An Overview of VOIP W.R.T to Voice Compression Technique
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Abstract— Nowadays VoIP enabled applications like Hangouts, GTalk are being used prolifically. VoIP technology is a significant factor contributing much to the way people use to communicate online. Even though it uses UDP protocol for the reason that it offers less overhead, it can be implemented in a real time environment. GNS-3, a simulator, can be deployed to implement VoIP technology for practical purposes. Along with this CISCO IP Phones needs to be configured and registered on a client and/or server system for its proper implementation. It can be implemented on LAN or WAN by making the client and/or server to act as CME or using wi-fi. Quality of Voice enabled IP Phones is determined by MOS which in turn depends on the bitrate it offers. Codecs like G.729 offers an effective and novel way through which voice can be compressed on the above mentioned platforms. Without any significant data loss, packet loss and with slight variation in jitter voice can be compressed on VoIP on the above mentioned platforms using codecs

Key words: W.R.T, CISCO

I. INTRODUCTION

VoIP stands for Voice over IP (Internet Protocol). It is a technology that delivers voice communications and multimedia sessions over the internet. The ubiquitous upsurge of Internet supported by a range of available bandwidths has made it easy for users to connect to each other easily and facilitates them to communicate over Internet. VoIP allows users to have telephonic conversation over Internet or over Ethernet without the use of ordinary phones. An IP phone registered over an IP address can facilitate VoIP communication. VoIP besides avoiding long distance charges of telephone companies also provides quality in terms of voice clarity on phone calls with minimal packet and/or data loss. VoIP telephone calls are similar to orthodox telephone calls and involve channel establishment, signaling, digitization and packetization of the analog voice signals.

UDP (User datagram Protocol) is deployed for establishing VoIP connection as it offers less overhead and also because it offers multicasting better than the TCP (Transmission Control Protocol) protocol. UDP is also preferred as it facilitates real time delivery of audio and video. Application layer protocols like and RTCP (Real-Time Control Protocol) and RTP (Real-Time Transport Protocol) are also used in VoIP [1]. While on one hand RTP operates at top of UDP on the other hand RTCP is primarily involved in statistics reporting and provides end to end delivery services.

II. DIGITAL AUDIO

It is well known fact that audible frequency range for human ears is 20Hz to 20,000Hz. Also frequencies of human voice range from 300Hz to 35,000Hz. According to the Nyquist Theorem which states that two times the sum of highest frequency of the analog signal must be less than the sampling rate to be able to reconstruct the original signal from the sampled data. Therefore peak frequency at which human voice can be captured turns out to be 2*3,500Hz = 7,000Hz (To avoid any conversion errors, human voice is usually captured at 8000 Hz in practice.) To digitize human voice data at a sampling rate of 8000 Hz using eight-bit numbers for each sample the available bandwidth is 8*8,000= 64000bps (64Kbps).

III. IMPLEMENTATION OF VOIP USING SIMULATOR GNS-3

A. VoIP Connection On LAN:

Steps for establishing VoIP connection on LAN
1) Embed the image of Router on GNS-3 (Graphical Network Simulator-3) simulator.
2) Drag and drop a Router and set its interface as f0/0 and then a Cloud and select its slot as Ethernet and add it.
3) Connect the Router and a Cloud in the workspace area of GNS-3 using Fast Ethernet Link
4) Start the Router and open it's Console page on Superputty.
5) Provide IP address to the Router at f0/0 interface.
6) Create a DHCP (Dynamic Host Configuration Protocol) pool and name it.
7) Define TFTP (Trivial File Transfer Protocol) address using ‘option 150’ command.
8) Create telephony service and add required number of e-phones and directory numbers. Also assign an IP address to it.
9) Assign a telephone number to each of the specified e-phones.
10) Connect both the system using LAN (Local Area Network) wire or if there are more than two system use switch for the same.
11) Set the IP address of Ethernet adapter corresponding to the IP address given to the Router on all the clients and the system to be used as CME (Call Manager Express).
12) On the installed CISCO IP Phones add the same TFTP address mentioned in the ‘option 150’ command on console.
13) IP Phone is now registered on CME. Restart all other CISCO IP phones.
14) Dial the required number from one of the client and start the communication.
B. VoIP Connection On WAN:
For practical purposes a Router with two interfaces (say f0/0 and f0/1) is selected and two clouds are used in WAN (Wide Area Network) connection. For 1st Cloud, its slot is set to VMware Net1, IP phone is registered virtually using VMware and its IP address is set accordingly and on 2nd Cloud, slot is set as Ethernet and Ethernet Adapter’s IP address is set accordingly.

