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**MINISTRY FOR DEVELOPMENT OF INFORMATION
TECHNOLOGIES AND COMMUNICATIONS OF THE REPUBLIC
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This graduation thesis is devoted to modeling Next Generation Convergence network using a Cisco Packet Tracer software. In this work, an overview of NGN architecture, multimedia services such as DVB, IPTV, HBB TV and setting devices of NGN network were studied. During the thesis delivery delay of packets between end devices and connecting different phones have been analyzed.

The thesis also reviewed issues of life safety.

Дня выпуска квалификационная работа посвящена моделированию Next Generation Convergence network с помощью программного обеспечения Cisco Packet Tracer. В данной работе представлена базовая архитектура NGN, мультимедиа услуги DVB, IPTV, HBB TV и настройки устройств в сети NGN. В работе рассмотрены вопросы задержки и доставки пакетов между конечными устройствами и соединениями разных телефонов.

В работе также рассмотрены вопросы безопасности жизнедеятельности.

Ушбу битиров маънавий иши Cisco Packet Tracer дастуридан фойдаланиб NGN архитектурасининг имитация моделини ишлаб чиқишга бағъ ишланган. Ушбу ишда NGN архитектураси, мультимедиа хизматлари DVB, IPTV, HBB TV ва тармоқдаги қурилмаларнинг соъланishi о`рганиб чиқилган. Таҳлил давомида тармоқдаги охириги қурилмалар о`ртасида пакетларнинг узатилиш ва кечикиш вақтлари ҳамда турли хил телефонларнинг о`зaro боғ`ланishi ко`риб чиқилган.

Shuningdek hayot faoliyati xavfsizligi masalalari ham ko`rilgan.

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INTRODUCTION

The role of Telecommunication in Uzbekistan shows the positive growth, and this tendency will proceed. In 2011 our turns grew twice more, and for us it is good indexes. Also as the President of Uzbekistan Islam Karimov adopted the resolution " о рх п дльнйшму вдрню и рзвитию сврмнных инфрмциинн-кммункициинных тхнлгий" in our country the very great value is attached to development of information communication technologies, in particular, to the direction of software and communication technologies products, providing of infocommunication services for population of Uzbekistan and support of this sphere we see from the government of our republic[1]. Opinions on NGN definition may differ in some ways, but the main principles of the NGN (Next Generation Networks) were formed when the idea of NGN itself emerged. The next two definitions from ETSI and ITU-T describe NGN in substance. According to ETSI NGN is a concept for the defining and establishing of the networks, allowing a formal distribution of functionalities into separate layers and planes by using open interfaces, making it possible for the service providers and operators to create a platform which can be gradually developed thanks to creation, implementation and effective management of innovative services.[4] ITU-T defines NGN as a network based on packet transfer, enabling to provide services, including telecommunication services, and is capable of using several broadband transmission technologies allowing guaranteeing QoS.

The functions related to services are at the same time independent of the basic transmission technologies. NGN provides unlimited user access to different service providers. It supports general mobility providing the users with consistency and availability of services. NGN network is designed to offer various multimedia communications, which implies that NGN network guarantees high broadband capacities, multichannel transport with high data rates, low latencies, low packet loss and QoS. NGN provides the capabilities (infrastructure, protocols, etc.) to converge voice services (traditionally provided by a circuit-

switched network in Time Division Multiplex mode) and data services (traditionally provided by a packet-switched network) into one common network infrastructure: IP/MPLS-based packet network. It represents new technology and services that all operators want to have at their disposal. NGN technology enables converged IP/MPLS network services, mostly focused on opportunities for service differentiation and service-oriented technology. The convergent services are based on packet switching rather than circuit switching technology. In this paper we will describe in details NGN network architecture, protocols, services but we will be focused in the main part of NGN network that is softswitch.

1. AN OVERVIEW OF THE NGN NETWORK

Softswitch is considered as platform for packet switching in NGN network. It is designed to replace class 4 and 5 switches which are based on circuit switching technology. Also, here we will describe the the main functions of softswitch in NGN network in TK and the role of softswitch in a simple call scenario in NGN network in TK. NGN is a safe technology for the future. Nowadays market demands always change. Customers require different services with high quality. In order to fulfill those requirements is developed NGN network with an advanced platform. Next Generation Networks as a platform is able to offer ubiquitous connectivity and intelligent interfaces for human and machine communication as well as pervasive services access, bringing value to human life for its improvement and new experiences. It is proved that NGN network offers more services compared to PSTN network. NGN network needs lower investments to achieve the same functions. By creating dedicated virtual communication environment, NGN is able to disappear distance barrier between two users, handover problems etc. NGN is packet based network, so it is oriented in VoIP services. The main features of NGN network are: packet based transmission; broadband capabilities with end-to end QoS, fixed-mobile convergence as well as wide range of services. Two international organizations as the IETF and ITU propose two different NGN network models, each one having its own set of features and protocols. According to IETF, NGN network topology model is based in Softswitch model which consists of[6]:

Media Gateways (MG): the role of media gateway is conversion of voice message from circuit format into packet format.

Media Gateway Controllers (MGC): the main role of MGC is to manage different connections in a certain packet network. Also, has the feature for call control[11].

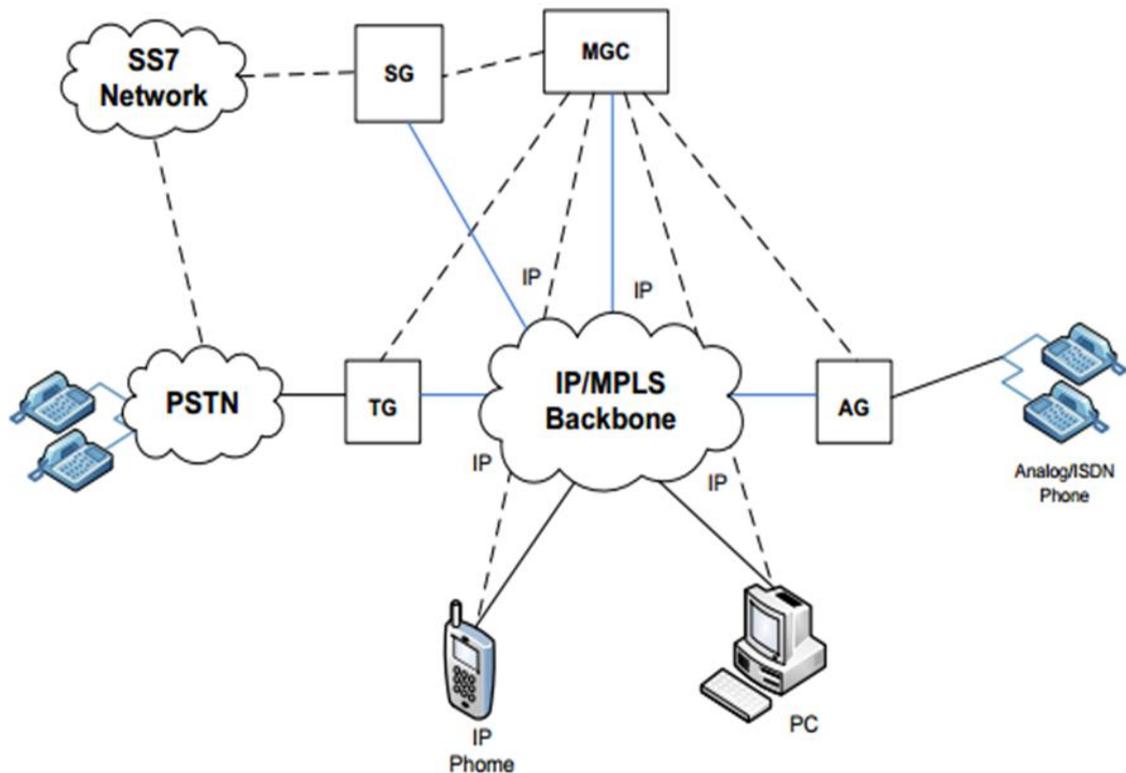


Figure 1.1. General NGN network architecture

1.1. NGN concept

Within the NGN concepts the standardization institutions are solving the following issues and problems:

- existing networks migration towards NGN;
- development in the field of access technologies;
- connection of other networks to IP networks;
- provision of services and development of new ones;
- interworking in the area of addressing;
- interworking of signaling systems;
- roaming a mobility;

There are many conceptual models and reference architectures for both the converged networks and VoIP architectures. Therefore, we have tried to find

common features and to define a suitable conceptual model for NGN.

An objective of the conceptual model is to determine functional layers (covering similar functionalities), their entities, reference points (interfaces) and information flows between them. Such a model then can be mapped more easily into the physical reference architecture (and it is independent of the physical entities, i.e. components of the architecture). In most analyzed cases the NGN conceptual model layers are from the point of view of functionalities divided into independent parts as follows: access (some reference architectures do not include it directly into the NGN model or replace it by the adaptation one), transport (transmission, switching), control (call/sessions control) and application.

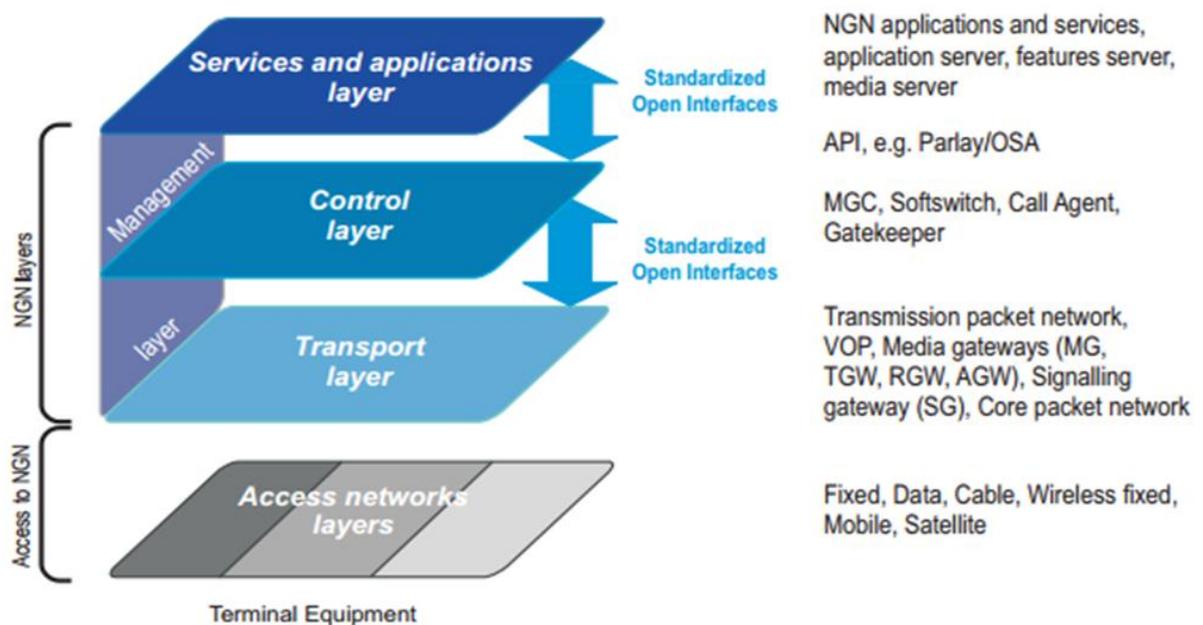


Figure 1.2 NGN conceptual model and its functional layers.

Conceptual model layers

The access layer provides the infrastructure, for example an access network between the end user and the transport network. The access network can be both wireless and fixed and it can be based on various transport media.

The transport layer ensures the transport between the individual nodes (points) of the network, to which are connected access networks. It connects physical elements deployed in the individual layers. It also enables the transport of

different types of traffic, media (signaling, interactive data, real-time video, voice communication, etc.)

The control layer includes the control of services and network elements. This layer is responsible for set-up/establishing, control and cancelling of the multimedia session. It ensures the control of sources as well, depending on the service requirements. One of the fundamental NGN principles is the separation of control logic from the switching hardware.

The service layer offers the basic service functions, which can be used to create more complex and sophisticated services and applications. It controls the progress of the service based on its logic. In the NGN it is required that the network control is not determined only by the terminal equipment applications, but that the network intelligence may carry out control over the network at all levels of the reference model. The network management reference model implies the following tasks for the network intelligence it has to ensure: Resource management (capacity, ports, and physical elements) and QoS in access to the network and in the transport network, as necessary[9].

1.2. Requirements of NGN network

High-capacity packet transfer within the transmission infrastructure, however, with a possibility to connect existing and future networks (be it the networks with packet switching, circuit switching, connection-oriented or connectionless, fixed or mobile).

Separation of managing functions from transmission features. Separation of service provisioning from the network and ensuring the access via an open interface and thus a flexible, open and distributed architecture.

Support for a wide range of services and applications by using the mechanisms based on the modular and flexible structure of elementary service building blocks. Broadband capabilities, while complying with the requirements for QoS (Quality of Services) and transparency. Possibility of a complex network

management should be available.

Various types of mobility (users, terminals, services). Unlimited access to a variety of service providers. Various identification schemes and addressing which can be translated to the target IP address for the purposes of routing in the IP network. (Flexible addressing and identification, authentication). Converged services between fixed and mobile networks (as well as voice, data and video convergence). Various categories of services with the need of different QoS and *classes of services* (CoS)[17].

Conformance to the regulation requirements, such as emergency calls and security requirements in terms of personal data protection. Cheaper and more effective technologies if compared to the current technologies.

1.3. Cisco Networking Academy.

Cisco networking academy global education initiative from Cisco Systems, offers networking programs, like the (Cisco Certified Network Associate) CCNA and (Cisco Certified Network Professional) CCNP courses, which prepare students for the certification exams of the same name, and other computer-related courses. Also see History of virtual learning environments for how Cisco Networking Academy has developed since 1997 relative to others within the VLE community. Courses are available in approximately 9,000 local academies, in over 165 different countries. As of 2010, there were over 900,000 active students (defined as students currently enrolled, students enrolled in a future course, and students who were enrolled in a course during the last five months).

Background

In 1993, Cisco embarked on an initiative to design practical, cost-effective networks. It quickly became apparent that designing and installing the networks was not enough, schools also needed some way to maintain the networks after they were up and running. Cisco Senior Consulting Engineer George Ward, based in

Cisco's Phoenix, Arizona, office, developed training for teachers and staff for maintenance of school networks. The first pilot of the program started with computer science instructors and students at Greenway High School in northwest Phoenix, Arizona. The students in particular were eager to learn and the demand was such that it led to the creation of the Cisco Networking Academy.

Packet Tracer is a cross-platform visual simulation program designed by Cisco Systems that allows users to create network topologies and imitate modern computer networks. The software allows users to simulate the configuration of Cisco routers and switches using a simulated command line interface. Packet Tracer makes use of a drag and drop user interface, allowing users to add and remove simulated network devices as they see fit. The software is mainly focused towards Certified Cisco Network Associate Academy students as an educational tool for helping them learn fundamental CCNA concepts. Students enrolled in a CCNA Academy program can freely download and use the tool free of charge for educational use. In addition to simulating certain aspects of computer networks, Packet Tracer can also be used for collaboration. As of Packet Tracer 5.0, Packet Tracer supports a multi-user system that enables multiple users to connect multiple topologies together over a computer network. Packet Tracer also allows instructors to create activities that students have to complete. Packet Tracer is often used in educational settings as a learning aid. Cisco Systems claims that Packet Tracer is useful for network experimentation.

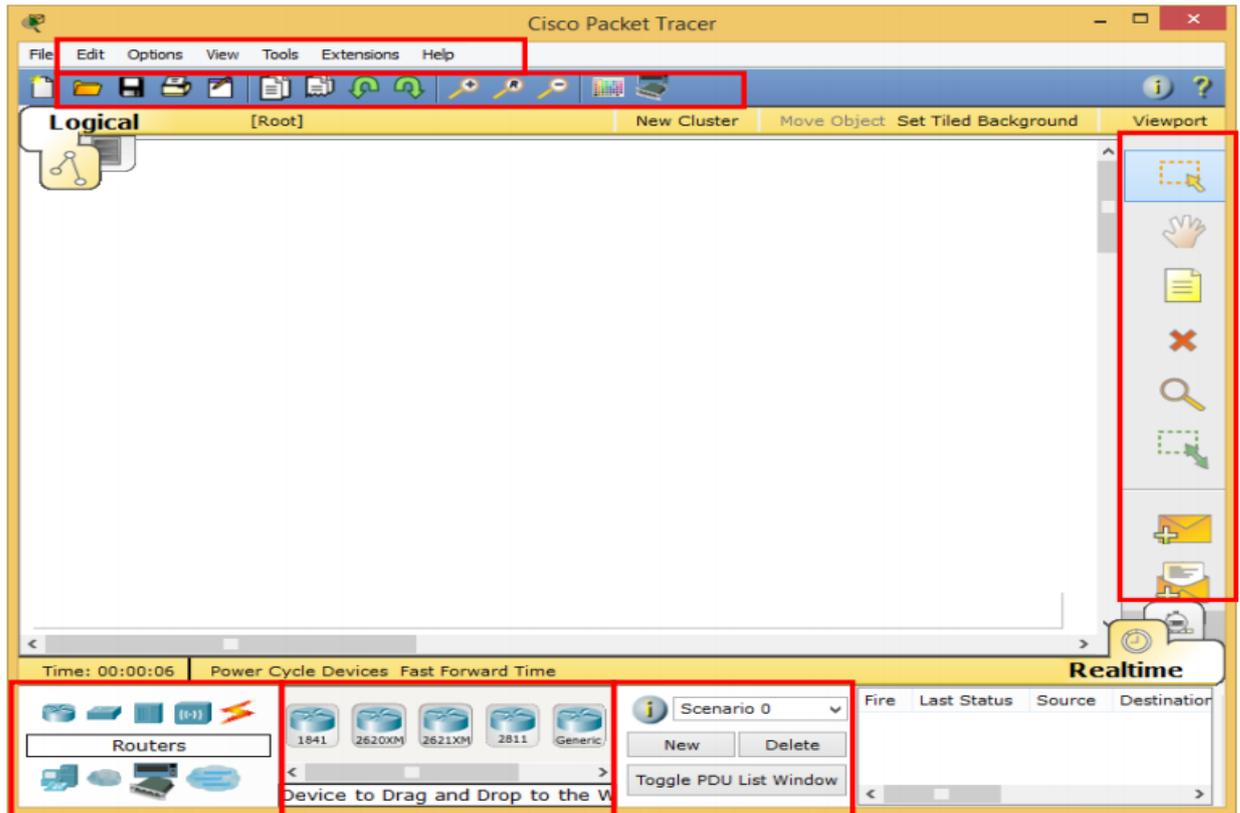


Figure 1.3. Cisco packet tracer (simulator)

Summary

This chapter has provided a clear picture of a converged, all IP communications environment, which fulfils almost all the expectations and requirements of a NGN system. Cisco Systems claims that Packet Tracer is useful for network experimentation. This “e-doing” capability is a fundamental component of learning how to configure routers and switches. The aim was to help the reader comprehend the concept of convergence, its drivers and enablers in NGN. Various issues of NGN are discussed and the current and future trends of standardization activities for NGN are presented in detail. IMS is depicted as a major enabler for achieving convergence. Some of the applications mentioned may be a bit advanced and not feasible from the perspective of Uzbekistan, they are certainly going to be deployed in near future.

2. NGN MULTIMEDIA SERVICES

As it has already been stated, the next generation networks are a vision of a converged network, meeting all the requirements for a converged universal packet network of the future. The main aim is to explain the deployment and functions of the individual components within the network intelligence and to give a brief characteristic of the individual layers of an NGN conceptual model. After introducing the first real solutions, the next generation networks are becoming a reality, not just a concept. That is why it is appropriate to look into their evolution and to outline their future trends and the open issues to be solved as well. Migration scenarios of different types of networks platforms are based on the idea to integrate TDM (Time Division Multiplexing) and IP (Internet Protocol) platforms into one converged NGN platform (from the point of network infrastructure, as well as services, Figure below).[14] The separation of processes of service control and providing from the physical network architecture and extension of telephone and multimedia services are two different NGN aspects. New concepts and architectures of new generation of ICT (Information and Communication Technology) based on converged ICT and NGN offer to operators new opportunities to implement and provide wide spectrum of multimedia services and applications.

