MINISTRY OF EDUCATION AND SCIENCE OF UKRAINENATIONAL AVIATION UNIVERSITY
FACULTY OF AERONAVIGATIONS, ELECTRONICS AND TELECOMMUNICATIONS
DEPARTMENT OF TELECOMMUNICATION AND RADIO ENGINEERINGSYSTEMS

ADMIT TO DEFENCE
Head of the Department

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"_____" ____________2022

DIPLOMA WORK

(EXPLANATORY NOTE)

BACHELOR'S DEGREE GRADUATE
BY SPECIALITY "TELECOMMUNICATIONS AND RADIO ENGINEERING"

Topic: «Model of VoIP service for private business based on Nextiva Business Phone System»

Performer: ______________________________ K. Zhukova
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Supervisor ______________________________ O. Holubnychyi
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NATIONAL AVIATION UNIVERSITY

Faculty of aeronavigations, electronics and telecommunications
Department of telecommunication and radio engineering systems
Speciality: 172 "Telecommunications and radio engineering"

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“_____” _____________2022

TASK
for execution of bachelor diploma work

Kateryna Zhukova
(full name)

1. 1.Topic of diploma work: «Model of VoIP service for private business based on Nextiva Business Phone System» approved by the order of the rector from «25» April 2022 №433/ст.

2. The term of the work: from 23 May 2022 to 17 June 2022.

3. Initial work data: IP Telephony standards; Security protocols of an IP Telephony

4. Explanatory note content: INTRODUCTION; CHAPTER 1 THE ARCHITECTURE OF VOIP SERVICE AND PROSPECTS FOR THE DEVELOPMENT OF VOIP TELEPHONY NETWORKS; CHAPTER 2 EVALUATION OF CUSTOMER’S NETWORK; CHAPTER 3 BUILDING A SIP TRUNKING ACCOUNT

5. List of required illustrative material: figures, table.

6. Work schedule

№ n/p	Task	Implementation term	Performance note
1	Develop a detailed content of sections of diploma (qualification) work	23.05.2022 - 25.05.2022	Done
2	Introduction	25.05.2022	Done
3	THE ARCHITECTURE OF VOIP SERVICE AND PROSPECTS FOR THE DEVELOPMENT OF VOIP TELEPHONY NETWORKS	26.05.2022 - 29.05.2022	Done
4	EVALUATION OF CUSTOMER’S NETWORK	30.05.2022 - 02.06.2022	Done
5	BUILDING A SIP TRUNKING ACCOUNT	03.06.2022 - 08.06.2022	Done
6	Elimination of shortcomings and defense of the thesis	09.06.2022 - 17.06.2022	Done

7. Date of issue of the assignment: April 25, 2022.

Supervisor ____________________ O. Holubnychiy
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Accepted task for execution ____________________ K. Zhukova
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ABSTRACT

Diploma work «Model of VoIP service for private business based on Nextiva Business Phone System» contains 44 pages, 20 images, 1 table, 13 sources.

The object of study: voice connection processes based on Nextiva Phone Systems

The subject of study: model and means of virtualization and network communications of VoIP telephony

The purpose of the diploma work: building a simulation model of VoIP telephony using Nextiva Phone Systems algorithms

Research methods: remote configuration and control of client network performance. Conducting research aimed at the operation and maintenance of VoIP telephony equipment.

Materials of diploma work are recommended to be used in building small businesses and enterprises as a VoIP solution.

Keywords: VoIP, IP TELEPHONY, SIP, SIP TRUNKING, PBX.

CONTENTS

LIST OF ABBREVIATIONS

VoIP – Voice Over IP

IP – Internet Protocol

PBX – Private Branch eXchange

SIP - Session Initiation Protocol

MGCP - Media Gateway Control Protocol

PSTN - public telephone network

UAC - User Agent Client

UAS - User Agent Server

UA - User Agent

DHCP - Dynamic Host Configurational Protocol

NAT - Network Address Translation

INTRODUCTION

Voice over Internet Protocol (VoIP) is a technology that allows you to make voice calls over a broadband Internet connection instead of a regular (or analog) telephone line. Some VoIP services may allow you to call only those users who use the same service, while others allow you to call anyone who has a phone number, including local, long-distance, mobile, and international numbers. In addition, although some VoIP services only work through the user's computer or dedicated VoIP phone, other services allow you to use a traditional phone connected to a VoIP adapter. [5]

Voice business services are the driving force in the VoIP services market, as the transition to IP has a positive impact on cloud, trunking, and managed services in all business segments. Large enterprises, in particular, continue to actively evaluate unified cloud communications, while moving to SIP trunking and its deployment based on their premises.

According to a GMI research report [6], the ITATS market segment is expected to grow the fastest by more than 15% by 2025, as IP-based telephone systems are hosted by service providers. Hosted VoIP eliminates the need to install

any equipment in place, reducing maintenance and training costs. Small and large businesses are increasingly relying on these solutions, which will allow them to manage their telephone systems while gaining access to advanced communication features such as call queuing, messaging, and car service. This will allow them to focus more on their core competencies without investing huge sums of inexpensive equipment [7].

The growth of the VoIP market is due to factors such as the efforts of public institutions and private companies to develop wireless infrastructure and increase the acceptance of VoIP cloud services through their cost-effectiveness.

Businesses with poor communication infrastructure face problems such as poor sound quality and long delays, which can negatively affect their performance. In this way, they are moving from traditional telephone systems to cloud telephone systems designed to handle voice mail and calls to ensure uninterrupted communication. Because the technology supports voice and video communications over the Internet, businesses are using such solutions extensively to ensure high business efficiency through more reliable and routed call services and reduced service. Another factor that is leading to demand in the VoIP market is the trend of increasing labor mobility. By implementing VoIP solutions, businesses can improve communication and collaboration between employees and remote users to increase business productivity. In addition, businesses are aware of the benefits of converged voice and data services to increase their productivity. It is expected that the unification of unified communication services and corporate VoIP to provide chat in real-time video conferencing and other calling capabilities will contribute to the market value of VoIP [6].

The purpose of the diploma work is to build a simulation model of VoIP telephony using Nextiva Phone Systems algorithms and consider an example of testing the network for the ability to run VoIP service.

The object of study is voice connection processes based on Nextiva Phone Systems.

The subject of study is the model and means of virtualization and network communications of VoIP telephony.

To achieve this goal, the following tasks are solved in the work:

1. Analysis of the VoIP telephony market, consideration of the main components of the packet-switched network communication system, existing voice over IP protocols, major types of calls, and services provided.
2. Reviewing principles and conditions for the creation of VoIP using

Nextiva Phone Systems.

3. Testing the client network for weaknesses such as Double NAT, SIP ALG, and Jitter.
4. Construction of a simulation model of a corporate network, and the configuration of VoIP equipment in the Nextiva Phone Systems environment.

The scientific novelty of the obtained results

1. Developed and built a simulation model of VoIP-telephony service based on Nextiva Phone Systems, which allowed creating a working network model and acquiring practical skills with network devices, completing VoIP telephony devices, testing, configuring, designing networks, and forming an idea of the principles of organization and operation of telephone services for small and large businesses.

2. The optimized structural implementation of the telephone network for small and large businesses is proposed, which provides for the reduction of tariffs for corporate telephone calls and ensures high reliability and versatility.

Approbation of research results

The results of the research are presented in the form of a report at the scientific-practical Internet conference of young scientists and students "Problems of operation and protection of information and communication systems".

CHAPTER 1
THE ARCHITECTURE OF VOIP SERVICE AND PROSPECTS FOR THE DEVELOPMENT OF VOIP TELEPHONY NETWORKS

1.1 Conditions for the creation of VoIP and its differences between fixed and mobile telephony

Wired telephony for its more than a century of development due to technical and economic obstacles could not become the common property of mankind. In developed countries, telephones have been installed in all government agencies, and commercial firms, in almost every city apartment and village house, but we have a landline telephone even at the end of the XX century remained inaccessible to many not only in remote and sparsely populated areas but also in large villages. But people need to be able to communicate "here and now" by voice to quickly solve many private and business issues.

Mobile telephony, which began its expansion about forty years ago, has significantly increased the availability of voice communications, especially in areas not covered by fixed communication. In a short time, it has become a technology of mass application.

Wired and wireless technologies have mastered their niches and coexisted successfully, which did not bother competitors, ten years ago when suddenly there was a third "rival" – VoIP technology. And this opponent turned out to be dangerous, first of all, because it provided an opportunity to pay significantly less for the same minutes of voice communication that could be spent in the networks of his predecessors. And VoIP can easily be "implemented" both in stationary and in mobile zones, carrying out communication with subscribers of both the first and the second technology of language communication.

To provide fixed telephony services, it was necessary to create an infrastructure in the form of cable communication lines of enormous length. For mobile - to build base stations for radio signal transmission. And the bodily surroundings for the VoIP era have turned out to be the Internet - companies of the latest telecommunications have now no longer needed to create their very own infrastructure: there may be the Internet - there may be the opportunity of providing VoIP services, that is pondered in the "name" of the era - Voice over Internet Protocol. The cost of building base stations is reimbursed to the operator in payment for calls on mobile phones.

VoIP providers - technologies that use built-in Internet networks, such as the cost of VoIP equipment did not carry, and, accordingly, in tariffs for communication services they are not present. However, the name VoIP does now no longer completely describe the generation that lets in the reception and transmission of now no longer most effective language (this section is named "IP telephony"), however additionally video content material, and, in general, any records offered in virtual form.