For establishing VoIP connection on WAN all the steps required for VoIP connection on LAN has to be applied. Besides that dial-peer voice connection for voice has to be established where destination pattern is set for each LAN according to the e-phone-number assigned to each of them. Then session target is set as the IP address of the LAN to which the connection has to be established. CUCM (CISCO Unified Communication Manager) can be deployed in case VoIP connection on WAN is to be established for large number of LANs.

![Fig. 2: VoIP connection on WAN using GNS-3](image)

![Fig. 3: Overview of VoIP connection on WAN using GNS](image)

C. Description Of Voip Connection Using GNS-3 Simulator:
VoIP connection implementation for practical purposes requires simulator GNS-3, CISCO IP Communicator, LAN wires and/or switch in case there are more than 2 clients. A CUC (CISCO Unity Communication) is a reliable and user friendly voicemail solution that provides users with flexible message access options. Communication of IP phones over single LAN is established using CME (Call Manager Express) [8]. CUCM is the Call Manager or the backbone of VOIP technology wherein it establishes dial–peer connection (connection between multiple LANs). IP Phones are connected to Power over Ethernet (POE) Switches which extract VLAN (Virtual Local Area Network) information from switches. These switches are connected to DHCP (Dynamic Host Control protocol) Gateways or Routers thereby enabling switches to extract subnet mask, default gateway and TFTP (Trivial File Transfer Protocol) information. The server virtualized with CUC and CUCM is connected to these Gateways. GNS-3 works on complex networks so it will be deployed to simulate ISR (Integrated Services Router) Routers, VOIP gateway and POE Switches. The GUI (Graphical Use Interface) of IP phones is provided by CISCO IP Communicator. In Fig 3. HQ denotes the Headquarter site while BR denotes the Branch site and the triangles denotes the IP Phones connected to the POE switches on different LANs.

On LAN one of the clients is made to act as CME (Call Manager Express) where all the IP Phones are registered and all other clients need to set their IP addresses and TFTP address accordingly as set in the system specified as CME. Owing to the Nyquist equation initially on LAN the bandwidth achieved is 64Kbps.

On WAN, CUCM can be deployed to establish dial peer connection when multiple LANs are to be connected. In WAN, each of the LAN connection needs to use the dial peer VoIP command for voice. They need to set the destination pattern according to the series of number that LAN’s CME has assigned to its clients. The last step is to set the IP address of the session target to which the connection has to be established.

IV. IMPLEMENTATION OF VOICE COMPRESSION AND ITS ROLE
After the VoIP connection has been established, a noteworthy point to focus is the bandwidth at which the communication is being performed. As mentioned earlier according to the Nyquist equation the bandwidth achieved is 64Kbps on LAN environment.

Codecs offer an intriguing technology that can be used when bandwidth comes into picture. An audio codec is a device or computer program capable of coding or decoding a digital data stream of audio. An audio codec implements an algorithm that compresses and decompresses digital audio data. The algorithm aims to represent the highly reliable audio signal with minimum number of bits while retaining the voice quality. This can effectively reduce the bandwidth required for transmission of the stored audio file.

When we talk about IP Phones, codec is the component in an IP phone that digitizes voice and converts it back into an analog stream of speech. The codec is mainly deployed to perform the voice compression and decompression. It incorporates sampling, packetization and analog to digital conversion in itself.

So to compress the digital audio at bandwidth of 64Kbps, a codec G.729 is deployed which selects best 1000 samples out of 8000 with 8-bit number for each sample, the bandwidth reduces to 8*1000Hz = 8Kbps. Thus voice is compressed from 64Kbps to 8Kbps without any significant packet loss and/or data loss but with very slight variation in jitter. Wireshark is also deployed to analyze this bandwidth change. It dissects the UDP packets as RTP [10].

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Mean Opinion Score (MOS) defines the true quality of a voice call. A score more than 3.5 is considered quite good enough to carry out communication. Voice compression will affect the MOS. Codec G.729 has an MOS equivalent to 3.93 out of 5.

V. CONCLUSION
An effort has been made to implement VoIP technology for practical purposes. After implementing the VoIP connection on GNS-3 platform working on connection less UDP protocol and RTP and RTCP as media protocols, its LAN and WAN implementation was analyzed in terms of bandwidth. The bandwidth at which the connection was established is considered and compressed accordingly using the codec G.729 operating at a MOS of 3.93 out of 5. It is observed that packet loss and data loss is minimal after the compression and jitter has slightly been changed without affecting voice quality.

REFERENCES