TONI JANEVSKI

NGN ARCHITECTURES, PROTOCOLS AND SERVICES

WILEY

NGN ARCHITECTURES, PROTOCOLS AND SERVICES

NGN ARCHITECTURES, PROTOCOLS AND SERVICES

Toni Janevski

Ss. Cyril and Methodius University, Macedonia

WILEY

This edition first published 2014

Registered office
John Wiley & Sons Ltd, The Atrium, Southern Gate, Chichester, West Sussex, PO19 8SQ, United Kingdom

For details of our global editorial offices, for customer services and for information about how to apply for permission to reuse the copyright material in this book please see our website at www.wiley.com.

Library of Congress Cataloging-in-Publication Data applied for.

ISBN: 9781118607206

Typeset in 10/12pt TimesLTStd by Laserwords Private Limited, Chennai, India

1 2014

To my great sons, Dario and Antonio, and to the most precious woman in my life, Jasmina

Contents

About the Author

Toni Janevski, PhD, is a Full Professor at the Faculty of Electrical Engineering and Information Technologies, Ss. Cyril and Methodius University, Skopje, Macedonia. He received his Dipl. Ing., MSc, and PhD degrees in electrical engineering, all from the Faculty of Electrical Engineering and Information Technologies, Ss. Cyril and Methodius University in Skopje, in 1996, 1999, and 2001, respectively.

During 1996–1999 he worked for the Macedonian mobile operator Mobimak (currently T-Mobile, Macedonia), contributing to the planning, dimensioning, and implementation of the first mobile network in Macedonia. In 1999 he joined the Faculty of Electrical Engineering and Information Technologies in Skopje.

In 2001 he conducted research in optical communications at IBM T.J. Watson Research Center, New York. During 2005–2008 he was an elected member of the Commission of the Agency for Electronic Communications (AEC) of the Republic of Macedonia, where he contributed towards the introduction of new technologies in Macedonia, such as WiMAX and 3G mobile networks, as well as new operators and services.

During the periods 2008–2012 and 2012–2016 he was an elected member of the Senate of the Ss. Cyril and Methodius University in Skopje. In 2009 he established the Macedonian ITU (International Telecommunication Union) Center of Excellence (CoE) as part of the European CoE network, and has served as its head/coordinator since than. He is the author of the book titled "Traffic Analysis and Design of Wireless IP Networks," published in 2003 by Artech House Inc., Boston, USA. Also, he is the author of the book "Switching and Routing," written in the Macedonian language, published in September 2011 by the Ss. Cyril and Methodius University in Skopje, for which in 2012 he won the "Goce Delchev" award, the highest award for science in the Republic of Macedonia.

He has published numerous research papers and led several research and applicative projects in the area of Internet technologies and mobile and wireless networks. He is a Senior Member of IEEE. His research interests include Internet Technologies, Mobile/Wireless and Multimedia Networks and Services, Traffic Engineering, Quality of Service, Design and Modeling of Telecommunication Networks, as well as Next Generation Networks.

1

Introduction

1.1 Introduction

The development of telecommunications and communication technologies in the twenty-first century, at least in its first half, has an unambiguous direction toward a single goal, and that is the Internet as a single platform for all services through a global network. However, the initial concept of telecommunications was based on real-time services such as voice communication between users over a telephone network (i.e., telephony), or diffusion of video and/or audio (i.e., television and radio). If we go even further in the past, one may mention the telegraphy in the nineteenth century as the first telecommunications technology for data transmission based on the usage of electrical signals.

However, the world of telecommunications or ICT (Information and Communication Technologies) is continuously evolving and changing, including the technologies, regulation and business aspects. Going from the telegraphy as main telecommunication service in the nineteenth century, then the telephony and television (including the radio diffusion) as fundamental telecommunication services in the twentieth century (and they continue to be nowadays), and Internet phenomenon by the end of the twentieth and beginning of the twenty-first century, telecommunications have changed and the technologies have changed. But, in such process was kept the backward compatibility for the flagship services, such as telephony and television, and their integration with the new services, such as Internet native services [e.g., World Wide Web (WWW), electronic mail (e-mail), etc.].

Regarding the development of the telecommunications so far, one may distinguish among four key phases:

1. The automation of the telephone exchanges and networks at the end of the nineteenth and the beginning of the twentieth century;
2. The transition from analog to digital telecommunication systems from the 1970s to 1990s;
3. The integration of the circuit-switched telephone networks, such as Public Switched Telephone Networks (PSTN) and Public Land Mobile Networks (PLMN), with the packet-based Internet in the 1990s and 2000s;
4. The convergence of all telecommunication services, including telecom-native services (such as telephony and TV/radio) as well as Internet-native services (such as WWW, e-mail, peer-to-peer services, etc.), over the broadband Internet as a unified global networking platform (regardless of the access network type, either wired or wireless), toward the Next Generation Networks (NGN), in 2010s and 2020s.

NGN Architectures, Protocols and Services, First Edition. Toni Janevski.