However, today the most popular is IP telephony, and this technology in addition to working on the Internet can be implemented in any dedicated digital channels that support the Internet protocol and components of the IP network.

The lack of investment in infrastructure, which should pay off for fixed and mobile operators, for which they are "invisibly present" in their tariffs - this is the main advantage of IP telephony is expressed in its cost-effectiveness. The second condition that allowed IP telephony companies to set a minimal stage of price for their offerings is that during public telephone networks, the price for a name is decided through its period and the duration of the committed channel. When using IP telephony, only the Internet connection and the amount of transmitted traffic are paid. The third item of expenses of fixed-line communication providers, which is included in the tariffs, is the payment of pauses in conversations.

The fact is that in traditional circuit-switched networks, payment is considered for the time of "renting" the channel. And the fact that pauses in the conversation, in fact - a waste of time, the billing system does not take into account, but simply counts the minutes of "rent" the channel and multiplies them by the tariff. In IP telephony there is a mechanism to block the transmission of pauses (dialogue, components, semantics, spent by the subscriber to find the right words, distractions from the conversation, etc.), which can be up to 40-50% of the time of the transmission channel.

1.2 Principles of voice information transmission in packet-switched networks

The first transmission of voice over Internet protocol (voice over IP, VoIP) took place in 1973, as a result of testing the experimental Network Voice protocol, specially created for the ARPANET. But then, until 1995, no significant steps were taken. However, without these multiple stories, there would be no main story.

The foundations for VoIP were laid in 1925-1928, even at the time when the Vocoder was invented - an electronic speech synthesizer based on AT&T. This synthesizer reproduced the semblance of human speech, studying the sounds made by a person. This device was actively used during the Second World War to transmit confidential information.

In 1988, another significant event took place - the appearance of the first famous codec G.722 (Wideband Audio Codec), the quality of which is comparable to speech transmitted over the PSTN. This wideband codec had a bit rate twice that of the previous G.711, and it produced excellent sound for those times.

In 1991, John Walker, the founder of Autodesk, put together a framework for VoIP that required only 32 kbps of bandwidth (64 kbps was the norm at the time) and publicly released the NetFone program (later renamed Speak Freely), which and became the world's first VoIP phone. But in fact, NetFone was originally used and created for - communication within Walker's company.

In 1993, the first system for video conferencing, the Telepresence System, appeared, which was called the Teleport (Teleport), but in

later renamed Tele Suite.

Along with this, in research centers and universities, parallel studies were carried out on speech transmission using packet data switching.

After the advent of the Internet, users did not have enough voice communication to communicate with each other. The solution to the requests and requirements of users was the emergence of VoIP - a packet voice protocol, in 1995. Under this protocol, packets were transmitted from one address to another using the Internet Protocol. This is how IP telephony appeared.

In general, at first, IP telephony was considered simply a cheap solution for long-distance and international communications. But it very quickly became in demand among users and businessmen.

1993 - 1994. Charlie Klein created the first PC program, Maven, which could transmit voice over the network. Around the same time, Cornell University's CU-See Me videoconferencing software for Macintosh PCs gained popularity. Both of these applications have gained immense popularity - with their help, the flight of the space shuttle Endeavor was broadcast on Earth. Maven could transmit audio, while CU-See Me could transmit an image. After some time, these two programs were combined into one common [1].

In 1995, the Israeli company Vocal Tec invented the very first Internet phone, simply called the Internet Phone, and it was available to the general public.

In 1996, the very initial version of the H.323 standard was released, which was intended for voice and video communications over the Internet, at the same time work began on an open standard for IP telephony - the SIP standard. Initially, SIP was created and implemented to connect several people in a conference mode and this standard had nothing to do with VoIP telephony. The first development of SIP knew only one command - to make a call, and only after three years, a total of six commands were mastered. But even then it was clear that in terms of its potential and volume it would overtake H.323.

1998 was one of the crises and turning points for IP telephony. Enterprises were able to realize all the advantages of this type of communication and began to develop their private solutions. In particular, businessmen have begun to move away from PC-to-PC products and implement PC-to-phone and phone-to-phone solutions for VoIP. IP-telephony began to connect to public switched telephone networks (public switched telephone network, PSTN).

In the same year, the first IP switches appeared - considered the first physical equipment for IP telephony, responsible for routing calls. Despite such technical breakthroughs, VoIP calls as of 1998 did not even reach 1% of all voice traffic. In 2000, this figure barely reached 3%, but in 2003 there was a sharp jump - up to 25%. Telephone calls over IP protocol quickly took on the image of free and very cheap calls to all directions, regardless of distance. At one time, commercial companies exploited this freeness in their way and could broadcast commercials at the beginning or middle of a conversation as a “payment” for a free connection. This practice was later discontinued.

In 1999, the first IP-PBX appears (a virtual PBX specifically for VoIP, because a virtual PBX for PSTN was created earlier) - Asterisk. As is often the case, Asterisk grew out of a company's need for a product it couldn't buy or didn't like. Thus, Mark Spencer, who has his own Linux technical support company, realized that he urgently needed a powerful PBX for a call center, but at that time this equipment cost a lot of money. After Asterisk gained popularity, Spencer changed the company's profile to supporting and developing hardware for Asterisk. Until now, Asterisk is very popular among developers and businesses. So, for example, when we integrated IP telephony with our Region Soft CRM, we chose Asterisk as the main virtual PBX. [2]

In 2000, Cisco was considered one of the technology leaders. And in the same year, she made the transition to IP telephony of all her headquarters in California, namely in San Jose. This whole process took about a year and 55 buildings with 20,000 people switched to IP telephony. This project was considered one of the largest. This experience could not but affect the profile of the company, because today Cisco provides exceptional opportunities and solutions in the field of IP telephony and network management.

The 2005 year. Calypso Wireless introduced the C1250i phone to the market, the very first mobile phone that could switch between a GSM cell tower and an available Wi-Fi 802 network using the Cisco Aironet Access Point and Calypso's own proprietary Wireless ASNAP technology. With this, users could organize

video conferencing and make VoIP calls. This phone was considered a smartphone on WindowsMobile.

In 2006, the first mobile application for IP telephony Truphone was developed. Initially, this application was created for Nokia cell phones, but then it was also released for iPhone, Android, and BlackBerry platforms. Truphone could make free calls within its network, send a text to another network, including Skype, and make calls to the PSTN.

The application used the SIP standard and made calls, not via GSM, but through a Wi-Fi network. Later, the company released several softphones, and at the moment it is engaged in profitable travel SIM cards. Gradually, these softphones learned to cooperate with PSTN phones, mobile phones, as well as faxes, and e-mail. Moreover, until now, when testing IP-telephony tools in large companies, the test plan necessarily includes tests for sending faxes from one device to another device, fax to mail, and fax between SIP, H.323, and SS7 protocols.

IP telephony has a lot of advantages that made it very popular and left the development potential:

- it is cheap - each user receives a common tariff, regardless of the distance;
- it has open standards;
- it is relatively simple for developers;
- it is supported by many devices and platforms;
- it is easily integrated into third-party applications, etc.

Today, IP telephony surrounds us everywhere, including at home and in business. She made a breakthrough in the world of communications - telecommunications giants were forced to lower prices and look for optimal solutions for their customers. There are a lot of interesting moments in the history of IP telephony: from encryption to SORM operation in VoIP, from protocols to non-standard equipment. The story continues and still promises to be interesting.

1.3 General information about IP telephony

IP telephony is a technology that uses a packet-switched network to conduct all types of calls and faxes in real-time based on the IP protocol. The most common such network is the Internet. Sometimes you can find such a term - VoIP (Voice Over IP), otherwise known as "voice over IP", which means the transmission of voice information over IP lines. For a long time, circuit-switched networks (ie telephone networks) and packet-switched networks (IP networks) could exist independently of each other.

Some channels we could use to transmit voice information, and others - for data transmission. IP telephony has allowed us to connect both networks with the help of the so-called gateway - a device that stands at the junction of telephone and IP lines.

The following is an example [3] of the principle of operation of an IP phone system based on Nextiva Phone Systems.

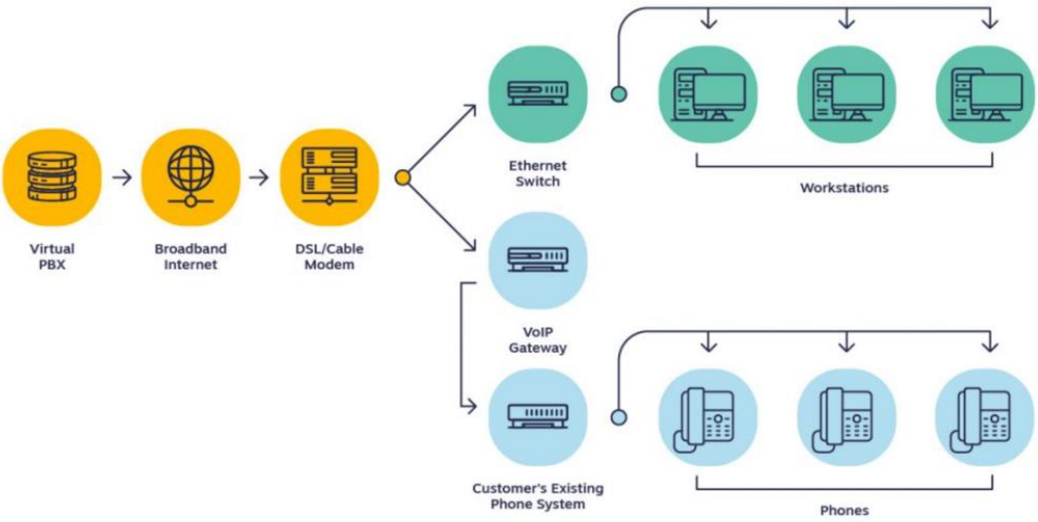


Fig. 1.1. Example of the Nextiva IP phone system

In the above example, an incoming call first contacts a virtual server hosted by the provider. The VoIP provider converts the call into an audio file and transmits the data to an IP phone. After that, we are connected to the caller by the Internet instead of a cell tower.

