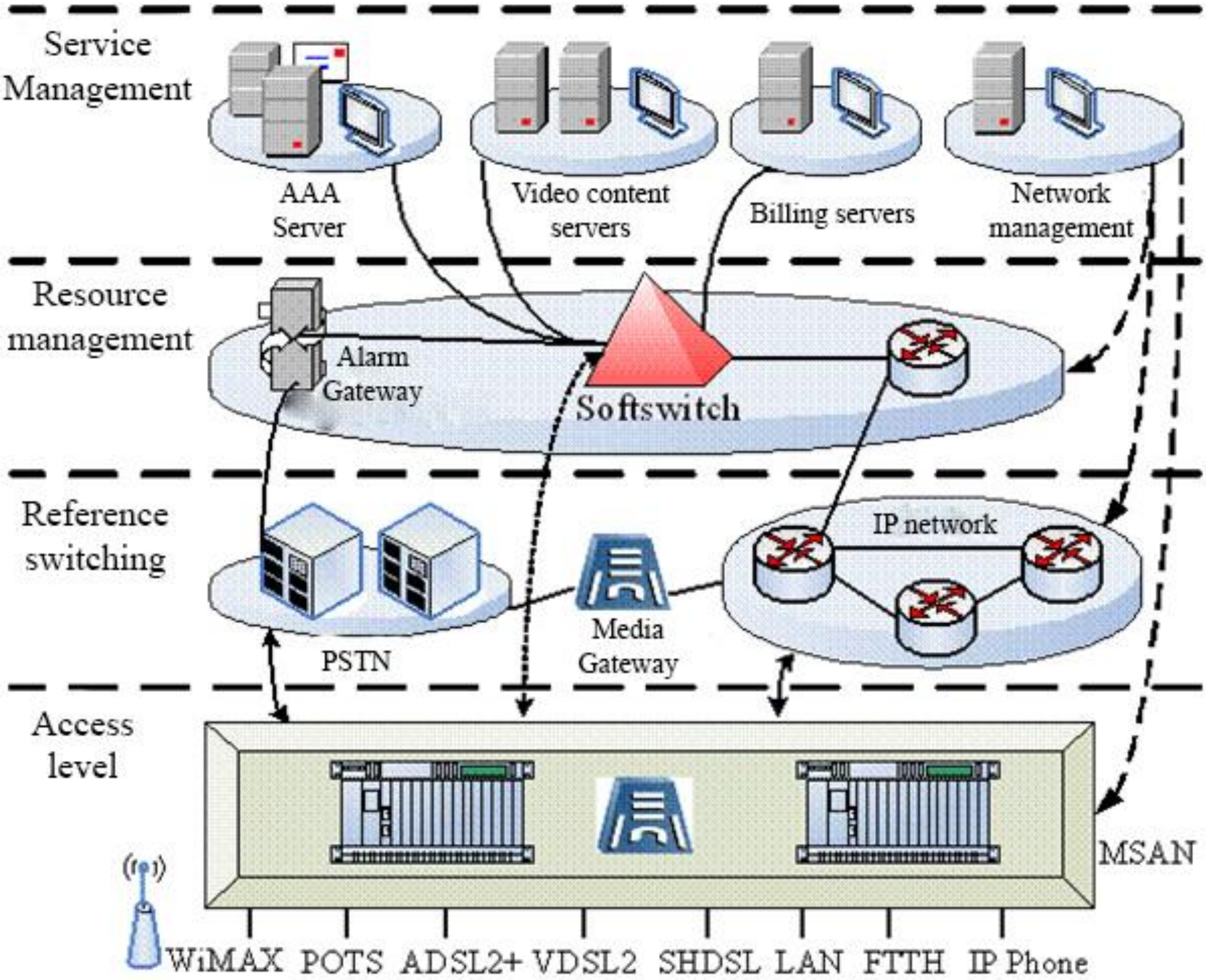


Figure 2.19 - NGN Network Architecture

If we imagine the NGN architecture as a set of planes, then below is the subscriber access plane based on the three transmission media: the metal cable, the fiber and the radio channels, then the switching plane - the switching of the channels and / or the packet switching, and above them the softswitch), which make up the resource management plane [22].

Even higher is the service management plane. NGN networks support one management platform and have a common core for both mobile and fixed communications. As a result, subscribers receive a single set of services: for PSTN, for IP-telephony, and for a mobile network.

The main feature of NGN is the provision of telephone services on the basis of SoftSwitch softswitches.

Softswitch is the central device in the telephone network that connects calls from different telephone lines, solely through software.

Softswitch manages:

- calls;
- Media Gateways (MG);
- distribution of the resources of the backbone network;
- processing of alarm messages;
- Authentication;
- the cost of services;
- provision of voice communication, mobile communication and multimedia communications services to subscribers.

Figure 2.20 shows the implementation model for Softswitch.

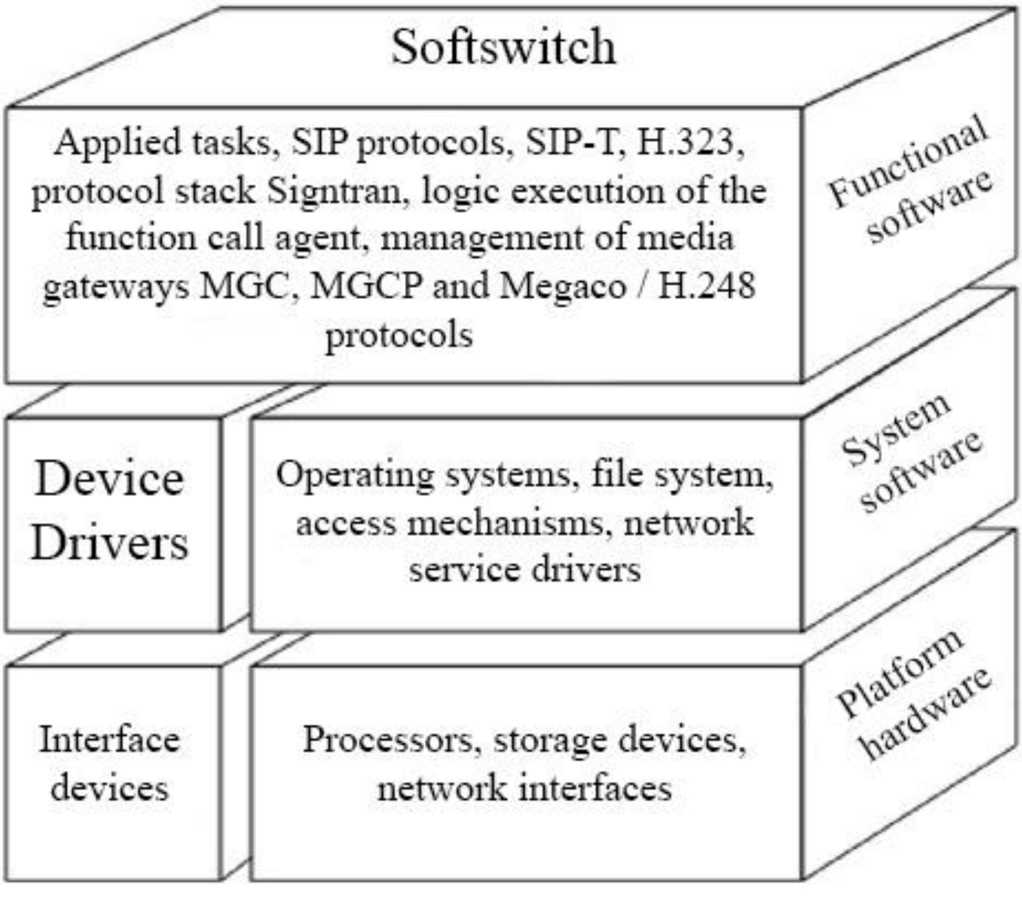


Figure 2.20 - Softswitch implementation model

Softswitch classes:

- class4 - transit Softswitch, for end-to-end transfer of traffic through the upper networks. Has a minimum of functions, high performance and flexible routing;
- class4 - local Softswitch, must support all services of traditional local PBX, as well as additional services to users.

Softswitch, as an IP-PBX, is a software and hardware complex that is designed for registration, management of end and node devices of an IP network. Softswitch also manages the routing of calls and various services (call forwarding, hold, call transfer, conferences, etc.), also monitors the duration of connections, etc. Based on the data received from this hardware and software complex, special software (billing systems) determines the cost of calls and the use of services that the operator provides.

Systems allow you to connect and manage many millions of client devices (IP phones, IP gateways, etc.). On the other hand, Softswitch is simply specialized software installed on the server.