The four phases of telecommunications development have resulted in exponential increase of number of telecommunication networks and number of users. The nineteenth century can be denoted as a century dedicated to telegraphy. At the end of the nineteenth century the telephony was invented and telephone networks started to be implemented around the world. The twentieth century was dedicated mainly to telephony as primary service in telecommunications worldwide. At the end of the twentieth century appeared the Internet for public usage. Nowadays, in the second decade of the twenty-first century, all telecommunications services are being transferred to Internet. Hence, from this point of view, one may say (or predict) that twenty-first century will be dedicated to Internet and will be information centric. The framework of such development is set by the ITU (International Telecommunication Union) in the NGN concept. The main requirement for accomplishment of such task is broadband access to Internet, including fixed broadband, as well as mobile broadband. The broadband is a term used to describe the Internet access data rates which can provide access to all existing telecommunication services at given time including the currently most demanding ones such as video or multimedia streaming services (e.g., TV over the Internet). The birth and rise of the Internet, as well as broadband access to the global Internet network, has influenced the "look" of the telecommunications (i.e., the ICT).

So, today we have several important segments in the ICT globally. Telephony is still one of the primary services, where one can distinguish between fixed telephony and mobile telephony. Further, Internet is usually identified by certain types of services such as WWW, e-mail, peer-to-peer services, and many more, provided in so-called best-effort manner. Best-effort principle is based on connection control by the end point of the communication (called hosts, such as computers, servers, mobile terminals, etc.), where network nodes perform basically routing of all packets from all services without differentiation among them. Finally, Internet requires broadband access, including fixed broadband and mobile broadband, with aim to provide capabilities for different types of services including the most demanding ones regarding the available data rates (i.e., the bandwidth). These five segments form the outlook of today's telecommunications. The number of users for fixed telephony, mobile telephony, individual Internet users, and users with fixed and mobile broadband access to Internet are shown in Figure 1.1 (for more details a reader may refer to [1]). It is obvious that mobile telephony has overtaken the number of telephone users from the fixed telephony a decade ago. Hence, the number of mobile users increases exponentially and it is targeting the total population on Earth. Mobile telephony is personal, while fixed telephony is related to a certain location (e.g., a home or an office). Hence, the market capacity for the fixed telephony is several times smaller than the market capacity for the mobile telephony. However, it is likely that mobile telephony will be soon saturated by reaching 100% of the world population. On the other side, the mentioned trend of integration between traditional telecommunications and the Internet, which have been developed separately at the beginning of each of them, is finally resulting in transition of telecommunication world into the Internet world, and vice versa. The number of Internet users is increasing exponentially in the past two decades, almost in parallel with the rise of number of mobile users, as it can be seen in Figure 1.1. The broadband is crucial for the Internet. However the exponential rise of the broadband access started 10 years ago, including fixed broadband access and mobile broadband, and currently it is in a similar position to mobile telephony a decade ago (Figure 1.1). Hence, the highest market potential (in the ICT world) currently is the broadband Internet. Similar to the mobile-to-fixed telephony comparison, the mobile broadband is personal and hence will have faster exponential

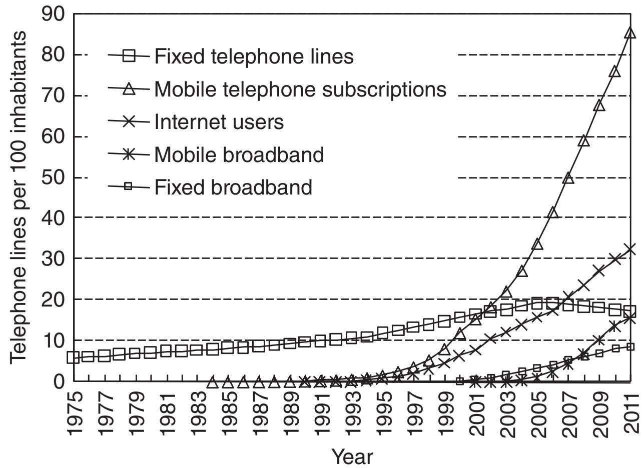


Figure 1.1 Global development of ICT

growth and higher penetration compared to the fixed broadband. On the other side, the fixed broadband will always have a higher capacity due to scarce radio spectrum resources over a given geographical area.

According to the discussions above, in the following part we will cover the main aspects of the traditional telecom world, and then the traditional Internet world, which converge nowadays to NGN as defined by the ITU (International Telecommunication Union). However, the ITU was established in 1865, when there was only the telegraphy present as a telecommunication service, and hence it was originally founded as the International Telegraph Union. Later the word "Telegraph" in the name of the ITU was replaced with the word "Telecommunication", with the aim to cover the broader range of services after the invention of telephony and later the invention of radio broadcast and television. Today the ITU is part of the United Nations (UN), as a specialized UN agency for ICT. It is main international organization for telecommunications, which provides harmonization regarding the radio frequency spectrum worldwide through the sector International Telecommunication Union-Radiocommunication (ITU-R). Also, ITU develops technical standards as well as provides harmonization for usage of ICT technologies globally via its sector International Telecommunication Union-Telecommunications (ITU-T). Finally, ITU strive to improve the access to ICTs to developing countries and underserved communities worldwide through its sector International Telecommunication Union-Development (ITU-D), because everyone in the world has a fundamental right to communicate.

