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DEPLOYING A SIMPLE VOICE OVER IP NETWORK USING A SIMULATION TOOL



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Voice over IP is a major advancement in the field of IP communications systems technology since the advent of Internet. It is a communication technology which enables a device to transmit and receive voice traffic with the help of an IP-based network such as the Internet. Various types and deployments of Voice over IP are prevailing due to its popularity since its origin. Since its advent, it has managed to evolve and has given a platform to be benefited with its numerous advantages not only to large-scale companies or enterprises but also to small-medium companies and ordinary people.

Even though Voice over IP technology has been used by many large and small companies, there are still many small companies which might need to have Voice over IP services implemented in their offices. Even a large company might need to establish multiple branches and needs to deploy a simple Voice over IP network for its branches. The main objective of this thesis is to encourage, clarify and guide the reader to have an understanding of the concept of Voice over IP, knowledge of purpose, planning and preparation for its implementation and skills towards the configurations of the devices needed to make the Voice over IP network fully functional. This thesis focuses on the deployment of a simple voice network with a switch, a router, four IP phones, two PCs and laptops.

To have a practical approach for the configuration of the devices of the Voice over IP network, we use a virtual simulation software called Packet Tracer. It enables us to virtually simulate a real Voice over IP network giving us the facility to configure each device in the network topology so that it can be implemented later.

KEYWORDS: Voice network, IP telephony, VoIP, IP phone, Voice, Packet Tracer, VLAN, DHCP, PoE, CME, QoS.

CONTENTS

LIST OF ABBREVIATIONS (OR) SYMBOLS.....	5
1 INTRODUCTION.....	7
2 THE CONCEPT OF VOIP.....	9
2.1 VoIP technology overview.....	12
2.2 Pros and Cons of VoIP.....	14
3 UNDERSTANDING THE DEPLOYMENT OF VOICE NETWORK.....	19
3.1 Purpose for Voice implementation in LAN.....	20
3.1.1 Cost savings.....	20
3.1.2 Increased efficiency and productivity.....	22
3.2 Planning for voice support in LAN.....	24
3.3 Preparing a LAN infrastucture for VoIP.....	28
3.3.1 Voice VLANs.....	30
3.3.2 QoS for voice traffic from IP phones.....	31
3.3.3 PoE.....	32
4 PREPARING THE DEVICES USING A SIMULATION TOOL.....	35
4.1 Switch configuration.....	37
4.2 Router configuration.....	42
4.2.1 Subinterface configuration.....	42
4.2.2 DHCP pool configuration.....	44
4.2.3 CME configuration.....	48
5 CONCLUSION.....	52
REFERENCES.....	54

APPENDICES

Appendix 1. Running-configuration of the switch.....	55
Appendix 2. Running-configuration of the router.....	56

FIGURES

Figure 1 IP telephony topology of small network with IP phones and PCs.....	35
Figure 2 IP phones powered by PoE.....	36

LIST OF ABBREVIATIONS (OR) SYMBOLS

IP	Internet Protocol
VoIP	Voice over Internet Protocol
WAN	Wide Area Network
MAN	Metropolitan Area Network
LAN	Local Area Network
SMS	Short Message Service
PSTN	Public Switched Telephone Network
ATA	Analog Telephone Adapter
POTS	Plain Old Telephone System
ATM	Asynchronous Transfer Mode
SONET	Synchronous Optical Network
UPS	Uninterruptible Power Supply
QoS	Quality of Service
DoS	Denial of Service
SOHO	Small Office Home Office
PDA	Personal Digital Assistant
TDM	Time Division Multiplexing
PBX	Private Branch Exchange
ROI	Return Of Investment
VLAN	Virtual Local Area Network
PoE	Power over Ethernet
IVR	Interactive Voice Response
TCP	Transmission Control Protocol
UDP	User Datagram Protocol
DHCP	Dynamic Host Configuration Protocol
CUCME	Cisco Unified Communications Manager Express

CME	Communications Manager Express
ISR	Integrated Service Router
CUCM	Cisco Unified Communications Manager
CDP	Cisco Discovery Protocol
PVID	Port VLAN ID
VVID	Voice VLAN ID
DSCP	Differentiated Services Code Point
CoS	Class of Service
LLDP	Link Layer Discovery Protocol
GUI	Graphical User Interface
CLI	Command Line Interface
TFTP	Trivial File Transfer Protocol
DNS	Domain Name System

1 INTRODUCTION

IP telephony is a technology to make voice communication possible using Internet Protocol and IP networks, such as the Internet. Many companies are still using traditional telephone services as the typical and old form of voice communication. These services are still prevalent these days as well. This thesis serves to generate the awareness of eliminating the use of traditional landline telephones over IP telephony deployment for voice communication within a small LAN. Inspiration for IP telephony deployment and deploying a simple voice network in small LAN is the main purpose and motivation of this thesis. It is a useful guide for anyone who wants to eliminate the legacy telephone services with landline telephones and needs to create a well functioning converged voice network with data network for the implementation of IP telephony with IP phones.

The focal points of this thesis are to encourage small companies to deploy voice network and to provide the necessary configurations for router and switch with IP phones and PCs attached to support IP telephony over a small LAN. Prevalence of the use of traditional telephone leased-lines has prevented several companies to harvest the benefits of converged voice and data network for IP telephony deployment. The idea of elimination of this restriction is the seed for this thesis. With the rise of the idea, the need of knowing about voice over IP network with its advantages and disadvantages, the purpose, plan and preparation of its implementation and the commands used to configure switch and router with IP phones and PCs to support voice facility within a small LAN emerged.

In addition, a virtual simulation software is used to simulate the voice network with the desired topology and the necessary configurations. To address the needs, this thesis has been constructed so that the second chapter details the basic concept, technological overview and pros and cons of VoIP network. The third chapter then clarifies the purpose, planning and preparation of voice

network for VoIP network. Furthermore, the fourth chapter illustrates and explains the steps for configuring the switch and router with IP phones and PCs with packet tracer. At last, the conclusion wraps up the whole thesis in the fifth chapter.

2 THE CONCEPT OF VOIP

The term VoIP (Voice over Internet Protocol), for which some say "Voice over IP", is pronounced as "vee-oh-eye-pee" or just "voyp". Internet Protocol (IP) telephony is commonly referred as VoIP and is used interchangeably throughout this thesis. IP telephony or VoIP applies digital networking technology to create, transmit and receive telecommunication sessions over IP networks.[1] This technique is an advancement in the field of internet technology and internet usage made for the convenience of supporting voice facility within a networked infrastructure, be it a small company with few employees or a large company with many employees within different branches, which lets a user communicate with another user in real time using various devices customized to support this technology. The word user here is a person using the end device to make a voice communication over any IP-based network regardless of size of the network, for instance, a Wide Area Network (WAN), Metropolitan Area Network (MAN), Campus Area Network (CAN), Local Area Network (LAN).

VoIP is a set of technologies allowing voice applications such as telephony, teleconferencing and voice messaging. It is also commonly referred to IP telephony, internet telephony, broadband telephony and broadband phone. IP telephony specifically refers to a methodology provisioning communications services such as voice, fax, voice instant messaging and Short Message Service (SMS). The technology with which we can implement voice communication using Internet Protocol and IP networks, allowing us to make voice calls over a broadband internet connection, rather than over a traditional analog telephone system- referred to as Public Switched Telephone Network (PSTN), with the help of IP phones, Softphones or Analog Telephone Adapter (ATA) is understood as VoIP. It is an advanced or extended form of IP communication which supports voice for communication purposes, usually to and from users over the internet. VoIP over a LAN using IP telephony for a

small sized company is nothing but basically a user communicating with another user with the help of IP phones connected to a networked infrastructure.

VoIP competes with voice facility of mobile phone services as it offers free or lower cost IP call to anywhere in the world via a Wi-Fi or an Internet connection. If a company considers saving costs regarding massive typical old telephone bills then implementation of VoIP telephony in the office is the best solution. It provides a way to implement voice communication for about no cost at all as it is managed within a network which makes data communication. As a new and emerging technology for IP communication, VoIP telephony is considered to be one of the most low cost and flexible voice communication technologies. It can be implemented on a private network with an option of having connection to global telephone network. The main reasons for choosing a VoIP telephony implementation are the extra capabilities of VoIP telephony and the flexibilities provided with its implementation to the users and to the company as a whole.

Since the origin of VoIP or internet telephony, many technology-based businesses and companies have satisfied the needs of their customers and benefited from VoIP technology by developing new standard and protocols and utilizing the availability of high speed internet and broadband services to provide voice communication services to large, small-medium companies and to the general public.[1]

In 1995, PC-to-PC internet telephony was developed by hobbyists from Israel to communicate with each other with the help of their computers and internet connection. Internet Phone Software was developed by Vocaltech in 1995. Later in 1998, many companies emerged being able to setup VoIP gateways to provide services such as PC-to Phone and Phone-to-Phone IP calling by using internet telephony. During VoIP's evolution, many businessmen and entrepreneurs soon realised the market and business value of it and made it more profitable by providing new standards and protocols to satisfy consumer needs. In the early stages, VoIP used to be connected in between PSTN for long distance calls to minimize the expenses needed to deploy internet telephony for it.