Protocols are what IP telephony itself consists of, they organize a telephone conversation, manage calls, and transmit traffic over a network. There are three large families of standards that differ in approaches to building a network: H.323, SIP, and MGCP.

The H.323 IP telephony standard is considered the oldest and is necessary for VoIP telephony and video conferencing. This is a huge set of protocols and elements that can allow media to be transmitted over packet networks with non-guaranteed bandwidth. The composition of the H.323 recommendation assumes various communication options - from simple telephony to video conferencing with further media transmission.

One of the strengths of the H.323 standard is its bridging function, which allows different manufacturers' devices to cooperate and communicate with each other.

The most basic network elements are terminal (Terminal), gateway (Gateway), gatekeeper (zone controller, Gatekeeper), and conference control unit (Multipoint Control Unit MCU). [4]

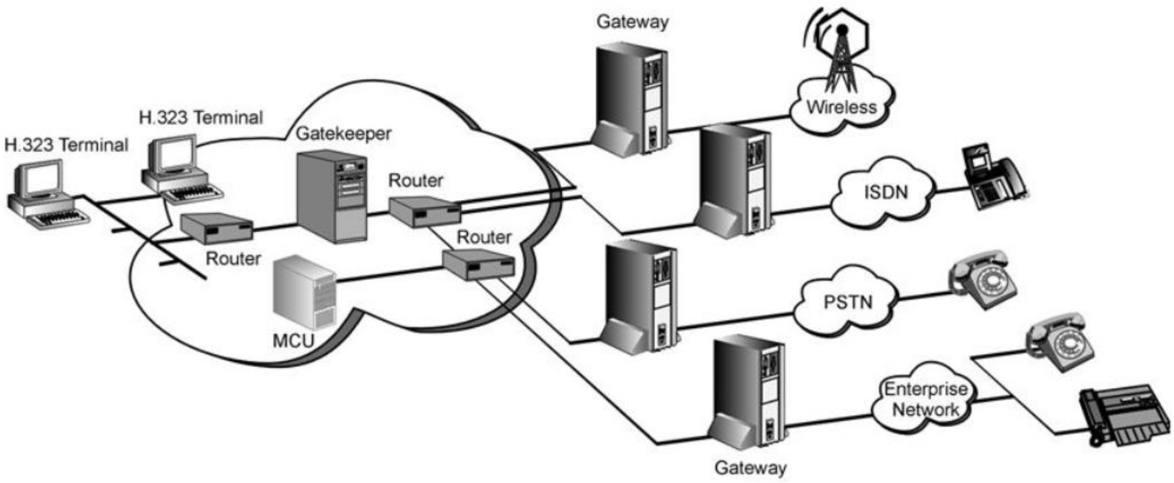


Fig 1.2 Network architecture and components of H. 323

Terminal - the terminal device of the user of the IP telephony network, providing two-way multimedia communication with other H.323 terminals, a conference control device, or a gateway.

The main purpose of the gateway is to convert the voice information that comes from the PSTN into a form that would be suitable for further transmission over networks with IP packet routing. In addition, the gateway converts SS7 and DSS1 signaling messages into H.323 signaling messages and performs the reverse conversion per ITUH.246 recommendation.

The gateway allows you to compress information (voice), convert it into IP packets, and send it to the IP network.

The conference control server (MCU) communicates between two or more terminals - H.323. All H.323 terminals that participate in the conference can connect to the MCU.

The conference control device must have one mandatory element of the conference controller (MC), and it may include several processors for processing user information (MP). The conference controller element can be combined with a zone controller, gateway, or conference control device, and the conference control device can be combined with a gateway or zone controller.

1.4 Ensuring Security in IP Telephony

VoIP technology is the present and future of our telephone communications. But at the same time, we cannot ignore the threats it poses to their security. VoIP has its drawbacks inherent in any IP service, given its complex structure and real-time service requirements. Most of these problems are addressed with secure IP PBXs and phones, deployment of VoIP-optimized firewalls (FWs), security-aware infrastructure upgrades, and general-purpose security tools. But before resorting to security recommendations, we need to identify existing vulnerabilities and threats.

IP telephony, being a follower of the IP network, inherits from it both advantages and disadvantages of security. IP telephony is considered a unique service of its kind, but it is no better protected than any other IP service, such as e-mail. These services are often the target of attacks, as they have their flaws and vulnerabilities.

A VoIP network requires more elements and programs than other classical circuit-switched networks. These elements include support servers, switches, routers, DOE, IP-PBX, cable systems, and softphones. The vulnerability is high if there are many components.

Today, there are a lot of IP telephony standards, such as Session Initiation Protocol (SIP), H.323, Media Gateway Control Protocol (MGCP), H.248, and other proprietary protocols.

Implementation flaws in protocols can be software bugs. For example, attacks carried out by an attacker when trying to check the size of a protocol request:

-remote access - an attacker can gain remote access

to the system under administrator rights;

- denial of service - as a result of malformed

the request fails the service;

- denial of service due to huge load - a large number of requests can "paralyze" a vulnerable system.

Network users always rely on the privacy of telephone conversations, compared to email or instant messaging services, which are not expected to be confidential. Although not most, some VoIP calls are encrypted. Along with this, encryption without strong authentication is not a guarantee of confidentiality, since it does not prevent an attacker from using attacks and gaining access to the transmission medium.

Conclusions to the Chapter 1

1. The analysis showed that the IP telephony service provided on packet-switched networks is currently being actively implemented in corporate networks of the business sector. Saving long-distance voice calls, especially for corporations with large data networks, is an important driver of IP telephony. Transmitting voice traffic on a data network within a business located on a building or campus can also achieve significant cost savings as the operation of modern PBX installations work relatively inefficiently.

2. It is shown that an important advantage of voice traffic over data networks is the integration of programs for voice and data, which can increase the efficiency of the business process. Examples of such programs are integrated voice and e-mail, teleconferencing, support for their collaboration, and automated and intelligent call distribution. The flexibility offered by IP telephony due to the transfer of intelligence from the network to the end stations and the open nature of the construction of IP networks is also important advantages that allow you to create new services. Thus, it can be concluded that VoIP technology has revolutionized corporate communication by allowing organizations cost-effectively streamline various communication methods to increase business productivity.

3. The considered stack of H.323 protocols, which is designed to support multimedia services in IP - telephony networks and which describes: the rules of

encoding, decoding, and packing of audio and video signals; alarm and call management; opportunities to exchange information on the Internet. H.323 defines the VoIP network architecture, which contains four main components of a network communication system: terminals, gateways, zone controllers, and multipoint control units, which enables the implementation of four IP telephony call schemes and integrates VoIP services with conventional public telephone services.

CHAPTER 2

EVALUATION OF CUSTOMER'S NETWORK

2.1 Network requirements for VoIP setup

2.1.1 Bandwidth

The first thing that we need to check is the speed of the Internet connection. Employees do not have to experience issues during peak times. It is very important to upgrade to a faster connection before deploying VoIP. [9]

As a telephone solution that relies on internet connectivity, the quality and speed of office broadband must be considered when migrating to VoIP. Internet speeds of at least 90-100 kbps (kilobits per second) are required to achieve good quality voice calls. This amount should be multiplied for each user. For example, if the customer has 10 VoIP handsets, they will need about 1 Mbps of bandwidth. [8]

2.1.2 Network security and encryption

Telecom providers need to protect their call data while on the network. However, we need to make sure that our local network is also protected on a high level. Many network devices provide hardware encryption. And there are many more security software options. If you are using wireless networks, always make sure your wireless networks and devices are password-protected, as WiFi networks are the most vulnerable to cyber-attacks. [8]

Wired connections are more reliable and secure than wireless connections. If you are building your own VoIP network, we recommend that you use an Ethernet connection from the beginning. That way, you don't have to switch from a WiFi network to a wired network later. There is nothing wrong with including a WiFi connection for mobile devices. However, it is unwise to rely on wireless networks for all VoIP communications. Use these criteria to determine the Internet connection plan you need and the best internal network components to set up your VoIP network. Once we know what we need to do with our network, continue with the setup. [8]

2.1.3 Quality of Service

Quality of service (QoS) manages and prioritizes certain types of traffic on the network to reduce packet loss, delay, and jitter for certain types of network traffic or protocols such as VoIP technology. [10]

QoS controls and manages network resources by prioritizing certain types of data on the network. QoS optimizes your network by managing bandwidth and prioritizing applications that require more resources than other applications. QoS is an underutilized tool that allows system administrators to customize network settings and share available bandwidth between applications. For example, with proper QoS rules, you can prevent streaming Netflix from being buffered because large files are being downloaded at the same time, or slow down your work laptop the last time you try to respond. Can be done. Deadline for minutes while roommates are playing online games. [10]