1.2 Traditional Telecom World

Traditional telecom world is mainly based on the telephony, which is the most important service in it. Hence, on the way toward the NGN, the telephony is still one of the most influential services. The other important traditional telecommunication service is television (also, coming from the first half of the twentieth century, while main spreading of the television worldwide happened in the second part of the last century). However, from the beginning the television was not offered by telecom operators which provided the telephony. Instead, the television was provided via separate broadcast networks, either terrestrial or cable. Traditional telecommunication networks are in fact the telephone networks, and hence they are in the focus in the following sections.

1.2.1 History of Telephony

The current look of the telecommunications started in the nineteenth century with the invention of the telephone by Alexander Graham Bell in 1876. However, telephony as a service even at the beginning required large number of users to have telephones, so they could call each other. So, the telephony has never been a service that could be dedicated to a privilege group of users only, because in such case they could call only each other, not the rest of the people. And, the true meaning of the telephony is possibility to call anyone from anywhere in the world using a global interconnected telephone networks. But, when a large number of users already had telephones, then the problem was to connect all users in organized telephone networks, something that introduced the need for switching. One may define switching as a way to connect a certain input line to a certain output line using so-called exchange (i.e., switch) in the traditional telephony networks. However, first switches were manual, based on an operator (human) in a central switching room. The human operator provided the switching between the calling and the called user lines by plugging the jacks of the connector wires to the calling user's line on one side and to the called user's line on the other side, using a switchboard for such task. After the termination of one side of the connection, by putting the telephone on-hook (signaled usually by a lamp on the operator's switchboard), the operator would release the connection by pulling off the jacks. But, the manual approach was not incremental and thus needed automation.

The first automatic exchanges appeared at the end of nineteenth century, which were based on step-by-step switches called Strowger switches. Further, they were replaced by crossbar switches in the first half of the twentieth century, which stayed in operation worldwide until the 1990s (i.e., until the transition from analog to digital telephony). The exchanges were connected via transmission systems, which were based on multiplexing of analog voice signals over coaxial copper cables, while the access toward end subscribers (called local loop) was based on twisted pairs (i.e., a pair of copper wires between the user's telephone device and the exchange). So, for almost 100 years (from the invention of the telephone in 1876) the telephone networks were analog, which means that original analog signals (generated by human on one side of the connection) were carried over transmission systems and via established connections in the switching matrices (in each exchange on the way) to the called party, and vice versa (from called party to the calling party). Hence, the first really worldwide network for communication between people was the telephone network, consisted of many national telephone networks interconnected on national and international levels, thus providing truly global network in which each user could reach every other user on Earth. However, telephony is one of the most demanding services regarding the delay end-to-end, since it is a conversational type of service which happens in real-time, similar to two (or more) people conversation face-to-face. Telephony is two-way (full duplex) end-to-end communication. For that purpose it requires signaling before the voice connection is established, with aim to find the called party based on certain address (i.e., a telephone number) and to provide channels end-to-end (between the calling and called users) in all transmission systems and all exchanges on the way.

Further, one of the major phases in traditional telecommunications was the digitalization, which meant a transition from analog signals to digital signals in the telephone networks. That was accomplished by representing the analog voice signals with digits (mainly the digits were bits, due to simplicity of handling only two logical and physical levels, representing the binary "0" and binary "1," than multiple ones). This was made possible by the development of electronics and computer systems in 1960s. As a result of this development first appeared hybrid switching systems, characterized with computer-based management, while the switching remained analog. Such systems were called quasi-electronic systems.

Later, in the 1970s/1980s and especially in the 1990s the switching was done with digital systems. That paved the ground for telecommunications networks to become digital networks. However, the local loop in digital telephone networks remained to be analog. One may say that it happened because it was a simpler solution (due to backwards compatibility with the analog telephone handsets) as well as cheaper one (to perform analog-to-digital conversion, and vice versa, on the side of the telephone exchanges rather than the user side).

The digitalization of the subscriber lines began with the spread of ISDN (Integrated Services Digital Network) in the 1990s, a technology which provided digital local loop (over twisted pairs) with basic rate interface of 144 kbit/s, consisted of two 64 kbit/s bearer channels (for telephony or data via dial-up modem) and one 16 kbit/s signaling channel (i.e., delta channel). However, ISDN has not experienced success as one might expected in the 1990s. The main reason for that was the appearance of the Internet and its exponential growth in the world since 1993 when the WWW (which is the dominant Internet service until today) started to be used globally via browsers (i.e., WWW clients). The need for higher data speeds (i.e., rates) grew continuously, because the Internet provided different services (e.g., WWW, video streaming, multimedia, voice, peer-to-peer services, gaming, e-mail, chat, etc.) over the same network. ISDN could not satisfy such requirements for higher data rates for the emerging Internet services. That resulted in the replacement of ISDN with the xDSL technologies where DSL means Digital Subscriber Line. In the 2000s and 2010s the most widespread DSL was Asymmetric Digital Subscriber Line (ADSL), providing higher data rates in downlink compared to lower data rates in the uplink, which corresponds to the traffic behavior of the Web (i.e., the WWW) as the dominant Internet-native service.