Media gateways are installed on the border of NGN and traditional PSTN and connect VoIP environment and PCM lines or analog telephone lines. Media gateways perform compression and voice packetization, transfer of compressed voice packets to the IP network, and conducts a reverse operation for calling telephone network users from the IP network, and converting all telephone signals ("call", tone dialing, busy) to packet traffic. One softswitch can manage a number of media gateways, including geographically distributed ones. The distances between these two devices can in principle reach thousands of kilometers, although they are trying to optimize their mutual placement, and the work of program switches is duplicated. Media Gateways, managed by the software switch, forward the streams to the access network, converting them into the appropriate format.

The alarm gateway is located on the border between the PSTN and the IP network. The signaling gateway broadcasts signaling information from the PSTN through the packet network to the softswitch or other signaling gateways.

At the backbone switching level, packet switching is performed using routers or Layer 3 IP switches in which processing of packets is performed in hardware. A packet data network with high capacity is usually built using MPLS technology, which provides a flexible, fast and efficient transmission medium. Data of any type (voice, video, security and fire alarm system information, etc.) are delivered to subscribers quickly and with high quality.

At the access level in the NGN network, as a rule, the nodes of multi-service access (Multi Service Access Node, MSAN) are installed. MSANs connect to the IP backbone almost always with an optical cable and very rarely with a copper cable. To connect the IP network to MSAN, in the vast majority of cases, GE technology is used, and very rarely SDH or E1 paths are used for this purpose. MSANs can be interconnected by an optical cable and connected to a backbone IP network using a tree topology, a linear circuit, but more often a more tenacious ring topology is used. The length of an optical cable between nodes of such a network can be from one to 100 km, which is especially important for rural communication networks. MSANs can be installed either in street maintenance cabinets, or in the entrances of residential buildings, or in the form of an IP-gateway directly in the apartment of the subscriber.

Depending on the needs of subscribers, certain blocks are installed in the MSAN enclosure that provide the necessary interfaces, for example:

- POTS interfaces (for connecting a conventional analog phone);
- interfaces ADSL, VDSL, SHDSL for connection of DSL-modems;
- Ethernet interfaces (for connection of local computer networks);
- WiMAX, LTE interfaces (to enable wireless access for mobile users);
- with the help of FTTH interfaces it is possible from MSAN to get the fiber directly to the subscriber's apartment.

3 Architectural solution of the IMS network

3.1 Basic Properties of the IMS Architecture

IMS is a hardware-software complex that is a key component of almost all next-generation IP networks supporting SIP-telephony applications and is intended to provide standardization of multimedia services in all interconnected networks.

Due to the universal architecture, the same IMS platform can be used for applications and services in mobile networks of all generations (2G, 3G, 4G), as well as in fixed networks.

The main properties of the IMS architecture are:

- multi-level - the levels of transport, management and applications are highlighted;
- independence from the access environment - allows operators and service providers to converge fixed and mobile networks;
- support for multimedia personal information interchange in real time (voice and video telephony) and an analogous exchange of information between people and computers (games);
- Integration of real-time and non-real-time multimedia applications (streaming applications and chat rooms);
- the possibility of interaction of various types of services;
- the ability to support multiple services within a single session or the organization of several simultaneous synchronized sessions.

What does IMS do?

- Provision of the required QoS - IMS application can set the QoS class when establishing a session;
- the possibility of tariffication of the service at the discretion of the operator (flat rate, time-based charging, event-based, QoS based), supports charging both online and offline;
- combined services;
- interaction with other networks (stationary and mobile (2G) networks with circuit switching);
- access invariance - assuming the use of any access technology that can provide the transport of IP traffic between user equipment and IMS objects;
- roaming;
- security.

The architecture of the IMS network, which consists of three parts, is shown in figure 3.1:

- the level of application servers (contains a set of application servers that may not already be IMS elements, and includes both multimedia IP applications based on the SIP-telephony protocol and applications implemented in mobile networks based on a virtual home environment) ;

- Session management level (it manages communication sessions);

- the transport layer and subscriber units (organizes a session using the Session Initiation Protocol signaling and provides transport services with the convergence of voice from an analog or digital signal to IP packets using RTP protocol).

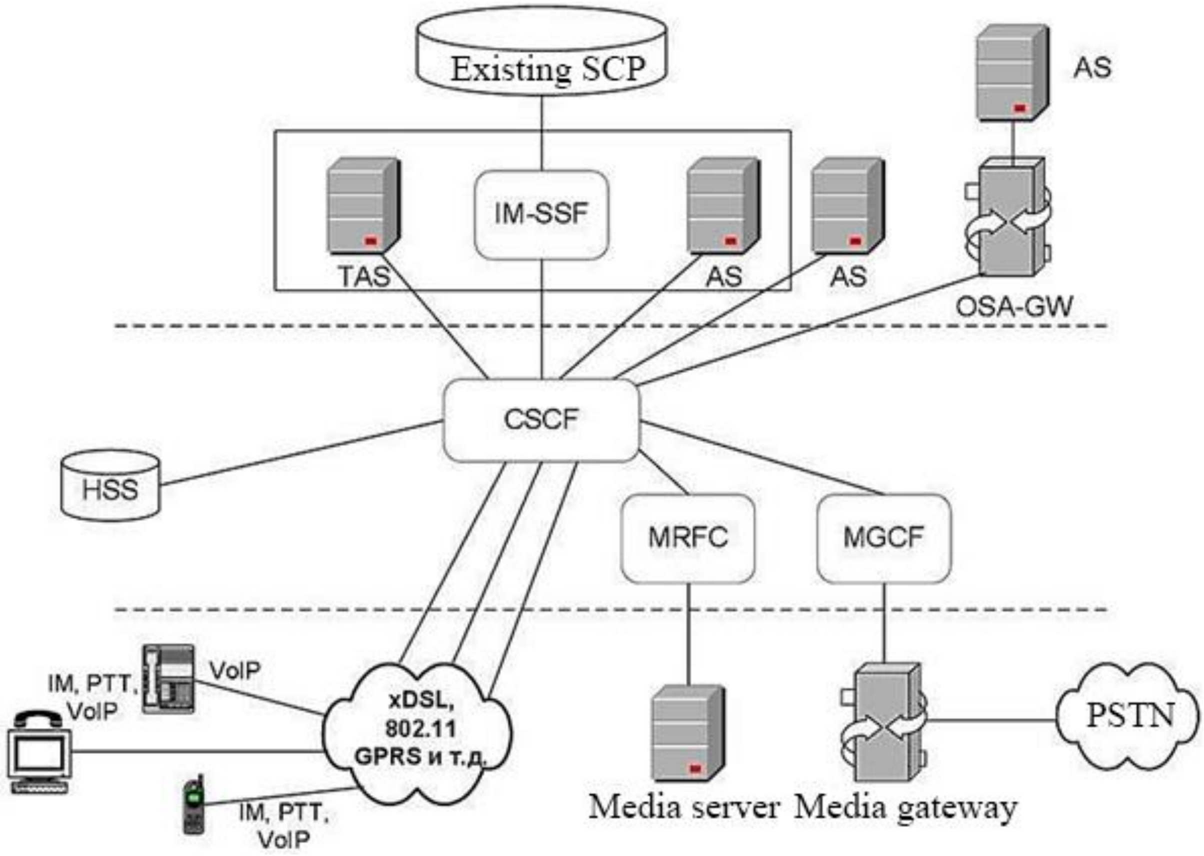


Figure 3.1 - IMS network architecture

A network based on IMS technology supports a variety of application servers that provide both traditional telephone services and new services (instant messaging, instant multi-drop communication, video streaming, multimedia messaging, etc.).

The application server layer performs the following functions:

- application server (AS);
- a telephone application server (TAS);
- functions of service concatenation (IM-SSF);
- gateway to the Parlay API (OSA-GW).

The session management level performs the following functions:

- Session and call management functions (CSCF);
- subscriber data server (HSS);
- media server management functions (MRFC);
- Gateway management functions (MGFC).

The level of transport and subscriber devices performs the following functions:

- Media server (MRFP);
- Media Gateway (MGFP);
- subscriber access.

Network security is implemented at three levels:

- management level - in case of threat of configuration change, unauthorized access, unauthorized actions on the network, access to accounts;
- level of services - in case of threat of opening the network topology, interception of SIP messages, DoS attacks of SIP applications;
- data level - if there is a threat of interception or loss of traffic.

3.2 Advantages of the IMS network

The main advantages of the IMS network are:

- ensuring the interaction of different types of networks;
- the ability to develop and rapidly introduce new services, including VoLTE;
- quality assurance of service provision (QoS);
- accurate invoicing;
- Reduction of operating costs;
- scalability of solutions.