Before 2005, the development of VoIP was just only on non-commercial and small scale. But in 2005, the level of its progression reached to the point where it could be used to challenge the telecommunication industry to satisfy normal consumer needs.[1] It started to become a lucrative technology for most entrepreneurs and businesses since the availability of high speed internet and faster broadband internet services. Hardware manufacturers such as Cisco and Nortel produced advanced hardware for VoIP that was capable of switching the voice data packet into something readable by PSTN and vice versa which was the turning point in VoIP's history. The expansion on its usage and popularity started when the hardware became cheaper and VoIP implementation for internal IP network was possible for large companies, utilizing it to save on both long distance and infrastructure costs.

As technology is always changing and emerging to its fullest, there is always a need to make technology easy to implement, easily available, accessible and as cost effective as possible amongst all the previous ones. These qualities are fulfilled by VoIP. Since, the invention of internet, many traditional technologies which opt out the use of public internet have been disappeared as the internet technology industry has always managed to make alternatives to the old fashioned technology. Dependency on a most popular technology in today's world that is the Internet and easily available, accessible and manageable IP based networks have helped a lot of companies to make the transition from PSTN to VoIP telephony possible. The trend of relatively low fees for the use of a broadband internet and demand for even faster internet access has only made the hype of VoIP telephony grow even much more in time among customers of companies providing communications services.

The expansion of VoIP to residential users and it being a mainstream alternative technology to standard telephone service have helped it to become more and more popular in time. The hassles of implementing a legacy telephone network for a company and providing connection to its consumers includes installation of much more devices and wires. In addition, much more manpower and cost is needed to manage the facilities connecting an end user to PSTN than in the

implementation of VoIP telephony over public internet or even within a complex IP network. Furthermore, VoIP implementation for a company willing to provide voice communication for its customers is a less expensive solution over typical landline telephone service. Fundamentally, we can say that the migration from PSTN to VoIP and popularity of VoIP since its origin are solely based on imparted above-mentioned benefits of VoIP to an end user and companies providing voice communication services.

2.1 VoIP technology overview

VoIP is a technology for converting analogue voice data which we get to hear during phone calls to digital voice data to traverse throughout an IP based network over the Internet. It is an advance IP communication technique based on the routing of voice packets over public Internet through IP based networks for voice communication purposes. It is an important technology used on several platforms to be benefited with voice communication. These platforms support voice application on a desktop (softphone) or a mobile/smartphone to reach another similar user with similar voice application installed on his/her desktop or mobile/smartphone over the Internet or IP based networked infrastructure like a home, entities like a school, campus or university and companies.

Due to the easy availability of the Internet, VoIP applications are free or nearly free of cost to download and many companies and vendors facilitate free service to Internet-only voice calls but with optional access to PSTN, they charge a modest fee to place a voice call. For VoIP telephony calls to originate, the process involves steps and principles similar as traditional digital telephony with involvement of digital signaling, setup of a channel, digitization of the analog voice data and analog signals generated from IP phone, and encoding of analog signals to digital signals. A voice communication is possible on a VoIP telephony, after the encoding takes place, when the digital information is packetized and IP packets are transmitted over a packet switched network such as IP networks and Internet which travel through different IP based devices to

reach another end, instead of using a traditional approach and transmission occurring over a circuit switched network.

Various flavors have been taken for VoIP implementation since its origin and it has only been emerging with newer technology.[1] VoIP takes four different ways to make a voice call:

- ATA: Analogue Telephone Adapter

An ATA acts as a mediator or an intermediary device between a standard traditional phone and a computer or a router for internet connection to allow the standard phone to connect to internet and make a VoIP call easily. The vital work for a VoIP call to occur using the standard phone is done by ATA; as we speak on the telephone, ATA converts the analog signals received from the telephone into digital signals capable of being transmitted over the Internet. Using an ATA as an intermediary device between a standard telephone and Internet is just the simplest way it can be used. Setting up an ATA is as simple as plugging the telephone cable into the ATA and ATA cable to the Internet but some ATAs need to be configured with additional software installed on the computer connected to it.

- IP phones

IP phones are the best solution for Voice over IP network with a device with all necessary softwares and hardwares inbuilt on set with all the capabilities of a standard telephone and an ATA. These phones are designed like a normal telephone with handset, cradle and buttons but rather they are equipped with RJ-45 Ethernet connectors instead of standard telephone RJ-11 connectors, with which the IP phone connects directly with a router or switch to place an IP call over the Internet. Wi-Fi IP phones can be implemented to let users make an IP call from any Wi-Fi hot spots.

- Computer-to-Computer

This approach to IP calls are the most cheapest, simplest and common these days. With an internet connection as fast as cable or DSL modem, softphones,

microphone, speaker plus a soundcard installed on both end computers, a user can communicate with another user by IP calls over the Internet. Some companies provide softphones free of cost or it may cost a very low fee to purchase a softphone. This type of IP call is free for most of the distance calls but some calls can be charged a low usage fee for using PSTN services.

- Mobile-to-Mobile

Making a Mobile-to-Mobile IP call is similar to Computer-to-Computer instead a software or application to initiate voice communication is installed on mobile phones or smartphones these days supporting Wi-Fi, 3g/4G cellular networks. Different companies and vendors are available nowadays and provide applications to make IP calls possible, among them some are free or may cost money to download it. It also includes a usage fee for PSTN services, else it is usually free to make any distant IP calls.

2.2 Pros and Cons of VoIP

With its increased popularity for decades, VoIP has not only managed to be a beneficial technology but also a technology which has to be taken care of seriously to be secure and yield a desired output from it.[2] The following sections shortly explain the pros and cons of it.

The development of VoIP has managed to provide advantages so that it can outnumber the use of PSTN. Let us explore some of the many benefits of it.

- Saves money

With internet connection and broadband internet service available, one can save upto 40% on local calls and 90% on international calls. These savings occur because:

- VoIP uses an internet network to make voice communication and only the internet bill is the cost for numerous long IP calls that is it is not necessary to pay for telephone bill per minute.

- No extra cost for voice infrastructure that is only one network exists for voice and data communication over the Internet
- Paying low taxes to government
- Calling all over the world is free and usually much more cheaper than standard telephone
- Conference call

Calling a number of people at a time is called a conference call. VoIP enables us to call more than two persons making it possible by setting up a conference for the whole team. As all the simultaneous calls operate with no extra costs, it eliminates the need to call each and every person individually and thus saving a lot of time. This happens as data packets are compressed during transmission causing an access line to handle more voice packets or calls at a time.

- Affordable hardware for PC

One needs to buy a sound card, speakers and a microphone besides a computer and internet connection which are much affordable compared to buying standard telephone, cables and manpower to install the telephone service. This benefit utilizes softphones for PCs specifically designed to transmit audio, data and video traffic like Viber, Skype, Facebook Messenger etc.

- Undeniable features of IP phones

Many additional functions and features are provided by IP phones which are useful for proper utilization of VoIP. To name some, they are caller ID, contact lists, voice mail, extra-virtual numbers, call management, fax, 3-way calling, local number portability, do not disturb, speed dialing, SMS, anonymous call block etc. This services enhance the productivity and efficiency of an IP call far more than calls made by a traditional telephone even with an ATA.

- Not only voice

VoIP supports the transmission of other media types like images, videos and text along with voice communication. For instance, one can simultaneously send images and video while talking.

- Increase in bandwidth efficiency

Efficient use of available bandwidth is done by VoIP as it manages to utilize the silence between voice conversation so that bandwidth is not wasted, that is bandwidth is not provided to a user while silent but is rather provided to other consumers or to transmit data.

- Network layout flexibility

Flexibility in VoIP is gained as the network is not mandatory to have a particular topology. This enables an organisation to deploy technologies such as Asynchronous Transfer Mode (ATM), Synchronous Optical Network (SONET), Ethernet, Wi-Fi etc.

- Portability

The following reasons make VoIP portable:

- One can still access VoIP facilities from home using an organisation's intranet or extranet.
- It is not location or distance dependent that is one can make or receive calls wherever the broadband internet access is acquired.
- Elimination of network complexity of PSTN

The simplicity of a VoIP network yields an integrated and flexible network to support other communication. The standardization of a VoIP system requires less management of equipment and thus is more fault tolerant.

VoIP has managed to become popular very fast but it is necessary to understand its drawbacks too.[3] Here, we list some of those:

- Service outage during power shortage

Whenever a blackout occurs, the traditional telephone service is not interrupted because it is powered by the current supplied by the phone line which is rather not possible for VoIP but the whole network needs to be supported with Uninterruptible Power Supply (UPS) or a generator.