From the start of the digitalization era in telecommunications in the 1970s one may notice the fast exponential increase of fixed telephony users worldwide until 2005, and afterwards it is slight decline (Figure 1.2; for more details a reader may refer to [1]). This is mainly due to exponential growth of mobile telephony, which is more personal and hence has higher market potential compared to the fixed telephony.

1.3 Public Switched Telephone Networks

Traditionally, telecommunication services have been associated with the existence of a certain network infrastructure. The main goal in traditional telecommunication networks is to ensure accurate and timely transmission of information from the source to the end user (i.e., the destination). The information is transmitted through copper, optical cables, or through wireless networks, which are located between the transmitter and receiver. The end-to-end channel is consisted of the transmitter, receiver, and the transmission system for transfer of

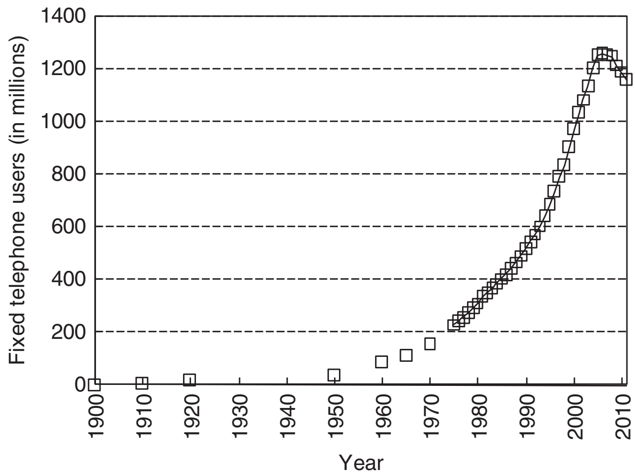


Figure 1.2 Telephone users statistics since 1900

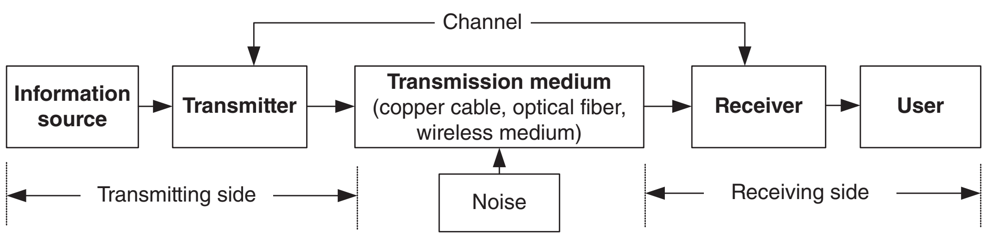


Figure 1.3 Model of communications channel

the information (shown in Figure 1.3). Transmitted signals are degraded by the noise in the communication channel. Usually, transmitters and receivers are packed into a single terminal device, such as phone, mobile terminal, personal computer, lap-top, and so on. The end terminal can be used by a human or another machine (e.g., Web server, e-mail server, automatic voice machine, etc.).

In practice, the communication channel model is not simple as shown in Figure 1.3. Namely, the path from source to destination in most cases is divided into several sections, interconnected via exchanges (i.e., switches) in traditional telephony or routers in Internet.

In classical telecommunication for the transmission of voice (i.e., telephony) telecommunication systems can be divided into two main groups:

- Switching systems;
- Transmission systems.

PSTN and PLMN, as they have existed at the end of the twentieth century, are based on switching and transmission of digital signals. Because main service was telephony, all systems were designed to fit the requirements for end-to-end delivery of voice in both directions.

1.3.1 Pulse Code Modulation

In digital telephone networks the globally accepted standard for performing A/D (analog to digital) conversion is Pulse Code Modulation (PCM), standardized by ITU-T as Recommendation G.711 [2].

Using the PCM, the standard analog voice signal in the frequency bandwidth 0−4000 Hz is transformed into digital binary signal with data rate 64 kbit/s. There are three basic steps for the PCM, and they are:

- sampling;
- quantization;
- coding.

Sampling is process of taking samples of the analog voice signal at regular time intervals. According to the Nyquist theorem for sampling of analog signals, the frequency used for sampling must be at least two times bigger than the highest frequency in the analog signal. Since the analog signals for voice are in the range 0−4000 Hz, the highest frequency is 4000 Hz = 4 kHz. Hence, the sampling frequency adopted for PCM in 2×4000 Hz = 8000 Hz, which results in taking samples every 125 µs (1/8000 Hz = 125 µs).

The range of the electric amplitude (voltage) of the analog signal is further divided into limited number of levels, called quantization levels. For voice it is set to 256 levels which can be presented using 8-bit codes ($2^8 = 256$). So, the quantization is a process of converting the amplitude of a given voice sample to one of the 256 discrete levels. However, due to the fact that probability for low-amplitude signals is higher than probability for high-amplitude signals, usually the number of levels is more concentrated at low-amplitude values, with aim to reduce their distortion (i.e., quantizing noise).

Finally, encoding is a process of assigning 8-bit code to each quantized sample. The code belongs to one of the 256 discrete signal levels. That gives as outcome the data rate for single voice channel, which is 8 bits/125 µs = 64 kbit/s. This data rate, obtained with the PCM, is further the basic rate for all transmission systems (i.e., multiplexing hierarchies) in digital telecommunication networks. Therefore, the 64 kbit/s is also denoted as Digital Signal 0 (DS-0), and it is basic granularity in circuit-switched telephone networks.