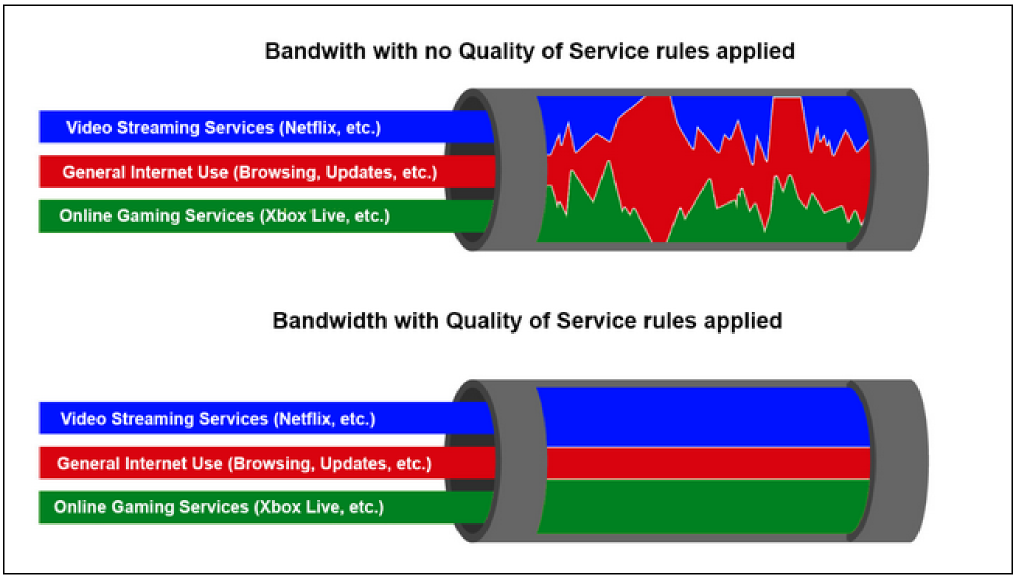


Fig. 2.1 Example of Bandwidth with QoS and without QoS

2.1.4 IP Addresses and IP Management

The Internet Protocol address (IP address) is a numeric specification assigned to each device that participates in one device. Function: Host or network interface identification and location addressing. [11]

Internet Protocol designers have defined IP addresses as 32-bit numbers, and this system is well known. Internet Protocol version 4 (IPv4) is still in use. Due to the expected depletion of the Internet and available addresses, a new version of IP (IPv6) was developed in 1995 that uses 128 bits for addresses. IPv6 was standardized as RFC2460 in 1998 and its deployment. It has been running since the mid-2000s. IP addresses are usually written and displayed in a human-readable format. Notations such as 172.16.254.1 (IPv4) and 2001: db8: 0: 1234: 0: 567: 8: 1 (IPv6). [12]

2.2 Network Tests

There are a couple of tests that we can use to help identify network-related issues, which measure certain aspects of the network connection:

2.2.1 Ping

Ping is a computer network administration software utility used to test the reachability of a host on an IP network, as well as to measure the round-trip time for messages sent from the originating host to a destination computer and back. The name comes from active sonar terminology which sends a pulse of sound and listens for the echo to detect objects underwater. Generally, I recommend less than 100ms150ms ping, but this includes the entire path. It is also important to remember that ping is only a crude measure because latency is influenced by packet type, packet size, Quality of Service (QoS) settings, and how much congestion the network has at any given time (Congestion is where bandwidth requirements become relevant).

2.2.2 Latency

Latency is the time interval between the stimulation and response, or more generally. Seen as a time lag between the cause and effect of physical changes in the system. Network delay in packet-switched networks is measured in either direction. A source that sends a packet to a destination that receives the packet or a round trip (one-way delay from the source).

2.2.3 Packet loss

Packet loss occurs when the network connection loses information during transmission. This can make our network connection seem slower than it should be and reduce the reliability of network communications with both local and remote devices. Stopping packet loss should be at the top of the list for anyone looking to improve a troubled network.

2.2.4 Jitter

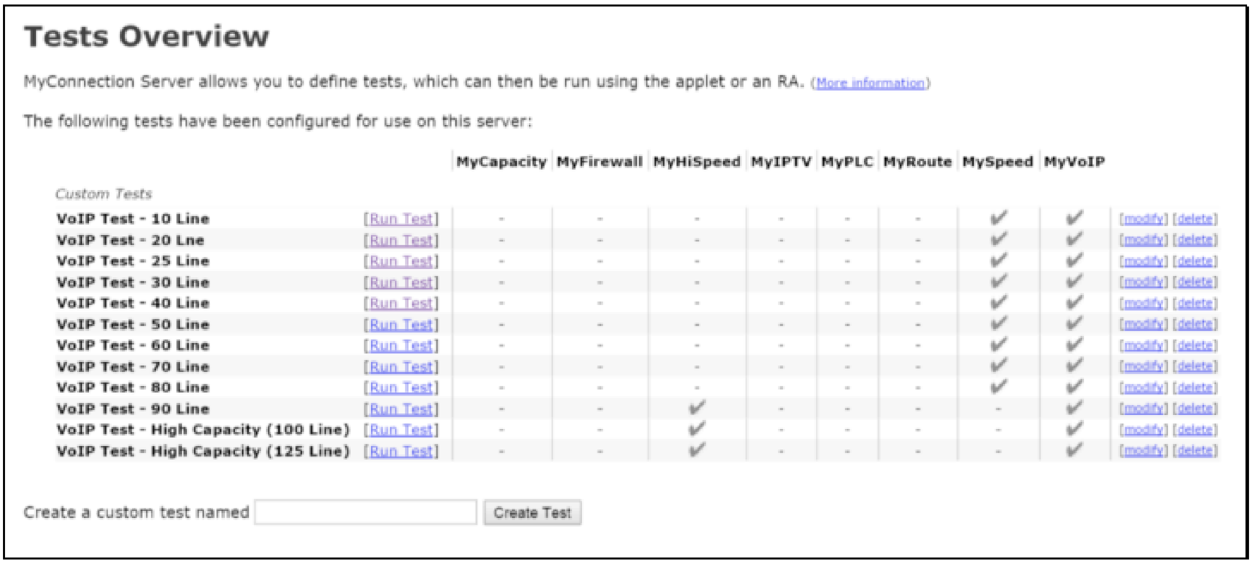
Jitter: Jitter is defined as a statistical variance of the data packet inter-arrival time. This may sound similar to latency because most VoIP-enabled devices have a built-in jitter buffer. This means that when packets arrive out of order the phone will attempt to put the packets in the correct order, even waiting for missing packets, causing latency in the process. This will usually occur with excessive network traffic when RTP packets are trying to jostle for position. For a Quality phone call, the jitter rating should not be higher than 15-20ms.

2.2.5 Nextiva Voiptest

Voiptest.nextiva.com: While there are numerous tools and methods to help measure the above and identify potential issues, Nextiva offers its VoIP-specific network test (Fig. 2.2). The Nextiva VoIP Test offers significantly more-realistic results than most other speed tests for a couple of reasons:

While most speed tests select a server close to your physical location, our VoIP Test connects to Nextiva points of presence (PoPs), providing more accurate results.

The Nextiva VoIP Test includes various sub-tests for SIP and other important criteria that regular speed tests do not.



Tests Overview

MyConnection Server allows you to define tests, which can then be run using the applet or an RA. (More information)

The following tests have been configured for use on this server:

Custom Tests		MyCapacity	MyFirewall	MyHiSpeed	MyIPTV	MyPLC	MyRoute	MySpeed	MyVoIP	
VoIP Test - 10 Line	[Run Test]	-	-	-	-	-	-	✔	✔	[modify] [delete]
VoIP Test - 20 Lne	[Run Test]	-	-	-	-	-	-	✔	✔	[modify] [delete]
VoIP Test - 25 Line	[Run Test]	-	-	-	-	-	-	✔	✔	[modify] [delete]
VoIP Test - 30 Line	[Run Test]	-	-	-	-	-	-	✔	✔	[modify] [delete]
VoIP Test - 40 Line	[Run Test]	-	-	-	-	-	-	✔	✔	[modify] [delete]
VoIP Test - 50 Line	[Run Test]	-	-	-	-	-	-	✔	✔	[modify] [delete]
VoIP Test - 60 Line	[Run Test]	-	-	-	-	-	-	✔	✔	[modify] [delete]
VoIP Test - 70 Line	[Run Test]	-	-	-	-	-	-	✔	✔	[modify] [delete]
VoIP Test - 80 Line	[Run Test]	-	-	-	-	-	-	✔	✔	[modify] [delete]
VoIP Test - 90 Line	[Run Test]	-	-	✔	-	-	-	-	✔	[modify] [delete]
VoIP Test - High Capacity (100 Line)	[Run Test]	-	-	✔	-	-	-	-	✔	[modify] [delete]
VoIP Test - High Capacity (125 Line)	[Run Test]	-	-	✔	-	-	-	-	✔	[modify] [delete]