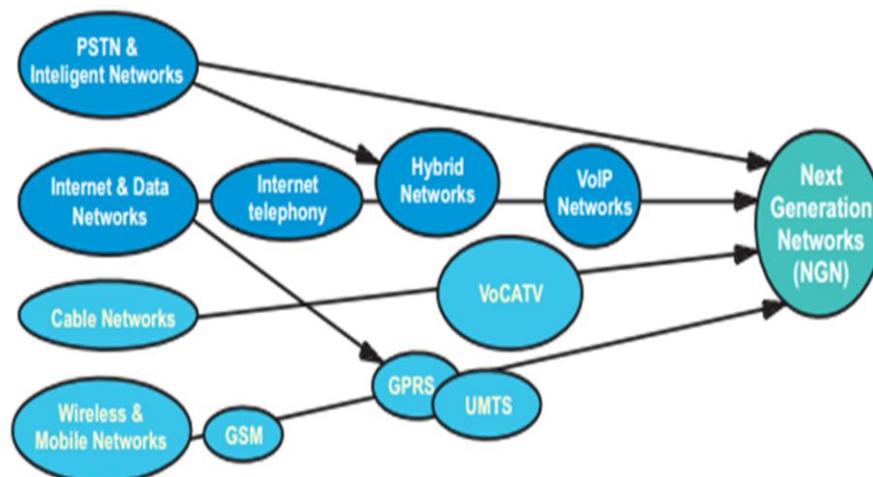


Figure 2.1. Migration scenarios

Therefore operators can move from vertical silo architecture where each type of service has dedicated access, transport, control and application infrastructure per service to horizontally oriented architecture more independent from provided services. The main idea of NGN based IPTV is to include functionalities and infrastructure required for any of multimedia NGN services specially here the IPTV type of services to NGN architecture.[18]

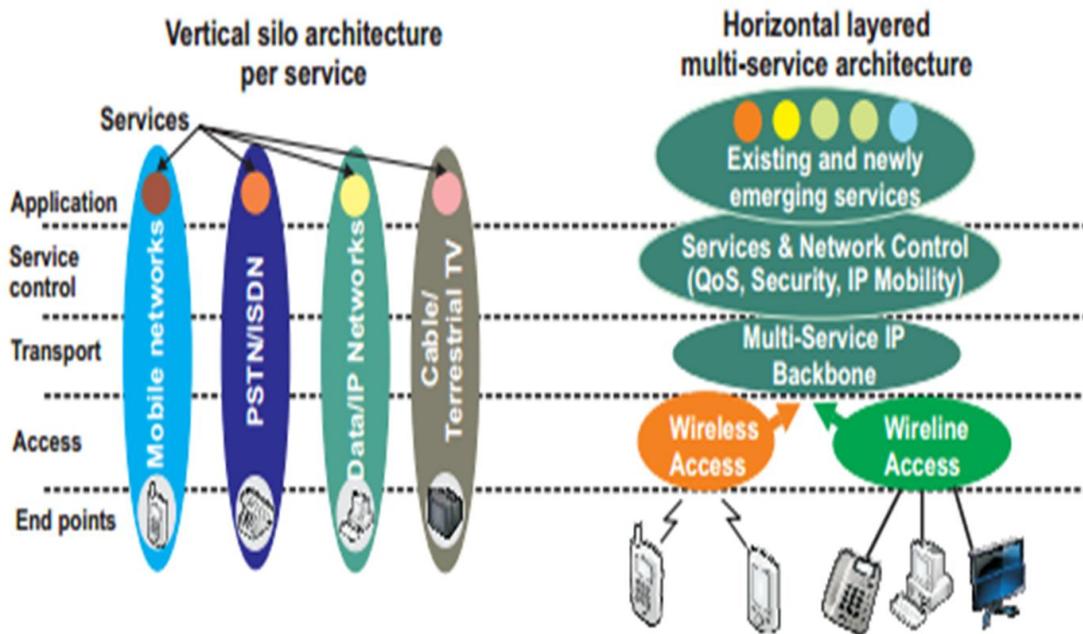


Figure 2.2. From vertical silos to horizontal NGN architecture

Table shows some of the main parameters and features of network concepts: NGN, PSTN/IN (*Public Switched Telephone Network/Intelligent Network*) and Internet (simplified and generalized interpretation).

Table 2.1.

Comparing the features of PSTN/IN, Internet and NGN

	PSTN/IN	Internet	NGN
Multimedia services	NO	YES	YES
QoS support	YES (Voice)	NO	YES
Network intelligence	YES	NO	YES
Intelligent terminal equipment	NO	YES	YES
Integred Supervision and control	YES	NO	YES
Reliability	High	low	high
Service creation	complex	ad-hoc	systematic
Simplicity of services use	medium	high	high
Modularity	low	medium	high
Time of service introduction	long	short	short
Openness of architecture	small	high	high

2.1. IPTV (Internet protocol Television)

The end user finally perceps the quality and IPTV (*Internet Protocol Television*) service portfolio, as well as the usability in order to satisfy his requirements. Several actors are responsible for the delivery of the content from their originators such as TV stations and studios but probably also from other users. In this section the IPTV domains and services are described together with the explanation of how the standardization develops from requirements to architecture. End to end chain for delivery of the IPTV content to the end user usually contains these 4 main domains that are involved in the provision of an IPTV service (Figure below);

- Content provider;
- Service provider;

- Network provider;
- End-user;

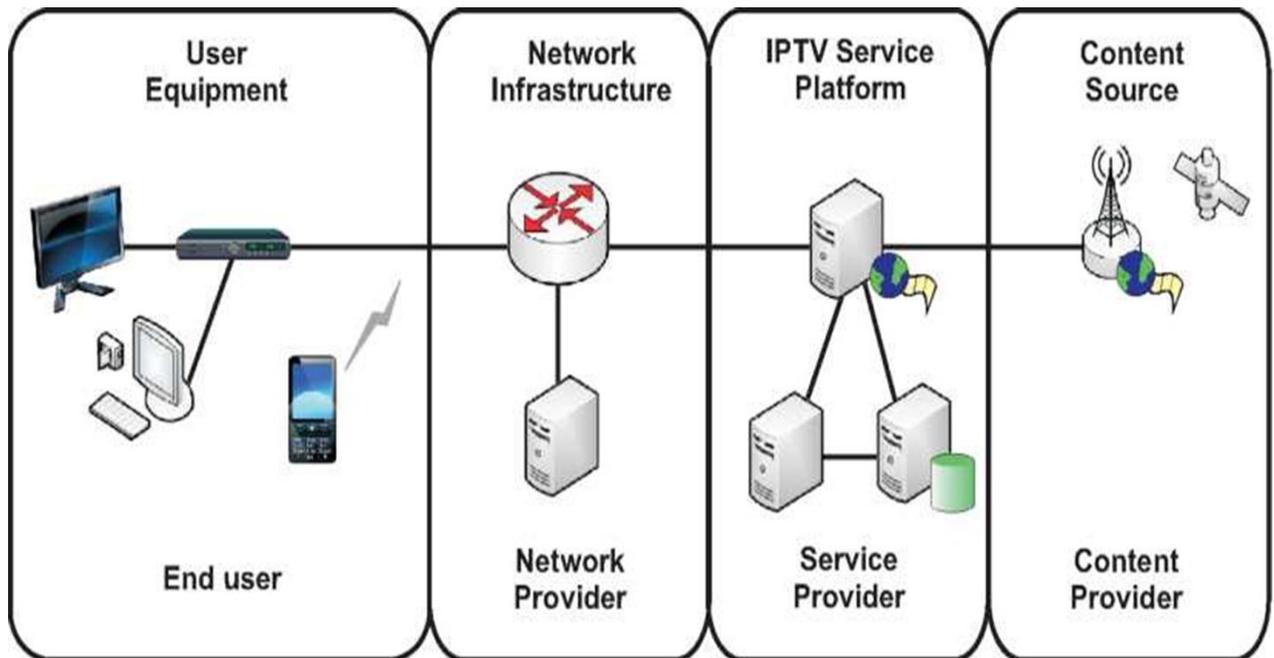


Figure 2.3. IPTV Domains

The four IPTV domains definitions could be provided by the ITU-T or ETSI TISPAN specification. Most of the standardization bodies follow the same schema to produce end to end solution specifications that apply also to the IPTV. First of all it is necessary to specify all requirements for the service but also from the UE and network capabilities point of view (stage 1)[18]. Secondly it is the specification of the functional architecture, functional entities and their task, relevant reference point among the functional entities as well as high level procedures for services (this is done usually in stage 2). In the final stage 3 it is required to conclude all details needed from the implementation perspective as for example the protocol models and detailed protocol procedures. There are two main aspects of the IPTV. First one is technological one resulting to the IPTV architecture and second one is the user's perspective aspect which can be seen from the provided IPTV services and user experience. From the user's perspective is not really important what architecture the IPTV service provider selects, but it is surely more important which services are provided. Most of the existing non-NGN

solutions provide only basic set of services like linear TV (live TV channels), *video on demand* (VoD), and some of them also PVR (*Personal Video Recording*). New NGN based IPTV solution should therefore provide much more services, features but most important also new user experience in watching TV with more interactivity, personalization, mobility and last but not least comfort in consumption of the right content in the right time and right way[13].

Architecture of non-NGN Based IPTV

The general Triple Play architecture (Figure below) usually consists of the following parts;

- Service platform domain including IPTV middleware (non-NGN);
- Transport network;
- Access network;

Home network and CPES;

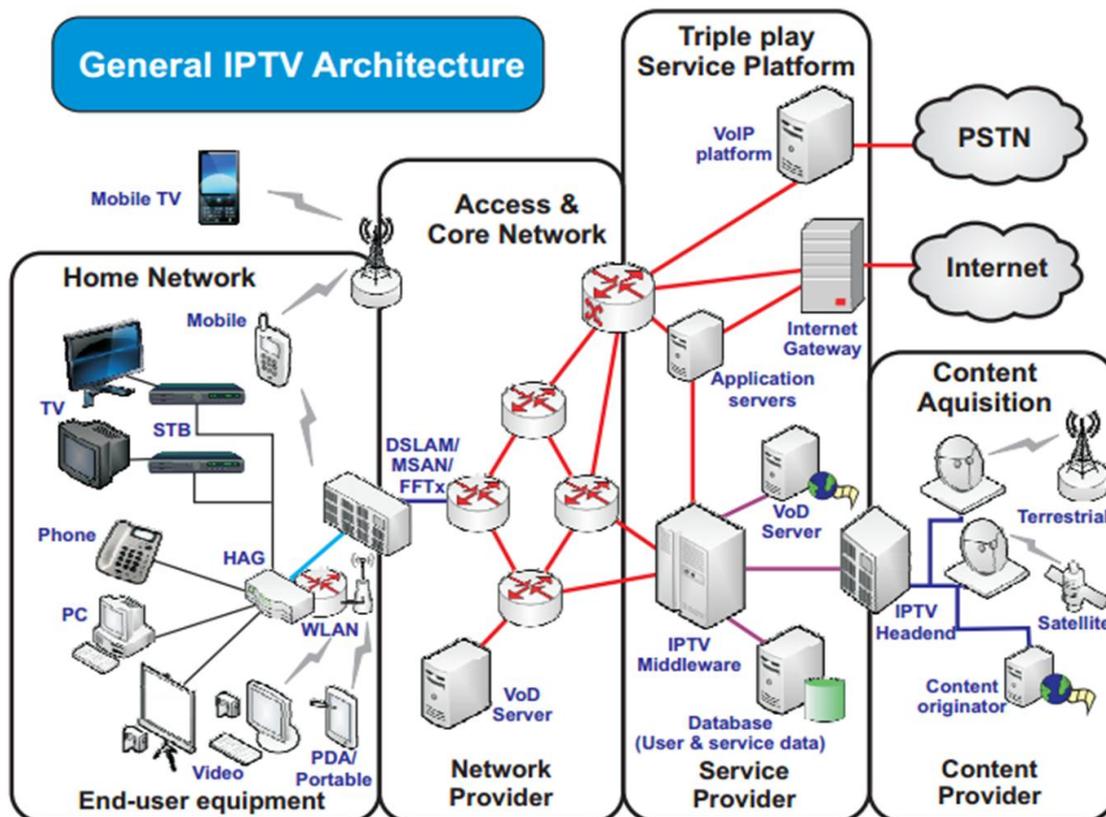


Figure 2.4. General IPTV architecture

The Triple Play service platform usually contains several less independent parts of complex service architecture:

Content acquisition subsystem which allows to receive, process, and encode content from external sources to defined media coding and encapsulation (receiver and decoders infrastructure, IPTV headend, VoD import and pre-processing). Content distribution subsystem responsible for retrieving, protecting, distributing, storing and delivering of the content by preferred way to the end user's system (user equipment).

IPTV middleware contains the application servers which control and manage the whole IPTV infrastructure (servers, databases, frontend, backend systems, interfaces to external systems e.g. OSS/BSS), users and services. Part of the application platform could also be additional IPTV applications or gateways allowing limited interaction with other systems (e.g. VoIP, NGN).

Service selection and discovery subsystem which allow the user to browse and find via user TV portal an appropriate content or service information (metadata) which he would like to watch (could be part of IPTV middleware).

VoD, nPVR or other subsystems - specialized subsystem infrastructure required for dedicated services (Video on Demand or network based personal video recording service)[21].

For the Triple Play contains tree type of services - video, voice, data - the connection to internet services and voice service platform is required (e.g. over VoIP gateway). There is no single approach to the IPTV service provisioning. Due to huge costs involved in the network equipment, operators usually follow incremental approaches to network upgrading, always relying on existing premises and procedures. Therefore the way a new NGN service is provisioned, it clearly depends on the history of the operator. Therefore there are a lot of differences from solution to solution and also to operator specific transport, access and home network design[23].

Architecture of NGN Based IPTV

The major players in any IPTV delivery chain consist of content providers, service providers, network providers, end-users. Content provider is a source of content as for example TV stations, studios, content aggregators, etc. The IPTV platform usually own by service provider has to provide all functions necessary for control and delivery of IPTV services over network infrastructure (network provider) to end user[23].

Main blocks NGN based IPTV platforms are following (Figure below):

- Application functions;
- Service control functions and User profiles;
- Media control and delivery functions;
- Supporting, management and security functions;

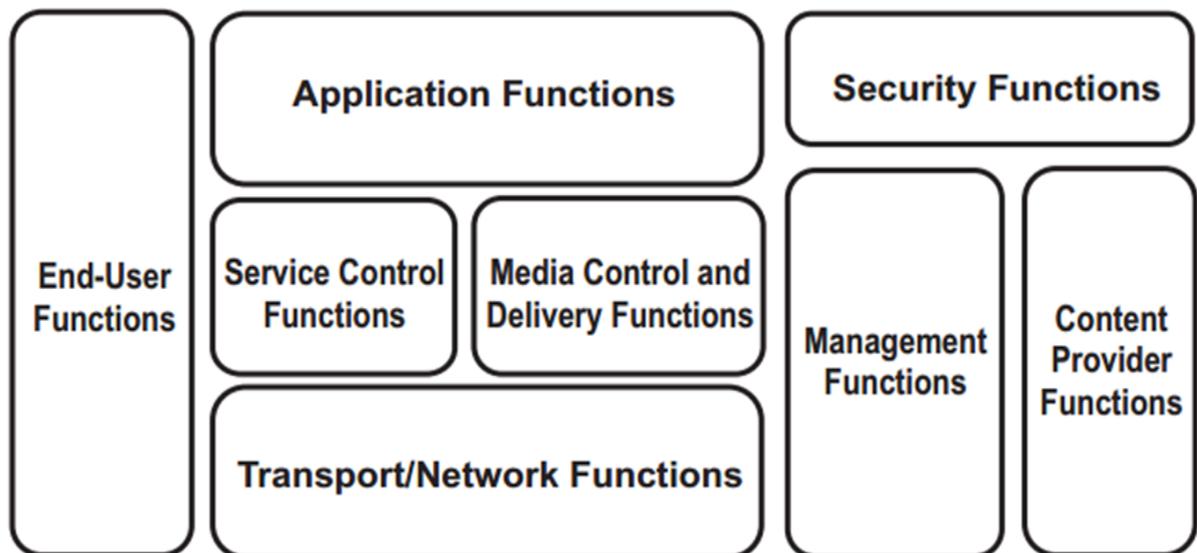


Figure 2.5. High-level architecture of NGN IPTV functional

Application functions can include several service logics of the IPTV services, mechanisms for service discovery and selection to find right services and content, also help to interact with other application and external systems.

Service Control Functions provides functionality for authentication,

authorization of service requests. This function is also responsible for the setup and control of all the IPTV services. It can also reserve resources towards transport control functions.

User Profiles contain user data and user profiles related to user's services. **Media Control and Delivery functions** has received content and media streams from content provider and then control and provide media processing, media delivery, content storing, transcoding and relaying of content.

End-User Functions represent home network and user equipment as for example end devices (e.g. TV with set-top-box, mobile, etc.) but also home networking part including Home Access Gateways. The greatest advantage of NGN based IPTV architecture is possibility integrate. IPTV services with other NGN services, reused existing NGN capabilities, better utilized resources, personalization of services and mobility[22].

HbbTV (Hybrid Broadcast Broadband TV)

HbbTV (*Hybrid Broadcast Broadband TV*) is a new industry standard providing an open and business neutral technology platform that seamlessly combines TV services delivered via broadcast with services delivered via broadband and also enables access to Internet only services for consumers using connected TVs and set-top boxes. You can find a full range of connected TVs nowadays. Every major brand has its own IPTV platform, such as Panasonic has VieraConnect or Samsung has Smart TV, they are offering a mixture of catch-up services and additional content, such as movie trailers, YouTube access, and other applications. HbbTV is similar to MHEG (*Multimedia and Hypermedia Expert Group*) it means, that an application can be delivered to the TV using a data carousel, too. That application can later work with or load additional content from the internet. HbbTV is also an ETSI standard (TS 102 796).



Figure 2.6 .HbbTV doesn't depend on a particular broadcast link or on a particular IP link –it'll work with either.

HbbTV applications, however, can be downloaded from an application portal in the TV and delivered solely over the internet. So it can support content services from providers who don't own broadcast channels[23].

Other advantage is that, it is based on technologies that are well known to web developers, mainly CE-HTML, a specification that includes XHTML, Ajax, CSS, and Javascript (jQuery). Because of that, content providers can build and release their apps more quickly. HbbTV's Javascript API was extended to provide TV functionality, such as handling channel changes. This standard expects that host TVs have a minimum display resolution of 1280x720.