- Emergency calls

Without involvement of physical addresses and since essentially VoIP is a data packet transfer between two IP addresses over Internet, it is hard to trace any originating call which makes emergency service useless for VoIP users.

- Reliability

These points affect the reliability of VoIP:

- Broadband internet connection's reliability and quality
- Limitations of end VoIP devices
- Highly congested network
- VoIP voice quality

VoIP Quality of Service (QoS) depends upon:

- Broadband connection
- Hardware
- Service provided by provider
- Destination of call
- Security issues

Since VoIP is all based on Internet, it has also drawbacks like other Internet technologies. Identity theft and service theft, viruses and malware, Denial of Service (DoS), spamming, call tampering and phishing attacks are some of the security issues which VoIP faces.

3 UNDERSTANDING THE DEPLOYMENT OF VOICE NETWORK

IP telephony is an internet communication technology to implement voice support over IP networks or computer networks such as Internet. It follows the laws of packet-switched network rather working on the principles of circuit-switched network as a traditional telephony system does. Implementation of telephony with packet-switched networking can be referred to as packet telephony. A packet-switched network is a communications network that functions on the principles of packet switching, as opposed to circuit switching that standard telephone systems use. Packet switching is a communications technology that combines all the transmitting data forming them into suitably sized blocks of data called network packets which are then transmitted over digital networks via a medium that may be shared by many simultaneous sessions of communication. Network packets are made up of a header and payload where the header possesses information that they need to reach their destination and the payload makes them usable by applications.[4]

LAN is a locally manageable computer network which interconnects computers and other network devices with a server within a certain geographical area such as school buildings, office buildings or even Small Office Home Office(SOHO).

This chapter focuses and guides a reader on the purpose, planning and preparation of the implementation of IP telephony for LAN. It also further explains in detail the importance of implementation of IP telephony, how should the implementation be planned and how should the implementation be prepared for IP telephony. In this chapter, the implementation of IP telephony strictly signifies the voice communication through IP networks over Internet.

3.1 Purpose for Voice implementation in LAN

Voice integration on the data network is considered as an advanced technology for LAN. This technology, as time progresses, is trending toward becoming a standard deployment technology for LAN of any business or company. Cost savings for a company which wants to make profit cutting the huge expenses on phone bills, and increased efficiency and productivity of the network from voice services are the overlying purposes for using VoIP in a LAN. A company's LAN with voice integration can be highly benefited using a single converged network for voice communication. The following points further explain about the purpose of VoIP:

3.1.1 Cost savings

IP telephony not only saves the cost to deploy the POTS but also saves a lot with its different advantages, features and services.[5] If we consider national and international calls made by many employees to customers for just a minute or more without the deployment of VoIP on the LAN, a company can definitely save a lot of expenditure on telephone bill per annum with the installation of VoIP. Since, the expenditure on IP calls is almost nothing or negligible, the total expenditure a company has to allocate for voice communication is only the broadband Internet connection fees. However, it might be needed to make sure the broadband Internet connection is faster than what it was before implementing voice which increases the cost of Internet connection. This increase in cost is not much as compared relatively to the expenditure on telephone calls. The use of bandwidth for the broadband Internet should be determined on the basis of a company's requirements and the number of the users using the service simultaneously over the LAN.

The best way VoIP can help to improve a company's cost-effectiveness is in terms of the amount of savings accounted on the basis of charge for a call per minute. The cost of calling throughout a working day racks up for an employee of a company as the call will invariably end up being charged on minute-to-minute basis if the traditional telephone system is used, but, on the other hand,

VoIP service provides the same voice facility to a company with free calls among two VoIP users. This basically means that if customers or business partners also have a VoIP connection then they can stay on the line as long as they need to be and of course without paying for the voice privilege. It helps an organisation to manage voice communication between their employees with a proper working VoIP service which helps the institution to be more efficient and productive due to the healthy collaboration and synchronization inside the institution. This situation is more beneficial if a company or an institution has one or more business partners, customers or branches spread within a country or around the globe whom are needed to be contacted regularly. Also, it is particularly useful if some employees need to work from home and need to be contacted regularly or that employee needs to contact other customers on regular basis. The following list explains more about cost saving:

- Telephony network transmission for lower cost

As per traditional telephony, combining a 64-kbps channel into high-speed links for transportation across the network needs a substantial amount of costly equipment. Instead in packet telephony, voice traffic is multiplexed alongside data traffic, which means that, voice traffic and data traffic are transmitted together with proper utilization of bandwidth since both coexist on the same network infrastructure. This feature provides substantial savings on transmission, equipment and operational cost.

- Low expense on consolidation of voice and data network

Data network becomes the major traffic carrier if it functions as a separate network from the voice network. The creation of the underlying voice network to use packet-switched architecture within a single integrated communications network results a common switching and transmission system. This consolidation is the root of significant cost-savings on the network.

- Revenue increase from new services

New integrated services such as broadcast-quality audio, unified messaging, real-time voice and data collaboration etc. are enabled by packet telephony which increase employee productivity, efficiency and profit margins above the basic voice services. These services also help companies and service providers to be different among others and to improve their position in the market.

- Flexible pricing structure

Transformation of services and pricing structure for companies and service providers can be achieved with packet-switched networks. Network usage is also necessary to be measured in minutes or distance because with packet-switched network, network bandwidth can be allocated dynamically providing companies and service providers the flexibility to alter the volume and breadth of communication service they are serving to meet the required needs of each customers in such a way that can be beneficial for them.

3.1.2 Increased efficiency and productivity

IP telephony can handle calls and manage calls making it more efficient, that is, without any hassles of hard work needed or complicated on-premise technology, it can initiate a whole range of routing and answering call functions. This complicated on-premise technology is maintained and hosted remotely by a VoIP service provider. The call forwarding facility provided by VoIP can be used to forward call to any appropriate compatible device so that the calls can be routed and handle in such a fashion that they reach the right person which makes the LAN of a company to answer more calls and grab a business opportunity with future potential clients which might result in huge business profit for the company. The integration of networks makes VoIP easy to integrate with existing set-ups and provide compatibility to make calls to non-VoIP users and vice versa. This enables a LAN to call any number and receive calls from them which make a company accessible from anywhere around the globe.

Remote working can be very much productive for a company and it can make all the employees efficient as well.[6] With remote work, members of staff can operate effectively like they are in their office when they are out of the office either wirelessly via a mobile device or from home. Since remote working allows a business to be much more flexible, VoIP in LAN fits well for remote working and it is a boon for boosting efficiency. It also helps an employee to cope up with the changes occurred and keep in touch with the work whenever he or she can and wherever he or she happens to be. It even makes sure that the business continuity is not halted whenever there is a break or disaster allowing the company to maintain the stability and may be increase the overall productivity and efficiency, which on the contrary the older telephone system struggles to offer.

Most companies have realized the lucrative nature of IP telephony in LAN and have jumped onto the VoIP bandwagon for the better future of their businesses. [5] Those companies are successfully deploying VoIP and have enjoyed a successful implementation of voice communication in their LAN. Each of these companies has benefited a lot from VoIP deployment as it has brought huge cost savings and has dramatically increased user productivity.

The following list further captures the inspiration behind the deployment of voice in LAN.

- Use of bandwidth and equipment efficiently

Unlike traditional telephony, IP telephony does not use circuit-switched network, instead packet-switched network is used. For each voice call, bandwidth of 64-kbps channel is used by traditional telephony. Bandwidth is shared among multiple logical connections by packet telephony and much traffic volumes are offloaded from existing voice switches.

- Ability to use new communication devices

Packet telephony enables companies and service providers to make use of devices, such as computers, wireless devices, Personal Digital Assistants

(PDA), household appliances, cable set-top boxes etc., that are mostly inaccessible to the Time Division Multiplexing (TDM) infrastructure of today. Thus, packet technology with intelligent access to these devices enables companies and service providers to increase communications service by maximizing the volume of communications they deliver, the breadth of facilities they offer, and the amount of subscribers they serve. It also enables companies to market new and advanced technologies and devices such as videophones, multimedia terminals and advanced IP phone.

3.2 Planning for voice support in LAN

Now as the purpose of the implementation of voice in LAN is reviewed and understood critically, voice support in LAN should also be planned in a proper manner. From the perspective of implementation, all the voice devices and applications software used by end users are connected to the access layer of the LAN to initiate voice communication that is IP phones are plugged into the nearest LAN port or switch port and softphones are installed into the computers which will connect to a switch port.