3.3 Comparison of Softswitch and IMS

Softswitch and IMS architectures have a tiered division, and the level boundaries run at the same places. For the Softswitch architecture, the network devices are primarily depicted, and the IMS architecture is defined at the function level. The idea of providing all IP-based services and the separation of call control and switching functions are also identical. The functions of the OSA gateway and the subscriber data server are added to the already known Softswitch functions.

Differences between systems. Evaluating lists of functions in both architectures, you can see that the composition of the functions is practically the same. It could be concluded that both architectures are almost identical. This is true, but only in part: they are identical in the architectural sense. If we analyze the contents of each of the functions, then there will be significant differences in the Softswitch and IMS systems. For example, the function CSCF: from its description already seen the difference from similar functions in Softswitch. In addition, if in the architecture of Softswitch functions have a fairly conditional division and description, then IMS documents give a rigid description of the functions and procedures for their interaction, as well as defined and standardized interfaces between system functions.

The difference begins with the basic concept of systems.

Softswitch is primarily a converged network equipment. The gateway management function (and accordingly MGCP / MEGACO protocols) is dominant in it (SIP protocol for the interaction of two Softswitch / MGC).

IMS was designed within the 3G network, completely based on IP. Its main protocol is SIP, which allows to establish peer-to-peer sessions between subscribers and use IMS only as a system that provides service functions for security, authorization, access to services, etc. The gateway management function and the media gateway itself are only a means for connecting 3G subscribers with fixed network subscribers. And we have in mind only PSTN.

IMS also focuses on IPv6 targeting: many experts believe that the popularity of IMS will serve as an impetus to the protracted implementation of the sixth version of the IP protocol. But for now this presents some problem: UMTS networks support both IPv4 and IPv6, while IMS - usually only IPv6. Therefore, at the entrance to the IMS-network, there must be a gateway that converts the header format and address information. This problem is inherent not only in IMS, but also in all IPv6 networks.

The SIP protocol was developed and specified by the IETF committee, but for use in IMS it was partially modified and modified. As a result, there may be a situation where when receiving SIP requests or sending them to external networks, the sub-function of the S-CSCF can detect the lack of support for the corresponding SIP extensions and / or refuse to establish a connection, and also process it incorrectly.

One of the strengths of the IPCC approach is currently its prevalence: in the world there are many networks that have followed this path of development, and extensive experience has already been accumulated on the implementation of SoftSwitch-architectures. A large number of supported technologies enable the operator to select the equipment that best meets its requirements and allows to interact in the optimal way with existing network resources. SoftSwitch solutions are relatively easy to scale, starting with the simplest architecture serving the corporate sector, and ending with large-scale projects of the inter-regional operator. Thus, the operator can minimize the initial investments in the network of BSC. This same feature allows the operator, creating a large-scale project, to use new network resources (and, therefore, to profit) immediately after their installation. However, the IPCC solution has another side. The variety of equipment presented in this market segment raises the problem of its compatibility. Numerous centers for the provision of system interaction help to solve it only partly, because often the tests do not keep up with the updating of software versions and can not cover all possible combinations of devices operating in the networks of operators. This also generates a broader problem of operator interaction with each other and negates the opportunities provided by many technologies to provide mobility of the user and services. Some equipment manufacturers provide proprietary network management systems that do not always correctly and fully work with third-party equipment when integrating it into the operator's network, since there are differences not only in implementation but also in the functionality of many systems.

In the IMS, the compatibility problems of equipment are partially mitigated, since the interaction of the functional modules is regulated by the standards. A new approach to the provision of services was extremely successful and provided roaming services, which should bring additional profit to the operator. The use of a uniform IMS system in wired networks of the MSE and mobile networks of 3G allows to see in the future the possibility.

4 FMC networks

4.1 Development trends of broadband wireless communication

Telecommunication technologies have recently migrated in two directions:

- from voice transmission services, to the services of transmission of large information flows over high-speed channels;
- from the stationary subscriber to the mobile, which can only be achieved with wireless communication.

Broadband wireless access networks combine these two areas of development of telecommunication technologies.

Figure 4.1 shows the development trends of broadband wireless communication.

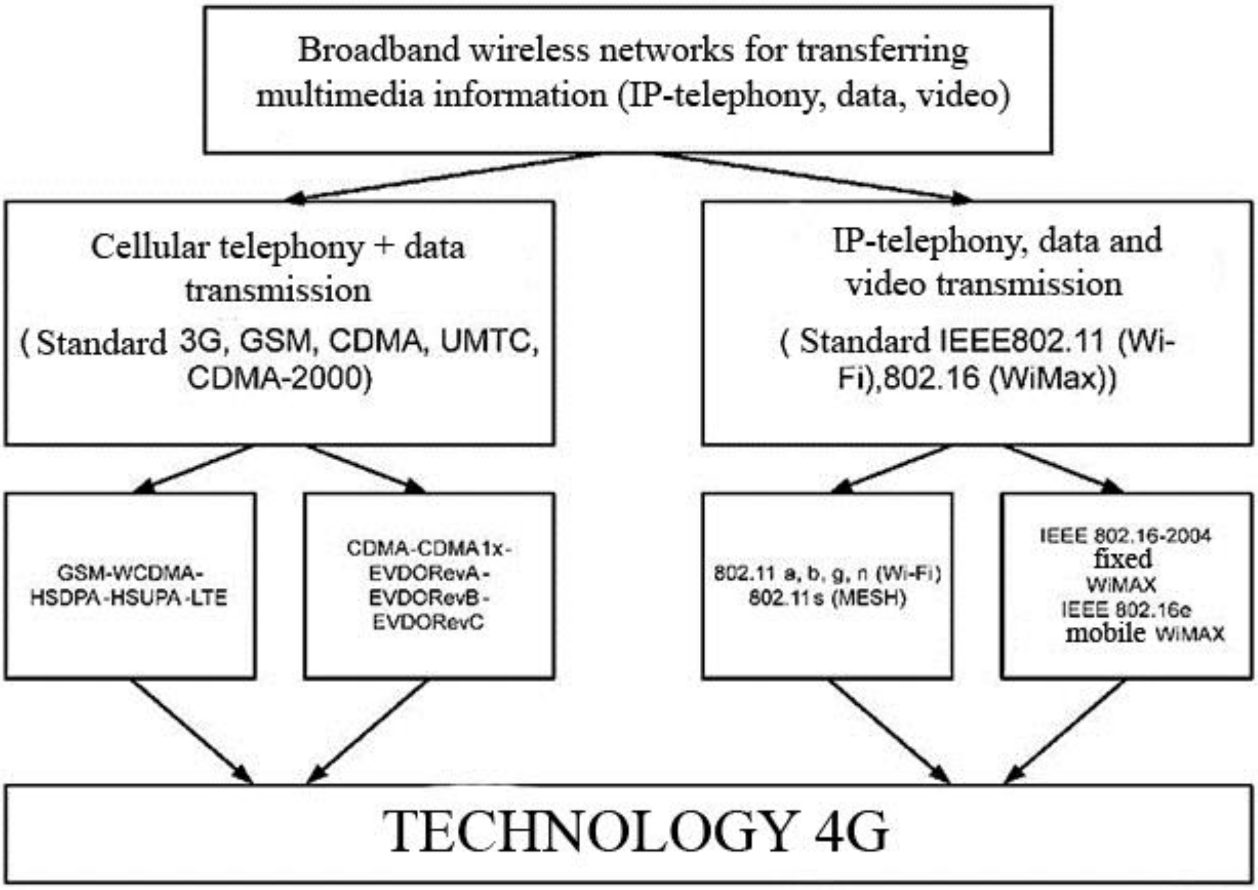


Figure 4.1 - Directions for the development of broadband wireless communication

4.2 About FMC technologies

FMC (Fixed Mobile Convergence) allows you to connect two types of communication networks:

- fixed

- mobile with a general plan for short numbering that allows the company's employees to communicate with each other via intracorporate short numbers in the access zone of any of these networks.

That is the idea of FMC is to combine fixed office and mobile phones into a single intracorporate network with a single numbering plan. An employee can call a short number from an office phone to another person's mobile phone, which also has a short number, and an employee who has a mobile phone connected to the FMC can also call an office phone using his short number. There is no need, on the one hand, to dial federal mobile numbers and pay for a call at general rates, on the other hand, dial directly to a company employee bypassing secretaries or voice menus while dialing four or five-digit numbers.

In a single plan of corporate numbering, office phones "know" about short numbers for which mobile numbers are available, and mobile phones "know" that office numbers are available for short numbers. Moreover, even being in roaming, a mobile subscriber using FMC can dial up to any internal number as well as in the home region - simply dialing a short number.