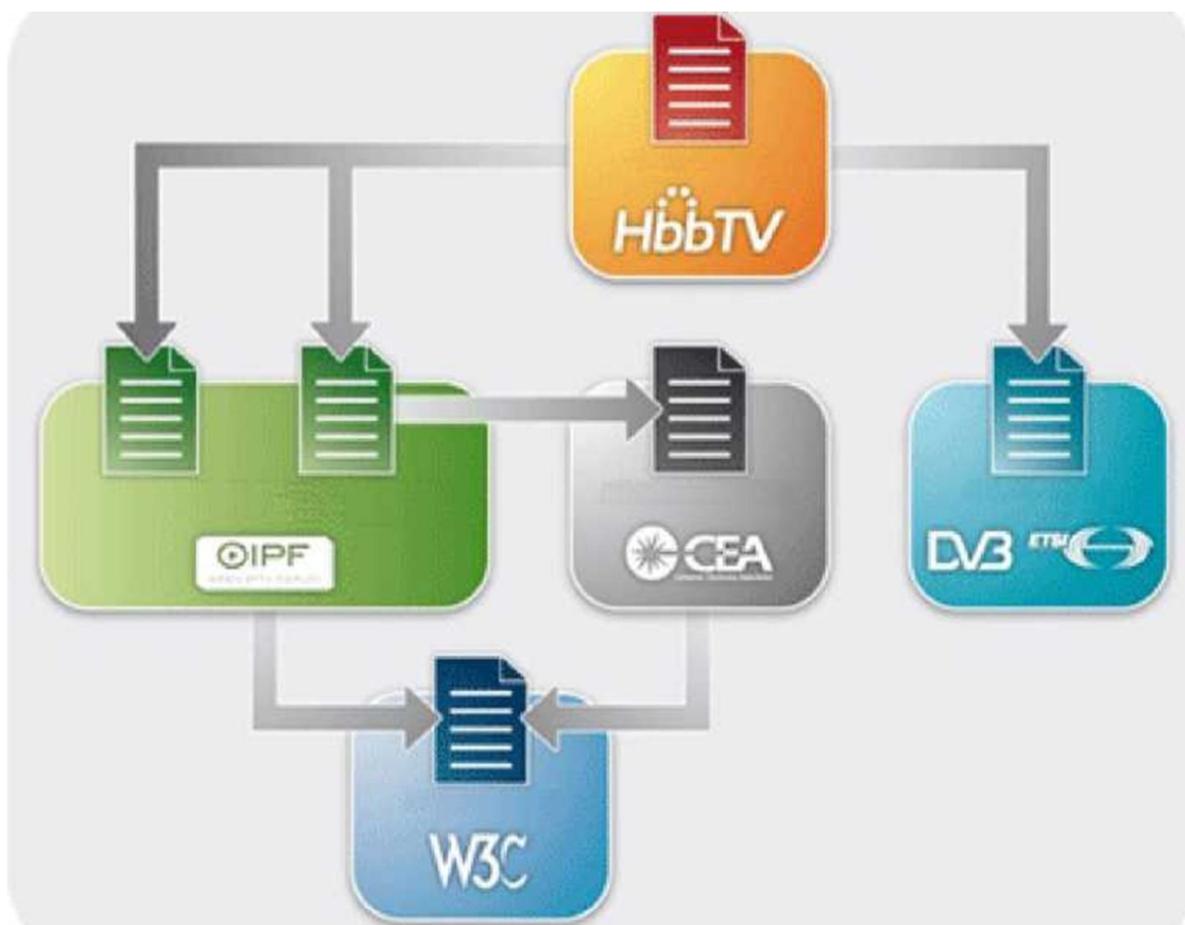


Figure 2.7. The HbbTV specification corrals standards , including CE-HTML, DVB and JavaScript, from many existing organizations , such as the Open IPTV Forum and the W3C.

The HbbTV specification also contains a contribution from the Open IPTV Forum, which has defined a set of audio and video formats that should be supported. HbbTV relies on the AVC (H.264) codec for both standard and high-definition video, with either E-AC3 or HE-AAC for audio. Audio streaming services use either MP3 or HE-AAC.



Figure 2.8. IPTV Internet Protocol Television

Services

Services delivered through HbbTV include.

- enhanced teletext;
- catch-up s;
- ervices and video-on-demand (VOD);

Catch up TV is a term used to describe VOD in which TV shows are available for a period of days after the original broadcast.

- Electronic program guides (EPG);

provide users of television, radio, and other media applications with continuously updated menus displaying broadcast programming or scheduling information for current and upcoming programming

- interactive advertising;
- personalisation;
- personal video recording (PVR);

similar to a VCR but records television data in digital format

- voting and games;
- social networking, other multimedia applications;

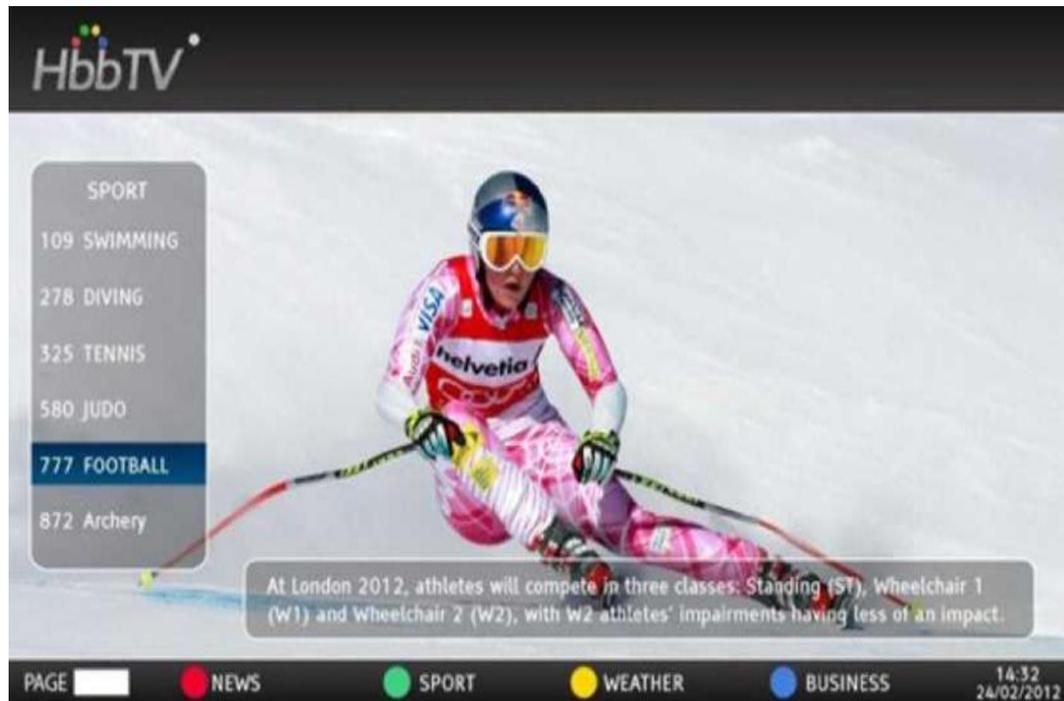


Figure 2.9. Example of HbbTv services

2.2. VoIP (Voice Internet Protocol)

This chapter presents a high-level overview of several basic models that you can use in designing your IP telephony network. This overview provides some guidance with respect to when and why a particular design should be selected. Subsequent chapters delve into each network model in greater detail, beginning with the simplest model and building to increasingly complexity models. This chapter includes the following major sections:

- General Design Models;
- Single-Site Model;
- Multiple Sites with Independent Call Processing;
- Multisite IP WAN with Distributed Call Processing;
- Multisite IP WAN with Centralized Call Processing;

- General Design Models;
- Figure 2.10 provides a composite scenario that illustrates the goals of the network design models discussed in this guide. This scenario represents what is possible with Cisco CallManager Release.

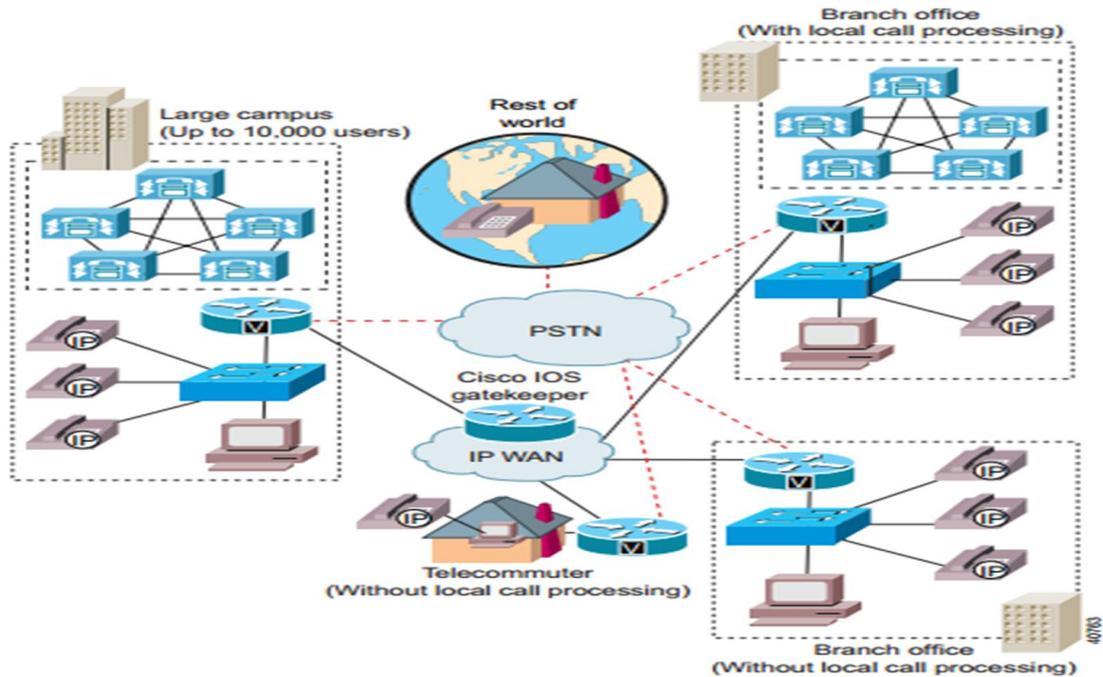


Figure 2.10. Composite model

The overall goals of an IP telephony network are as follows:

- End-to-end IP telephony;
- IP WAN as the primary voice path with the Public Switched Telephone Network (PSTN) as the secondary voice path between sites;
- Lower total cost of ownership with greater flexibility;
- Enabling of new applications;

For IP telephony networks based on Cisco CallManager Release 3.0(5), there are four general design models that apply to the majority of implementations:

- Single-Site Model, page1-3;
- Multiple Sites with Independent Call Processing;
- Multisite IP WAN with Distributed Call Processing;

- Multisite IP WAN with Centralized Call Processing, page1-10;
- The following sections summarize the design goals and implementation guidelines for each of these models;

Single-Site Model

Figure 2.11 illustrates the model for an IP telephony network within a single campus or site.

Single-Site Model

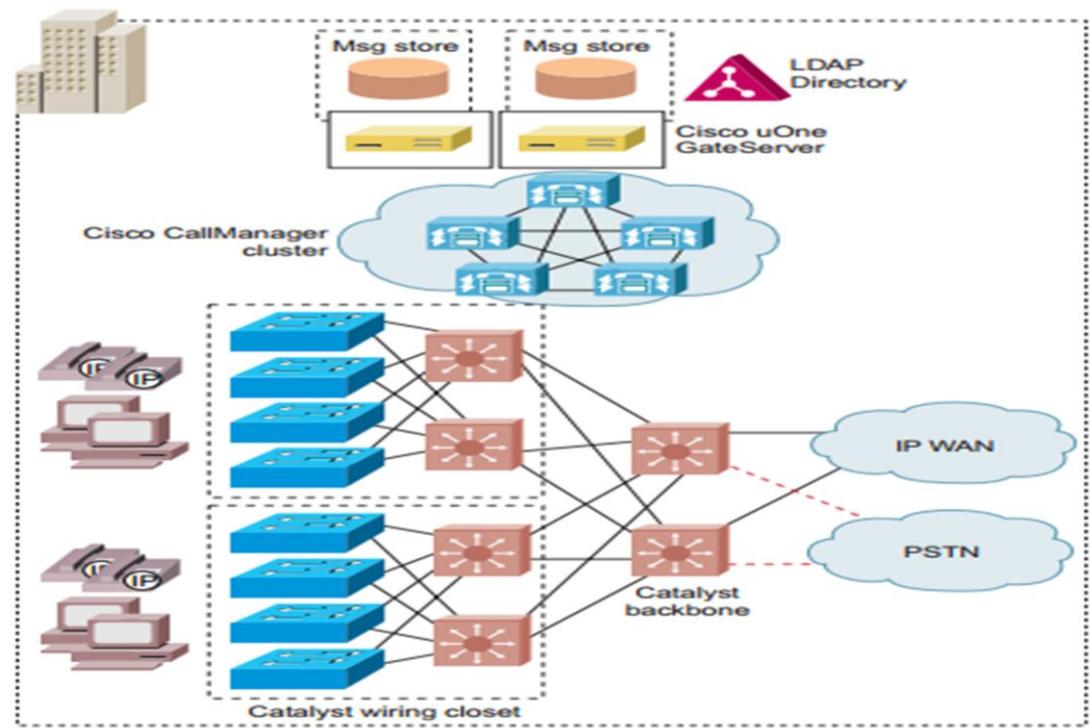


Figure 2.11. Single-Site Model

The single-site model has the following design characteristics;

- Single Cisco CallManager or Cisco CallManager cluster;
- Maximum of 10,000 users per cluster;
- Maximum of eight servers in a Cisco CallManager cluster (four servers for primary call processing, two for backup call processing, one database publisher, and one TFTP server);
- Maximum of 2,500 users registered with a Cisco CallManager at any time.
- PSTN only for all external calls;

- Digital signal processor (DSP) resources for conferencing;
- Voice mail and unified messaging components;
- G.711 codec for all IP phone calls (80 kbps of IP bandwidth per call, uncompressed);
- To guarantee voice quality, use Cisco LAN switches with a minimum of two queues. See Chapter 2, “ Campus Infrastructure Considerations,” for more detail;
- Multiple Sites with Independent Call Processing:[1]

Figure 2.12 illustrates the model for multiple, isolated sites that are not connected by an IP WAN. In this model, each site has its own Cisco CallManager or Cisco CallManager cluster to handle call processing for that site.

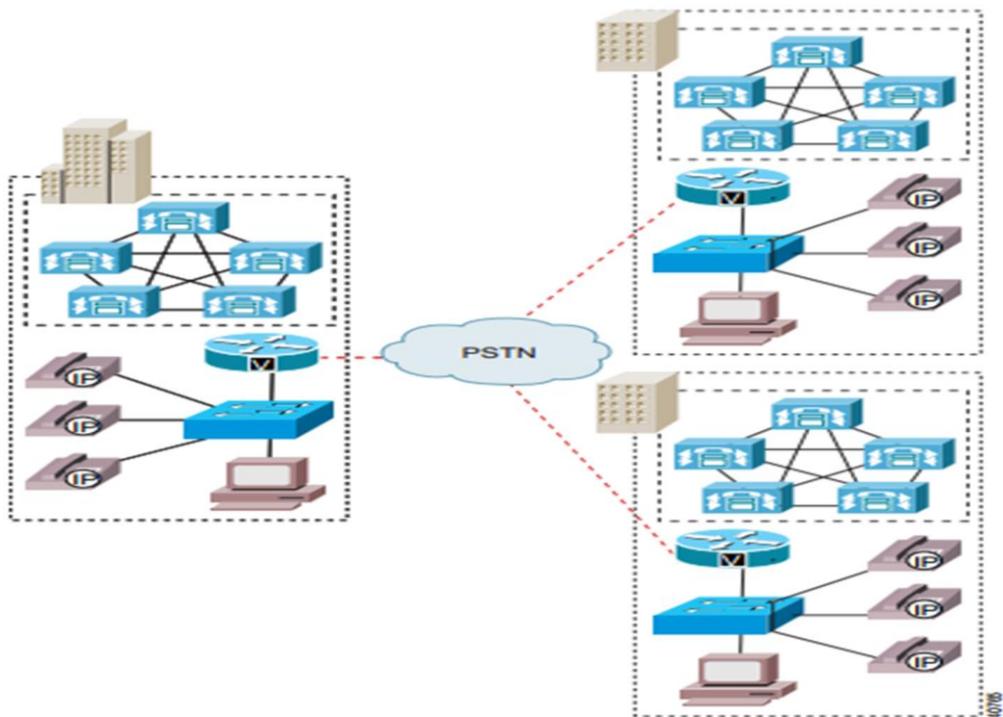


Figure 2.12. Multiple Independent Sites

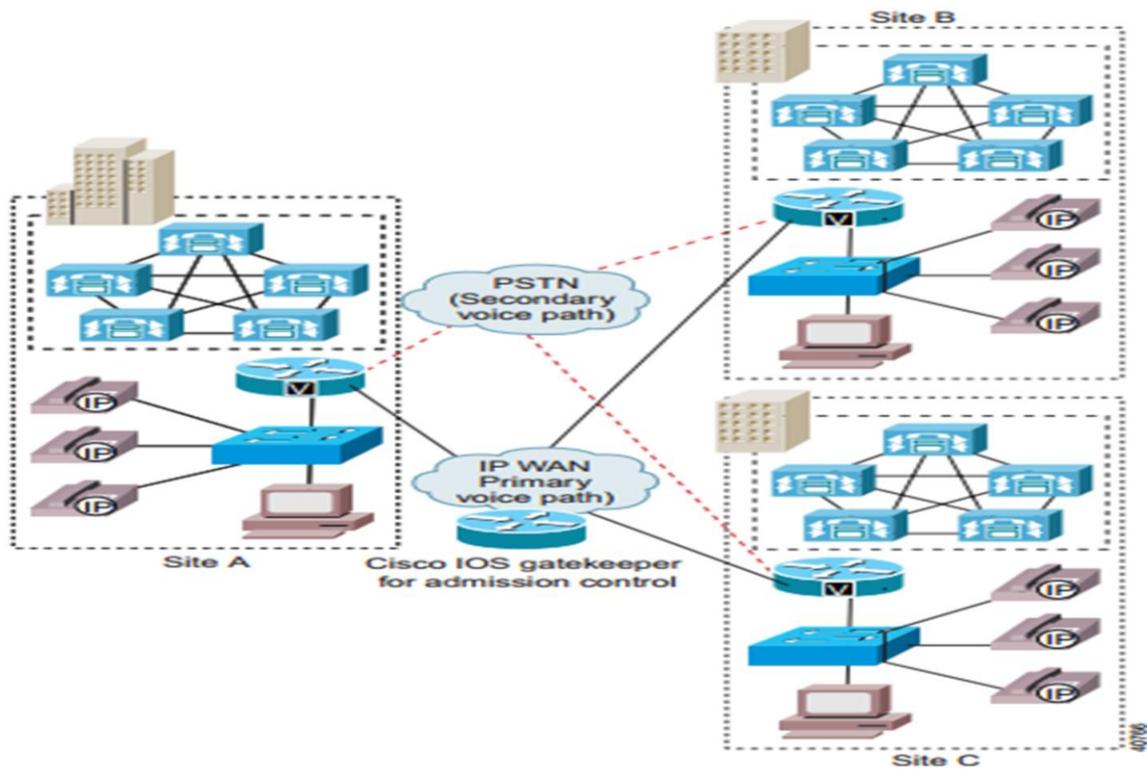
The model for independent multiple sites has the following design characteri:

- Cisco CallManager or Cisco CallManager cluster at each site to provide scalable call control;
- Maximum of 10.000 IP phones per cluster;
- No limit to number of cluster.;

- Use of PSTN for networking multiple sites and for all external calls;
- DSP resources for conferencing at each site;
- Voice message or unified messaging components at each site;
- Voice compression not required;

Multiple IP WAN with distributed Call processing

Figure 2.13 illustrates the model for multiple sites with distributed call



processing.

Figure 2.13. Multisite Model with Distributed Call Processing

The multisite IP WAN with distributed call processing has the following design characteristics.

- Cisco CallManager or Cisco CallManager cluster at each location (10,000 users maximum per site);
- Cisco CallManager clusters are confined to a single campus and may not span the WAN;
- IP WAN as the primary voice path between sites, with the PSTN as the

secondary voice path;

- Transparent use of the PSTN if the IP WAN is unavailable;
- Cisco IOS gatekeeper for E.164 address resolution;
- Cisco IOS gatekeeper for admission control to the IP WAN;
- Maximum of 100 sites interconnected across the IP WAN using hub and spoke topologies;
- Compressed voice calls supported across the IP WAN;
- Single WAN codec supported;
- DSP resources for conferencing and WAN transcoding at each site;
- Voice mail and unified messaging components at each site;

Minimum bandwidth requirement for voice and data traffic is 56 kbps. For voice, interactive video, and data, the minimum requirement is 768 kbps. In each case, the bandwidth allocated to voice, video, and data should not exceed 75% of the total capacity.