Create a custom test named Create Test

Fig. 2.2 Voiptest.nextiva.com Home Screen

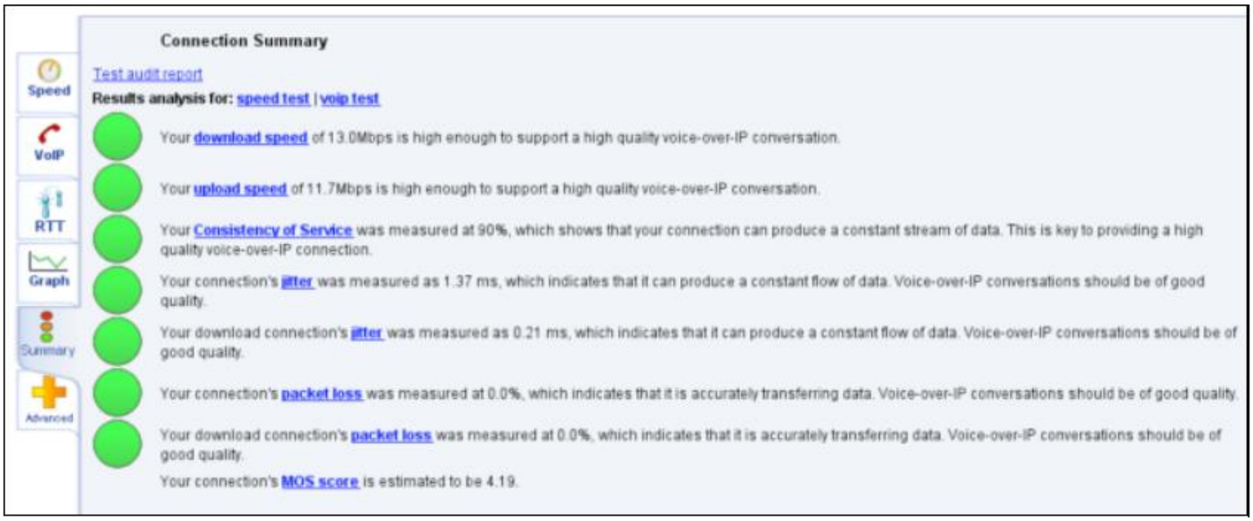


Fig. 2.3 Voiptest.nextiva.com Results Screen

From Fig. 2.3 we can see the summary of the Test audit report. Results are shown for the following options:

The Download speed is 13.0 Mbps - this is enough to support a high-quality VoIP conversation.

The Upload speed is 11.7 Mbps – this is a good result for VoIP service.

The Consistency of Service was measured as 90%, it is a high index to provide a constant stream of data. This is the most important thing to provide a high-quality VoIP connection.

The Connection’s jitter is 1.37 ms – it is an allowable index to produce a constant flow of data.

The Download connection’s jitter is 0.21 ms, which indicates that it can produce a constant flow of data.

The Packet loss was measured as 0.0% - which means that it is accurately transferring data.

The Download Packet loss was measured as 0.0% - which means that it is accurately transferring data.

The MOS score - Mean Opinion Score is estimated to be 4.19

Based on the results of the network analysis above, we can conclude that no problems were found at the network verification stage. Thus, all equipment connected to this network should work without delays and interruptions. Having checked the network, we can move on to an overview of network equipment.

2.3 Identifying Network Devices

First of all, we must ensure that the computer we are connected to is on the same network that the phones are connected to.

Then we need to open the command prompt. How to do this is dependent on the customer's computer's operating system.

Windows XP: Start Menu -> Run -> Enter "cmd" and press OK

Windows 7: Start Menu -> Type "cmd" into the search bar and hit enter.

Windows 8: Open the start menu -> type "cmd", it will show a command prompt as a program in the search.

Mac: Select the "Spotlight" search icon in the top right -> Search "terminal", and open the terminal program.

Run ipconfig (Windows) or ifconfig (Mac/Linux). Obtain the following pieces of information:

IPv4 Address - This is the local IP address of the customer's computer.

Default Gateway - This is the address of the device that assigned them their local IP address

Open up a web browser and type in the Default Gateway that you had acquired from the ipconfig.

The local gateway address should open up the customer's router. Usually, this will require an admin login, the default varies by the make/model of the device.

Look for the connection details, within the connection details it should provide the WAN IP address for that router. If the WAN IP address is a local IP (ranges listed above), then the device is being managed by another NAT.

Once inside the router, we can also use the device and/or DHCP client lists to identify what devices are connected to their network. The following sections outline the steps for finding devices in common routers.

Netgear: Netgear has an “Attached Devices” option on most of their “Basic” menus (Figure 2.4). To see the DHCP reservation list, go to the “Advanced” tab, then click on “Setup” and “Lan Setup”.

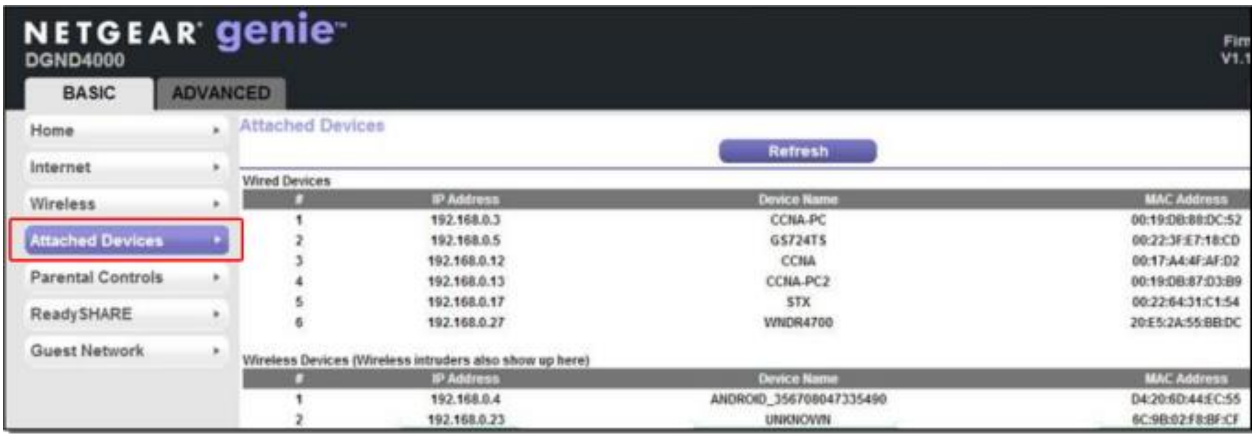


Fig. 2.4 Netgear Attached Devices

Fig. 2.4 Netgear DHCP Reservation List

Conclusions to the Chapter 2

A large number of electronic devices that cause interference, and poorly performed breeding - all this can significantly reduce the speed of the local network. That's why you need to measure the speed inside the network to identify "vulnerabilities".

The speed on the local network depends primarily on the bandwidth of the router, cable compression, and devices on which you plan to watch the video. A modern router, a well-prepared cable, and modern gadgets are the key to a good home network.

The connection speed can be significantly increased by upgrading or replacing obsolete network equipment, respectively, to increase the data transfer rate in the home network.

The network was analyzed using Nextiva Voiptest, based on the results we can say that the network is suitable for use on its VOIP service. There is also an example of how to identify the number of devices connected to the router.

CHAPTER 3

BUILDING A SIP TRUNKING ACCOUNT

Before we start building a SIP Trunking account, let’s consider another protocol widely used in VoIP - the SIP protocol (which works in VoIP networks in conjunction with the popular H.323 protocol, described in section 1).

The SIP protocol was proposed by the IETF MMUSI Working Group to provide multimedia sessions and provides the following features:

- User position. SIP allows you to detect the end user's position to establish a communication session or send a SIP request. User mobility is initially supported by the SIP protocol;
- User capabilities. The SIP protocol allows you to find out the capabilities of the transmission medium and the devices involved in the session;
- User availability. The SIP protocol allows to find out the readiness of the end-user to establish communication;
- Setting up sessions. The SIP protocol allows you to set session parameters for the parties involved. The SIP protocol allows you to modify, transfer and end an active session.

3.1 SIP network elements

1. User Agent (UA). A logical function in a SIP network that initializes or responds to SIP transactions. The UA agent can act as a client or server of SIP transactions. The UA agent may or may not interact directly with the user (person). Agent UA has a fixation stateful, ie it is ablecanin the state of the session or dialogue.

2. User Agent Client (UAC).

A logical function that initializes SIP requests and receives SIP responses. Examples of UAC agent work are initiating a SIP telephone request on behalf of a user or redirecting a SIP request to a proxy server on behalf of a UAC.

3. User Agent Server (UAS). A Boolean function that accepts SIP requests and sends back SIP responses. The SIP phone, for example, accepts requests such as INVITE.

4. Proxy server. A proxy server is an intermediate entity in the SIP network that is responsible for redirecting SIP requests to the target UAS agent or another proxy server on behalf of the UAC agent. But first of all the proxy server carries out routing in the SIP network. The proxy server may also be responsible for supporting network policies, such as authenticating the user before providing services. The proxy server can work without locking the status, locking the status of transactions, or locking the status of calls. Typically, proxy servers work with recording the status of transactions, ie they maintain the state for the duration of the transaction (approximately 32 seconds).

5. Redirect server. A redirection server is a UAS agent that generates SIP Class 300 responses to received requests by redirecting a UAC agent to an alternative set of Uniform Resource Identifiers (URIs).

6. Registrar server. A UAS agent accepts SIP REGISTER requests and transfers information from the request to the location database.

7. Back-To-Back User Agent (B2BUA). An intermediate object that handles incoming SIP requests as a UAS agent. To respond to incoming SIP requests, the B2BUA agent acts as a UAC agent, recovering the SIP request and sending it over the network. The B2BUA agent must maintain the status of the dialogue and participate in all dialogue transactions.