Operators provide subscribers with a personal web-cabinet through which you can manage the FMC service. For example, you can configure devices for reditection, depending on the time, which numbers and in which order to call (figure 4.2).

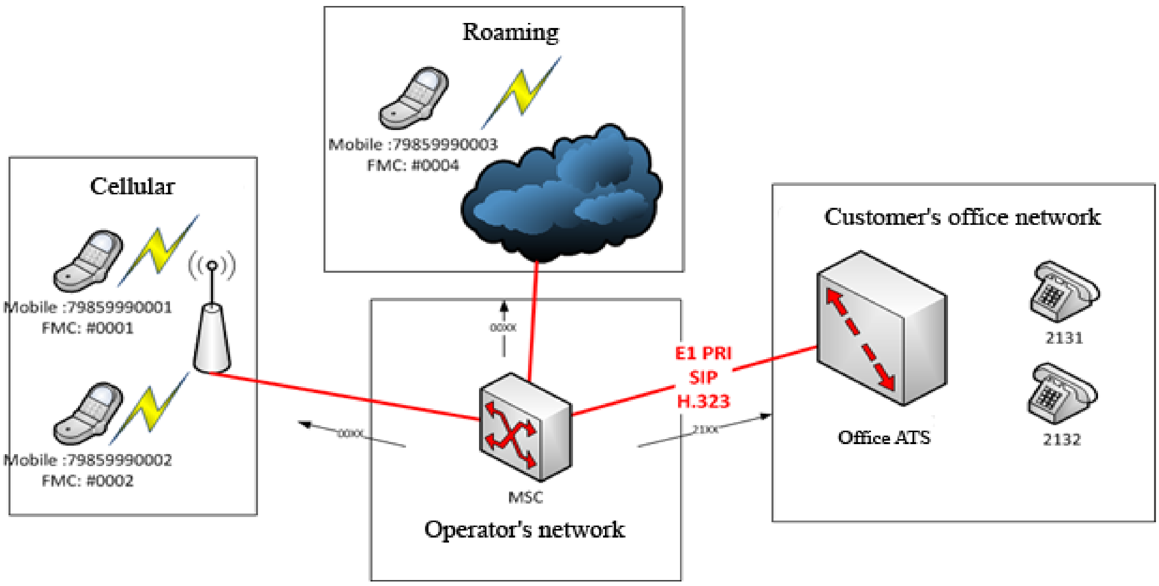


Figure 4.2 - Example of implementing the FMC network

4.3 FMC Network Architecture

FMC provides convergence of fixed and mobile networks. Therefore, the concept of NGN is classified as a concept in relation to the concept of FMC, because of FMC is based on IMS.

The concepts of convergence and FMC are given in the ITU-T - Telecommunication Standardization Sector Study Period 2005-2008. NGN GSI Rapporteur Group Meeting: Geneva, 12-22 May 2008, which addressed the implementation of roaming users when providing them with basic or additional services, as well as the network architecture for the implementation of FMC, based on IMS.

Convergence is the coordinated evolution of formally separate networks to unity in supporting the provision of services and applications.

FMC - the use of wired or wireless access technologies in conjunction with a backbone network based on IMS.

The IMS platform will provide for FMC:

- access to basic IMS services with different terminal identifiers;
- continuity of the service at the moment of movement between the zone created by the fixed communication equipment and the mobile communication network.

The purpose of FMC is to provide users with the same services in environments - fixed and mobile.

Figure 4.3 shows the architecture of the FMC network.

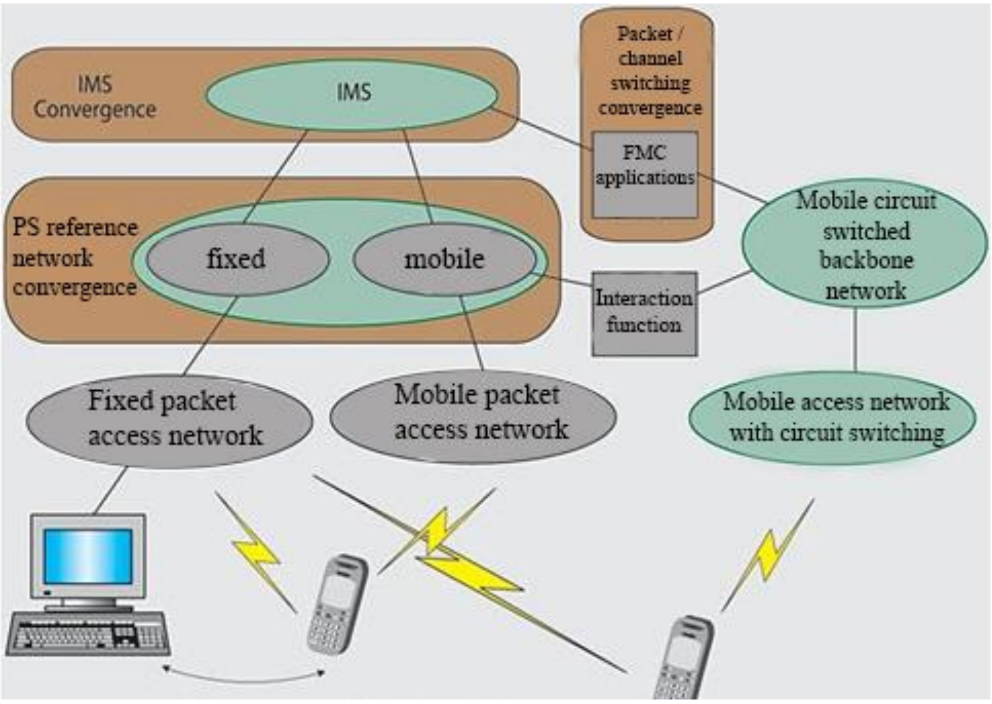


Figure 4.3 - FMC Network Architecture

The architecture of the FMC network consists of three main levels:

- network convergence;
- convergence of services;
- application convergence.

At the network convergence level, operational costs are reduced by converging various fixed and mobile networks into a single IP / MPLS backbone network that supports a wide range of access methods: traditional telephony, DSL, dedicated channels, Metro Ethernet, wireless networks (WLANs) and radio access networks (RAN) in the networks of mobile operators. Convergence of the backbone and access networks is the most obvious and well-developed step in the process of merging fixed and mobile platforms. This concept also covers the convergence of the fixed and mobile networks, including the transfer of significant amounts of voice traffic over the same IP backbone, over which broadband data, GPRS and UMTS services are delivered, the so-called transfer of transit traffic to the IP network. For mobile operators, converged networks usually begin by translating SMS and MMS traffic from traditional platforms and a signaling network to an IP network - this speeds up the convergence of signaling protocols from IP. When 2.5G and 3G radio access network traffic is transmitted through an optimized IP access network, the network convergence provides a penetration depth up to the access network of the mobile operator.

At the convergence level of services, session management functions are performed. It is this level that makes it possible to deploy high-profit, next-generation IP-based services, such as mobile data access, audio and video conferencing, voice and instant messaging. Awareness of and control over each session ensures the availability of the service at any subscriber terminal and through any access method, allowing you to switch between different types of access without adversely affecting active sessions. In addition, it is the level of convergence of services that ensures that any IP service is allocated appropriate network resources, and any service is properly charged.

One of the main indicators of the functionality of the converged network is to ensure the continuity of the service, in case of crossing the boundaries of fixed and mobile networks. The concept of service continuity is quite specific for each of the areas - voice, data transmission and multimedia traffic. However, technologies such as converged voice devices (phones, smartphones, PDAs, laptops, etc.), voice convergence architectures (for example, UMA or IMS) and data convergence protocols (in particular, Mobile IP) are the connecting link between fixed and mobile platforms. Another key element is the awareness of the platform through which the service is delivered, the sessions being transferred, and its ability to perform specific policy enforcement actions, regardless of the location of the session participants and their access method, whether wired through xDSL or mobile data in the UMTS environment. Service convergence is the fundamental level that ultimately provides consumers with the convenience of using services, carrying out data and voice data transients unnoticed for subscribers between the terrestrial and wireless broadband domains. At the same time, the network dynamically adapts its policies for allocating resources and ensuring quality of service, taking into account the fact of the terminal's mobility and in what transmission environment the terminal is at the moment.

The level of convergence of applications includes the services themselves, with which operators enter the market and which they are going to advertise as the end product. In particular, continuous data services provided through any access network, voice services for enterprises with dual-mode terminals (eg, Wi-Fi / GSM), etc. Application convergence is the process of delivering applications across many different media in a format, considering the difference in access speeds that these environments provide. The domain of convergent applications is supported and provided mainly by the functionality of the SIP protocol, which takes into account the mobility of subscribers and the dynamism of their state (registration) on the respective servers. As one example of convergent applications, simultaneous delivery of a video stream to a 3G terminal and a personal computer via a content distribution network from the same service center can be called. More generally, application convergence is the provision of voice, data and video services to consumers through all available types of networks using innovative methods (WiFi, WiMax, WiBro, HSDPA / HSUPA). The implementation of each of the considered levels provides significant advantages. Network convergence creates opportunities to save operating costs and capital costs, application convergence - to offer new service packages and improve marketing.