- Remote sites can use Cisco IOS as well as gateways based on the Skinny Gateway Protocol.

Multisite IP WAN with centralized Call Processing

Figure 2.14 multisite illustrates the model for multiple sites with centralized call processing.

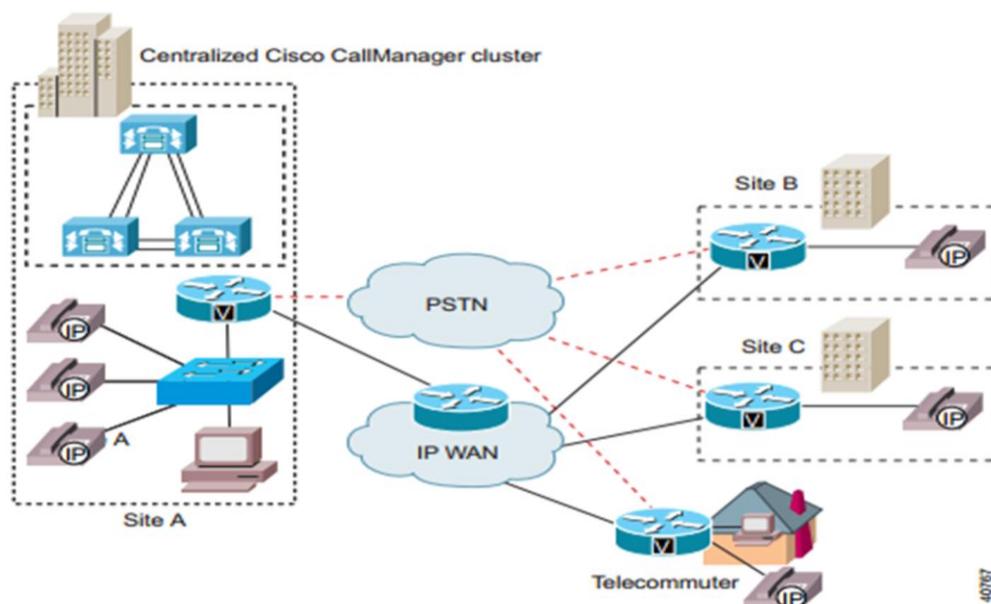


Figure 2.14. Multisite Model with Centralized Call Processing.

2.3. NGN protocols

The best way to show the functions of individual protocols in the hierarchy of NGN protocols platforms supporting voice transport over the packet networks or controlling of elements in NGN architecture is shown on Figure below with depicting the individual protocols and the OSI reference model layers they belong to.

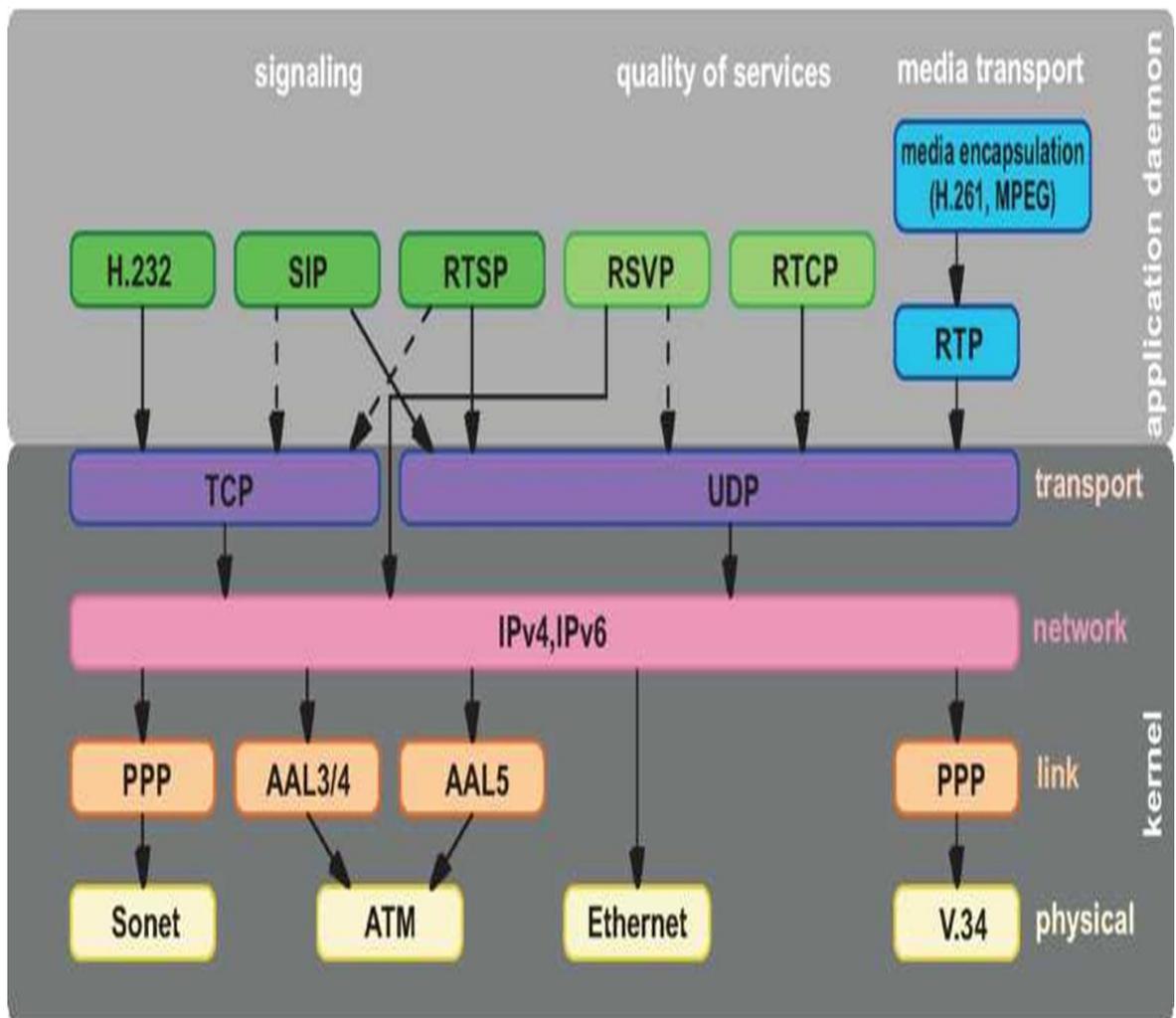


Figure 2.15. Protocols for NGN

The protocols for the converged technologies and NGN platform can be divided into the following groups.

- call control protocols (VoP signaling from the telecommunication point of view): SIP/SDP, H.323;
- media gateway control protocols (components of the distributed VoP architecture): MGCP, Megaco/H.248 (protocol approved by both IETF and ITU-T);
- protocols for signaling transport: SIGTRAN, BICC, SIP-T, SIP-I;
- transport protocols: RTP, RTCP (in the sense of media transfer not RM OSI, as otherwise TCP/IP or UDP/IP is used for all);
- protocols for QoS support: RSVP, RTCP (RTCP is a transport one, but allows QoS support as well);
- Other support protocols;
- DHCP, ENUM, DSN, COPS;
- RTSP (Real-Time streaming Protocol)- protocol for creation of streams in the real time;

Fundamentals of NGN protocols

SIP

Session Initiation Protocol (SIP) is an application-layer control protocol that handles the setup, modification, and tear-down of multimedia sessions. Media can be added to (and removed from) an existing session. SIP is used in combination with other protocols to describe the session characteristics to potential session participants. SIP is based on a request and response transaction model similar to HTTP. Each transaction consists of a request that invokes a particular method or a function on the server and at least one response.

SIP supports five facets of establishing and terminating multimedia communications:

- User location: determination of the end system to be used for communication;
- User availability: determination of the willingness of the called party to engage

in communications;

- User capabilities: determination of the media and media parameters to be used;
- Session setup: "ringing", establishment of session parameters at both called and calling party;
- Session management: including transfer and termination of sessions, modifying session parameters, and invoking services;

SIP is a text-based protocol suggested and standardized in RFC 3261. SIP has been proposed as a part of a unit based of following protocols.

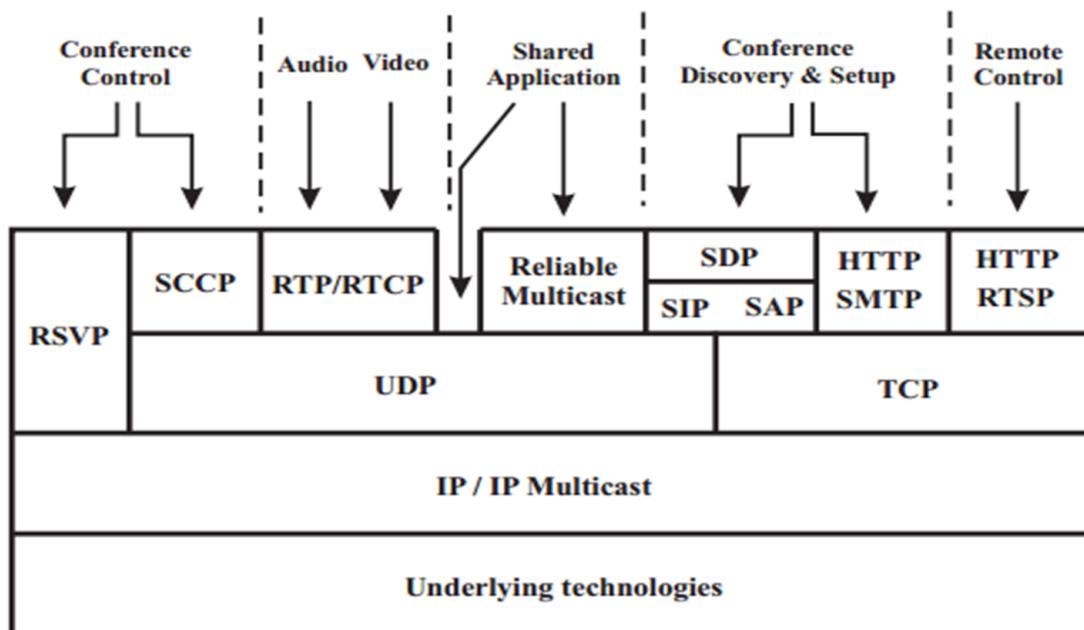


Figure 2.16. SIP protocol stack

Consequential protocols

A lot of SIP functions depend from other protocols. SIP defines establishment, termination and call modification and SIP use other protocols as *Real-time Transport Protocol (RTP)* for transporting real-time data and providing QoS feedback, the *Real-Time Streaming Protocol (RTSP)* for controlling delivery of streaming media, the *Media Gateway Control Protocol (MEGACO)* for controlling gateways to the *Public Switched Telephone Network (PSTN)*, and the

Session Description Protocol (SDP) for describing multimedia sessions. Therefore, SIP should be used in conjunction with other protocols in order to provide complete services to the users. However, the basic functionality and operation of SIP does not depend on any of these protocols.

SDP

SDP (*Session Description Protocol*) [4] is intended for describing multimedia sessions for the purposes of session announcement, session invitation, and other forms of multimedia session initiation. When initiating multimedia teleconferences, voice-over-IP calls, streaming of video, or other sessions, there is a requirement to convey media details, transport addresses, and other session description meta data to the participants. SDP provides a standard representation for such information, irrespective of how that information is transported. SDP is intended to be general purpose so that it can be used in a wide range of network environments and applications. However, it is not intended to support negotiation of session content or media encodings: this is viewed as outside the scope of session description.

An SDP session description includes the following:

- Session name and purpose;
- Time(s) the session is active;
- The media comprising the session;
- Information needed to receive those media (addresses, ports, formats, etc.)

As resources necessary to participate in a session may be limited, some additional information may also be desirable;

- Information about the bandwidth to be used by the session;
- Contact information for the person responsible for the session;

RTP

The goal of this part is to present **RTP** (*Real Time Transport Protocol*) in the way that it will be easily understandable also by a beginner. It contains deeper description of the main RTP protocol, but also protocols that stand next to RTP,

cooperate with it and fill its gaps. The structure of RTP will be shown, to introduce the protocol body to the reader with a close view of its parts and their functions. RTP provides end-to-end delivery services suitable for applications transmitting real-time data, such as audio, video or simulation data, over multicast or unicast network services. Real-time means that not only correct results are required, but also a sufficient time in which the result is delivered. That is why the delivery of the audio or video data is typically delay sensitive. According to this RTP use timestamp and control mechanisms for synchronizing different streams with timing properties.

RTCP

RTCP (*Real-time Transport Control Protocol*) is an application layer protocol designed to control of data delivery in real-time and to measure the QoS. It is defined in RFC 3550 published in July 2003. RTP protocol uses the RTCP protocol, which transports the following additional information for the management of the session. RTCP is based on the periodic transmission of control packets to all participants in the session. The underlying protocol must provide multiplexing of the data and control packets, like UDP protocol that allows the multiplexing of RTP data packets and RTCP control packets. RTCP protocol requires the sending of information periodically by the participants of the session. RTP packets only transport user's data, whereas RTCP packets only transport in real time the supervision.

Protocol RTCP performs these principal functions:

Provide the information about the quality of the session (QOS) by means of feedback, which include the number of lost packets, the time return ticket and the gigue. Keep a trace of all the participants by a persistent transport-level identifier called CNAME (Canonical Name). Because SSRC (Synchronization Source Identifier) may change if a conflict or program restart occurs. Control the media flow and adapt it to all the participants of the RTP session. By having each participant send its control packets to all the others, each can independently observe the number of participants. This information is used to calculate the rate

at which the packets are sent.

DIAMETER

DIAMETER is a member of “AAA” protocols collection, derived from its predecessor RADIUS protocol. It is a peer to peer protocol , used for handling service requests such as user validation, network resource control, connection and session management, wireless or roaming charging, billing applications etc. Diameter sessions consist of exchange of commands and AVPs between servers and clients and unlike Radius, uses peer to peer architecture rather than more classic client/server scheme. Each node may initiate a message (request) at any time, as example, server may abort a service to specific user. Diameter is defined in terms of base protocol and a set of applications. This design allows protocol to be extended for new access technologies. The base protocol provides basic mechanism for reliable transport, delivery and error handling.

MEGACO/H.248

This protocol has been established to cover the need of IP networks and services to interoperate with traditional networks (e.g. PSTN) and provide the same services over both types of networks (IP, Traditional). This enables separation of call control from media conversion. Megaco/H.248 is defined as master / slave architecture based protocol which is used for communication between MGC (*Media Gateway Controller*, sometimes called a call agent or softswitch, which dictates the service logic of that traffic) and one or more decomposed MGs (*Media Gateways*), which converts circuit-switched voice to packet-based traffic. Megaco/H.248 instructs an MG to connect streams coming from outside a packet or cell data network onto a packet or cell stream such as RTP. Megaco/H.248 is similar to MGCP from an architectural standpoint and the controller-to-gateway relationship, but Megaco/H.248 supports a broader range of networks, such as ATM.

SIGTRAN

In view of functionality and performance the user make high demands on modern telecommunication networks. Using IP (*Internet Protocol*) signaling

messages will be transmitted over TCP (*Transmission Control Protocol*) or UDP (*User Datagram Protocol*). These transport protocols are not designed to meet the requirements given by a signaling system used in a circuit switched network like PSTN/ISDN (*Public Switched Telephone Network/Integrated Services Digital Network*). So the working group SIGTRAN was founded by the IETF (*Internet Engineering Task Force*) to develop a new protocol, based on IP, in consideration of given requirements by the existing switched telephone network.

This protocol, named SCTP (*Stream Control Transmission Protocol*) has some advantages in comparison to TCP. The SCTP offers a fundament to initiate and run secured transport connections using IP networks to transmit signaling information. Based on SCTP, several adaptation layers enable the transmission of upper layer protocols, i.e. ISUP (*ISDN User Part*), SCCP (*Signaling Connection Control Part*) and DSS1 (*Digital Subscriber System No. 1*).

Supporting NGN protocols

DHCP

DHCP (*Dynamic Host Configuration Protocol*) evolved from the BOOT protocol (BOOTP). Both protocols are described in RFC 2131 (DHCP) and RFC (BOOTP). DHCP includes all features known from BOOTP, that means an *Internet Software Consortium* (ISC) DHCP includes a BOOTP server and additional features along with a dynamic address assignment. Both protocols act for IP address assignment to nodes. Therefore, the un-configured IP node sends a request for an IP address to a DHCP server. Then this DHCP server assigns an IP address to the client. Furthermore, this answer includes e.g. domain-name, IP address of the name-server or IP address from a router. The transmission of all configuration parameters will be proceeded automatically, depending of the chosen method.

DNS

The DNS (*Domain Name System*) protocol is used to link IP addresses with domain names. Usually, it is more convenient for people to remember names (ngnlab.eu) than IP addresses (147.175.103.213). IP addresses are required by the

third layer of the network model to deliver the application data through networks. This chapter will highlight SIP/IMS specific use of DNS and not introduce the protocol itself.

Besides only storing a mapping between IP address and domain name, DNS contains additional information using various record types. DNS can handle for example certificate records, location information records, service information records and much more.

HTTP

The *Hypertext Transfer Protocol* (HTTP) is application protocol using the request response mechanisms and is one of the most used protocols on Internet for web services. A client sends a request to the server in the form of a request method, URI, and protocol version, followed by a MIME-like message (*Multipurpose Internet Mail Extensions*) containing request parameters and body content over a TCP connection with a server (HTTP session). Server is replay with response containing status line including the message's protocol version and a success or error code, followed by a MIME-like message containing server information, metadata and body content.

SIGTRAN

In view of functionality and performance the user make high demands on modern telecommunication networks. Using IP (*Internet Protocol*) signaling messages will be transmitted over TCP (*Transmission Control Protocol*) or UDP (*User Datagram Protocol*). These transport protocols are not designed to meet the requirements given by a signaling system used in a circuit switched network like PSTN/ISDN (*Public Switched Telephone Network/Integrated Services Digital Network*). So the working group SIGTRAN was founded by the IETF (*Internet Engineering Task Force*) to develop a new protocol, based on IP, in consideration of given requirements by the existing switched telephone network.

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and run secured transport connections using IP networks to transmit signaling information. Based on SCTP, several adaptation layers enable the transmission of upper layer protocols, i.e. ISUP (*ISDN User Part*), SCCP (*Signaling Connection Control Part*) and DSS1 (*Digital Subscriber System No. 1*).

Multimedia Services Control Protocol

MPEG 4

MPEG-4 and H.264 have a common heritage within the ISO and ITU standards committees. As a result the overall coding approach is quite similar. Both algorithms are based on a common heritage of DCT based, hybrid image coding, first used in H.261 and MPEG-1. A number of comparison tests have been performed between H.264 and MPEG-2/MPEG-4 on standard MPEG test material. For standard resolution (704x480, 60 Hz interlaced) video sequences to achieve a PSNR level of 28, NBA must be coded at a rate of 5 Mbps using MPEG-2, but only 1.8 Mbps using H.264.

RTSP

Real-Time Streaming protocol (RTSP) is an application-level protocol for control over the delivery of data with real-time properties and its goal is streaming of multimedia over multicast and unicast in "one to many" applications. RTSP provides an extensible framework to enable controlled, on-demand delivery of real-time data, such as audio and video. Sources of data can include both live data feeds (e.g. Live TV channels) and stored clips (e.g. Video On Demand). RTSP establishes and controls single or several data delivery time synchronized media sessions, provide a means for choosing delivery channels such as UDP, multicast UDP and TCP, and provide a means for choosing delivery mechanisms based upon RTP and control mechanism of streams upon RTCP. RTSP is not tied to RTP and RTCP. There is no notion of an RTSP connection - instead, a server maintains a session labeled by an identifier. An RTSP session is in no way tied to a transport level connection such as a TCP connection. During an RTSP session, an RTSP client may open and close many reliable transport

connections to the server to issue RTSP requests. Alternatively, it may use a connectionless transport protocol such as UDP. The streams controlled by RTSP may use RTP, but the operation of RTSP does not depend on the transport mechanism used to carry continuous media. The protocol is intentionally similar in syntax and operation to HTTP/1.1 so that extension mechanisms to http can in most cases also be added to RTSP.