Figure 3.1 shows the SIP network, which contains SIP proxy servers and user agents connected to the public telephone network (PSTN). The SIP UAC agent, proxy servers, and SIP gateway are located within the IP network. Solid black lines in Figure 3.1 indicate SIP requests, and green lines indicate SIP responses, as well as dashes, are for media.

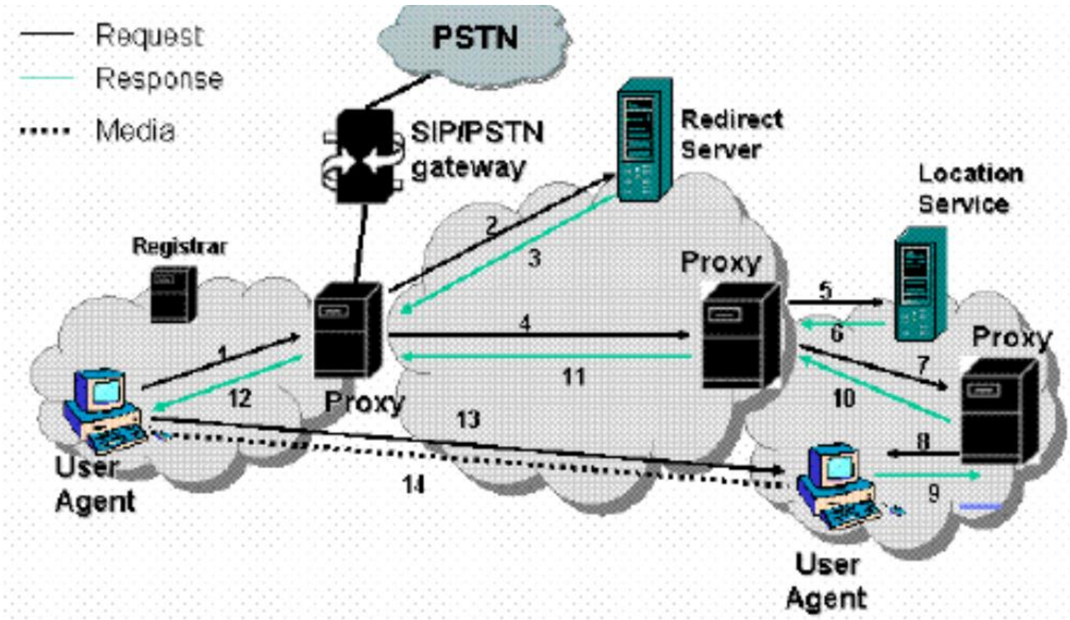


Fig. 3.1 Query and response paths in the SIP network

3.2 Detailed analysis of the PBX

A PBX is a telephone system that enables the communication between telephones within an organization. Nextiva's NextOS SIP Trunking Service provides the internal PBX with a connection to our carrier, which allows us to share this connection between all the phones in the organization.

An in-house PBX allows companies to call two or more phones to each other without using Nextiva's network. For example, if a phone connected to an Avaya® PBX calls another phone connected to the same PBX, the phone will not be routed through Nextiva's network. Instead, the calling phone forwards the call to the PBX and then directly to the receiving phone.

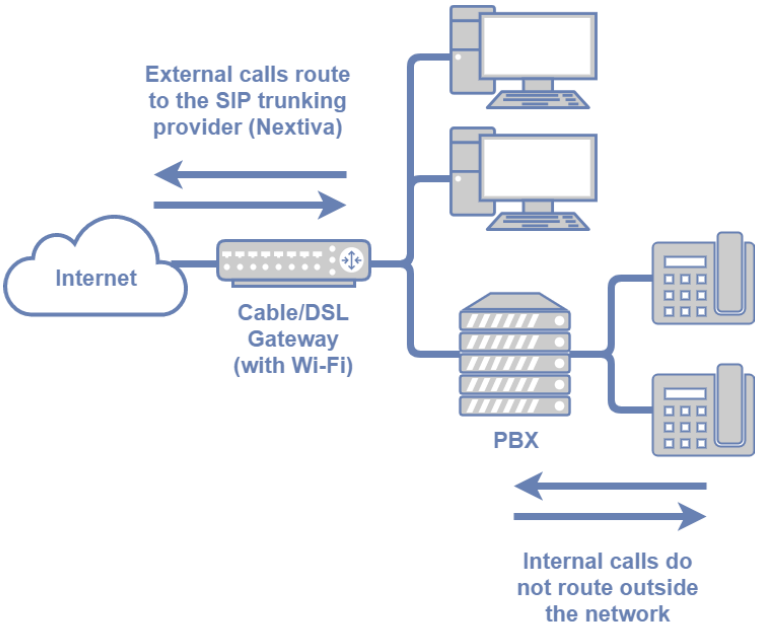


Fig 3.2 An example of a network diagram with an in-house PBX

3.3 Advantages of an in-house PBX

Less monthly cost

With in-house PBX, customers can save money over Nextiva's hosted PBX solutions. For example, a customer of 20 employees with a NextOS Enterprise account would have to pay $ 699.00 per month and 20 lines of service with fees and taxes. However, with the NextOS SIP Trunking Account, the number of SIP trunks a customer needs to purchase only needs to match the maximum number of concurrent calls expected. If the maximum number of simultaneous calls is 5, the monthly charge to the customer is $ 124.75 (5 x $ 24.95) plus fees and taxes. [11]

Better control over PBX and phone configurations

With an internal PBX, customers do not have to contact Nextiva Support to make phone configuration files or make global changes everywhere at once. [11]

3.4 Configuring a Sip Trunking Account

Setting the Trunking Call Capacity

Call capacity refers to the number of simultaneous calls that the customer can make or receive at one time. Call capacity is specified in multiple places on a NextOS SIP Trunking account. An admin with system-level access needs to set the Trunking Call Capacity at the Enterprise level.

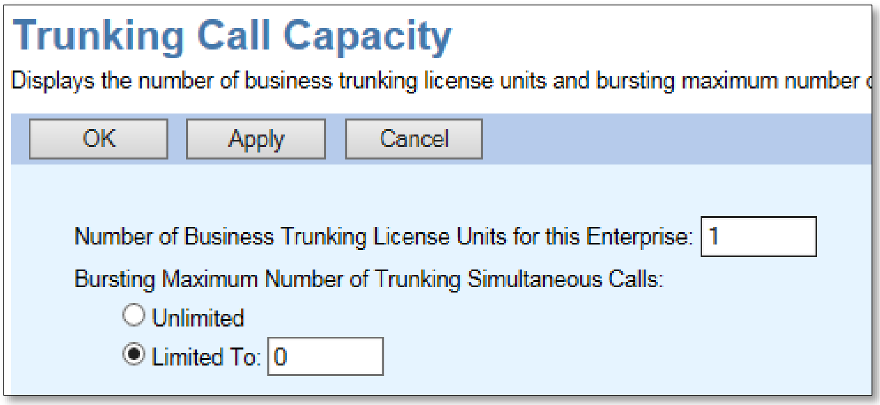


Fig. 3.3 The Enterprise Trunking Call Capacity

A NextOS SIP Trunking account can be set up using either the bt.voipdnsservers.com domain or the pai.voipdnsservers.com domain. By default, only the bt.voipdnsservers.com domain is added to a NextOS SIP Trunking account. Adding the pai.voipdnsservers.com domain may or may not be necessary depending on the make and model of the customer's PBX. Table 3.1 below lists common make and model PBXs and their corresponding domains.

Table 3.1

Common PBXs and their corresponding domains

Make and Model	Domain
Asterisk (any version)	bt.voipdnsservers.com
Avaya (any model)	pai.voipdnsservers.com
Elastix	bt.voipdnsservers.com
FreePBX	bt.voipdnsservers.com
Grandstream (any	bt.voipdnsservers.com

model)	
Panasonic (any model)	pai.voipdnsservers.com
Trixbox	bt.voipdnsservers.com
Yeastar (any model)	bt.voipdnsservers.com
Zultys (any model)	pai.voipdnsservers.com