4.4 SoftSwitch Reference Architecture

According to the reference architecture developed by the International Packet Communication Consortium (IPCC), SoftSwitch is a technological solution that provides hierarchical execution of hierarchically different functions in four functional planes:

- transport,
- management of call and alarm services;
- services and applications;
- operational management (figure 4.4).

The implementation of the "FMC business model is best implemented using the NGN concept on the IMS technology platform using SoftSwitch technology." This model is focused on the interaction of networks (roaming), the performance of mutual settlements between different networks of providers (billing), as well as accounting for the place of registration of subscribers in the home and in the guest network. Moreover, along with the functions of calculating the cost of services in the business model of any operator, the function of assessing the quality of the services provided must necessarily be included.

The services that are provided within the framework of the NGN concept have their own peculiarities:

- multimedia, a wide variety of types of services;
- use of several broadband transportation technologies;
- universal mobility of the service user (handover between access subnets), networks of which are built on different technologies.

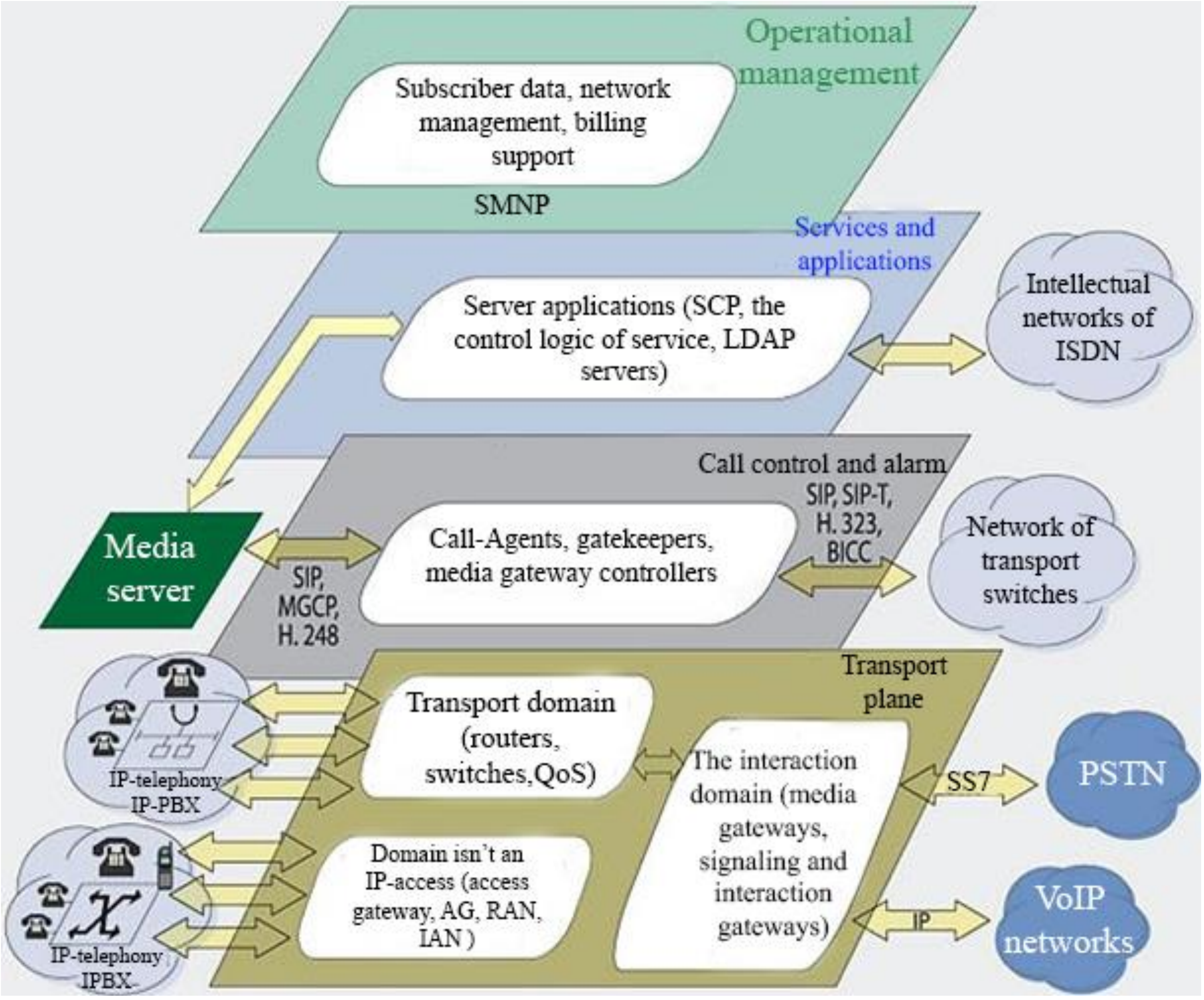


Figure 4.4 - SoftSwitch Reference Architecture

5 Quality of service in IP networks

5.1 Network third level device

Packet switching is used to transmit variable-speed, throttling traffic and is not sensitive to delays (sending text documents, browsing Web pages). At the same time, there is no guarantee of throughput, it is difficult to transmit real-time streaming traffic (speech, video) due to the occurrence of variable delays. Packet switched networks can operate in datagram mode or virtual channel mode. The datagram method of data transmission is based on the fact that all transmitted packets are processed independently of each other.

The mechanism of virtual channels takes into account the existence of data streams on the network and makes a single route for all packets of the stream.

The switch (swinch) is a network device (specialized, universal computer with the built-in software switching mechanism) of the second level of the network model, capable of performing data switching operations from one interface to another (figure 5.1). The switch monitors the network traffic and controls its traffic, analyzes the destination addresses of each frame [23].

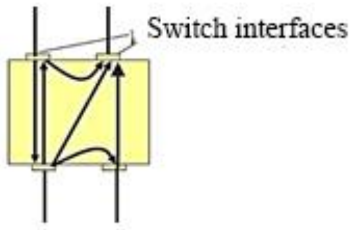


Figure 5.1 - The Switch

A router is a network device of the third level of the network model, based on the information about the network topology and certain rules, decide on forwarding network layer packets between different network segments. Calculation of the route and processing of packets is completely carried out by the network-level programs of the OSI model (figure 5.2).

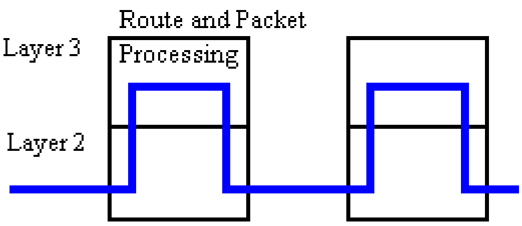


Figure 5.2 – Router

A router can be a specialized (hardware) device, or a computer that performs the functions of a router.

5.2 Basic Models and Mechanisms of Ensuring Quality of Service in Networks

Methods of QoS quality assurance occupy an important place in the technologies of packet-switched networks. Ensure the stable operation of modern multimedia applications, IP telephony, interactive distance learning, etc. QoS methods are aimed at improving performance and reliability of the network, reducing delays, variations in latency and packet loss. The attributes of packet-switched networks are:

- queues;
- Buffering of packets;
- Delay and packet loss.

The mechanisms for ensuring the quality of service are:

- network with excessive bandwidth;
- Queue management technique;
- methods of traffic condensation;
- methods of feedback;
- Reservation of network resources;
- methods of engineering of traffic;
- reducing the constant load on the network.

The main criteria for classification of traffic are:

- relative predictability of data transmission speed;
- traffic sensitivity to packet delays;
- the sensitivity of traffic to the losses and distortions of packets.

Predictability of data transmission speed (figure 5.3):

- applications with streaming traffic generate a uniform data stream that is delivered to a network with a constant bit rate (CBR);
- applications with pulsating traffic are highly unpredictable, traffic is characterized by variable bit rate (VBR).