IGMP

Internet Group Management Protocol (IGMP), allows Internet hosts to participate in multicasting. IGMP allows users to announce their intention to join particular multicast groups. These groups are identified by their unique Class-D IP addresses. When a workstation wants to participate in a multicast group, it sends an IGMP “join” message to its local router. If multiple routers exist on a single segment, they can mutually elect a “*Designated Router (DR)*” to manage all of the IGMP messages for that segment. After a router receives one or more “joins” for a specific group, the router will forward any packets destined for that group to the appropriate interface. The router should only forward one copy of the data packet per interface.

If multiple receivers exist on a single interface they will all receive the same information by monitoring common multicast MAC and IP addresses. If the multicast group has receivers spread over several router interfaces, the router must replicate the packet and deliver a copy to each interface that contains registered users. This type of transmission activity can be immense, which is why IGMP is highly useful as a stateful protocol. The designated router regularly verifies that the attached workstations want to continue to participate in their respective multicast groups. The designated router sends periodic “queries” to the receivers.

These queries are transmitted to a well-known multicast address (224.0.0.1) that is monitored by all systems. If the receivers are still interested in that particular multicast group, they will respond with a “membership report” message. When the router stops seeing responses to queries, it will delete the appropriate group from its forwarding table and other

Summary

Main summary of this chapter is to acquire basic knowledge in the area of Next Generation Network architectures and network components from point of current and future platforms. Participants also become familiar with mobile and optical communication technologies as well as with technologies for digital video delivery such as DVB and IPTV systems. Moreover, it will dispose with knowledge about state of the art technologies like content delivery networks (CDN) and hybrid broadband broadcast television (HBB TV) systems.

3. MODELING NEXT GENERATION CONVERGENCE NETWORK USING A CISCO PACKET TRACER SOFTWARE.

Cisco networking academy global education initiative from Cisco Systems, offers networking programs, like the (Cisco Certified Network Associate) CCNA and (Cisco Certified Network Professional) CCNP courses, which prepare students for the certification exams of the same name, and other computer-related courses. Also see History of virtual learning environments for how Cisco Networking Academy has developed since 1997 relative to others within the VLE community. Courses are available in approximately 9,000 local academies, in over 165 different countries. As of 2010, there were over 900,000 active students (defined as students currently enrolled, students enrolled in a future course, and students who were enrolled in a course during the last five months).

3.1. The model of NGN architecture on Cisco Packet Tracer software

Firstly, all devices of NGN network were located in Cisco Packet Tracer software.

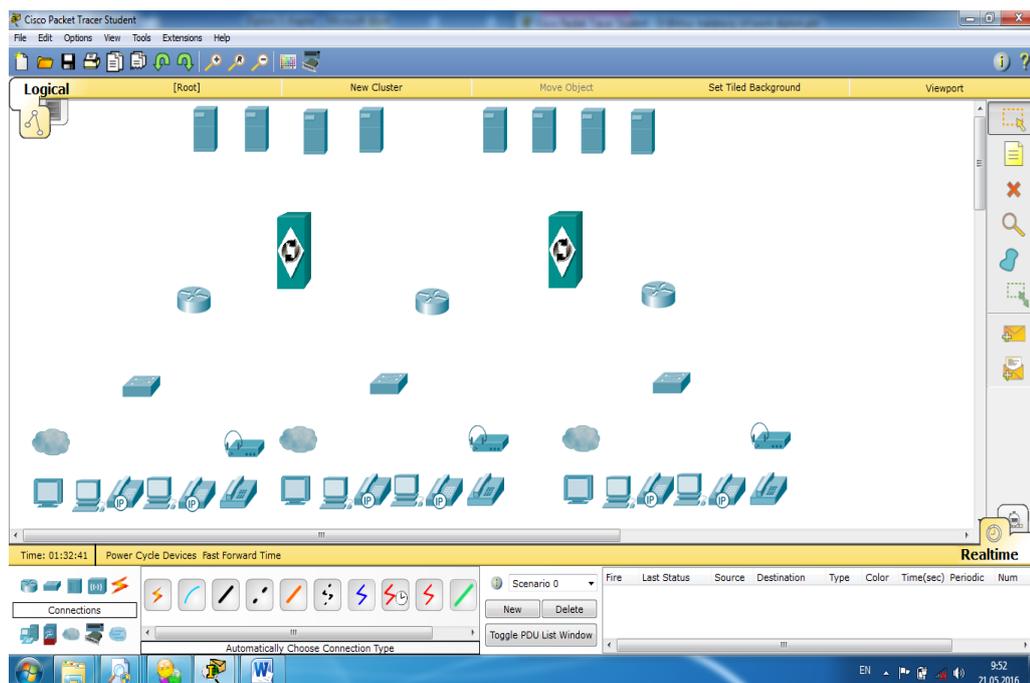


Figure 3.1. All devices of NGN in CPT

For constructing NGN network, firstly, we put necessary devices on

Cisco Packet Tracer software. In this case, we need NGN architecture which consist of 4 layers: 1-layer access, 2-layer transport, 3-layer control services, 4-layer application. After that we put devices on each layer. 1-layer: PC, ip-phone, analogue phone, IPTV, IAD. 2-layer transport devices: switch, router, L3 switch. 3-layer control device Softswitch. 4-layer application devices: Servers such as AAA RADIUS server, DNS server,HTTP server, FTP server, SNTP server Email server, TUIT.uz server, TI.uz server and others.

In next step, we provide connection between these devices. For providing connection, we select different types of cable such as fiber, coaxial, copper and ect. to connect different kind of devices with each other. For example, between PC and switch, we use copper straight-through cable; for connecting router with switch, we use copper cross-over or fiber cables.

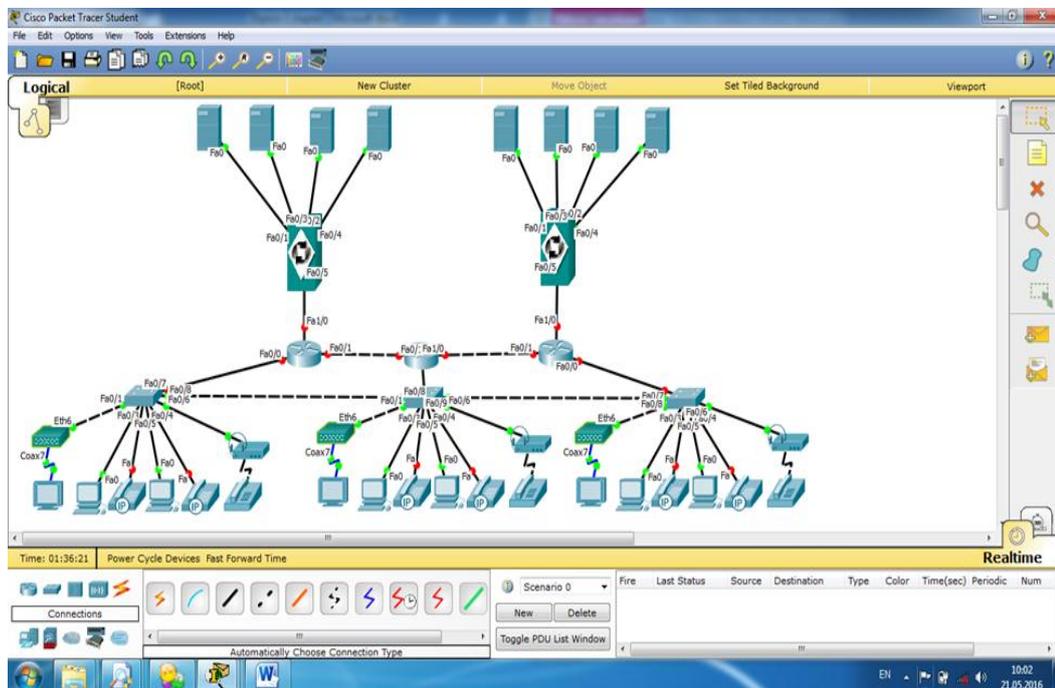


Figure 3.2. Cisco packet tracer’s behind devices connect

Router configuration

We click “CLI” menu and push “ENTER” on Router1and we wait for loading software.

Router> enable

Router#?

<1-99>	Session number to resume
auto	Exec level Automation
clear	Reset functions
clock	Manage the system clock
configure	Enter configuration mode
connect	Open a terminal connection
copy	Copy from one file to another
debug	Debugging functions (see also 'undebug')
delete	Delete a file
dir	List files on a filesystem
disable	Turn off privileged commands
disconnect	Disconnect an existing network connection
enable	Turn on privileged commands
exit	Exit from the EXEC
logout	Exit from the EXEC
more	Display the contents of a file
no	Disable debugging informations
ping	Send echo messages
reload	Halt and perform a cold restart
resume	Resume an active network connection
send	Send a message to other tty lines
setup	Run the SETUP command facility
show	Show running system information
ssh	Open a secure shell client connection
telnet	Open a telnet connection
terminal	Set terminal line parameters
traceroute	Trace route to destination
undebug	Disable debugging functions (see also 'debug')

write

Write running configuration to memory, network, or terminal

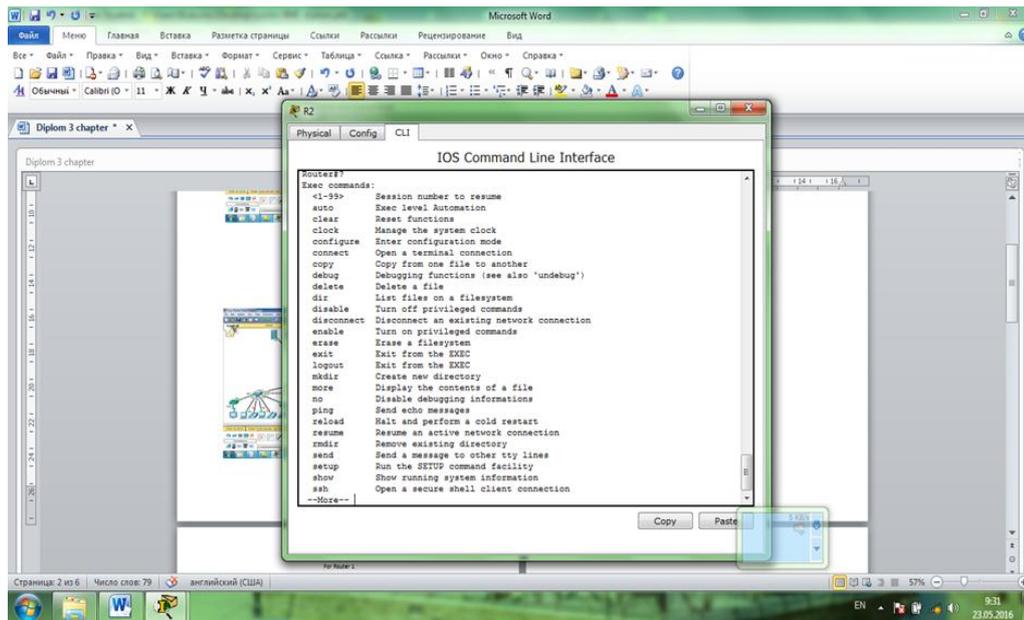


Figure 3.3. Router commands

We write “no shutdown” command on Router to switch on devices which connected with router’s ports.

3.2 Setup devices on Cisco Packet Tracer software

```
Router(config)#interface fa0/0
```

```
Router(config)#no shutdown
```

```
%LINK-5-CHANGED: Interface FastEthernet0/0, changed state to up
```

```
%LINEPROTO-5-UPDOWN: Line protocol on Interface FastEthernet0/0, changed state to up
```

```
Router(config-if)#ip address 192.168.1.1 255.255.255.0
```

```
Router(config)# enable password R1
```

```
Router(config)#do show run
```

```
Router(config)#exit
```

```
Router(config-if)#end
```

```
Router(config-if)#write memory
```

Telnet

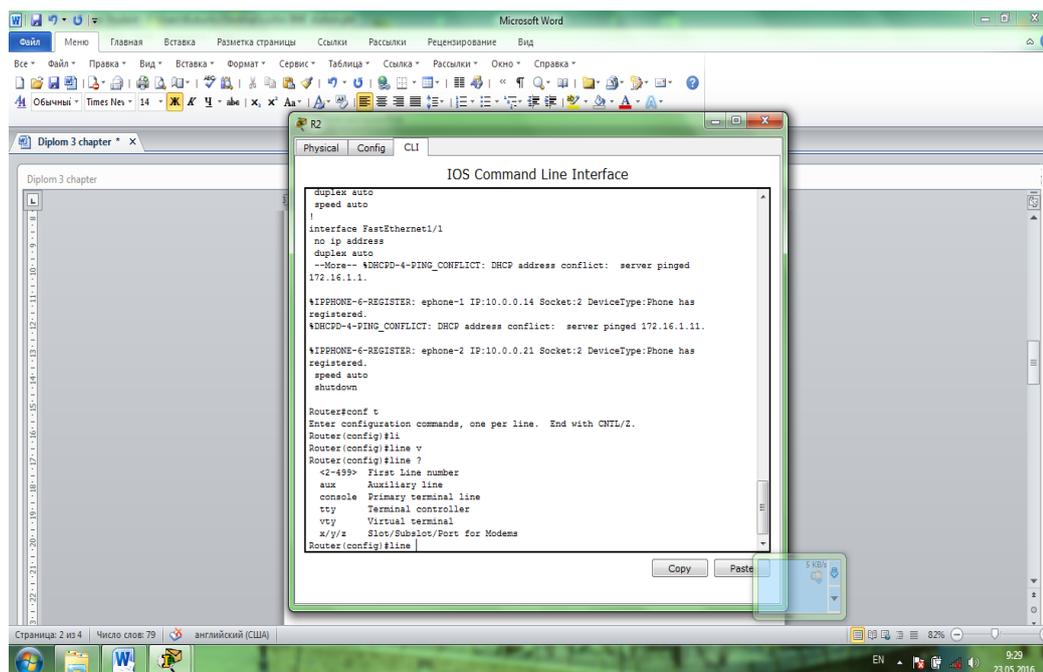


Figure 3.4. CLI menu of Router

With Telnet, we can control devices over the network at a distance. We need follow commands to switch on telnet protocol:

Router >**enable**

Router #**configure terminal**

Router (config)# **line vty 0 4**

Router (config)# **password telnet**

Router(config)# **aaa new-model**

Router(config)#**Username lochin secret 12345**

DHCP(dynamic host control protocol)

Router(config)#**ip dhcp pool Phone**

Router(config)#**default-router 192.168.1.1**

Router(config)#**network 192.168.1.0 255.255.255.0**

Router(config)#**option 150 ip 192.168.1.1**

Router(config)#**ip dhcp excluded 192.168.1.1 192.168.1.10** (for phones)

Router(config-dhcp)#**exit**

Router(config)#**ip dhcp pool Computer**

```
Router(config)#default-router 192.168.1.1
```

```
Router(config)#network 192.168.1.0 255.255.255.0
```

```
Router(config)#ip dhcp excluded 192.168.1.10 192.168.1.20
```

These commands for configuring ip address on Ip-phone automatically.

IP telephony

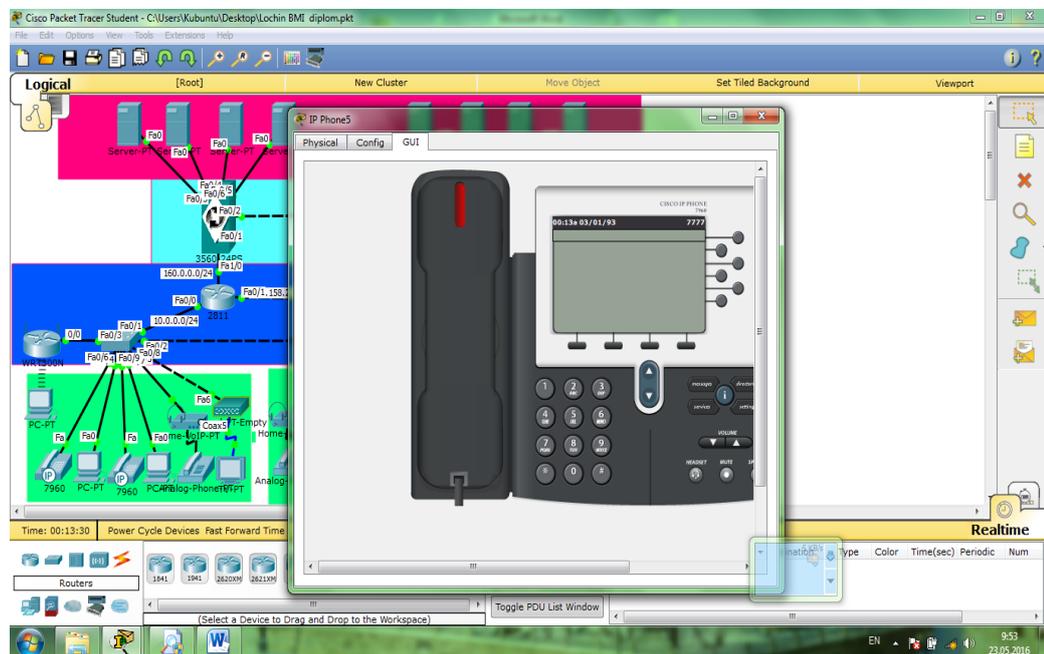


Figure 3.5. GUI menu of IP Phone

Now we click CLI menu on switch and insert “switch>enable” command

First opening switch cmd

```
Switch>enable
```

```
Switch#configure terminal
```

```
Switch(config)#interface range fa0/1-10
```

```
Switch(config)#switchport mode access
```

```
Switch(config)#switchport voice vlan 1
```

```
Switch(config)#switchport access vlan 2
```

```
Switch(config)#interface fa0/24
```

```
Switch(config)#switchport mode trunk
```

```
Switch(config)#switchport trunk allowed vlan 1,2
```

```
Switch(config)#vlan 1
```

```

Switch(config)#name voice
Switch(config)#vlan 2
Switch(config)#name data
Switch(config)#do show vlan brief
Switch(config)#end
Switch(config)#write memory

```

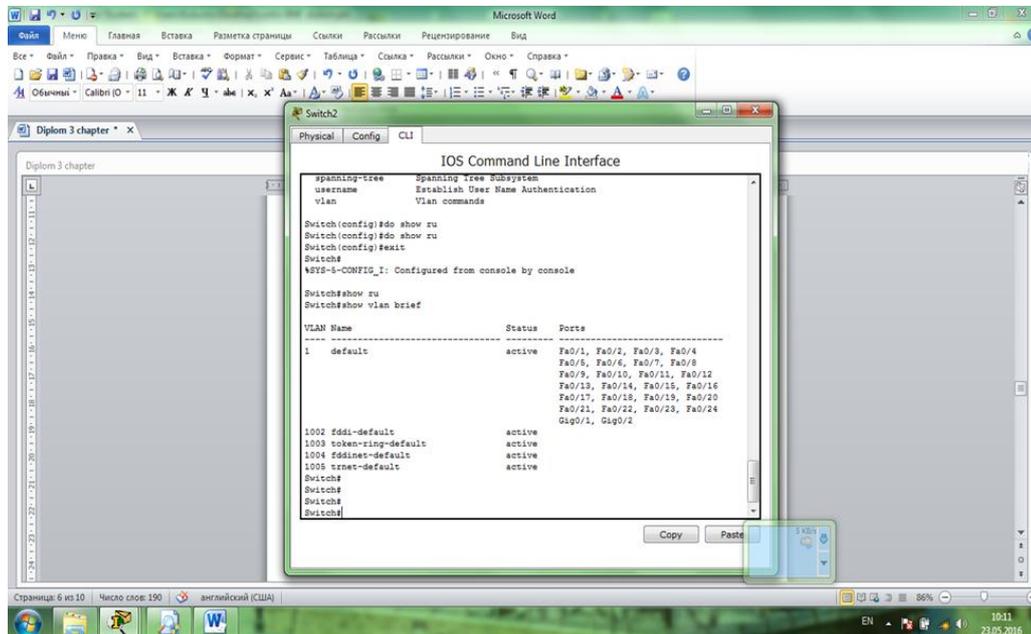


Figure 3.6. CLI menu of switch

In this step, we insert commands for Ip-phone and analogue phones on router. After loading ip-phone commands, we select ip-phone numbers.