Making doubts clearer, here, we do not consider planning of either hosted VoIP (off-site VoIP) or self-hosted VoIP (on-site VoIP) systems and all the technical requirements for their deployment. Hosted VoIP system, also called Hosted PBX, is a VoIP system hosted by VoIP service providers to a company with all the hardwares and Private Branch Exchange (PBX) hosted and managed at an off-site location. Unless a company is a VoIP service provider, it is suitable for the company because from all the necessary technical configuration to set-up and installation of hardwares and PBX is done by VoIP service providers with lower cost contrast to self-hosted VoIP where all the hosting, implementing, configuring and managing needs to be done on-site by self-hosting company. Here, we are planning the implementation of VoIP in LAN only and will not focus on the complication for making hosted VoIP LAN or self-hosted LAN work. We also do not discuss about softphones for voice communications.

Data networks with voice-enabled services running both simultaneously in the LAN are becoming a standard for company networks over traditional telephony networks.[7] There are many reasons why this change has occurred which we have previously discussed, among which cost-savings is the major reason for voice deployment in the LAN. From a technology perspective, since voice in the LAN runs on the top of TCP/IP, voice deployments in the LAN are referred to as VoIP deployments. VoIP deployments also require an initial start-up cost, so Return Of Investment (ROI) is a very important aspect associated with VoIP. However, over the life of the VoIP installation, the initial start-up cost will be returned from cost-savings.

From the planning perspective of VoIP, voice and data traffic must co-exist because telephony services associated with VoIP runs over LAN. That is why, we must deploy mechanisms in place to differentiate the voice and data traffic from each other and to prioritize the processing of delay sensitive voice traffic. Without these deployments and careful planning set, the situation of poor audio quality, jitter, echo and audio drop might be experienced by the voice services in LAN.

The problem of differentiating voice and data traffic is solved by the Quality of Service (QoS) policies used by switches to mark and qualify traffic. Further, to ensure the voice traffic is transmitted through the network safely, with special care and with minimum delay by keeping the voice traffic separate from the other data, a specific Virtual Local Area Network (VLAN) is utilized by switches. The best practice for the deployment of voice in LAN is it should include QoS, a separate VLAN for voice traffic and Power over Ethernet (PoE) for IP phones.

While planning for the implementation of VoIP in a network, VoIP architecture needs to be reviewed entirely. Cisco defines Unified Communications as its unique architecture and collection of VoIP products and services related to voice. Unified Communications include devices such as Cisco IP phone, video-conference station, call agent, multi-point control unit, application server and router/gateway. Other components such as software for voice, softphones and Interactive Voice Response (IVR) systems might also be included in the Unified

Communications deployment for extra additional services in LAN to achieve communication goals. The deployment of Unified Communications is outside the scope of this thesis.

Voice traffic is far more sensitive than compared to the data traffic. This sensitivity makes the voice traffic to have extremely strict QoS requirements and renders the voice traffic to be especially handled with care, priority and with minimum delay.[7] As a network is converged with voice and data traffic, stringent management of voice traffic is needed within the converged network because with the proper dedicated management of voice traffic, the demand of bandwidth by the voice traffic is generally smoothly generated and minimal impact on other traffic is caused. The dedicated management of voice traffic excludes every obstacles that the traffic faces when traveling from source to the destination IP address over IP network.

The traffic requirements that should be considered when planning the deployment of VoIP are the following:

- The size of voice packets are small and typically they must be between 60 bytes and 120 bytes.
- The voice quality can be poor when there is drop or delay which VoIP cannot tolerate.
- Since voice traffic communication is real-time opposed to data traffic, the Transmission Control Protocol (TCP) retransmit ability is not worth for voice traffic, therefore, it is better to use User Datagram Protocol (UDP) for voice data transmission.
- For voice traffic, the delay should not be more than 150 ms one way for best voice quality. This is managed only by enabling of QoS for voice packet.
- Not a single instance of packet loss is ideal for voice communication, but 1 percent of voice packet loss is acceptable.

Voice traffic as compared to data traffic differentiates itself and demonstrates delay and drops sensitive characteristics whereas data application traffic is tolerant to some average delay and high drop rates in packets transmission. Additionally, other than being sensitive to frame drops and latency, voice traffic generally requires low bandwidth whereas user application traffic requires generally variable amount of bandwidth. Instead of being bandwidth-intensive, voice traffic requires low and steady bandwidth.

Network Infrastructure analysis, also termed commonly as IP Telephony Readiness Assessment checks whether network infrastructure is ready to carry the converged traffic and is the main purpose of IP Telephony Readiness Assessment. Basic LAN switching, IP routing with power and environmental analysis etc. are covered in IP Telephony Readiness Assessment. It is vital to identify the gaps in the network infrastructure and make necessary changes and recommendations before a network engineer moves forward with the IP telephony deployment.

During the planning phase of the IP telephony deployment, the analysis of the above network infrastructure components and services is required. It aids to recognize the gaps in the current infrastructure and clarify those gaps to support the extra voice traffic over existing data traffic. After the gaps are identified, making the appropriate changes in the network is required, such as implementing QoS in LAN, upgrading the switches for QoS, and in-line power support. This analysis clarifies the technologies and best practices used to design an optimized infrastructure carrying both real-time, delay-sensitive and drop-sensitive voice and non real-time, delay-tolerant data traffic over the same IP telephony infrastructure.

This network infrastructure analysis is divided into five subsections:

- LAN infrastructure

In a small LAN, access, distribution and core layer switches are not necessarily deployed in a multilayer fashion due to factors such as lack of space, lack of

necessity to upgrade and implementation cost, and are generally collapsed in a single switch or switches depending upon the size of the network.

- QoS in LAN infrastructure

QoS for any network with voice and data traffic must be configured to avoid any voice related issues in a converged network. It renders the devices to prioritize and preferentially treat voice traffic than other traffic which is very useful for a network to forward the voice traffic during periods of network congestion and link failures.

- Inline power for IP phones

PoE-enabled Cisco switches provide inline power to IP phones which are capable of accepting power through Ethernet cables. These Ethernet cables are used to transmit the application data throughout the network and reduces the mess of using external power source and cable for each IP phone.

- Dynamic Host Configuration Protocol (DHCP)

Network services are very important and are critical for any IP telephony deployment. In an IP telephony network, DHCP helps each IP phones to acquire their own distinct IP addresses dynamically instead of providing IP addresses manually to many IP phones installed.

- Power infrastructure

Since, IP phones receive power from attached switches, switches of an IP telephony network need to be provided with power redundantly to make sure the high availability of IP phones. Backup power such as UPS and power generator should also be deployed to provide power to switches during the times of power outages.

3.3 Preparing a LAN infrastructure for VoIP

Close cooperation between a network engineer and voice specialist is required for the proper implementation of voice in LAN. Thorough planning is needed for the voice integration into an existing data network to integrate the voice services seamlessly.[7] During the implementation stage, configuration of access switches for VoIP support is needed. This topic illustrates the steps involved while configuring the LAN switch to support VoIP.

Devices commonly found in IP telephony LAN are reviewed in the following list:

- IP phone

It is the active end device which supports incoming and outgoing calls made by a user in IP telephony with inbuilt hardware and capability to encode and compress voice to IP and vice versa. It also provides Internet access and user directory lookup services. It is powered by either external power supply or through network connections.

- PoE enabled switches

PoE-enabled switches are switches that are similar to traditional switches but are manufactured and are enabled to provide inline power from the LAN ports through Ethernet cables to IP phones connected to those ports. These switches also perform some QoS mechanisms for the voice traffic generated from IP phones like classifying voice packets for preferential and prioritized treatment of voice routing throughout the network.

- Call processing manager

For small networks, Call-processing solutions are managed with the router-based approach offered by Cisco which is called Cisco Unified Communications Manager Express (CUCME) in short Communications Manager Express (CME). Since all the call-processing load is carried by Cisco Integrated Service Routers (ISRs), it is a cost-effective solution in contrast to the centralized Cisco Unified

Communications Manager (CUCM) while implementing IP telephony network over small network.

- Voice gateway

Generally understood as voice-enabled router or switch, voice gateway as the name suggests in IP telephony contributes as a translator between VoIP network and non-VoIP network, such as PSTN, and provides physical access for IP phones or switches. It also facilitates voice-to-IP coding and compression, IP voice packet routing, backup call processing and other voice services. CUCME is supported by voice gateways and take over call processing functions whenever a primary call-processing manager fails.

Configuration of Cisco switches are divided and applied by following three configuration options for the deployment of VoIP. They are:

3.3.1 Voice VLANs

A unique feature called Voice VLANs, alternatively named auxiliary VLAN, is offered by Cisco switches. Overlaying a voice topology onto a data network topology seamlessly is enabled by the voice VLANs. Even though the voice and data infrastructure are the same physically, the voice VLANs functions for the logical topology between two networks. Voice VLANs leads to the separation of voice traffic from IP phones or voice enabled devices logically from the physical topology same as the data traffic and treats the voice traffic seamlessly throughout the network. The Cisco voice VLAN feature provides advantages of convergence of physical infrastructure while logically maintaining voice and data traffic over separate logical topology. Voice VLANs are considered essential for any LAN and are strongly recommended during IP telephony deployment though they are optional.