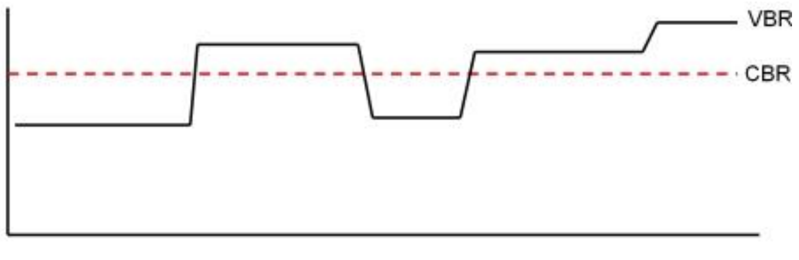


Figure 5.3 – CBR and VBR

Almost any traffic has a non-zero pulsation coefficient. The values of the pulsation coefficients of streaming and pulsating traffic differ significantly. Classification of applications by traffic type:

- asynchronous applications;
- interactive applications;
- isochronous applications;
- extremely sensitive to delays applications.

Classification of applications to packet loss and distortion:
- applications that are sensitive to data loss;
- applications that are resistant to data loss.

Table 5.1 lists the application classes.

Table 5.1 - Application classes

Class	Characteristics
A	Constant bit rate, sensitivity to delays, transmission with connection establishment (voice traffic). Parameters QoS: peak data rate, delay, jitter.
B	Variable bit rate, sensitivity to delays, transfer with connection establishment (compressed video). QoS parameters: peak data rate, ripple, average data rate, delay, jitter.
C	Variable bit rate, elasticity, transmission with connection establishment (computer network traffic). Parameters QoS: peak data rate, ripple, average data rate.
D	Variable bit rate, elasticity, transfer without connection (traffic of computer networks). QoS parameters: not defined.
X	The type of traffic and its parameters are determined by the user.

The user formulates his requirements to the quality of service in the form of certain limiting values of QoS characteristics, which should not be exceeded. The overall performance of each resource must be divided among the different traffic classes unevenly. If the cause of the overload is the insufficient performance of the processing unit of the network device, the unprocessed packets are temporarily accumulated in the input queue of the corresponding input interface. There can be several queues. If the cause of the overload is the limited bandwidth of the output interface, the packets are stored in the output queue of the interface. In the FIFO queue, in the event of an overload, all packets are placed in one common queue and are selected from it in the same order as they were received. It differs in its simplicity of implementation and in the impossibility of differentiated processing of packets of various flows.

Priority service is as follows:
- priority queues;
- traffic classification is carried out.

Traffic conditioning mechanisms control the current parameters of traffic flows, such as its average speed and ripple. The main principle of the routing protocols in packet-switched networks is the choice of a route based on the network topology without taking into account information about its current download.

Methods of engineering of traffic determine:

- characteristics of the transmission network - topology;
- information on the proposed network load.

Modified routing protocols should propagate over the network information about two parameters of free ability - for each traffic class separately. If the problem is generalized for the case of transmission of several classes of traffic across the network, then there must be as many counters associated with each resource as there are traffic classes in the network. The routing protocols must propagate the vector of free bandwidths of the corresponding dimension.

Network operation in underload mode assumes constant availability of redundant bandwidth (figure 5.4).

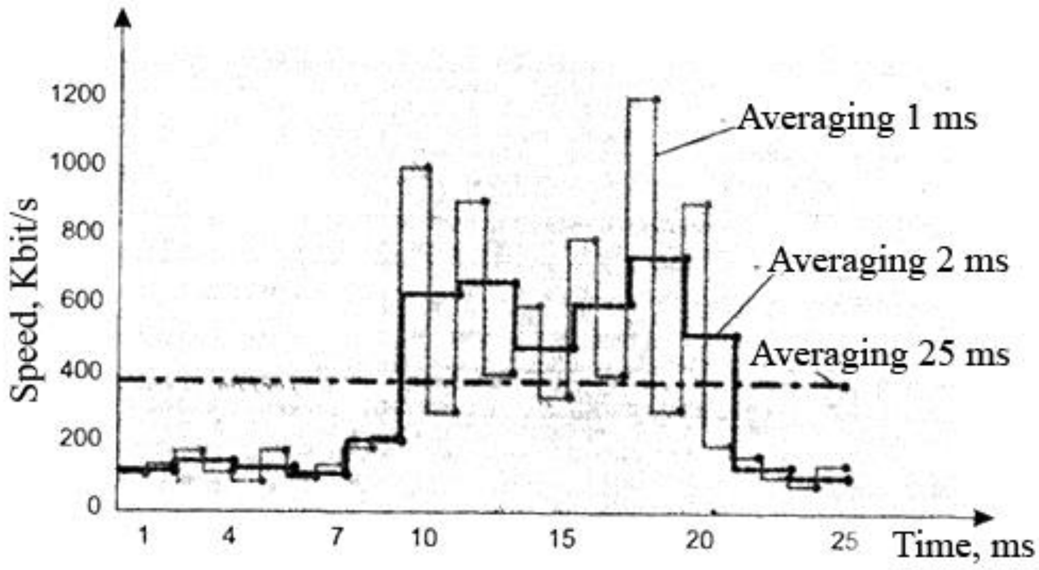


Figure 5.4 - Traffic averaging

The main QoS models are:

- Best effort - processing information as quickly as possible, but without additional effort (FIFO, drop tail). Allows the occurrence of overloads in the event of sudden surges of traffic;
- Soft QoS - service with priority maintenance, QoS parameters values depend on traffic characteristics. Relatively small number of classes of service;
- Hard QoS - guaranteed service. It is based on the preliminary reservation of resources for each thread.

5.3 Quality of Service evaluation in VoIP networks

In IP-telephony technology, the process of information transfer occurs in real time, where the dynamics of signal transmission is important, which is provided by methods of encoding and transmission of information. The main factors that affect the quality of IP telephony are the quality of the IP network (maximum throughput,

latency, jitter and packet loss) and the quality of the gateway (required bandwidth, delay, jitter buffer volume, packet loss, and echo cancellation function).
In IP networks for voice transmission, the UDP protocol is used. The loss of a small number of voice packets is considered acceptable and can be compensated by the encoding / decoding mechanism and various methods of interpolating speech, that is, by filling out missing sounds with the help of a DSP technology that analyzes the shape of the sound oscillation and predicts the missing sound. Lost packets in IP telephony violate speech and create distortion of the timbre. Losses of up to 5% of packages are permissible, and over 10-15% are unacceptable.
The primary criterion for the quality of audio and video information is the perception of the quality of the service by the user. There are subjective and objective assessments. The average subjective assessment of MOS (Mean Opinion Score) was developed by ITU-T and is based on a five-point scale: excellent (5), good (4), acceptable (3), poor (2), and unacceptable (1).
Score of 3.5 points and above corresponds to high (standard) speech quality, 3.0 ... 3.5 - acceptable, 2.5 ... 3.0 - synthesized sound. For the transmission of speech with good quality, it is advisable to focus on the MOS score of at least 3.5 points.

5.4 Delay and mitigation measures

The ITU-T G.114 standard recommends delay in the transmission of voice in one direction, which should not exceed 150 ms. A delay of 150 to 400 ms is acceptable if the speaker and listener understand the delay and are ready to accept it. If the delay reaches 400 ms or more, it becomes noticeable and is considered unacceptable. The delay in communicating through the satellite in one direction is approximately 170 ms without taking into account the delay that occurs in devices located on the ground.

The resulting delays in the IP network are changed randomly. This fact is a problem in itself, but it also exacerbates the echo problem. Classification of delay:

- algorithmic delay (the collection of frames of speech samples by an encoder), depends on the type of codec and varies from 0.125 μs to ms;
- delay processing (encoding and collection of encoded samples in packets) depends on the speed of the processor and the type of processing algorithm;
- network delay, which depends on the physical environment and protocols used in the transmission of voice data, as well as the buffers used to remove packet jitter at the receiving end.

The voice gateway delays when it processes signal transformations coming from the subscriber (figure 5.5):

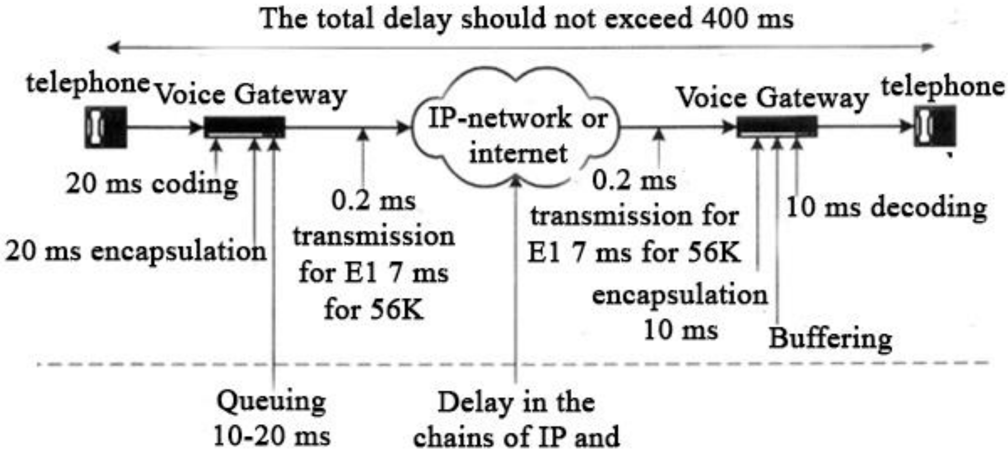


Figure 5.5 - IP network delays in speech processing

- the maximum time of digitization is up to 20 ms;
- the maximum encapsulation time is 10 ms;
- packet service by the channel is from 10 to 20 ms.