Setting Call Processing Policies

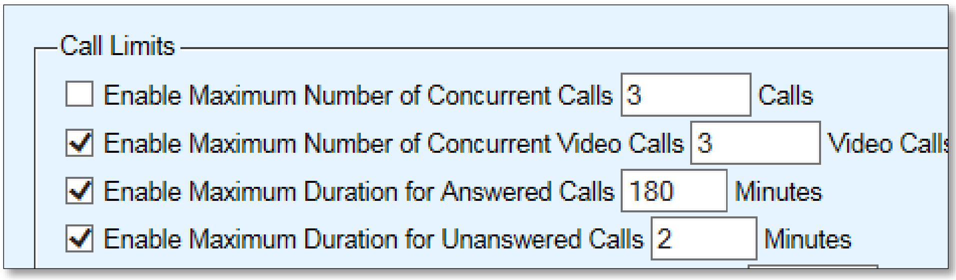
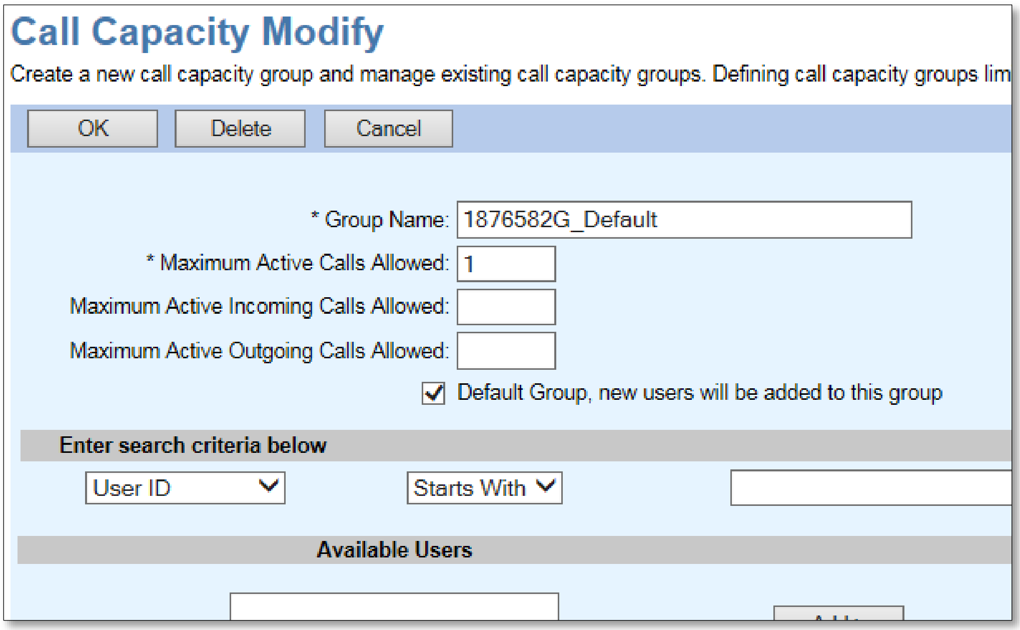


Fig. 3.4 The Call Processing Policies

By checking the next 3 boxes which are shown in Fig 3.4, the customer can be on 3 calls simultaneously, and also we can define the maximum duration of answered calls and the maximum duration of unanswered calls according to request.

Setting the Group Call Capacity

If there is a hybrid Trunking account, first of all, we need to increase the Group Call Capacity to allow hosted users to make and receive calls. As a rule, add three additional calls for every hosted user. As an example, if the account has ten trunks and six hosted users (including the account manager), the total call capacity should be set to 28 [10 + (3 x 6)].



4.1 The Group Call Capacity

Checking the Available Services

The purpose of checking the available services is to ensure that there is more than one trunk available if setting up multiple trunks and that there is no limit on the services available to create users.

The number of allowed trunk groups is set to 1 by default. If we need to set up more than one trunk group (because the customer has more than one PBX), increase the number of allowed trunk groups at the enterprise level first.

Increasing the Number of Allowed Users

Each phone number on a NextOS Trunking account is assigned to a different user. The purpose of increasing the number of allowed users is so you can set up as many users as there are phone numbers while continuing to have a user for the account manager.

Change the number of allowed users to the number of phone numbers the customer has, plus one for the account manager. For example, if the customer has two phone numbers, then enter three in the Limited to the text box, so the customer can have two users for each phone number in addition to their account manager.

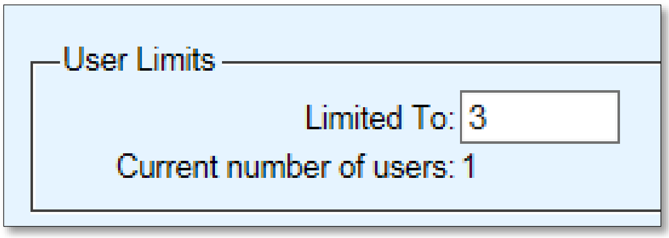


Fig. 3.6 The Group Profile user limits

If there is a hybrid trunking account, we need to further increase the number of allowed users to allow the creation of hosted users. As an example, if the account has ten phone numbers and six hosted users (including the account manager), the total number of allowed users should be set to 16 (10 + 6).

Assigning Numbers to the Group

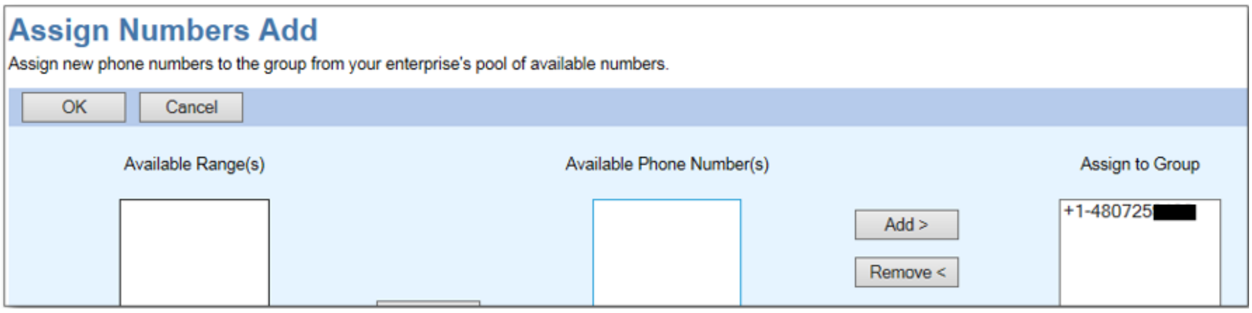


Fig. 3.7 The Assign Numbers

Adding a New Device Profile

Adding an Identity/Device Profile is similar to adding a device to a hosted account; it is necessary to generate authentication details for the PBX to register with. However, unlike a hosted account, you do not add the device to the NextOS 3.0 portal.

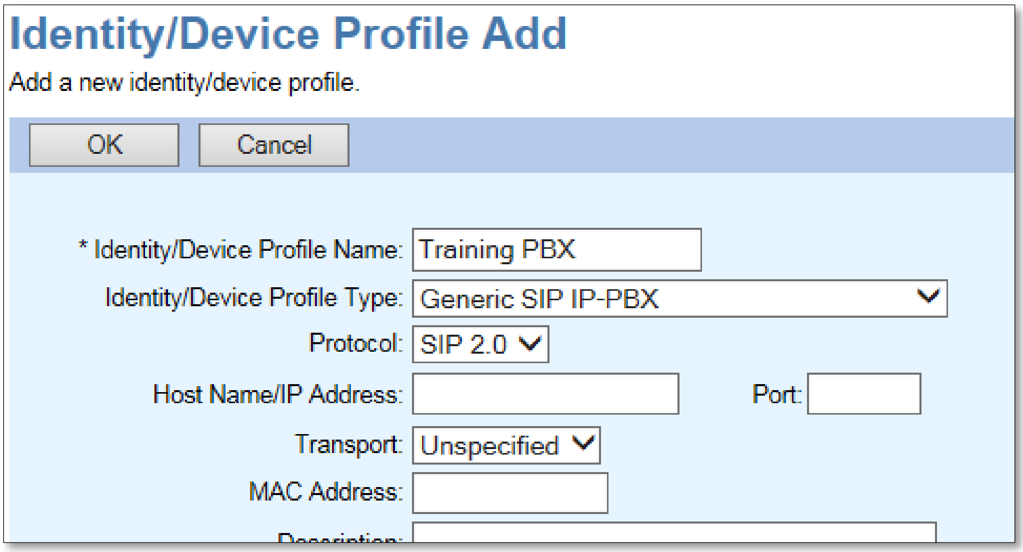


Fig. 3.8 The Identity/Device Profile

Creating a New Trunk Group

A Trunk Group is a group-level service that contains the authentication details that the PBX uses to register. A trunk group also specifies the maximum number of active calls allowed and controls whether the customer can change their outgoing caller ID number.

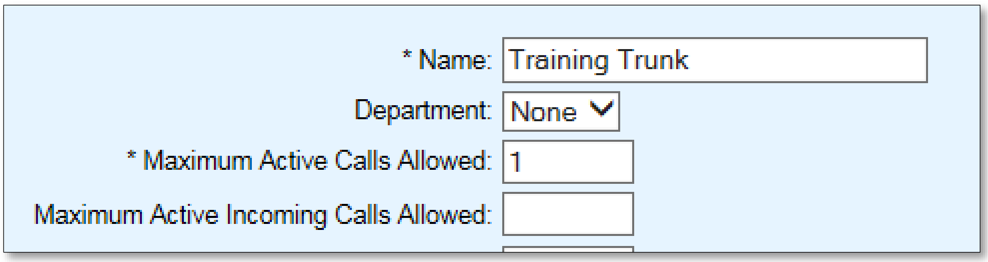


Fig. 3.9 The Trunk Group Add

The number we enter should be the customer's "main" phone number, which we need to get from them. If the NextOS SIP Trunking account is being built with temporary numbers, it does not matter which phone number we entered.

The next step is manually typing in a random number in the Type new authentication password and Re-type new authentication password text boxes. Make sure to document the number, as it will be used to register the PBX.