As an IP phone is inserted to a LAN port of a Cisco switch, without any intervention of end user to the IP phone, the phone is placed into and assigned its own VLAN with the help of voice VLAN. Assignment of IP phones to different IP subnets is easy using the DHCP feature and even if a phone is transferred to

another location, the assigned VLAN can be maintained seamlessly. Which means a user can simply plug an IP phone into the switch and the necessary VLAN information will be provided by the switch via Cisco Discovery Protocol (CDP).

The advantages of network segmentation and control, and preserved maintenance of the existing network IP topology for data end stations are gained by placing the phones into their own voice VLANs and IP subnets. The placement of IP phones into their own IP subnets and VLAN not only makes easier to identify and troubleshoot network problems but also to create and regulate QoS and security policies over the IP telephony network.

Multiple VLANs can be supported by voice VLAN. An IP phone can act as an end-user device but also can support attachments of the end-user workstation. The latter deployment lets a Cisco IP phone that is connected to a Cisco switch to switch application data traffic from the end-user workstation through the phone on the native VLAN by default, that is, on Port VLAN ID (PVID) to the Cisco switch but the phone switches the data traffic through the phone on voice VLAN, that is, on Voice VLAN ID (VVID) if a voice VLAN is configured.

3.3.2 QoS for voice traffic from IP phones

As we know that voice traffic requires special treatment with steady and constant bandwidth to travel through the network without delay, jitter and packet loss issues, the implementation of QoS is the first requirement for voice deployment in the network to make that happen because QoS prioritizes voice packets as they traverses through the network.

Within a QoS implementation, voice frames at defined boundaries can be accepted or altered depending upon the existing QoS values of the frames. Configurations at these trust boundaries are enforced by Cisco switches for voice traffic as it traverses into the network with QoS classification and marking. The trust boundary is ideally implemented at the very first switch that any voice devices or IP phones send voice traffic to. We can say that if a switch trusts an

entering frame then it accepts the Differentiated Services Code Point (DSCP) and Class of Service (CoS) value of that frame. As the traffic enters these trust boundaries, the traffic is optionally allowed to retain its original QoS qualities or marking or it is optionally ascribed with new additional marking or qualities by a switch due to the policies or services regulated at the boundaries of that switch. Either acceptance or modification of DSCP and CoS marking of entering frames is fulfilled with configuration of QoS on the switch which receives a frame.

Establishment of a strong border at the switch for the entering traffic is done by trust boundaries. The traffic entering a switch is either handled and prioritized accordingly with markings received or trusted as it is received originally when it entered the trust boundary. At the device with implemented QoS trust boundary, QoS values of frames are only trusted if they guarantee to accurately match the exact traffic type and preside over the processing of traffic as it needs to pass through the network. In the case of unreliability of traffic, the traffic is modified to make it easy to travel through the trust boundary with QoS values appropriate as the existing QoS values at the point where it enters.

3.3.3 PoE

IP phones need a source of power to function. Mobile IP phones or Wi-Fi enabled IP phones have battery as a source of power. Most of the desktop or wall-mount IP phones usually receive power with AC adapter connection. At a small LAN office, though AC/DC power sockets are readily available, using the power sockets for each and every IP phones might result in the waste of physical space and resources.

Power over Ethernet (PoE) on the other hand offers a logical solution to the hassles of connecting every IP phones with AC adapter by receiving power and also connectivity from the Ethernet cable connected with PoE supporting switch and with PoE enabled port which the IP phone connects. This solution is often seen as a best solution because it diminishes the cost and complexity of IP telephony installation.

If a situation ever pops up that the switch cannot provide power to the IP phone, as a solution an intermediate device can be installed between the switch and the IP phone. This device with its own power outlet is called an intermediate device because it joins the switch with one cable and the IP phone with another cable. A cable is connected from IP phone to this device and connected to the switch to the port which the IP phone is supposed to connect. This device is called power injector and provides power to the IP phone along with traffic transmission to and from the switch.

A major benefit of PoE is that an electrician is not needed because anyone knowledgeable of Category 5e cable with its requirements of maximum 328 feet or 100 meters can deploy the cabling required to power the PoE enabled devices. Other vendor IP phones than Cisco can connect to Cisco switches and be recognized and powered when attached to the network because the switches are implemented with Link Layer Discovery Protocol-Media Endpoint Discovery (LLDP).

PoE is implemented with two common methods: Cisco inline power and the IEEE 802.3af standard. The IEEE 802.3af standard differs from the Cisco inline power because it uses power detection mechanism for the detection of necessity of power for the connected device. The IEEE has enhanced the 802.3af standard to new standard 802.3at with extra powering capability for a connecting device. Even though Cisco IP phones utilize less than 15W of power easily powered by a 802.3af switch, Cisco E-series switches with enhanced PoE feature can provide up to 20W of power to any connected device.

A device that does not need any power can still be connected to the PoE enabled port which basically means that it does not consume any power and only power requiring devices utilizes the benefit of PoE. The planning of the overall power consumed by all the devices connected to the switch needs to be carried out although automatic detection of power supply amount is carried out by each port. A dedicated maximum power is allocated by every switch for PoE. So, as a device uses certain power, the used power is learned dynamically via CDP and is deducted from the total available power. However it is not always

that the total amount of power that is needed by a PoE device is consumed which eases for the optimization of actual power consumption by all the devices. For instance, IP phones can only use 6 to 7 Watts which is less than what it uses when fully utilized.

Some switches enabled with the IEEE 802.3af protocol can provide power with 15W on all ports but other switches do not. In addition, adequate amount of power is not always available on all ports which makes, for instance, IP phone to be powerless if the available power is less than 15W.

4 PREPARING THE DEVICES USING A SIMULATION TOOL

Building a working network with all the network devices up and working without having the real physical equipment irrespective of the network's size and complexity is achieved by the use of a simulation tool or software. We use Cisco Packet Tracer as the simulating tool to simulate our voice network topology.

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. Here in this thesis, we use the software to imitate the physical and logical voice network topology by using Command Line Interface (CLI) for router and switch. There is also a possibility using the software to configure the devices with Graphical User Interface (GUI). The imitation is achieved by making a virtual physical topology and configuring the devices using CLI for each devices. The software helps the user to implement a virtual environment of the desired voice network topology to make sure that the network is functioning well logically and the devices are connected well physically.

At first, it is always recommended to use a virtual simulating program to implement the desired network topology virtually and check all the necessary cablings, ports connecting the devices and configurations required and entered to make the network function properly. Secondly, as the whole network is checked thoroughly with the necessary devices, cablings, ports, and each configurations entered to make sure that the network is without any errors and will function properly in real environment, only one can enter into the real environment, buy the devices and deploy the virtually simulated network topology with real devices. This way of approach is safe to deploy a voice network topology because it will let us first to create a virtual network and as

everything is functioning well as expected and desired, then the real-environment deployment of the network can be started without any hesitation.

For instance, here we will make a simulation of deployment of voice network with packet tracer. This simulation will be of a voice network with router, switch and end devices such as IP phones, PCs and Laptops. This simulation of an IP telephony topology serves as physical and logical guidelines to deploy a basic voice network for IP telephony which explains the commands to configure the router and switch, and end device configurations.

Now, let us have a look at the voice network topology that we use throughout this topic.

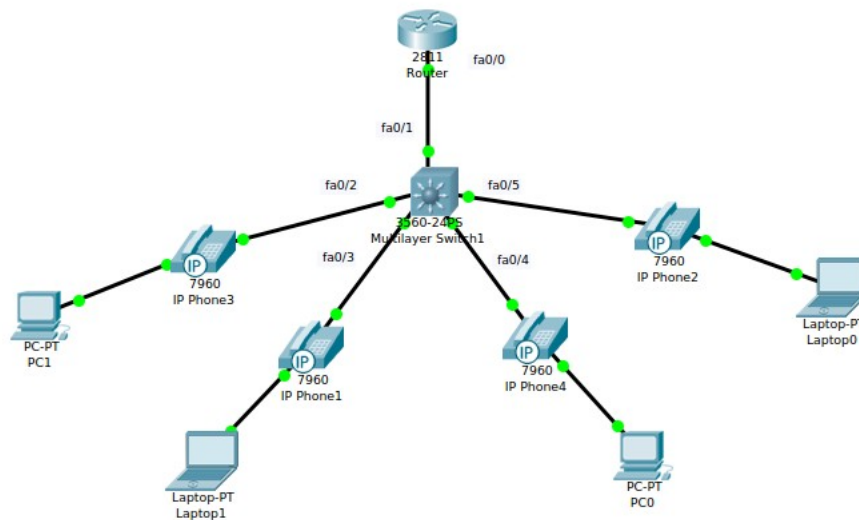


Figure 1. IP telephony topology of small network with IP phones and PCs

Here, we have a voice network topology with a router, a switch, four IP phones, two PCs and two laptops. The overall cabling for this topology is built with a copper straight-through Ethernet cable (RJ45). The network topology design leads to deployment of IP phones required for a user to use VoIP services within an IP telephony network.