The delay of message packets on the network depends on the speed of information transfer. The higher the speed, the less the delay. When packets arrive, the voice delay gateway is 30 ms for 10 ms for buffering, encapsulation for decompression and decoding.

Classification by time levels of delays in the transmission of speech:

- up to 200 ms - excellent communication quality;
- up to 400 ms - good quality of communication;
- up to 700 ms - acceptable quality of communication in the conduct of non-business negotiations (transmission of packages over satellite communication).

Delays in IP networks also have an impact on telephone signaling, which can lead to disturbances in the functioning of the alarm system when it interacts with the telephone network.

5.5 Jitter and the causes of its emergency

When voice data is broken into packets, have a certain constant speed, after the corresponding processing during the advancement over the IP network, packets often arrive at the destination at different times and with different sequences. This creates a spread in packet delivery time-jitter, which results in specific speech impairments (perceived as cracking and clicking).

Types of jitter:

- jitter dependent on the DDJ data appears in case of a limited bandwidth or when there are violations in network devices;
- distortion of the working cycle of DCD due to propagation delays between the upward and downward transmission of the interaction of open systems;
- RJ random jitter is the result of thermal noise.

The reasons for the occurrence of jitter are as follows:

influence of the network. Depends on the load of this network. With a small load, SSs handle speech packets almost instantly. When overloaded, packets are waiting in line for service. The more SS and lines in the route through which the packet passes, the longer its time lag and the greater the variation of this time, i.e., jitter;

- the impact of the operating system (OS). IP-telephony client applications are programs that run on Windows or Linux that access the UE (voice processing boards, dedicated alarm cards) through the application interfaces to interact with the drivers of these devices, and access to the IP network through logical ports. In this case, the OS can not control the distribution of CPU time between different processes with an accuracy exceeding several tens of ms, and can not process more than one interrupt from external devices in the same time. This leads to the fact that the lag in the progress of data between the network interface and the external voice output device makes, regardless of the used speech-processing algorithm, a value of the same order or even more. Consequently, the choice of OS is an important factor. To this end, gateway and IP phone manufacturers use the real-time OS (VxWorks, pSOS, QNX Neutrino, etc.) with sophisticated processor time sharing mechanisms to ensure rapid response to interrupts and more efficient exchange of data flows between processes. In the Dialogic, Audiocodes, Natural Microsystems for VoIP applications, a different approach is used: all functions, necessary performances in a rigid time frame (data exchange between voice codecs and the network interface, RTP support, etc.) are transferred to a specialized processor;

- influence of jitter buffer. This problem exists only in packet networks. The transmission of voice packets to the network is carried out at fixed intervals, but when passing through the network, packet delays are not the same. To compensate for the effect of jitter, devices use an optimized jitter buffer that stores the incoming packets in memory for a time determined by its volume;

- the effect of the codec and the number of frames transmitted in the packet. Codecs transmit information frames - therefore, it takes time to accumulate a certain frame length by counts. There are codecs that perform preliminary analysis of more voice information than should be contained in the frame included in the total budget of the packet delay duration. Due to the significant amount of overhead that is transmitted in RTP / UDP / IP packets, the transmission of short-length frames is inefficient, so if you use codecs with a short frame length, they are packed several frames into one packet.

5.6 Echoes, devices to limit its influence

Echo, this is a voice leak from the transmission path to the reception path.

The phenomenon of the echo causes difficulties in the conversation between the speaker and the listener. The speaker hears with a certain delay his own voice that the cause of the audible echo is the downside. If the signal is reflected twice, the listener hears the speaker's speech twice (the second time - with a weakening and a delay).

The echo can be electrical and acoustic in nature.

There are echoes of the speaker (the user speaking on the phone and hears his own voice) and the echo of the listener (the user hears the interlocutor's voice twice). To some extent, the echo is always present. In case the delay is low, as it happens when making inter-city calls from a fixed telephone, it can not be recognized even if the echo effect is present. If the propagation delay of the signal in the network is small, as in traditional telephony, then such a reflected signal is invisible and does not cause unpleasant sensations. If the delay is 15-20 ms - the effect of a "huge empty room" occurs. With a further increase in the delay, the MOS score deteriorates sharply, up to the total impossibility of extending the dialogue. A delay of 50 to 300 ms is observed in the IP-telephony of the cellular network and in the satellite network. In this case, the user does not feel a delay unless he has a visual contact with the opposite side. For these delays, as a rule, it is sufficient to use a simple echo suppression system. A delay of 300 to 800 ms takes place when several networks are docked, with large transport delays (many jumping satellite paths). At the same time, communication is difficult, since the reverse side hears the interlocutor's voice with a long delay. In such systems, it is necessary to have complex and costly echo suppression systems.

Leaks from the transmitting part to the reception room are acoustic and electric. Acoustic echoes appear when using speakerphones. Electrical leakage is possible when switching from digital trunks to analog trunks and using differential systems.

6 Application of neural networks in telecommunication systems

6.1 The main directions of neural networks application

The main fields of application of neural networks are:
- automation of the classification process;
- forecasting automation;
- automation of the recognition process;
- automation of the decision-making process;
- management, encoding and decoding of information;
- approximation of dependencies, etc.

Using neural networks successfully solved an important task in the field of telecommunications – design and optimization of communication networks (finding the optimal path for traffic between nodes). In addition to managing the flow routing, neural networks are used to obtain efficient solutions in the design of new telecommunications networks. Speech recognition is one of the most popular applications of neural networks.

6.2 Neural networks in telecommunications

Modern digital computers are superior to humans in their ability to produce numeric and symbolic computations. However, a human can easily solve complex tasks of perceiving external data (for example, to recognize in the crowd of a friend only through his flashed face) with such speed and accuracy that the most powerful computer in the world will be hopeless in comparison with him. The reason for such a significant difference in their performance is that the architecture of the biological neural system is completely different from the von Neumann architecture (Table 6.1), and this significantly affects the types of functions that are more efficiently executed by each model. The advantages of the neural network approach are the following [24]:
- parallelism of information processing;
- a unified and effective learning principle;
- reliability of operation;
- the ability to solve unformalized tasks.

Artificial neural networks (ANNs), like biological ones, are a computer system with a huge number of parallel, simple processors with many connections. Despite the fact that when constructing such networks, assumptions and simplifications are made that distinguish them from biological analogs, ANN demonstrate an amazing number of properties inherent in the brain: it is learning from experience, generalization, extraction of essential data from redundant information.

Neural networks can change their behavior depending on the state of their environment. After analyzing the input signals (possibly together with the required output signals), they are self-tuning and trained to ensure the correct reaction.

Trained network can be resistant to some deviations in input data, which allows it to correctly "see" an image containing various interference and distortions.

Table 6.1 - von Neumann machine in comparison with the biological neural system

Comparison Options	von Neumann machine	Biological neural system
Processor	Complicated	Simple
	High speed	Low-speed
	One or more	A large number
Memory	Separated from the processor	Integrated into the processor
	Localized	Distributed
	Addressing not by content	Addressing by content
Calculations	Centralized	Distributed
	Consistent	Parallel
	Stored programs	Self-study
Specialization	Numeric and symbolic operations	Perception problems
Operating environment	Strictly defined	Poorly defined
	Strictly defined	Without restrictions

Today, there are a large number of different configurations of neural networks with different principles of functioning. As an example, consider a multilayered, fully connected neural network of direct distribution (figure 6.1), which is widely used for searching patterns and classifying images. A fully-connected neural network is a multilayer structure in which each neuron of an arbitrary layer is associated with all the neurons of the previous layer, and in the case of the first layer with all inputs of the neural network. Direct signal distribution means that such a neural network does not contain loops.

The ability to learn is the main property of the brain. For ANN, training is understood as the process of configuring the network architecture (the structure of connections between neurons) and the weights of the synaptic connections (affecting the coefficients signals) for the effective solution of the task.

Usually, the training of a neural network is performed on a certain sample. As the learning process, which takes place according to some algorithm, the network should respond better and better (more correctly) to the input signals.