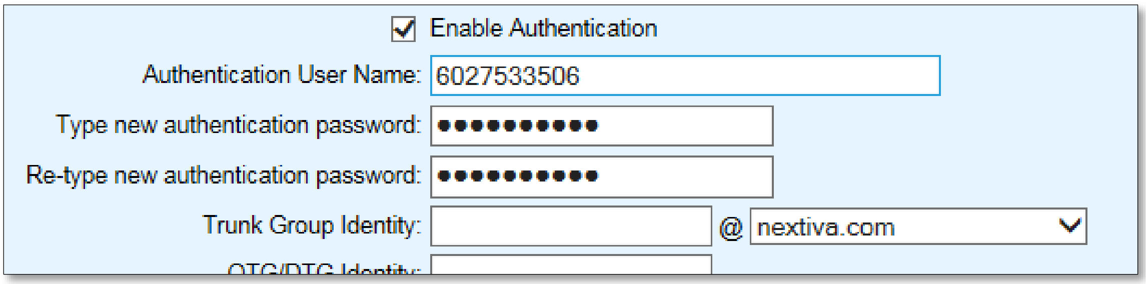


Fig 3.10 The authentication details for the trunk

Then we need to check the Allow Unscreened Calls check box toward the bottom of the page. This is necessary to allow the customer to spoof their outgoing caller ID number (Fig. 3.11).

And also check the Allow Unscreened Emergency Calls check box toward the bottom of the page, just underneath Allow Unscreened Calls. This is necessary to allow the customer to make 911 calls.

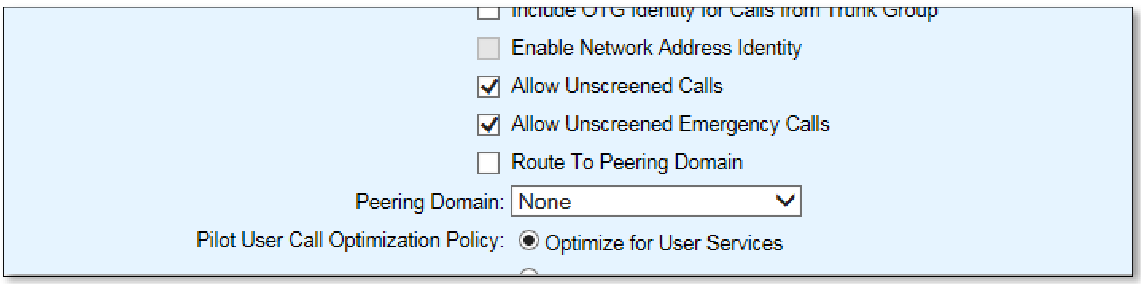
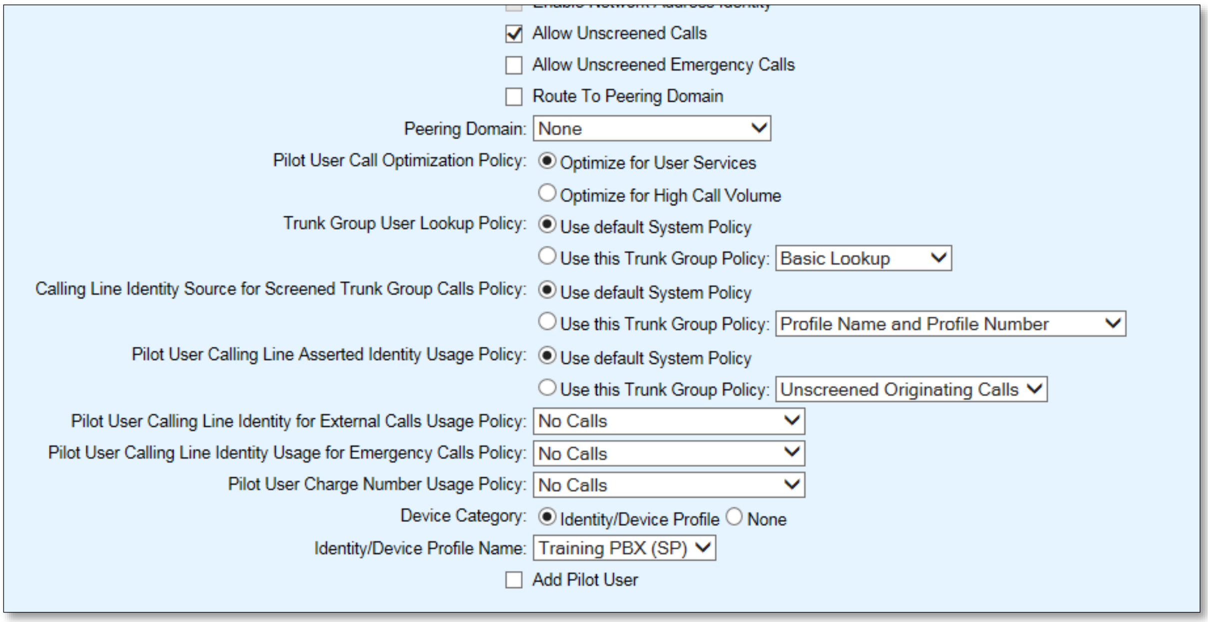


Fig. 3.11 The Allow Unscreened Calls and Allow Unscreened Emergency Calls checkboxes

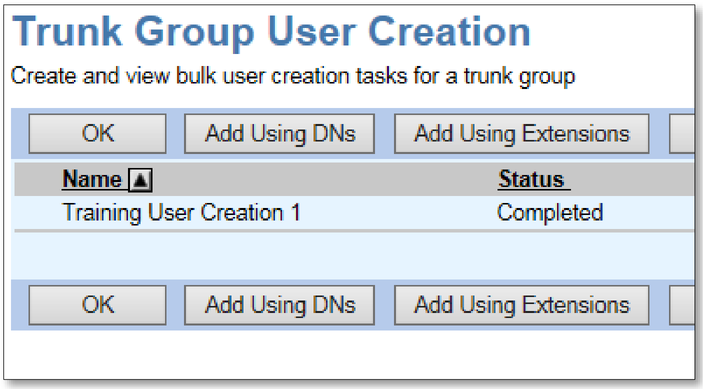
The next step is to select the Identity/Device Profile created earlier in the drop-down box below the Identity/Device Profile radio button.



3.12 The Trunk Group Add screen with an Identity/Device Profile selected

Assigning phone numbers to a Trunk Group is not like assigning phone numbers on a hosted account. Each phone number is assigned to a user, which is created through an automated process.

Trunking users are created automatically in the CP portal, as opposed to hosted users which are created manually in the NextOS 3.0 portal. It is not possible to assign numbers to a trunk group manually, either as direct numbers or as alternate numbers. It is also not necessary to obtain a user list from the customer before setting up the users.



3.13 The Trunk Group User Creation

Conclusions to the Chapter 3

SIP-Trunk is a virtual communication channel between a company providing digital telephony services and a client station. It allows you to connect the required number of numbers. Moreover, each of these numbers can have an unlimited number of channels (simultaneous negotiations on one number).

A SIP trunk is not a point, but a connecting line - usually multi-line.

During the analysis of the use of SIP Trunk, the following features can be distinguished:

- organization of the work of the call center and internal communications of employees;
- ensuring high-quality and, most importantly, reliable communication;
- saving money and resources

CONCLUSION

The analysis of the VoIP telephony market is carried out, the main components of the network IP architecture, protocols, and recommendations for voice transmission over the IP, and the main types of calls and services provided in VoIP - telephony are considered. The conditions of VoIP telephony creation are shown, and the differences between fixed and mobile telephony are given. It is shown that the important advantages

VoIP technology is the open nature of building IP networks, integrating voice and data applications and its flexibility by moving intelligence from the network to the end stations. This can allow organizations to cost-effectively organize different ways of communication to increase business productivity.

A simulation model of VoIP telephony service based on Nextiva Phone Systems was built, which allowed to create a working model of VoIP network and acquire practical skills of working with VoIP telephony devices, to set up a SIP Trunking account.

An optimized structured implementation of the telephone network for small and large businesses is proposed, which involves the use of a SIP Trunking account based on Nextiva Phone Systems. This solution allows you to easily and efficiently optimize the company's telephone network, get a wide range of VoIP - telephony features and reduce the cost of paying for used user.

REFERENCES

1. Гольдштейн В.С., Пинчук А.В., Суховицкий А.Л. IP-Телефония. — М.:Радио и связь, 2001. — 336 с.: ил. (date of access: 24.05.2022).
2. IP-телефония в компьютерных сетях : учебное пособие / И. В. Баскаков, А. В. Пролетарский, С. А. Мельников, Р. А. Федотов. — 3-е изд. (date of access: 24.05.2022).
3. https://www.nextiva.com/blog/ip-phone-systems.html (date of access: 24.05.2022).
4. https://slideplayer.com/slide/3542891/ (date of access: 24.05.2022).
5. Azarova, AA Computer networks and telecommunications: textbook / Azarova AA, Lysak NV - Vinnytsia: VNTU, 2012. - 293 s. (date of access: 24.05.2022).
6. VoIP market size forecast for 2019-2025. Report on the analysis of the share of the industry, date of publication: April 2019, 400 p. Report ID: GMI2989 (date of access: 24.05.2022).
7. VoIP Market Size Forecast 2019-2025. Industry Share Analysis Report - Name from the screen. URL: https://www.gminsights.com/industry-analysis/voice-over-internet-protocol-voip-market (date of access: 24.05.2022).
8. https://telnyx.com/resources/voip-network (date of access: 24.05.2022).
9. https://voipstudio.com/blog/what-are-the-network-requirements-for-voip/ (date of access: 24.05.2022).
10. https://www.nextiva.com/downloads/guides/Nextiva-Clarity/Nextiva-Clarity-User-Guide-v2.1.pdf (date of access: 24.05.2022).
11. https://www.nextiva.com/products/pbx-sip-trunking.html (date of access: 24.05.2022).
12. https://www.tek-tools.com/network/qos-monitoring-tools (date of access: 24.05.2022).