4.1 Switch configuration

Now let us prepare our IP telephony network. First we make sure that every IP phones receive power. For this we have taken a 3560 switch with the PoE feature enabled in all the ports. This PoE feature enables the IP phones to receive power from the ports connected to IP phones respectively.

```
Switch>en
Switch#show power inline
Available:370.0(w)  Used:25.2(w)  Remaining:344.8(w)
```

Interface	Admin	Oper	Power (Watts)	Device	Class	Max
Fa0/1	auto	off	0.0	n/a	n/a	15.4
Fa0/2	auto	on	6.3	IP Phone	2	15.4
Fa0/3	auto	on	6.3	IP Phone	2	15.4
Fa0/4	auto	on	6.3	IP Phone	2	15.4
Fa0/5	auto	on	6.3	IP Phone	2	15.4
Fa0/6	auto	off	0.0	n/a	n/a	15.4
Fa0/7	auto	off	0.0	n/a	n/a	15.4
Fa0/8	auto	off	0.0	n/a	n/a	15.4
Fa0/9	auto	off	0.0	n/a	n/a	15.4
Fa0/10	auto	off	0.0	n/a	n/a	15.4
Fa0/11	auto	off	0.0	n/a	n/a	15.4
Fa0/12	auto	off	0.0	n/a	n/a	15.4
Fa0/13	auto	off	0.0	n/a	n/a	15.4
Fa0/14	auto	off	0.0	n/a	n/a	15.4
Fa0/15	auto	off	0.0	n/a	n/a	15.4
Fa0/16	auto	off	0.0	n/a	n/a	15.4
Fa0/17	auto	off	0.0	n/a	n/a	15.4
Fa0/18	auto	off	0.0	n/a	n/a	15.4
Fa0/19	auto	off	0.0	n/a	n/a	15.4
Fa0/20	auto	off	0.0	n/a	n/a	15.4
Fa0/21	auto	off	0.0	n/a	n/a	15.4
Fa0/22	auto	off	0.0	n/a	n/a	15.4
Fa0/23	auto	off	0.0	n/a	n/a	15.4
Fa0/24	auto	off	0.0	n/a	n/a	15.4

```
Switch#
```

Figure 2. IP phones powered by PoE

The above figure is the result of the command **show power inline**. The command shows the statistics of the power used by the all the powered devices and the total capacity and remaining power of the power supply. It also illustrates the configuration of how the power is provided by the switch to IP phone. The interface column shows all the interfaces of the switch. The admin column shows the power management modes of ports. The operation column shows the operational status of a port. The power_(watts) column shows the supplied power by the power supply in watts. The device and class columns

show the powered device's name and class. The last max column shows the maximum power supplied by the power supply to any powered devices through a single port.

Since PoE is configured at port level, the command **power inline auto** is all that is needed to make PoE-enabled ports and detection of power requirements automatically.

As the network topology is ready for the deployment, we will now configure the switch to make it capable to support voice all over the network. The following steps will highlight the commands to make the switch ready to switch voice and data traffic over the network by creating voice and data VLANs respectively.

As we have connected all the devices with ethernet cables, the IP phones receive power from the switch with the help of PoE and the PCs and laptops are powered by default. We connect the switch, router, IP phones, PCs, laptops as shown in the above network topology.

To configure the switch, we click on the switch icon and as the configuration window for the switch appears with physical, config and CLI tabs. The physical tab shows the physical appearance of the switch and helps to customize the icon for physical and logical view. The config tab is for the purpose of configuring the switch using GUI which lets a user to configure the switch in a graphical easy way with few clicks and some typing. The tab CLI provides a user an interface where one can enter different commands that will configure the switch. Here we focus upon the CLI tab.

The CLI tab should be clicked to start the configuration. As the "press RETURN to get started" line is seen, enter should be pressed. By default, the hostname of the switch is Switch and the command prompt is in user-exec mode which is Switch>. To change the command prompt to privileged-exec mode, the command **enable** should be typed in user-exec mode and enter should be pressed to make the prompt as Switch#. Now to configure the switch, the command **configure terminal** should be typed in the privileged-exec mode to

enter into the global configuration mode which will change the prompt to Switch(config)#.

As there are data and voice traffic traversing throughout the network, the creation of different VLANs for each traffic is the solution for the problem to manage the traffic. Creating different VLANs manufactures the separate path for the traffic and leads each traffic to and from their designated port and devices throughout the IP telephony network. To achieve this, the following commands in global configuration prompt should be typed:

The words and numbers in bold signify the command. // depicts the starting of comments for the command.

Switch(config)#**vlan 10** // This command creates a vlan with number 10.

Switch(config-vlan)#**name data** // This command names the above vlan 10 as data.

Switch(config)#**vlan 20** // This command creates a vlan with number 20.

Switch(config-vlan)#**name voice** // This command names the above vlan 20 as voice.

Now as we have created VLANs for data and voice, we need to make sure they can travel throughout the network and connect properly with the router. Since the switch is connected to the router and different VLANs traffic are to be transported to and from the switch through router, a trunk link is configured on the interface fastEthernet 0/1 of switch which lets the switch and router to communicate more efficiently with the tagging and untagging of different VLAN traffic. The frames are tagged with dot1q tags and are distinctively communicated with network devices within distinct VLAN.

To configure trunking line on the switch interface connected to the router, the following steps need to be performed:

Switch(config)#**interface fastEthernet 0/1** // This command enters into fastEthernet interface 0/1 of the switch. Using this command changes the prompt to global interface configuration mode.

Switch(config-if)#**switchport mode trunk** // This command enables trunking on the above interface.

Switch(config-if)#**switchport trunk encapsulation dot1q** // This command makes dot1q distinct encapsulation tagging.

In this scenario, voice packets are sent tagged over the voice VLAN with distinct dot1q tagging associated with VLAN id 20 whereas data packets are sent tagged over access data VLAN with distinct dot1q tagging with VLAN id 10 and both tagged data frames and tagged voice frames are read by the switch and put into their respective VLANs so that it can better communicate with the router's subinterfaces. All the ports must be configured as access ports except fastEthernet fa0/1 and permission to pass through the designated ports for VLAN 10 and VLAN 20 should be given. This logical separation of virtually created network with the deployment of data and voice VLANs makes the transmission of both data and voice traffic possible through a single port to rest of the network.

The following commands should be configured to enter into the range of designated fastEthernet interfaces and to permit data and voice vlan over the interfaces.

Switch(config)#**interface range fastEthernet 0/2-5** // This command allows access to make changes to the fastEthernet interface 0/2 upto 0/5. Note the term **range** between the command should not be missed.

Switch(config-if)#**switchport mode access** // This command changes the mode of the switchports to access.

Switch(config-if)#**switchport access vlan 10** // This command permits vlan 10 with data traffic through the interfaces.

Switch(config-if)#**switchport voice vlan 20** // This command permits vlan 20 with voice traffic through the interfaces.

The switch will be the QoS trust boundary for the network. Referring to the Cisco configuration guide is recommended for any specific model. To enable QoS, the following global configuration command should be entered:

Switch(config)#**mls qos** // This command enables QoS globally.

Voice data which come on the switchports from the phone will either have the CoS values trusted or altered based on the information received on the ports. A Cisco IP phone has a value of 5 by default. CoS that is advertised will not be trusted at any port which may have a device other than a Cisco phone. Cisco Discovery Protocol (CDP) information is used to determine the availability of an IP phone when it is attached to a port. Configure fastEthernet interfaces 0/3 upto 0/6 to trust the CoS for recognized IP phones on the network. Commands for the interface fastEthernet 0/2 are shown:

Switch(config)#**interface fastEthernet 0/2** // This command enter into port fastEthernet 0/2.

Switch(config-if)#**mls qos trust cos** // This command configures the above interface to trust the CoS values received from the IP phone.

To configure other interfaces to trust CoS must not be forgotten.

For the verification of the commands entered above, the global configuration prompt should be exited with **exit** or **end** command and the following commands should be typed:

Switch#**show vlan brief** // This verification command shows all VLANs and the respective interfaces related to VLANs.

Switch#**show vlan id number** // This verification command shows the information of VLAN with specific id.

Switch#**show vlan name** *name* // This verification command shows the information of VLAN with specific name.

Switch#**show ip interface brief** // This verification command shows the information about all interfaces, such as IP-address assigned, status and protocol etc.

Switch#**show interfaces trunk** // This verification command shows the information of all the trunk links available.