There are three ways of teaching: with the teacher, self-study and blended. There are three ways of teaching: with the teacher, self-study and mixed. In the first method, the correct answers to each input example are known, and the weights are adjusted so as to minimize the error. Teaching without a teacher allows you to distribute the samples into categories by revealing the internal structure and nature of the data. Blended learning combines the above two approaches.

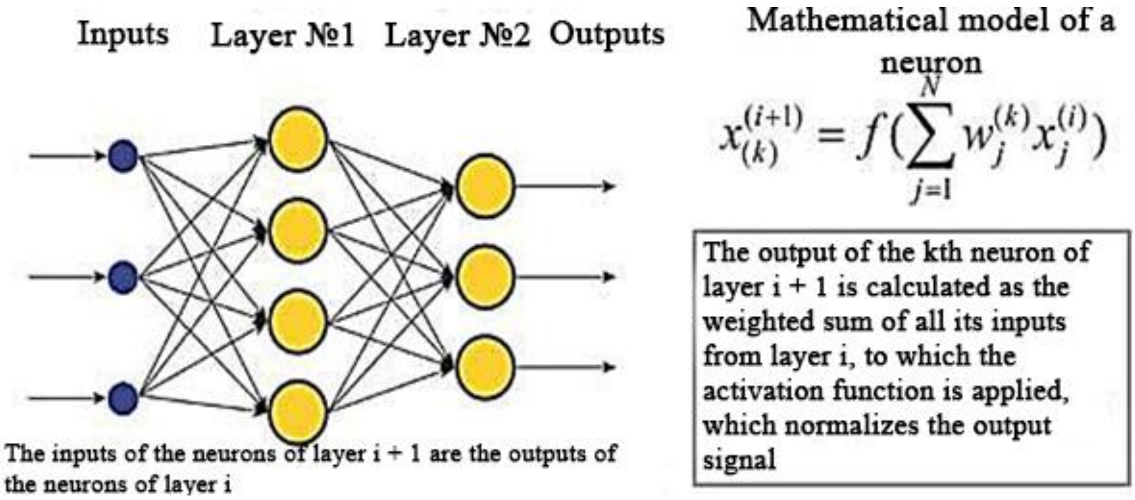


Figure 6.1 – Example of a multilayer fully connected neural network of direct signal distribution

There are a large number of learning algorithms that are focused on solving different tasks. Among them, there is an algorithm for back propagation of the error, which is one of the most successful modern algorithms. Its main idea is that the change in the weights of the synapses takes into account the local gradient of the error function.

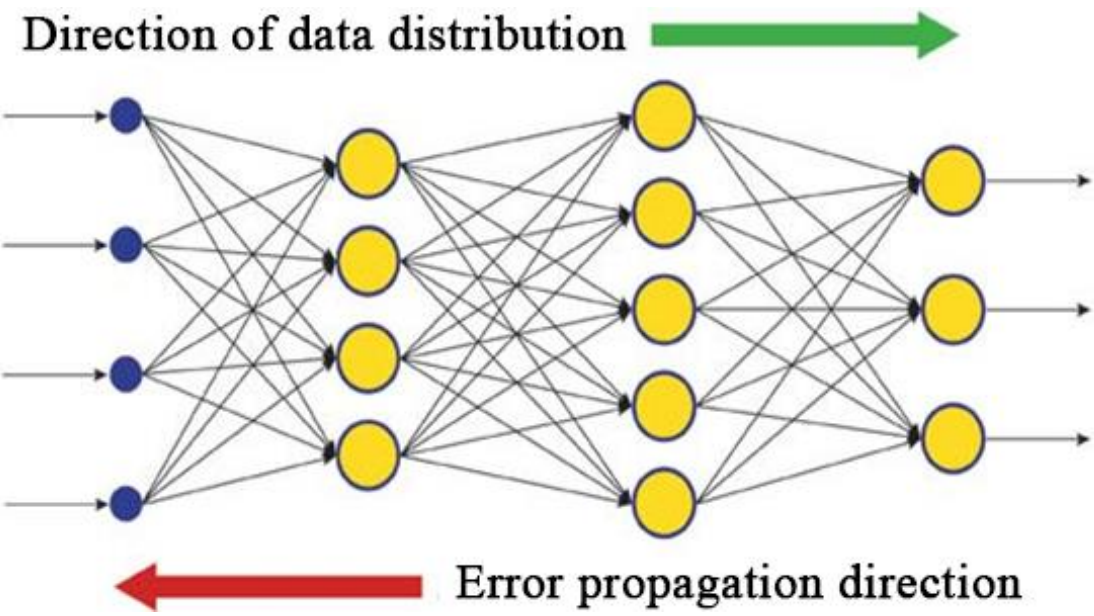


Figure 6.2 - Back propagation method for a multilayer, fully connected neural network

The difference between the real and correct answers of the neural network determined in the output layer, is propagated in the opposite direction (figure 6.2) –

upstream signals. As a result, each neuron is able to determine the contribution of each of its weight to the total network error. The simplest learning rule corresponds to the method of steepest descent, that is, changes in the synaptic weights are proportional to their contribution to the general error.

6.3 Application of artificial neural networks in TCS

Areas of application of neural networks are very diverse: text and speech recognition, semantic search, expert systems and decision support systems, prediction of stock prices, security systems, text analysis. With regard to telecommunications networks, there are four areas of application of ICS:

- switch management;
- routing;
- traffic management;
- channel allocation in mobile radio communication systems.

Routing is one of the important tasks for modern networks. The tasks connected with the choice of the route, the planning of the operation of communication equipment, etc., belong to the class of complex combinatorial-optimization problems, usually not having simple analytical solutions. In addition, the complexity of the necessary calculations exponentially increases with increasing number of nodes in the network. Therefore, at the present time, various heuristic algorithms and procedures obtained through the creative search, intuition and experience of the researcher are widely used. An alternative to existing methods of solving routing tasks is the use of neural network models, which, with a significant reduction in time, provide good suboptimal solutions. Thus, to solve combinatorial optimization tasks, models based on the Hopfield NS are widely used, first applied to solve the traveling salesman problem. These models were the beginning of the development of neural methods for solving complex optimization problems. When solving the traveling salesman problem, a certain group of cities with known distances between them is required to find the shortest route of visiting each city once with returning to the starting point. Cities are denoted by the letters A, B, C…, and the distances are – d_{AB}, d_{AC}…,d_{BC}… the solution is an ordered set of n cities. The sequence in which cities are bypassed is conveniently represented by a matrix nxn, the rows of which correspond to cities, and the columns of cities in sequence.

Thus, city C is visited first, city A - second, etc. The length of the route is $d_{CA}+ d_{AE}+\ldots+d_{DC}$. In each column and in each row of this matrix there can be only one unit, since at each moment only one city is visited and each city is visited only once.

$$
\begin{array}{cccccc}
 & 1 & 2 & 3 & 4 & 5 \\
A & 0 & 1 & 0 & 0 & 0 \\
B & 0 & 0 & 0 & 1 & 0 \\
C & 1 & 0 & 0 & 0 & 0 \\
D & 0 & 0 & 0 & 0 & 1 \\
E & 0 & 0 & 1 & 0 & 0
\end{array}
\qquad (6.1)
$$

A matrix of the form (6.1) can be perceived as a state of a neural network of $N = n^2$ neurons. The problem is to choose from $n! / 2n$ routes one with the shortest length. The state of each neuron is described by two indices, which correspond to the city and the ordinal number of its visit in the route.

List of abbreviations

ACS - automated control systems
ATS – automatic telephone station
DSC - dedicated signal channel
GII - global information infrastructure
CTN - city telephone network
ICS - infocommunication systems
ICT - infocommunication technologies
ANN - artificial neural networks
TSS - telecommunication standardization sector
OS - operating system
PC - personal computer
SLIC - super large integrated circuits
DTN - documentary telecommunication networks
MM - mass media
PTN - public telephone network
MN - mobile networks
RTN - rural telephone network
TV – television
TCS - telecommunication systems
PSTN - public switched telephone network
DTS - digital transmission systems
CBR - constant bit rate
FMC – fixed-mobile convergence
HW – hardware
IMS – IP Multimedia Subsystem
IP – internet protocol
ISDN – integrated services digital network
ISO – international standard organization
ITU-T – international telecommunication union
LAN – local area network
MCU – the device management conferences
ME – microelectronics
NGN – next-generation networks
PBX – private branch exchange
QoS – quality of service
SDH – synchronous digital hierarchy
SW – software
TCP/IP – protocol stack
USN – ubiquitous sensor networks
VBR – variable bit rate
VoIP – voice over IP protocol
WAN – wide area network

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