1.3.2 Architecture of the Telephone Network

The telephone network is consisted of end users who have phone handsets (i.e., telephones) connected via twisted pairs (twisted pair is a pair of copper wires) which connect the telephone in the home (or office) to the local exchange. Further, telephone exchanges are interconnected with transmission systems, following certain network hierarchy. When all switching and transmission is accomplished by using digital signals then we refer to it as a digital network.

Environment of exchanges in a telephone network is shown in Figure 1.4. It is consisted of subscribers' lines (twisted pairs), remote exchange concentrators, Public Branch Exchanges (PBXs), line concentrators, as well as circuits to and from other exchanges.

Subscriber line is the local loop. In digital telephone networks in twentieth century the subscriber line was usually analog and A/D conversion was completed at the local exchange premises (exchange to which the user was connected). We have still such networks today,

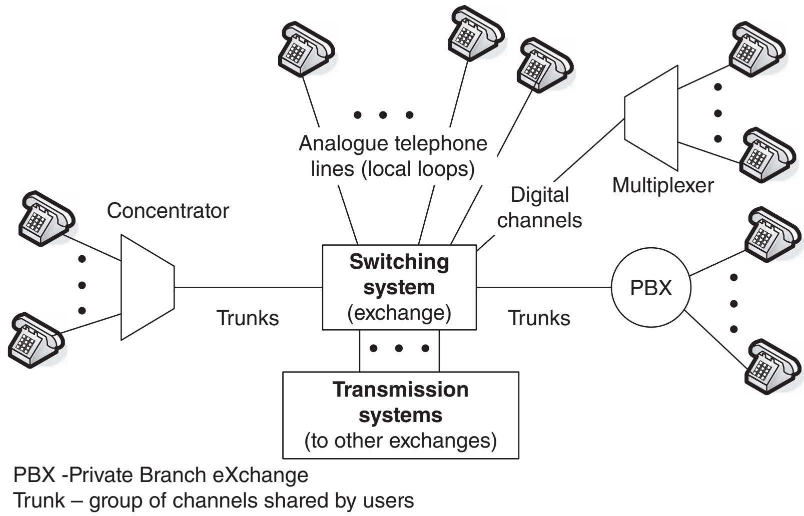


Figure 1.4 General environments for an exchange

mainly at the incumbent telecom operators, which were the national state-owned operators in the past. So, in traditional digital network, the local loop is analog. The telephone device receives its power supply from the local exchange. When a telephone is taken off-hook, the circuit between the telephone device and the exchange is closed, and electric current flows over it (which is DC – Direct Current). The two wires in the local loop are twisted to eliminate the crosstalk. Analog signals in local loop are coupled (between the telephone and the twisted pair on one side, and between twisted pair and the exchange on the other side) by using transformers. Further, analog signals are converted into digital signals at the local exchange. Then, the digital signals are transferred via the PSTN to end receiver for decoding (e.g., in local exchange on the other side which performs D/A conversion). However, due to the establishment of circuits in analog telephone networks, traditional telephony is called circuit-switching. The term continued to be used in digital telephone networks based on 64 kbit/s lines end-to-end (by using PCM). Hence, the PSTN are also referred to as circuit-switched networks.

The remote exchange concentrators and line concentrators are used to aggregate the traffic from many subscribers to smaller number of channels between the user premises and the local exchange. Such approach is possible in telecommunication networks because different users are active (have phone connections with other users) in different times. So, statistically speaking, there is always certain level of traffic in the network which is generated from a certain group of users, and it depends upon the activity of the users. Usually, the telephone networks are dimensioned by the traffic intensity (number of established telephone connections) in the so-called busy-hour (the hour with highest average traffic intensity during a day). Due to that, certain number of users can be served by a group of shared 64 kbit/s channels. For example, if we have 100 users, and average user activity in the busy hour is 20% (meaning that an average 20 out of 100 users have established connections during the busiest hour in the day), then that group of users can be served with 30 channels (with average call blocking probability of 1%, calculated by using Erlang-B formulae [3]). So, instead using 100 channels for 100 users the telecom operator may serve the users with 30 channels, thus saving the capacity and the costs for service provisioning. This simple example, in fact, explains the need and the usage of concentrators in telephone networks.

Additionally, there are also private telephone networks, called PBXs, which are used by companies with aim to manage the internal voice traffic (within the company) thus avoiding the need for switching the internal traffic via the PSTN owned by a telecom operator. However, PBX must be connected to PSTN to provide possibility for outgoing and incoming calls from other users.

Finally, each telephone network must be connected with other telephone networks via switching and transmission systems, with aim to provide global connectivity for voice communication since telephony was the main service in traditional telecommunication networks.

1.4 Signaling Network

In digital telephone networks prior to voice communications there is a need to find the called user by using destination telephone number and then to reserve 64 kbit/s channels in both directions between the local loops of the caller and called party, in all switching and transmission systems on the way. For such task there is a need of signaling in telephone networks, which existed since the analog era. Signaling can be done using analog signals (e.g., electrical signals) or digital signals (e.g., bits, bytes, or messages/packets).