Switch#**show interfaces** *interface-type interface-number* **switchport** // This verification command shows the information of specific switchport.

Switch#**show mls qos interface fastEthernet** *interface-number* // This verification command shows the information of QoS for a specific fastEthernet interface.

Switch#**show running-config** // Shows running configuration of a switch.

Typing **do** in front of the above privileged-exec commands shows the same desired output still being on global configuration prompt.

4.2 Router configuration

To configure the router, the router icon on the IP telephony topology should be clicked. As a new window named with the name of the router appears, the CLI tab must be clicked and enter should be pressed to get started. Here, we follow three steps to configure the router.

4.2.1 Subinterface configuration

Subinterfaces are associated with physical interfaces. Subinterfaces are enabled when the physical interface with which they are associated is enabled, and subinterfaces are disabled when the physical interface is shut down. Subinterfaces can be enabled and shutdown independent of the physical port with which they are associated. However, a subinterface of a physical interface that has been shut down can not be enabled. Subinterfaces are created by

subdividing the physical interface into two or more virtual interfaces which are assigned with unique Layer 3 network addresses such as IP subnets. Subinterfaces were invented as a method of virtually subdividing a physical interface into two or more interfaces so that the IP routing protocols would see the network connection to each remote networking device as a separate physical interface even though the subinterfaces share a common physical interface. The full subinterface number must be unique to the networking device.[8] For example, the first subinterface for fastEthernet interface 0/0 might be named Ethernet 0/0.10 where .10 indicates the subinterface.

From the topology point of view, the switch is connected with the trunk link at fastEthernet 0/0 port of the router. The creation of subinterfaces associated with the physical fastEthernet interface of the router are implemented to ease the communication between VLANs of switch and router properly. Here, the fastEthernet 0/0 physical interface is divided into two subinterfaces indicated with 0/0.10 and 0/0.20. These subinterfaces are associated and subdivided from the single physical interface fastEthernet 0/0 to communicate properly with the VLAN 10 and VLAN 20 of the switch, which means that subinterface 0/0.10 is used for VLAN data and subinterface 0/0.20 is used by VLAN 20 to function properly and carry traffic as needed by the network.

To create subinterfaces for interface fastEthernet 0/0, first we have to enable it. Then we create associated subinterfaces to it. To enable the interface and create subinterfaces for it, the following commands should be configured:

Router>**enable** // This command changes the prompt from user-exec mode into privileged-exec mode.

Router#**configure terminal** // This command changes the prompt from privileged-exec mode into global configuration mode.

Router(config)#**interface fastEthernet 0/0** // This command changes the prompt into global interface configuration mode for interface fastEthernet 0/0.

Router(config-if)#**no shutdown** // Interface fastEthernet 0/0 is enabled from shutdown state with this command.

Router(config-if)#**interface fastEthernet 0/0.10** // This command creates subinterface fastEthernet 0/0.10 associated with fastEthernet 0/0.

Router(config-if)#**encapsulation dot1q 10** // This command enables encapsulation with dot1q tagging for VLAN 10 traffic. The number 10 at the end of command below resembles the VLAN id.

Router(config-if)#**ip address 192.168.10.1 255.255.255.0** // This command configures a IP-address to the subinterface.

Router(config-if)#**interface fastEthernet 0/0.20** // This command creates subinterface fastEthernet 0/0.20 associated with fastEthernet 0/0.

Router(config-if)#**encapsulation dot1q 20** // This command enables encapsulation with dot1q tagging for VLAN 20 traffic.

Router(config-if)#**ip address 192.168.20.1 255.255.255.0** // This command configures IP-address to the subinterface.

Router(config-if)#**exit** // This command exits from the global interface configuration mode.

4.2.2 DHCP pool configuration

DHCP is used by the router to enable dynamic addressing scheme for the end devices such as IP phones, PCs and laptops. For this addressing to happen, a DHCP pool should be created at the DHCP server with proper IP subnetting. Here the router acts as the DHCP server and the end devices act as DHCP clients. The DHCP server assigns and manages the implementation of IP addresses and the DHCP configurations from specified DHCP address pools within the server to DHCP clients.

As a DHCP client sends a broadcast message about the discovery of a DHCP server's location to the DHCP server, the server offers a unicast message to the

client about the DHCP configurations parameters. Then the client sends a formal request as a broadcast message to the server about the previously offered DHCP configuration parameters and, finally, the DHCP server sends a unicast message to the client that it has confirmed the assignment of the DHCP configuration as an acknowledgement. In case, the DHCP configuration parameters within a unicast message that are offered by the DHCP server to DHCP client are invalid, a broadcast message is sent by the DHCP client to the DHCP server to decline the offer. Likewise, when a DHCP client requests the offered configuration parameters, the DHCP server sends a denial broadcast message to DHCP client saying that the offered parameters can not be assigned if errors occur during configuration negotiation or the client responds too slow to the offer given by the DHCP server.

The DHCP server follows the process defined by DHCP which defines the server to locate the IP subnet where the DHCP client resides. As the server locates the IP subnet of the client, it will assign a valid IP address to the client from the DHCP pool created for the subnet.[9] The identification of which DHCP pool to use and assignment of IP address from that pool to a DHCP client by a DHCP server are carried out as follows:

- If the client is not directly connected (the giaddr field of the DHCPDISCOVER broadcast message is non-zero), the DHCP Server matches the DHCPDISCOVER with a DHCP pool that has the subnet that contains the IP address in the giaddr field.
- If the client is directly connected (the giaddr field is zero), the DHCP Server matches the DHCPDISCOVER with DHCP pool(s) that contain the subnet(s) configured on the receiving interface.

In our IP telephony network topology, each IP phone sends a DHCPDISCOVER broadcast message to the router (which is DHCP server) and the router matches the DHCPDISCOVER message with the DHCP pool named voice which has the subnet that contains the IP address in the giaddr field. Likewise, each PC and laptop send a DHCPDISCOVER broadcast message to the router

and the router matches the message with the DHCP pool named data configured with the subnet that contains IP address in the giaddr field.

The DHCP pool is a collection of all IP addresses of a network or subnetwork which are allocated to be assigned to any DHCP clients which are ready to accept the IP addresses. An IP address of a DHCP pool is offered to a DHCP client when negotiation of DHCP configuration parameters between them is successful. The DHCP server assumes that every IP address in a pool is eligible to be allocated and ready to be assigned to the DHCP clients. This is not recommended because as all IP addresses are offered to DHCP clients, the network address and broadcast address of the network might also be offered to the DHCP client which should be totally avoided to be free from IP address mismatches in a network. Moreover, many scenarios will emerge where IP addresses for interfaces are required to be assigned or already assigned manually or statically and assigning these IP addresses to any of the DHCP clients are not free from occurrence of IP address mismatches. We must specify IP addresses which we want to exclude from the DHCP pool.

Cisco IP phones are required to download a specific configuration file, downloaded using Trivial File Transfer Protocol (TFTP), which provides them with specific and vital voice information, such as location of CUCME, availability of a newer firmware and codec to use. Configuring the IP address of the TFTP server manually into each IP phone is required if the IP address of the TFTP server is not provided using DHCP option 150. Here the number 150 provides a list of TFTP server's IP addresses. We can input as many as 8 TFTP server's IP addresses for this DHCP option. DHCP option 66 provides with a single IP address or a hostname of a single TFTP server and option 3 sets the default route.

As per our IP telephony network, the commands required to make DHCP implementation are follows:

Router(config)#**ip dhcp excluded-address 192.168.10.1 192.168.10.5** // This command excludes the IP addresses from 192.168.10.1 up to 192.168.10.5 to be assigned by DHCP server to DHCP clients.

Router(config)#**ip dhcp excluded-address 192.168.20.1 192.168.20.5** // This command excludes the IP addresses from 192.168.20.1 up to 192.168.20.5 to be assigned by DHCP server to DHCP clients.

Router(config)#**ip dhcp pool data** // This command creates DHCP server address pool named data and changes the mode to DHCP pool configuration mode.

Router(dhcp-config)#**network 192.168.10.0 255.255.255.0** // This command specifies a network for DHCP pool addressing. Here IP addresses from 192.168.10.1/24 network are assigned to DHCP clients.

Router(dhcp-config)#**default-router 192.168.10.1** // This command specifies IP address of a default router for DHCP client. Packets are sent to this IP address after a DHCP client has booted.

Router(config)#**ip dhcp pool voice** // This command creates DHCP server address pool named voice and changes the mode to DHCP pool configuration mode.

Router(dhcp-config)#**network 192.168.20.0 255.255.255.0** // This command specifies a network for DHCP pool addressing. Here IP addresses from 192.168.20.1/24 network are assigned to DHCP clients.

Router(dhcp-config)#**default-router 192.168.20.1** // This command specifies IP address of a default router for DHCP client. Packets are sent to this IP address after a DHCP client has booted.