Configuration of IP phone to Routers

```

Router(config)#telephony-service
Router(config)#max-ephones [5
Router(config)#max-dn 5
Router(config)#ip source-address 192.168.1.1 port 2000
Router(config)#auto assign 4 to 6
Router(config)#auto assign 1 to 5
Router(config) #dial-peer voice 1 voip
Router(config) #destination-pattern 11..

```

Router(config) #**session target ipv4:192.168.1.2**

Router(config)#**exit**

More-- %DHCPD-4-PING_CONFLICT: DHCP address conflict: server pinged 192.168.1.1.

%IPPHONE-1-REGISTER: ephone-2 IP:192.168.1.2 Socket:2 DeviceType:Phone has registered.

Router(config)#**ephone-dn 1**

Router(config-ephone-dn)#%LINK-3-UPDOWN: Interface ephone_dsp DN 1.1, changed state to up

Router(config)#**number 1111**

Router(config)#**ephone-dn 2**

Router(config-ephone-dn)#%LINK-3-UPDOWN: Interface ephone_dsp DN 1.1, changed state to up

Router(config) #**number 1122**

Router(config) #**ephone-dn 3**

Router(config-ephone-dn)#%LINK-3-UPDOWN: Interface ephone_dsp DN 1.1, changed state to up

Router(config) #**number 1133**

Router(config) #**ephone-dn 4**

Router(config-ephone-dn)#%LINK-3-UPDOWN: Interface ephone_dsp DN 1.1, changed state to up

Router(config) #**number 1144**

For checking loaded commands we insert “show ephones” command

Router(config)#**do show ephones**

ephone-1 Mac:0001.4329.7C51 TCP socket:[1] activeLine:0
REGISTERED in SCCP ver 12 and Server in ver 8
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0 caps:8
IP:192.168.1.5 1025 7960 keepalive 43 max_line 2
button 1: dn 2 number 3332 CH1 IDLE

ephone-2 Mac:0001.C767.A95D TCP socket:[1] activeLine:0

REGISTERED in SCCP ver 12 and Server in ver 8

mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0 caps:8

IP:192.168.1.2 1025 7960 keepalive 43 max_line 2

button 1: dn 3 number 3333 CH1 IDLE

ephone-3 Mac:00D0.FF8E.9901 TCP socket:[1] activeLine:0

REGISTERED in SCCP ver 12 and Server in ver 8

mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0 caps:8

IP:192.168.1.4 1029 ata keepalive 43 max_line 2

button 1: dn 4 number 3334 CH1 IDLE

Finally, we write “memory” command.

Router(config)#end

Router(config)#write memory

After setting IP Phone, this phone connected to network with UTP cable.

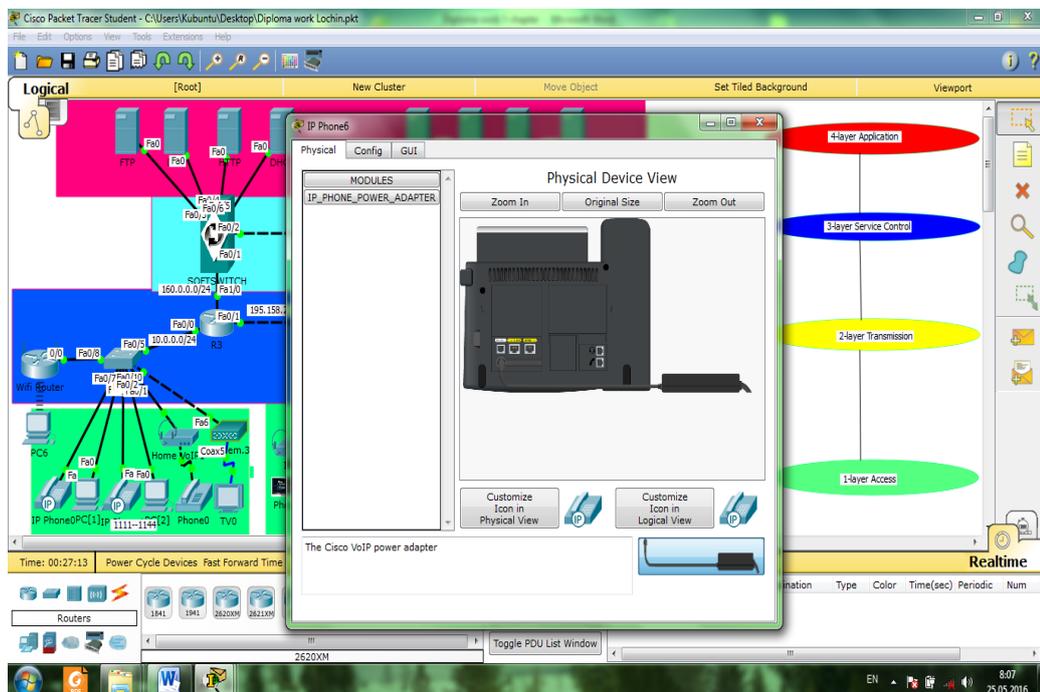


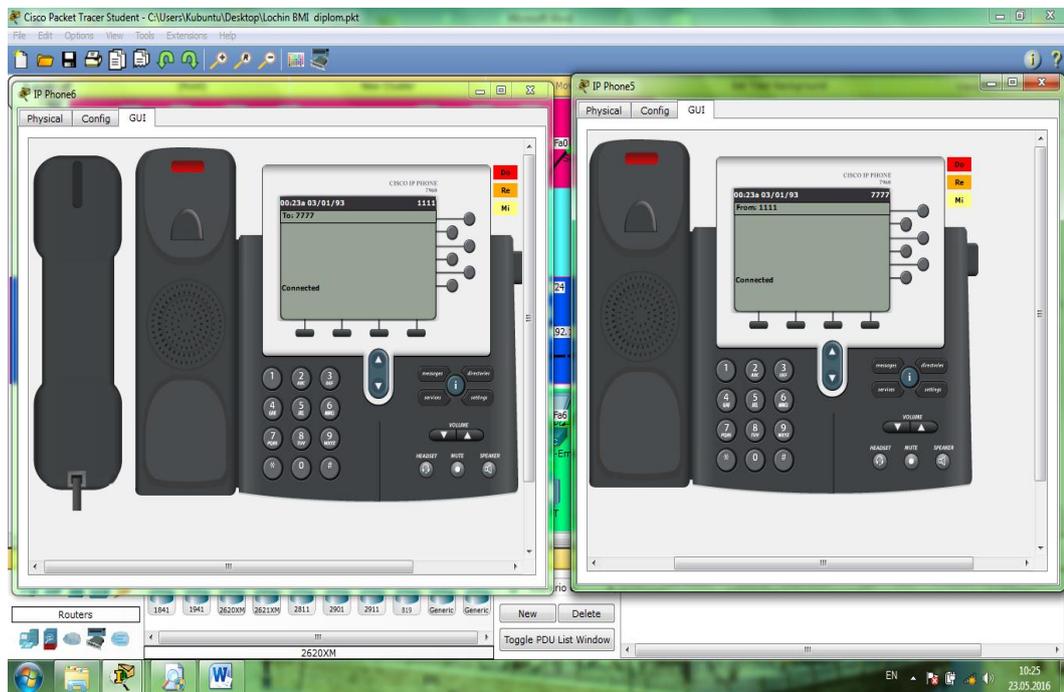
Figure 3.7. GUI menu of ip phone

IP phone was configured. Now we can connect Ip phone with another Ip Phone.



Figure 3.8. GUI menu of ip phone

In next step, we connect two ip-phones with each other.



x-

Figure 3.9. GUI menu of Ip phone

Now we can see that process is successful done. In this process we used H.323 protocol. When analogue phone is configured we insert network address into IAD device and connect 192.168.1.8 ip address to 1144 number.

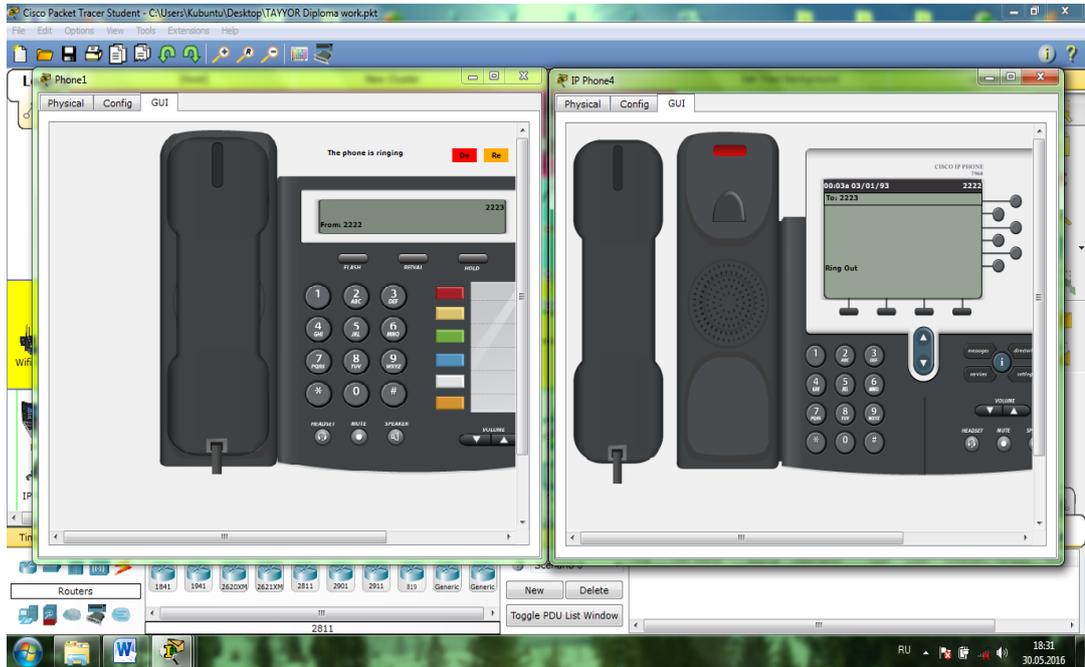


Figure 3.10. GUI menu of Ip phone

Now we connect Wi-Fi router with the switch and insert network address. On any laptop we click browser and insert 192.168.0.1 ip address. After that we complete username and password rows.

Wi-fi Router

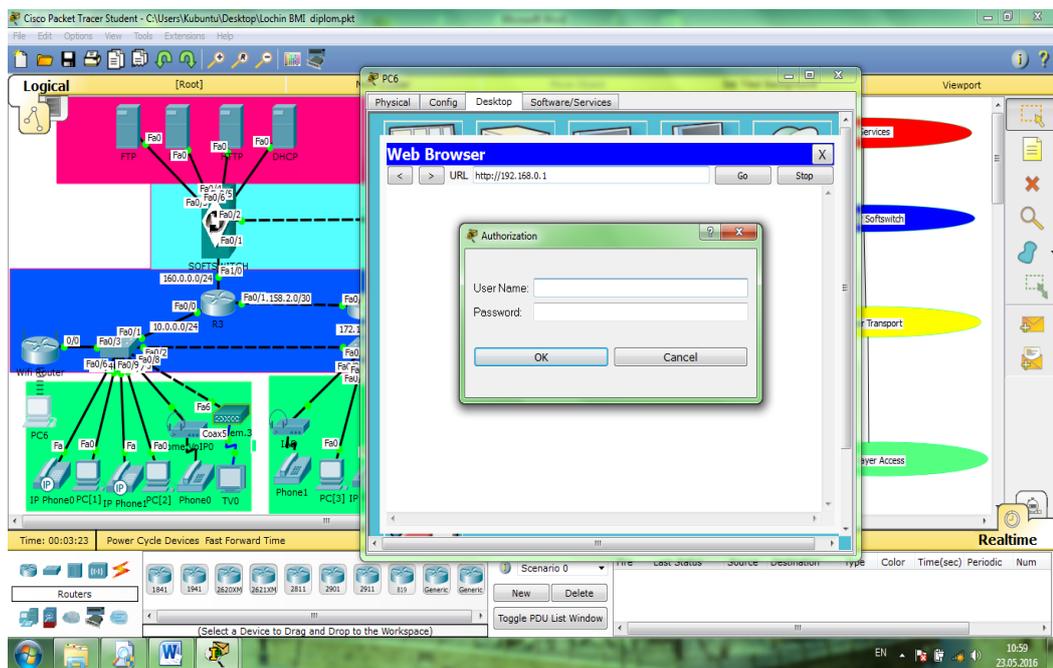


Figure 3.11. Desktop menu of PC

We click wireless command on wi-fi router and designate network name and password. After that we save this process.

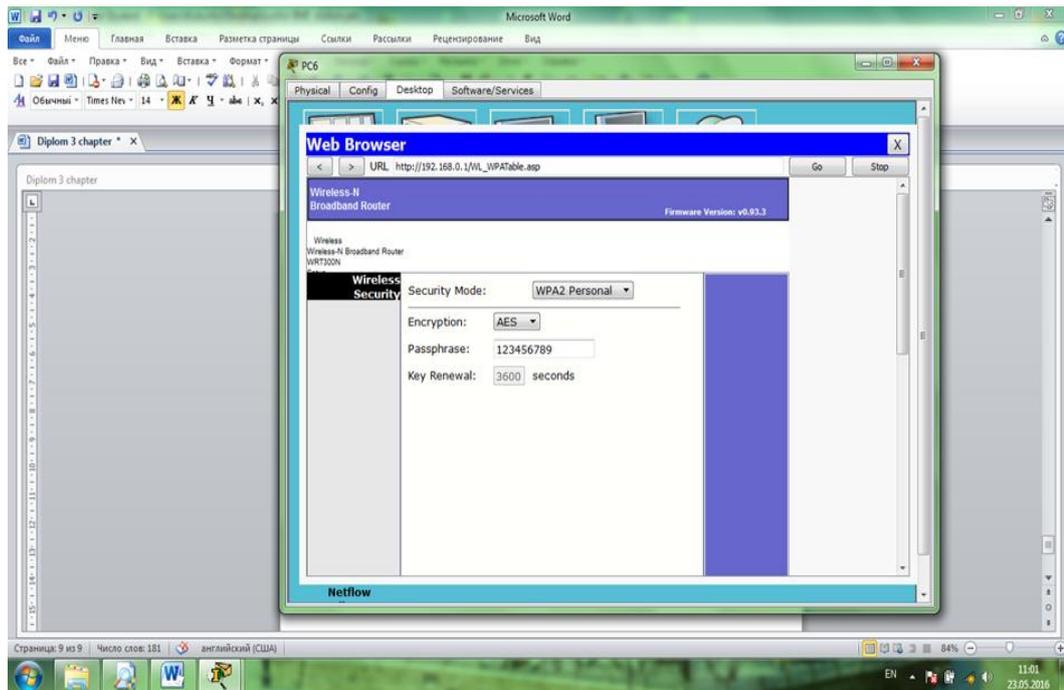


Figure 3.12. Laptop menu of PC

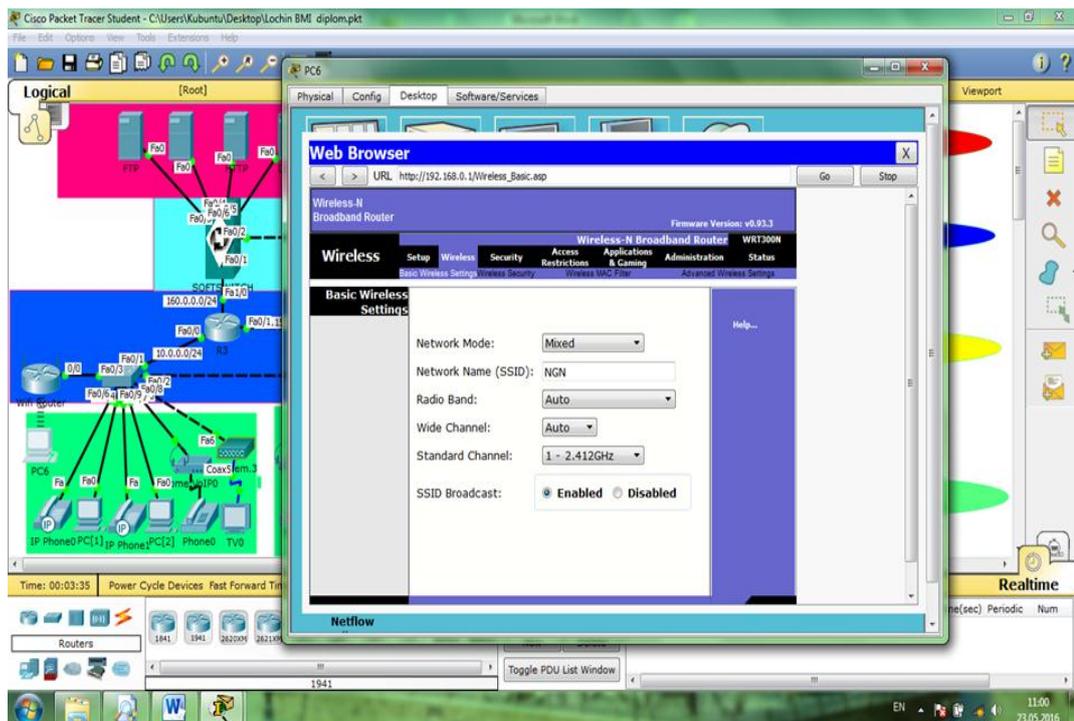


Figure 3.13. Laptop menu of PC

We use laptop in wireless network. We send “ping” from wireless network

to network using a ICMP protocol. Successful done!