Definiton 1.1: Definition of Signaling

Signaling is mediated exchange of control information signals using certain signaling alphabet (set of signals or messages).

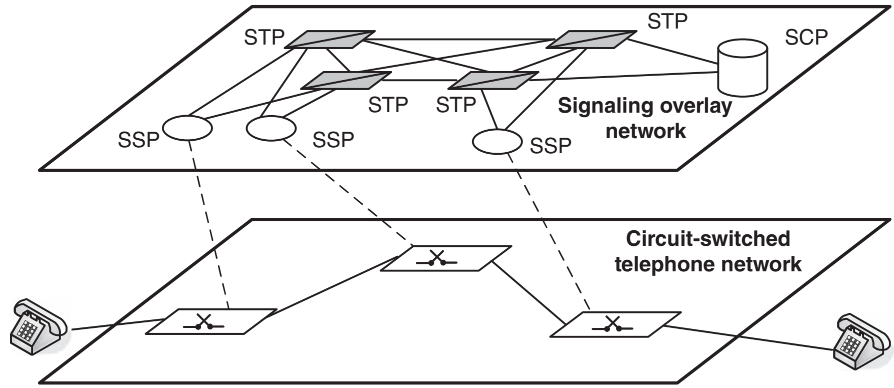
There is line signaling between the user telephone and the local exchange, as well as signaling between telephone exchanges. In many PSTNs the line signaling is still analog, while signaling between exchanges is digital.

Another general classification of signaling in PSTN is into two groups:

- Channel Associated Signaling (CAS), where certain signaling information is associated with the voice channels over the same transmission medium;
- Common Channel Signaling (CCS), where signaling information from many users is multiplexed over a common channel and can be carried separately from the voice traffic.

1.4.1 SS7 Architecture

In digital telecommunication networks, from 1980s the most used is Signaling System No. 7 (SS7), standardized by the ITU [4], and accepted globally. It belongs to the CCS type of signaling and it is used in all PSTN and PLMN worldwide. SS7 is packet-based signaling, so it has introduced packet-switching globally in telecommunication networks for the first time in 1980s, even before the Internet growth in the 1990s. Therefore, here is described SS7 with some important details regarding the network architecture and protocols.



SSP – Service Switching Point
STP – Signaling Transfer Point
SCP – Signaling Control Point

Figure 1.5 SS7 overlay network

SS7 logical network architecture defines three types of nodes (Figure 1.5):

- Service Switching Point (SSP);
- Signal Transfer Point (STP);
- Signal Control Point (SCP).

Each of the three node types in SS7 has specific functionalities. So, SSP is a network element in SS7, which is integrated with local telephone exchanges (with attached subscriber lines to them). SSP converts dialed number (called B-numbers or Global Titles according to SS7 terminology) into SS7 signaling messages and establishes signaling connection with the SSP of the called user. SSP establishes, manages, and terminates voice connections. It sends signaling messages to another SSP via the STP node to which it is connected.

STP is a router and/or signaling gateway in the SS7 network. This node (STP) has main task to route signaling messages between so-called signaling points in the network. Those STPs who act as gateway nodes actually connect signaling network of one telecommunications network (e.g., belonging to a telecom operator) with signaling network of another telecommunications network (e.g., belonging to another telecom operator).

SCP provides access to certain application in SS7. In fact, SCP can be viewed as a database with an appropriate interface for database access by other signaling points in SS7. Typical usage of SCP is for special B-numbers such as the 0800 series (when the called party is charged for calls), or for the provisioning of roaming in PLMN.

Signaling nodes are logically separated from the network for transmission of voice signals. Thus, they form a signaling overlay network as shown in Figure 1.5. Separation is logical, because over the same physical transmission media (copper, optical, or wireless medium) are transmitted signaling and voice channels. Also signaling nodes are located at the same physical locations and integrated in the switching systems (i.e., telephone exchanges).

1.4.2 SS7 Protocol Model

SS7 is composed of multiple protocols which are layered according to OSI (Open System for Interconnection) model. According to OSI there are seven layers of protocols which can cover all possible functionalities in telecommunication networks. The mapping between the OSI layering model and SS7 standardized protocols is shown in Figure 1.6.

SS7 is mapped into five layers of the OSI model, as shown in Figure 1.6. Physical level in SS7 (OSI layer 1) is MTP (Message Transfer Part) layer 1 (MTP1), specified in ITU Recommendation Q.702 [5]. This layer defines physical and electrical characteristics of the signaling link.

Layer OSI-2 in SS7 is MTP layer 2 (MTP2), standardized by ITU Recommendation Q.703 [6]. This level has task to provide reliable transfer of signaling messages from the source to the destination signaling point through a signaling link that directly connects two network nodes in the SS7 network.

Network layer in the SS7 is MTP layer 3 (MTP3), standardized by ITU Recommendation Q.704 [7]. MTP3 provides functionalities for routing of signaling messages between signaling points in SS7 network. As already mentioned above, the routing in the SS7 network is performed by the STP signal points.