Router(dhcp-config)#**option 150 ip 192.168.20.1** // This command specifies the IP address of a TFTP server. IP phone downloads voice configuration file from this server's address.

As DHCP is configured in the router, PCs and laptops should be checked if they accept IP addresses dynamically and are powered on. In addition, even though IP phones are powered with PoE in this IP telephony topology, it is recommended to check if the phones need power manually.

4.2.3 Cisco Unified Communications Manager Express (CME) configuration

As the configuration file is downloaded by the IP phone, it tries to reach to CME server (here router acts as a CME server). The provisioning of the IP address of the CME server is carried out through the Domain Name System (DNS) or it is provided within the configuration file to the IP phone. Like our IP telephony deployment, many small networks tend to utilize the benefit of assigning CME server as TFTP server to remove the need for DNS resolution and an external TFTP server. As the IP phone reaches and contacts the CME server, its registration occurs and obtains its extension number thus making the phone ready to place and receive calls.[10]

Let us configure CME on the router with the following three steps basic CLI configuration commands:

- Configuring telephony service

Router(config)#**telephony-service** // This command enters into telephony service configuration prompt.

Router(config-telephony)#**max-ephones 4** // This command sets the maximum number of IP phones supported by CME.

Router(config-telephony)#**max-dn 4** // This command sets the maximum number of IP phone directory numbers supported by CME .

Router(config-telephony)#**ip source-address 192.168.20.1 port 2000** // This command sets the source IP address and default port number for the IP phone registration.

Router(config-telephony)#**end** // This command exits to privileged-exec mode.

The maximum numbers of IP phones and extensions are dependent upon platform and IOS version. Port numbers from 2000 up to 9999 can be added to the command above.

- Configuring directory numbers

Router(config)#**ephone-dn 1** // This command configures a directory number 1.

Router(config-ephone-dn)#**number 101** // This command assigns 101 as an extension number for directory number 1.

Router(config-ephone-dn)#**exit** // This command exits from ephone-dn configuration prompt to global configuration prompt.

The following commands configure directory number 2 with phone number of 102.

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

Router(config-ephone-dn)#**number 102**

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

The following commands configure directory number 3 with phone number of 103.

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

Router(config-ephone-dn)#**number 103**

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

The following commands configure directory number 4 with phone number of 104.

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

Router(config-ephone-dn)#**number 104**

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

- Creating phones

Router(config)#**ephone 1** // Ephone configuration mode is entered with this command.

Router(config-ephone)#**mac-address AAAA.AAAA.AAAA** // Mac-address entered is associated with ephone 1 with this command.

Router(config-ephone)#**type 7960** // Type of the phone is configured with this command

Router(config-ephone)#**button 1:1** // This command associates the first button on the phone to the directory number 1. Here the first 1 is the first button on the phone and the second 1 is the directory number 1 created above.

Router(config-ephone)#**exit** // This command exits from ephone configuration mode into global configuration mode.

The ephone 1 with mac-address **AAAA.AAAA.AAAA** is linked with ephone directory number 1 with extension number 101.

With the help of the following commands, the ephone 2 with mac-address **BBBB.BBBB.BBBB** is linked with ephone directory number 2 with extension number 102.

Router(config)#**ephone 2**

Router(config-ephone)#**mac-address BBBB.BBBB.BBBB**

Router(config-ephone)#**type 7960**

Router(config-ephone)#**button 1:2**

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

With the help of the following commands, the ephone 3 with mac-address **CCCC.CCCC.CCCC** is linked with ephone directory number 3 with extension number 103.

```
Router(config)#ephone 3
```

```
Router(config-ephone)#mac-address CCCC.CCCC.CCCC
```

```
Router(config-ephone)#type 7960
```

```
Router(config-ephone)#button 1:3
```

```
Router(config-ephone)#exit
```

With the help of the following commands, the ephone 4 with mac-address **DDDD.DDDD.DDDD** is linked with ephone directory number 4 with extension number 104.

```
Router(config)#ephone 4
```

```
Router(config-ephone)#mac-address DDDD.DDDD.DDDD
```

```
Router(config-ephone)#type 7960
```

```
Router(config-ephone)#button 1:4
```

```
Router(config-ephone)#exit
```

For verification of the configuration of router, the following commands are used:

```
Router#show ip interface brief // This verification command shows the information about all interfaces, such as IP-address assigned, status and protocol etc.
```

```
Router#show ephone // This verification command shows all the information of all the IP phones configured on CME.
```

```
Router#show running-config // This verification command shows running configuration of a router.
```

5 CONCLUSION

VoIP is basically a group of technologies functioning in such a way as to provide voice and multimedia sessions transfer over the Internet and the transmissions occur between two IP addresses over IP networks. It has evolved so much over past decades and has revolutionized the world of telecommunication to bring people closer than it had been thought of before. It has gained a great amount of popularity in the world of information, communication and technology because of its various advantages with very few disadvantages for a company. Implementing a voice network is the best solution for a small company that needs to deploy voice facility to be benefited with voice services.

With all the benefits and advantages provided by IP telephony deployment, its deployment is an asset for a company. It is obvious and easy to understand why companies and enterprises have deployed, have been deploying and continue to deploy VoIP in their LAN. The most vital underlying inspirations and motivations behind implementing IP telephony in LAN are cost-savings, increased networking efficiency, and productivity boost due to voice services. Planning IP telephony implementation is very important for every network. The main foci for the planning of IP telephony deployment are QoS and VLANs for network convergence, PoE for IP phones, DHCP for IP phones and PCs, CME for call managing and processing, and management for redundant power supply. To utilize the best services of IP telephony deployment for a network, the needs and purposes of its deployment should be identified clearly and planned with all the best solutions available. It should also be prepared with proper devices and configurations based upon the requirements of the voice network.

Configuring IP telephony devices for small network with Packet Tracer simulation was helpful in order to check the validity of the configurations applied and authenticity of the IP telephony network to become ready to place and receive calls. The software - that is Packet Tracer, was an excellent choice configuring the devices with the necessary configurations they need. It was

quite easy to work with and very useful as well because working on it was like working with the real devices on real environment. The router was configured with basic DHCP commands to assign IP addresses dynamically, subinterfaces of a physical port to manage the traffic and basic telephony services to place and receive calls from IP phones. On the other hand, the switch was configured with VLANs to converge the voice and data networks and QoS features to give integrity to the traffic which needs the priority and preferential treatment for the smooth transmission of packets.

With all the configurations above applied to the router and switch, we managed to prepare an IP telephony network with 4 IP phones, 2 PCs and 2 laptops with DHCP addressing enabled. The network is well converged and is ready to place and receive calls from the IP phones.

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APPENDICES

Running-configuration of the switch

```
Switch#show running-config
< output omitted >
!
hostname Switch
!
mls qos
!
interface FastEthernet0/1
switchport trunk encapsulation dot1q
switchport mode trunk
!
interface FastEthernet0/2
switchport access vlan 10
switchport mode access
switchport voice vlan 20
!
interface FastEthernet0/3
switchport access vlan 10
switchport mode access
switchport voice vlan 20
!
interface FastEthernet0/4
switchport access vlan 10
switchport mode access
switchport voice vlan 20
!
interface FastEthernet0/5
switchport access vlan 10
switchport mode access
switchport voice vlan 20
!
< output omitted >
```

Running-configuration of the router

Router#show running-config

< output omitted >

!

hostname Router

!

ip dhcp excluded-address 192.168.10.1 192.168.10.5

ip dhcp excluded-address 192.168.20.1 192.168.20.5

!

ip dhcp pool data

network 192.168.10.0 255.255.255.0

default-router 192.168.10.1

ip dhcp pool voice

network 192.168.20.0 255.255.255.0

default-router 192.168.20.1

option 150 ip 192.168.20.1

!

< output omitted >

!

interface FastEthernet0/0

no ip address

duplex auto

speed auto

!

interface FastEthernet0/0.10

encapsulation dot1Q 10

ip address 192.168.10.1 255.255.255.0

!

interface FastEthernet0/0.20

encapsulation dot1Q 20

ip address 192.168.20.1 255.255.255.0

!

< output omitted >

!

telephony-service

max-ephones 4

max-dn 4

ip source-address 192.168.20.1 port 2000

!

```
ephone-dn 1
number 101
!
ephone-dn 2
number 102
!
ephone-dn 3
number 103
!
ephone-dn 4
number 104
!
ephone 1
device-security-mode none
mac-address 0030.F27D.5064
type 7960
button 1:1
!
ephone 2
device-security-mode none
mac-address 00D0.BC6B.7D6B
type 7960
button 1:2
!
ephone 3
device-security-mode none
mac-address 0040.0B07.73B3
type 7960
button 1:3
!
ephone 4
device-security-mode none
mac-address 00D0.FF0E.3102
type 7960
button 1:4
!
< output omitted >
```