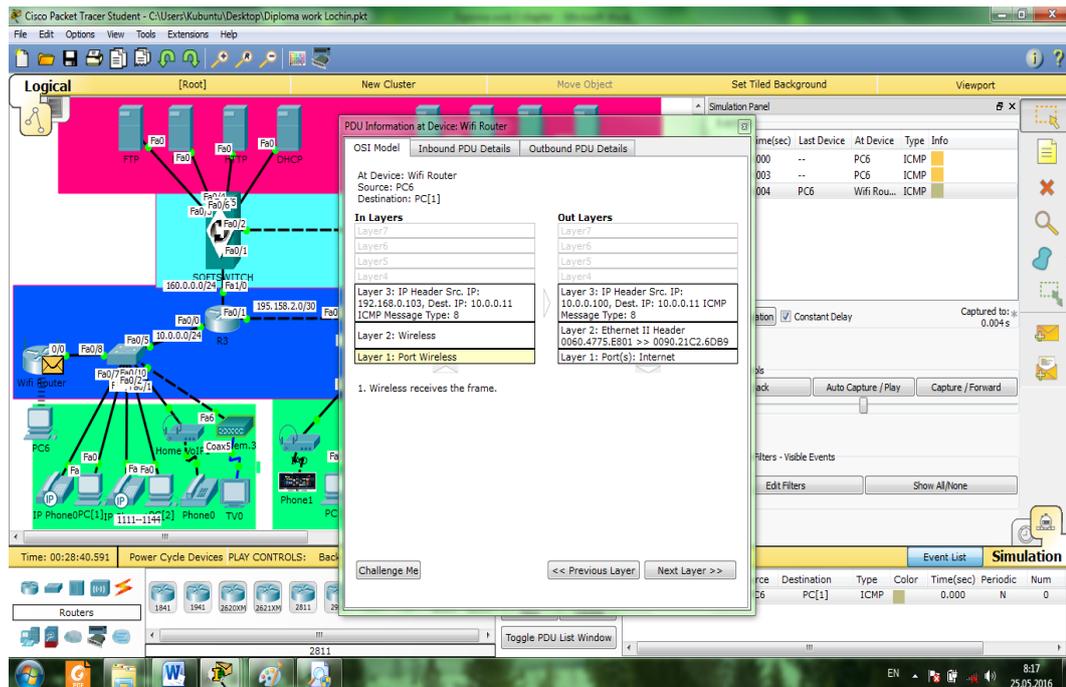


Figure 3.14. PDU menu of packet

In this process we create TUIT.UZ and TI.UZ using a HTTP protocol. We click services menu on server device and push “on” button. Then we create program using HTML web.

```
<html>
```

```
<center><font size='+2' color='blue'>Toshkent Axborot Texnologiyalar
Universiteti</font></center>
```

```
<hr>Welcome to Group 410-12 student Abdullaye.L
```

```
<p>Messages:
```

```
<br><a href='helloworld.html'>World messages</a>
```

```
<br><a href='copyrights.html'>Sport messages</a>
```

```
<br><a href='image.html'>IT messages</a>
```

```
<br><a href='cscoptlogo177x111.jpg'>new messages</a>
```

```
</html>
```

HTTP server

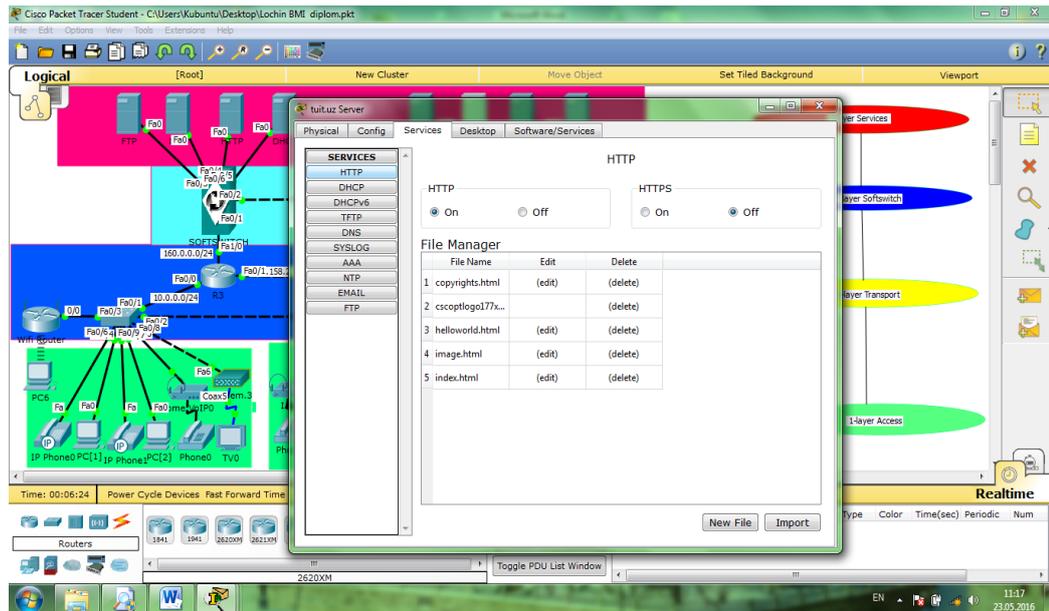


Figure 3.15. Services menu of HTTP server

We save inserted HTML program

```
<html>
```

```
<center><font size='+2' color='blue'>Telecommunication engenering  
faculty</font></center>
```

```
<hr>Welcome to Group 410-12 student Abdullaye.L <p>Messages:
```

```
</html>
```

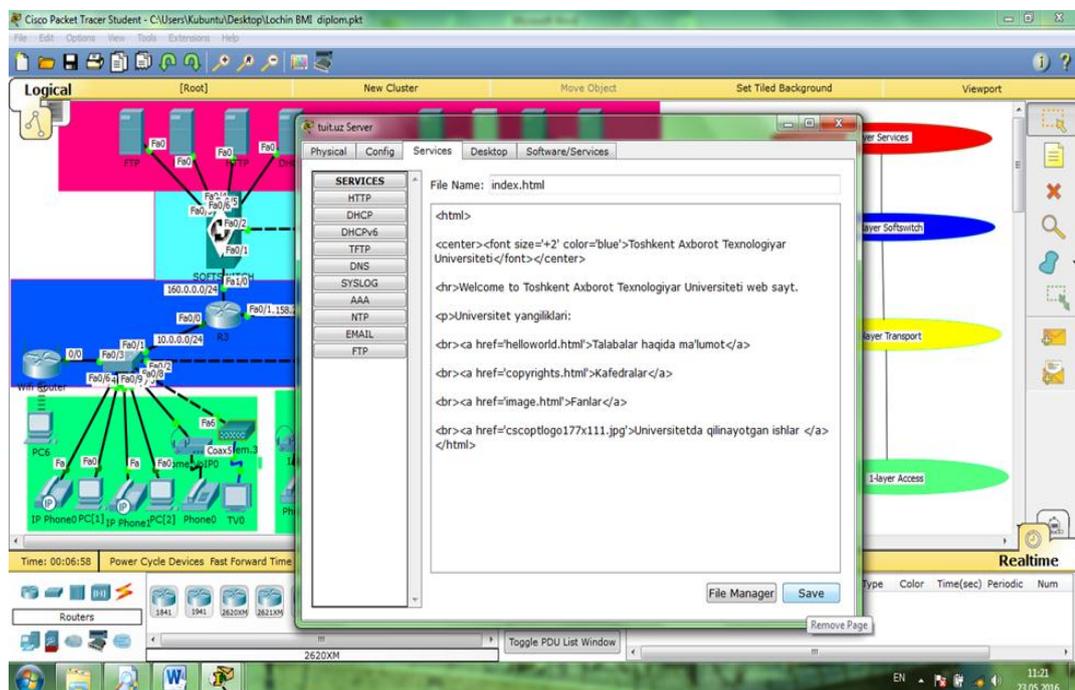


Figure 3.16. Services menu of HTTP server

We repeat this process again on TI.UZ website

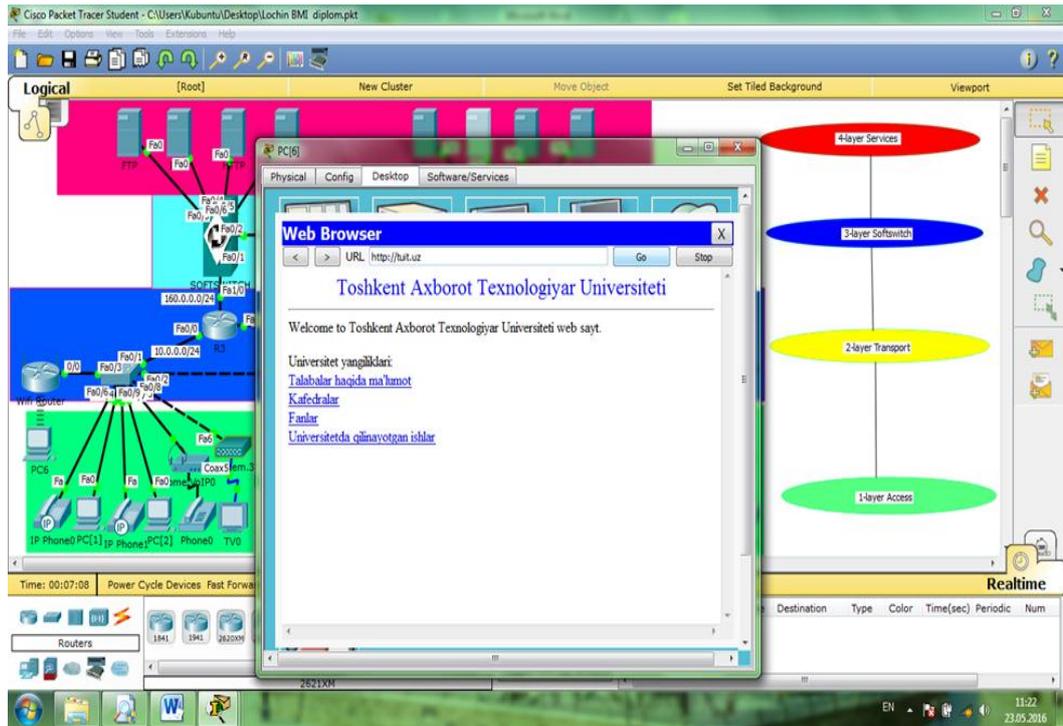


Figure 3.17. Web Browser menu of PC

We select Web Browser menu on PC and write “TI.UZ”. After that we can see created website on PC’s web browser.

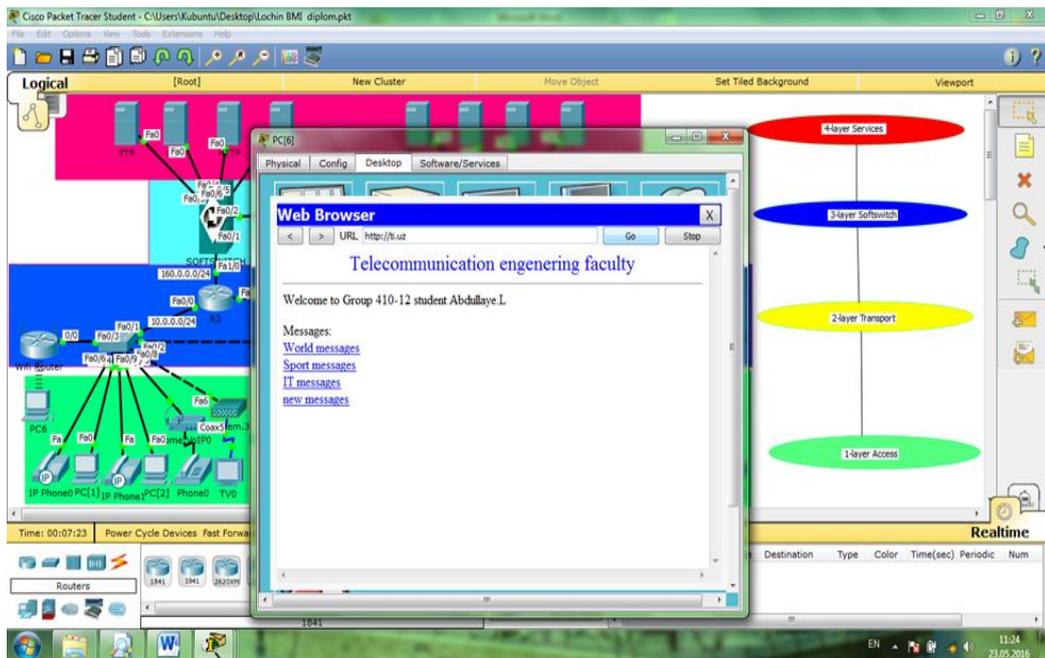


Figure 3.18 WebBrowser menu of PC

We use Email server between client and server. Email server works with

SMTP, POP3 and IMAP4 protocols. We click services menu on server and complete user's name and password.

EMAIL server

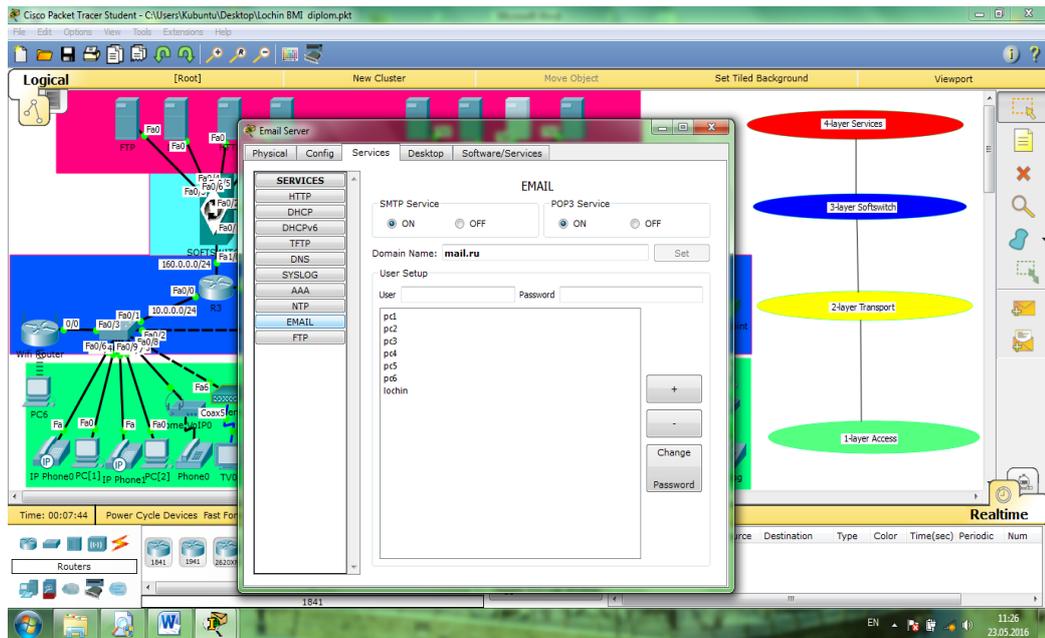


Figure 3.19. Services menu of Email server

3.3. Results

We can send message to another user on the internet using Email.

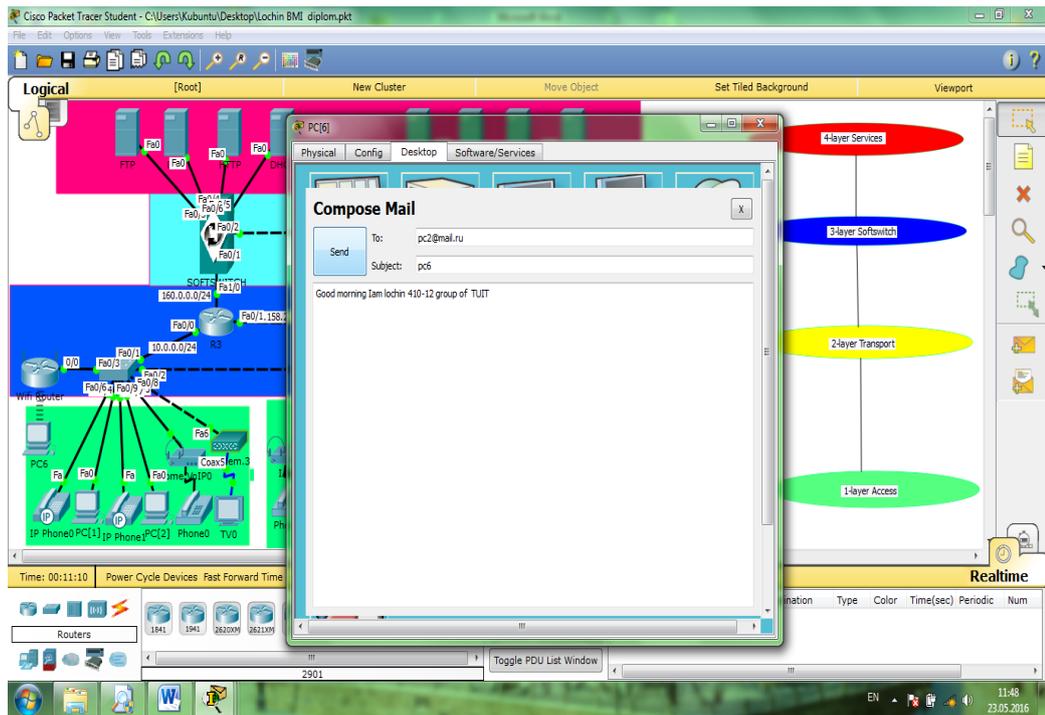


Figure 3.20. Desktop menu of PC

For checking received message, we click email menu on receiver PC and push “receiver” button.

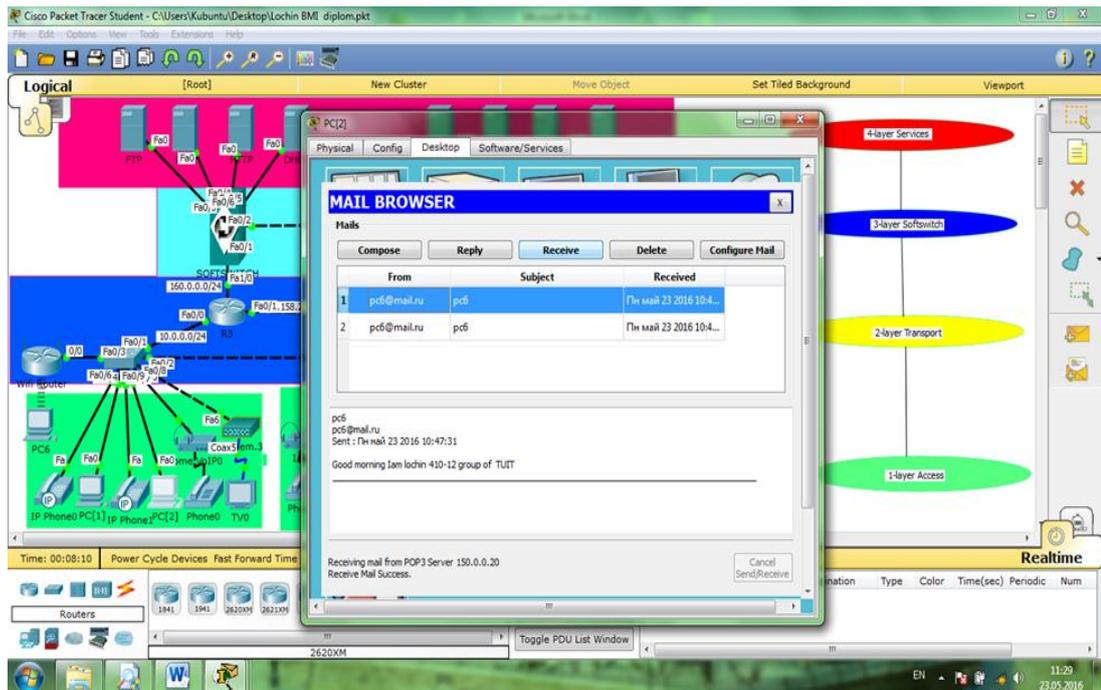


Figure 3.21. Mail Browser menu of PC

In this process, we use routing protocols to deliver packets.

Protocols Routing (**BGP**, **RIP**, and other).

RIP (Routing Information protocol).