Above the network layer in SS7 are defined customized signaling protocols that are designed for specific types of services (e.g., telephony, ISDN, mobile networks, etc.). For example, for ISDN subscriber line is used ISUP (ISDN User Part), for traditional telephony is used TUP (Telephone User Part), for services in a mobile network is used MUP (Mobile User Part), and so on. Internally SS7 network also uses TCAP (Transaction Capabilities), which typically uses SCCP (Signaling Connection Control Part) at OSI layer 4 in SS7 protocol model.

SCCP (defined by ITU Recommendations Q.711–719) provides a connection-oriented and connectionless (connectionless means that each message is independently routed through the signaling network) network services through transfer of MTP3 signaling messages between SSP nodes in the network. Thus, MTP layers allow transfer of signaling messages from one signaling point to another (i.e., from one node to another node in the SS7 network), while

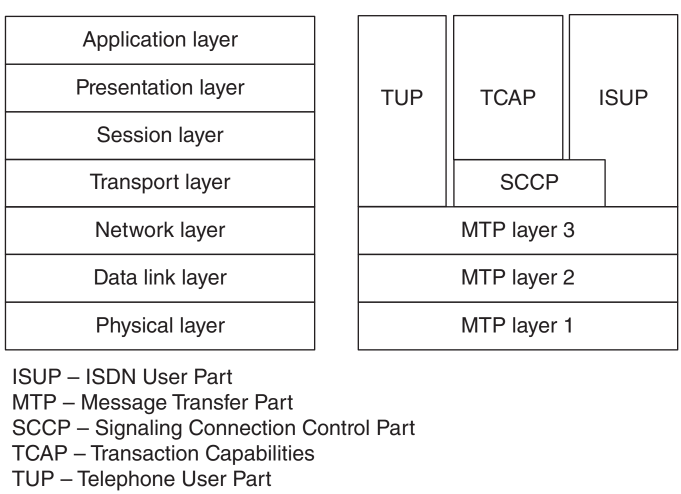


Figure 1.6 SS7 protocols mapped to OSI layering model

SCCP enables transmission of signaling messages end-to-end (from sender application in the sending signaling point to recipient application in the receiving signaling point).

TCAP (defined by the ITU Recommendations Q.770–Q.779) messages are used for applications in SS7. This protocol is used for communication between SCP nodes through STP nodes. Because SCP represents databases, it follows that TCAP is mainly used to access databases in the SS7 signaling network. Transmission of TCAP messages end to end is performed by using the SCCP.

At the end one may briefly summarize that SS7 is dominant and universal signaling system which is based on packet switching and transmission. The same system is used in fixed and mobile networks signaling for voice calls and supplementary services (e.g., call waiting, call forwarding, conference call, call barring, calling line identification presentation, etc.).

Due to its excellence as well as backward compatibility regarding the signaling for the telephony, SS7 is also adapted for usage in Internet environment by SIGTRAN (Signaling Transmission) family of specification created by IETF (Internet Engineering Task Force) for carrying SS7 signaling (for PSTN and PLMN) over IP networks.

However, best known equivalent of SS7 (and also its successor) in Internet environment is SIP (Session Initiation Protocol), which will be covered in the following chapters.

1.5 Transmission Systems

In telecommunications signals are carried from one node to another node in the network by using transmission systems, which are carrying signals, such as electric signals (over copper cables), optical signals (over fiber) and radio signals (over wireless links). With aim to reduce the costs of the transmission systems, as well as to manage them more efficiently, each transmission link is used to carry many signals from many users. Multiple usage of a given transmission medium is called multiplexing. Hence, systems that perform multiplexing are called multiplexers. Multiplexing is done at the sending side of a transmission link (there are always two ends – the sending and receiving ends), where many incoming signals are placed on the same transmission medium without interfering with each other. The reverse process called demultiplexing is being made always at the receiving side.

The transmission system is used to transfer N transmission signals instead of only one signal, as shown in Figure 1.7. This is done in order to more effectively use the expensive transmission links.

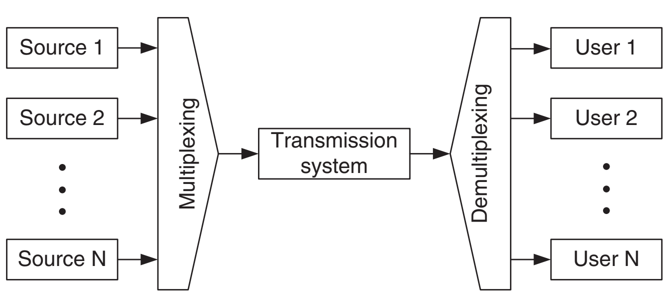


Figure 1.7 Multiplexing on a transmission system

Typically, network nodes (e.g., switches, routers) interconnected with transmission systems consist a telecommunication network. Moreover, the practice confirms that about two-thirds of the cost of a telecommunications network is related to the transmission systems and one-third of the cost is related to the switching systems. In this context, a reduction in transmission costs significantly affects the reduction of the total cost of the telecommunication network (including capital investments as well as operating costs).

Generally, telecommunication networks are designed by a trade-off between price and performance. Often it is not a question of whether something can be done or offered as a service, but it is more a question how much such service will cost and who or how many users will be able to pay for it. In that direction, to allow affordable cost of the telecommunication services to majority of the population globally, the main tendency is to minimize the overall cost of telecommunication networks as much as possible.