Router(config)#**router rip 1**

Router(config)#**version 2**

Router(config)#**network 192.168.1.0**

Router(config)#**network 192.168.2.0**

Router(config)#**network 192.168.3.0**

BGP (Border Gateway protocol)

Router(config)#**router bgp x**

Router(config)#**neighbor 192.168.1.0 remote-as 2**

Router(config)#**neighbor 192.168.2.0 remote-as 1**

Router(config)#**neighbor 192.168.3.0 remote-as 3**

Router(config)#**network 10.10.10.1**

Router(config)#**end**

Router(config)#write memory

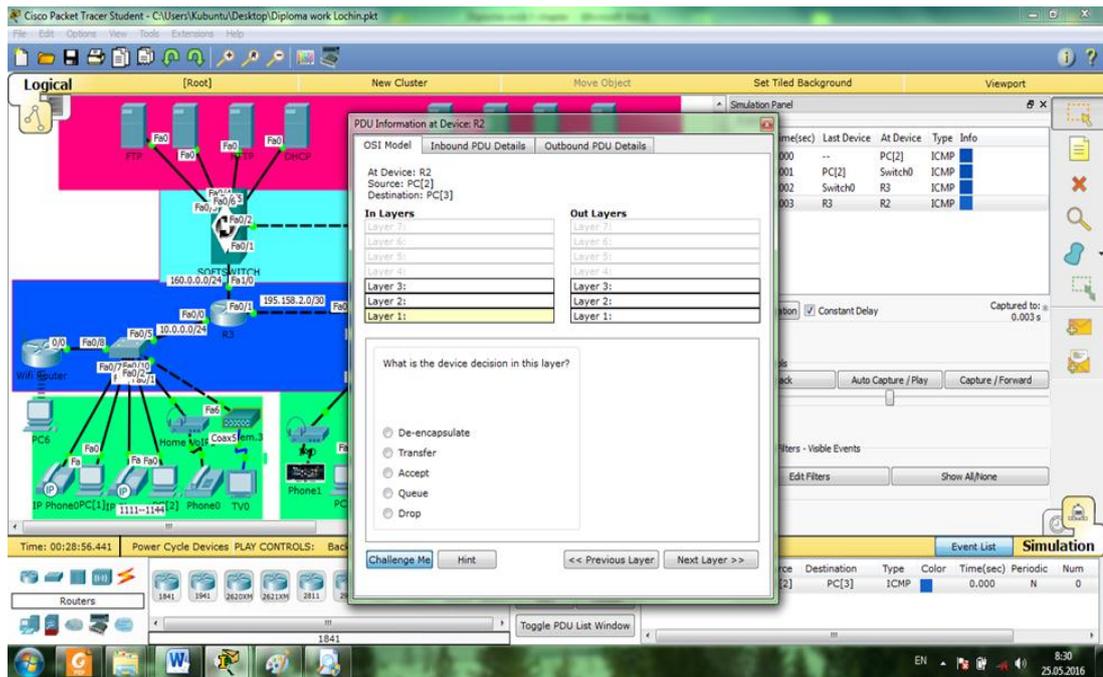


Figure 3.22. OSI model menu of packet

We can see virtual form of transmitting packets and all process at 7 layers of OSI model on CPT.

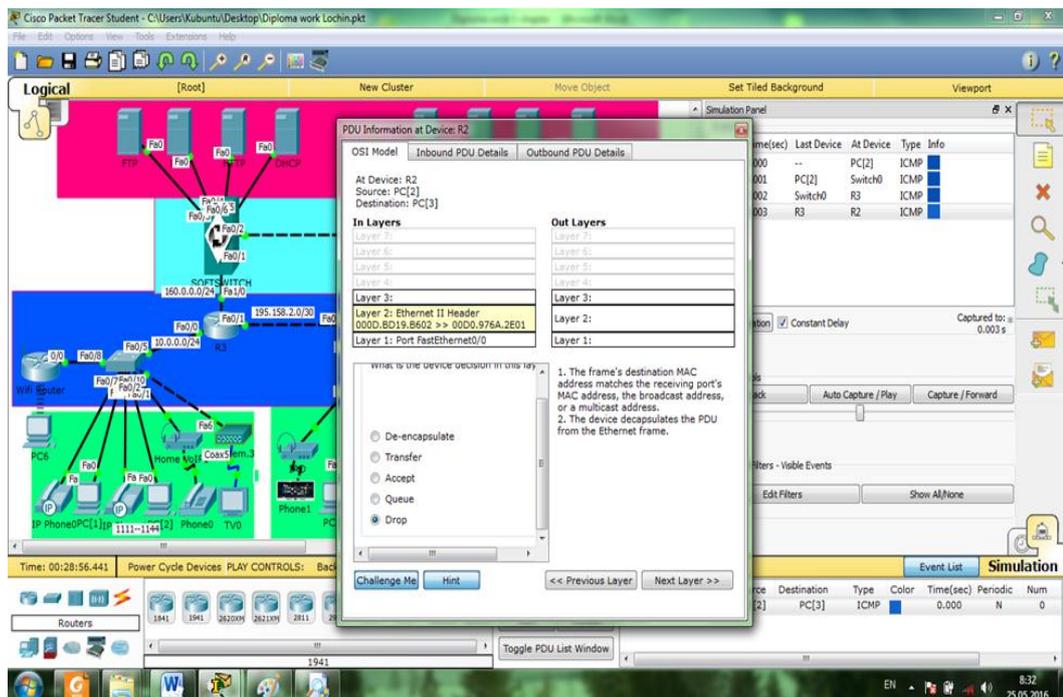


Figure 3.23. OSI model menu of PC

For checking network, we select cmd menu on PC and send ping. Also we

can see lost packets over the network.

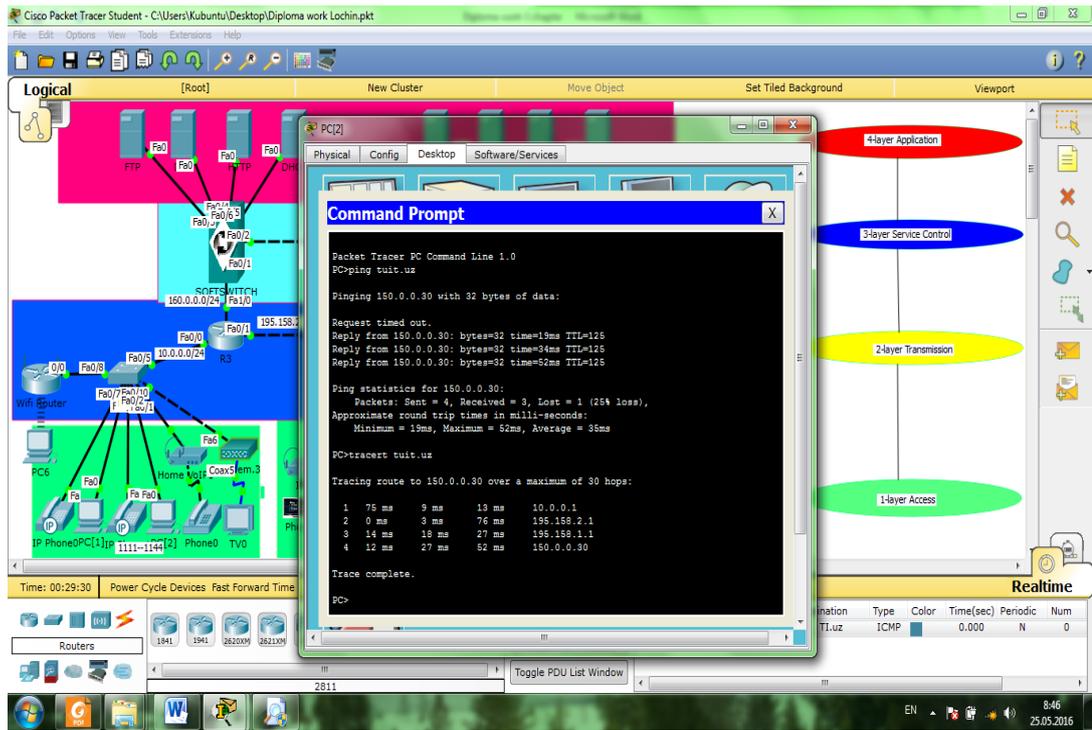


Figure 3.24. Command Prompt menu of menu

Now we exchange devices to real shape. To do it, we click “customize icon in physical view” on any device.

Main purpose of this diploma work is developing virtual model of NGN network on CPT.

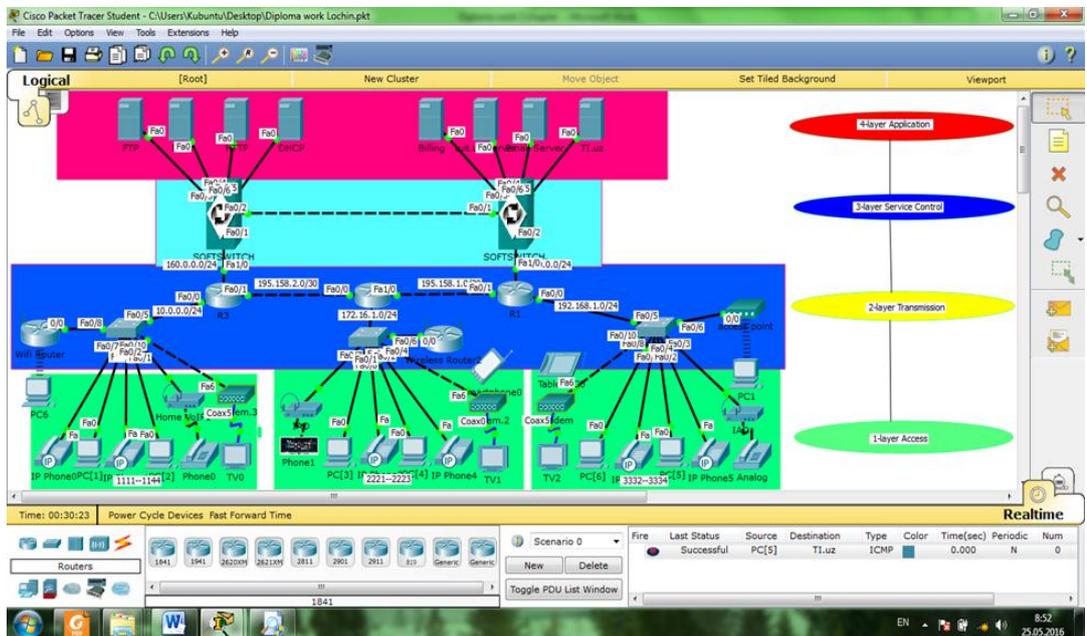


Figure 3.25. Cisco packet tracer NGN architecture

Physical view of devices in Cisco packet tracer were changed into view of real devices.

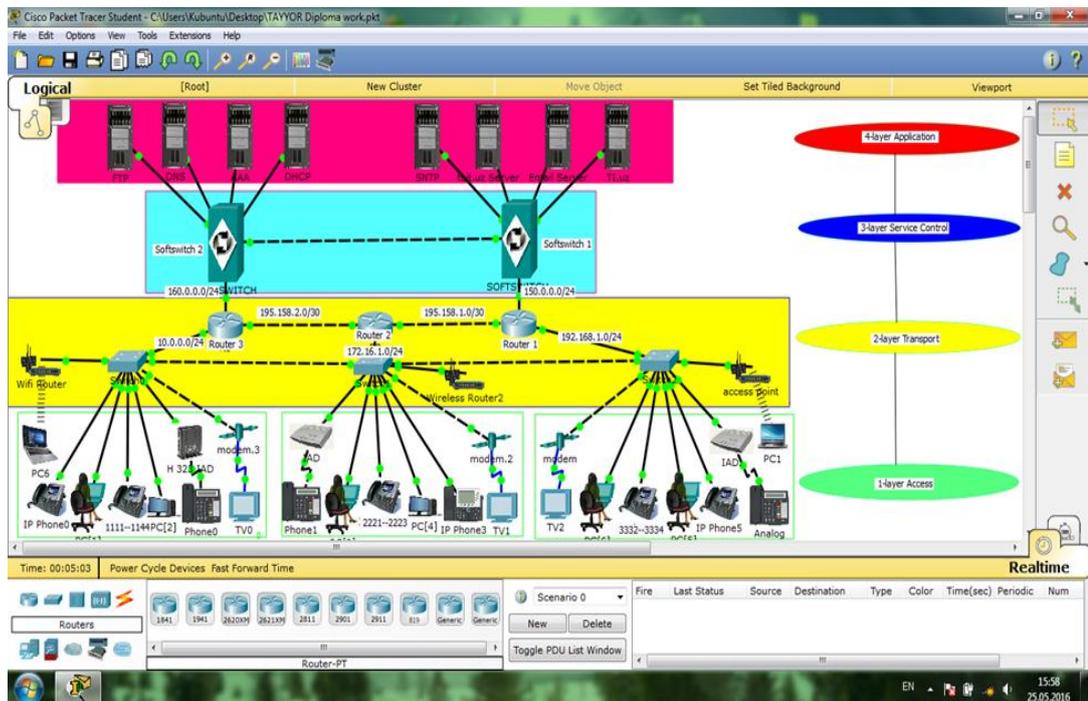


Figure 3.26. Cisco packet tracer NGN architecture

Summary

We developed imitative model of NGN using a Cisco Packet Tracer software. Chapter 3 includes establishing NGN architecture, setting different kind of devices, connecting devices with each other by various cables. Moreover, delivery delay of packets between end devices and connecting different phones analyzed in this chapter.

4. SAFETY ENGINEERING

4.1. General requirements of safety engineering during the work with personal computer

Requirements of work security, that are set forth here, are assigned on personal, that operating personal computers, programmers, PC users, who combines the work of operator with main work on PC less than their working time. Normal daily duration of the work time is eight hours, i.e. forty hours in week with two vacation days in Saturday and Sunday.

For the guaranteeing of optimal capacity for work and maintenance of health of professional users regulated breaks must be ranged during the working shift. Time of regulated breaks during the working shift is settled according to her duration, type and category of the work activity.

Duration of continuous work with PC without regulated breaks must not exceed two hours. During the eight and twelve hourly working shift regulated breaks are settled by fifteen minutes after every work hour.

During the work with PC in a night shift (from 1- p.m. hours to 6 a.m.), regardless of category and type of working activity, regulated breaks duration is exceeded to sixty minutes.

During regulated breaks for reduction of emotional stress, tiredness of visual analyzer, suppression of hypodynamia and hypokinesia influence, prevention of fatigue development personal must execute special training complexes.

If personal, that works with PC, has visual discomfort and other inauspicious feelings, insipid of maintenance of sanitary and hygienic, ergonomic requirements, work modes and vacation it's necessary to use individual approach in limiting the work time with PC, correction of break duration for vacation or change the working activity on other, that not associated with using PC.

During the work PC's user contacts with dangerous and harmful production factors (DHPF). Dangerous production factor is the factor, which's influence on the personal, in the definite conditions, could lead to injury or suddenly health

worsening, degradation or illness.

During the work process PC's user confronts with next DHPF.

Physical:

- increased electromagnetic radiation levels;
- increased X-ray radiation levels;
- increased ultraviolet radiation levels;
- increased infrared radiation level;
- increased static electricity level;
- increased suspended materials concentration level in the working zone;
- increased air ion maintenance in the air of the working zone;
- decreased negative air ion maintenance in the air of the working zone;
- decreased of increased air moisture;
- decreased of increased air mobility of the working zone;
- increased noise level;
- decreased of increased consecration level;
- increased straight brightness level;
- increased reflected brightness level;
- increased blindness level;
- brightness distribution irregularity in the field of vision;
- increased light image brightness;
- increased light flux pulsation level;
- increased voltage significance in electric chain, which's locking could pass throw the human body;

Chemical:

- increased carbon dioxide, ozone, ammonia, phenol, formaldehyde and polychloride biphenyl maintenance in the air of the working zone

Psychophysiological:

- sustained statical load;
- stereotyped working motions;
- intellectual overstrain;

- monotonous load;
- sensory loads;
- vision strain;
- attention strain;
- big information capacity, which processed in the unit time;
- work mode;

Irrational working place organization

Biological:

- increased microorganism maintenance in the air of the working zone

Contact with DHPF could bring to injuries or different illness development, that affects cardiovascular, respiratory, nervous systems, liver, kidney and so on.

Pregnant women do not admitted to implementation of all kinds of works, that concerned with PC using since pregnancy's ascertainment and in the baby breast-feeding period.

PC user should use individual protection facilities:

Preventive devices, display guard filter of the "absolute protection" grade, spectral glasses, implementation of the personal hygiene rules.

4.2. Requirements of safety before the work with PC

1. Inspect and put to rights the working place.
2. Adjust consecration on the working place, make sure in enough consecration, in absence of the reverberation on the screen, in absence of the counter light flux.
3. Check the equipment connection propriety in the electricity supply.
4. Make sure in the protective ground presence and screenful conductor connection to the processor case.
5. To wipe the screen surface and the guard filter with a special napkin.
6. Make sure in absence of the floppy disks in the disk drives of the PC processor.
7. Check the precision of the table arrangement, legs pedestal, equipment position, screen tilt angle, keyboard position, and, if it's necessary, carry out the regulation of the working table and the armchair, PC elements arrangement in

accordance with ergonomics demands for the purpose of uncomfortable poses and long body tiredness exception.

8. During the turning on PC's user must obey the next engaging equipment consecution:
9. Turn on the power unit.
10. Turn on the peripheral device (printer, monitor, scanner and so on)
11. Turn on the system case. (processor)
12. PC's user doesn't allowed to begin the work in view of:
13. Information absence of conditions of work attestation results on the present working place or in the presence of information of present equipment parameters discrepancy to sanitary standard demands.
14. Display guard filter of the "absolute protection" grade absence.
15. Disconnect ground conductor of the guard filter.
16. Equipment faultiness detection.
17. Guard ground PC device absence.
18. Carbon-dioxide or powder-type fire extinguisher and first-aid kit absence.
19. Hygienic regulations PC disposal violation.

4.3. Safety engineering during the work with personal computer

During the work PC user must:

- keep in clean and proper the working place during the whole of working day;
- open all device vent ducts;
- use "mouse" in the presence of special pad;
- correctly close all active tasks;
- Power off the power supply only in that case, if operator during the breaks after the PC work should to be near the display terminal (less than 2 m), otherwise you can not power off the power supply;
- execute the sanitary code and observe the routines of work and rest;
- Observe the computer engineering exploitation rules in accordance with

exploitation direction;

- During the work with text information you must choose physiological view mode of black symbols on the white background;
- According to daily time-table make up the eyes, neck, arms, body and legs relaxation;
- Observe the distance between the eyes and monitor in the range of 60-80 sm;

During the work PC user isn't allowed:

- Touch the monitor and the keyboard at the same time; touch the back; panel of the system case (processor) at powered conditions;
- Switch the peripheral devices interface cable at powered conditions;
- Put the paper and other items under the top panel;
- Admit the disarrange of the working place with paper for the purpose of barring accumulation of organic dust;
- Switch off during the executing the active task;
- Make the frequent power switches;
- Let the ingress of moisture on the system case(processor) surface monitor, keyboard working zone disk drives, printers and other devices;
- Power on the strongly chilled devices;
- Perform the opening and repairing the devices unassisted;

Summary

Requirements of work security, that are set forth here, are assigned on personal, that operating personal computers, programmers, PC users, who combines the work of operator with main work on PC less than their working time. Normal daily duration of the work time is eight hours, i.e. forty hours in week with two vacation days in Saturday and Sunday.

CONCLUSION

Main purpose of the thesis is to implement imitative model of NGN using a Cisco Packet Tracer software as an experimental model for TUIT students.

Chapter 1 provides the motivation for the development of all IP communications environment, which fulfils almost all the expectations and requirements of a NGN system. Cisco Systems claims that Packet Tracer is useful for network experimentation. This “e-doing” capability is a fundamental component of learning how to configure routers and switches. The aim was to help the reader comprehend the concept of convergence, its drivers and enablers in NGN. Various issues of NGN are discussed and the current and future trends of standardization activities for NGN are presented in detail. Some of the applications mentioned may be a bit advanced and not feasible from the perspective of Uzbekistan, they are certainly going to be deployed in near future.

Chapter 2 focuses on the explanation of basic conditions in the area of Next Generation Network architectures and network components from point of current and future platforms. participants also become familiar with mobile and optical communication technologies as well as with technologies for digital video delivery such as DVB and IPTV systems. Moreover, they will dispose with knowledge about state of the art technologies like content delivery networks (CDN) and hybrid broadband broadcast television (HBB TV) systems.

The experimental results are considered in chapter 3. We developed imitative model of NGN using a Cisco Packet Tracer software. Chapter 3 includes establishing NGN architecture, setting different kind of devices, connecting devices with each other by various cables. Moreover, delivery delay of packets between end devices and connecting different phones analyzed in this chapter. This model provides simulation, visualization, authoring, assessment, and collaboration capabilities to facilitate the teaching and learning of complex technology concepts. Most importantly, Packet Tracer helps students and instructors to create their own virtual “network worlds” for exploration, experimentation, and explanation of

networking concepts and technologies. In addition, students can build, configure, and troubleshoot networks using virtual equipment and simulated connections, alone or in collaboration with other students on Cisco Packet Tracer software. Packet Tracer offers an effective, interactive environment for learning networking concepts and protocols.

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APPLICATIONS
THE PRESENTATION SLIDES