Transmission systems, historically speaking, were analog before the 1970s and then digital in the past several decades. In the analog era the multiplexing was done of 4 kHz wide channels by using analog modulation techniques. The 4 kHz bandwidth is related to the dedicated spectrum to voice for the analog telephony, which is in the range 300–3400 Hz, thus resulting in 4 kHz bandwidth per voice channel in one direction. In digital systems the same frequency bandwidth is digitalized with the PCM, resulting in bit rate of 64 kbit/s per channel. The multiplexing of channels in digital transmission systems is done initially by using a PDH (Plesiochronous Digital Hierarchy), and later by using a SDH (Synchronous Digital Hierarchy) or SONET (Synchronous Optical Networking).

1.5.1 Multiplexing of Digital Channels

Multiplexing is a technique of placing many signals over one transmission medium (e.g., copper, optical fiber, or radio).

There are two basic schemes for multiplexing:

- *FDM (Frequency Division Multiplexing):* Different frequencies or frequency bands are assigned to different channels over the same transmission medium.
- *TDM (Time Division Multiplexing):* Different time intervals, called time slots, are assigned to different users over the same transmission medium and using the same frequency in the case of copper cables or radio transmission, or the same wavelength in the case of fiber as a transmission medium.

However, regarding the multiplexing schemes in telecommunications one have to mention also the CDM (Code Division Multiplexing), which is based on using different code sequences for different signals over the same frequency bands and during the same time intervals. However, CDM is not used in transmission systems so far, but it is mainly used in wireless and mobile networks (e.g., Direct Sequence Spread Spectrum – DSSS in WiFi, CDMA (Code Division Multiplexing) in IS-95 standard from second generation mobile networks in America, in 3G mobile networks as Wideband CDMA, etc.). When multiplexing schemes are used in the access networks (where end users access the network) they are called multiple access schemes, such as FDMA (Frequency Division Multiple Access), TDMA (Time Division Multiple Access) and CDMA (Code Division Multiple Access).

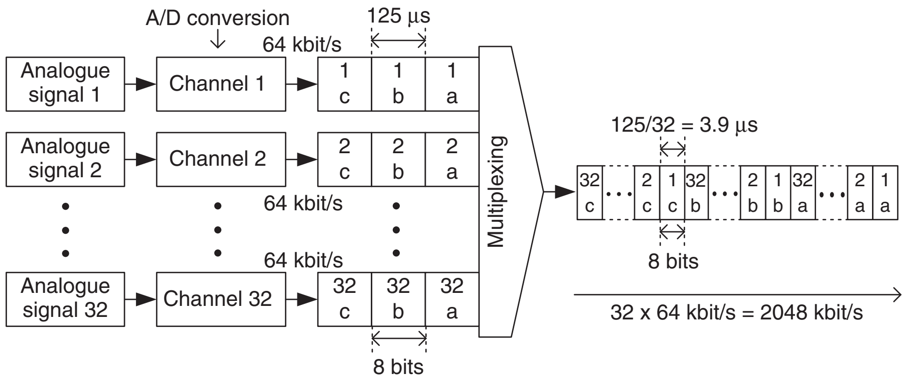


Figure 1.8 Principle of time division multiplexing

Regarding the multiplexing in transmission systems one have to mention also WDM (Wavelength Division Multiplexing). It is based on carrying user data on different wavelengths over the same fiber cable.

However, the main multiplexing techniques in transmission systems used in PSTN are based on the TDM (which is based on PCM coded voice).

1.5.2 Time Division Multiplexing in PSTN

When PCM is used for voice coding, the time duration for transmission of each 8-bits code must be less than 125 µs. However, if that time is much less than 125 µs, then there is possibility to transmit bits from other voice signals over the same transmission medium. That is called Time Division Multiplexing, illustrated in Figure 1.8, which is the basic approach in all digital transmission systems, either wired or wireless. However, it can be combined with other multiplexing techniques, such as FDM, CDM, and WDM.

ITU-T has standardized the hierarchy for bit rates in digital transmission systems [8]. In the digital hierarchy are defined four levels, where the first hierarchy level in Europe and most of the world is 2048 kbit/s, consisted of a frame with 32 time slots (32×64 kbit/s = 2048 kbit/s). Time slots are numbered as TS-0 to TS-31 (TS denotes Time Slot, which corresponds to single 64 kbit/s channel). From the total of 32 time slots, 30 time slots are used for voice (in telephone networks), first time slot (TS-0) in the frame is used for synchronization and alarming, and the TS-16 is mainly used for signaling related to voice connections.

In America the first hierarchy level is 1544 kbit/s and it is consisted of 24 channels, multiplexed with TDM. Although this version is developed first, the most used version on a global scale is the European primary multiplex of 2048 kbit/s.

The standardized hierarchical bit rates in digital networks are given in Table 1.1 [8].

However, to provide interconnection of digital networks based on the two different digital hierarchies, there was created hybrid hierarchy as given in Figure 1.9, which provided possibilities to map European to American digital hierarchy and vice versa.

The digital hierarchy rates in fact form the PDH, where the bit rate is controlled by a clock in the local equipment (e.g., exchange). The PDH was used in 1970s and 1980s, and later