

# VOICE COMPRESSION ON VOIP PLATFORM

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**Abstract**—Applications like Hangouts, GTalk, Asterisk and Apple's FaceTime which are being used prolifically use VoIP (Voice over Internet Protocol) technology for their proper implementation. VoIP technology is a consequential factor contributing much to the way people use to communicate online in this era. This paper proposes a novel method thorough which VoIP platform can be established and then voice compression could take place using the devised method. GNS-3 (Graphics Network Simulaotr-3), a simulator, can be deployed to implement VoIP technology for practical purpose of two way communication. Cisco IP Phones needs to be installed and registered on a host and/or server system for its felicitous implementation. It can be implemented on LAN (Local Area Network) or WAN (Wide Area Network) by making the host and/or server to act as CME (Call Manager Express) or utilizing wi-fi. Quality of Voice enabled IP Phones is tenacious by MOS (Mean Opinion Score) which in turn depends on the bitrate it offers. Codecs like G729 with an MOS of 3.93 which fundamentally works on CS-ACELP (Conjugate Structure-Adaptive Code Excited Linear Prediction) Algorithm offers efficacious and novel way through which voice can be compressed thereby reducing the bandwidth on the above mentioned platforms. Without any paramount data loss and packet loss and with remote variation in jitter voice can be compressed on VoIP on the above mentioned platforms utilizing codec.

**Keywords**-VoIP, GNS-3, codec G729, CS-ACELP Algorithm, CME, MOS

## INTRODUCTION

As we all know that there have been innumerable voice compression techniques known and over the years they have been deployed for the purpose of reducing the bit rate. VoIP communication set up on Cisco IP communicator provides a promising platform where voice compression techniques such as Codec G729 based on CS-ACELP (Conjugate Structure-Adaptive Code Excited Linear Prediction) algorithm can be applied. The encoding and decoding techniques used in CS-ACELP helps to achieve the same.

There has been a significant work carried out on various platforms on which VoIP has been deployed so far. Through this paper a novel way of simulated version of real-time VoIP connection has been portrayed forward using the simulator GNS-3, Cisco IP Phones, Superputty as the Console and Wireshark as a platform where the results of packet capturing and voice compression will be shown. IP Phones needs to be registered on CME (Call Manager Express) which facilitates voice communication. E-phone number is assigned to all IP Phones through which that particular user can be communicated over the network.

## WORKING OF VOIP

The fundamental principle of working of VoIP is Analog to Digital Conversion bolstered with sampling, quantization, encoding, digital signal generation and then followed by decoding at the end. On the simulated platform of GNS-3 a router is linked to IP Phones which can be configured on either LAN, WAN or on wireless platforms. For LAN platform, router is linked just to a cloud configured to Ethernet and same applies for the wireless connection establishment. But for WAN, router is connected to two cloud interfaces namely Ethernet and Virtual Machine such as VMware Workstation. A DHCP (Dynamic Host Configuration Protocol) pool needs to be created and renamed which automatically assigns IP address to the computer from a defined range of computers. We need to create telephony-service to add the desired number of e-phones and directory address The IP address set over the Ethernet and VMware adapter needs to be changed accordingly for the above purpose. Using the simulator GNS-3 and Cisco IP Phones, and Superputty one can easily register IP phone on CME. CME provides a platform where voice communication can be established and used hence forth.

## IMPLEMENTATION

### *Steps to connect VoIP on LAN*

- 1) On GNS-3(Graphical Network Simulator-3) capture the image of Router.
- 2) Select a Router, set its interface as f0/0 and a then a Cloud and make its slot as Ethernet and connect them using a fast Ethernet Link. Click the start button and double click on the Router.

- 3) At f0/0 interface of the Router configure its IP address.
- 4) Devise a DHCP pool and provide its name.
- 5) Using 'option 150' command, configure TFTP (Trivial File Transfer Protocol) address.
- 6) Add required number of e-phones and directory numbers on the created telephony service. Assign IP address to it.
- 7) Assign a short extension (telephone) number to each of the created e-phones.
- 8) Connect systems using LAN wire and/or switch if required.
- 9) Set the Ethernet adapter IP address corresponding to the Router's IP address on all the hosts and the CME (Call Manager Express).
- 10) On Cisco IP Phones write the same address of TFTP as assigned in the 'option 150' command.
- 11) All IP Phones are then registered on CME. Restart all IP Phones.
- 12) Dial the required number from one of the host and start the communication.

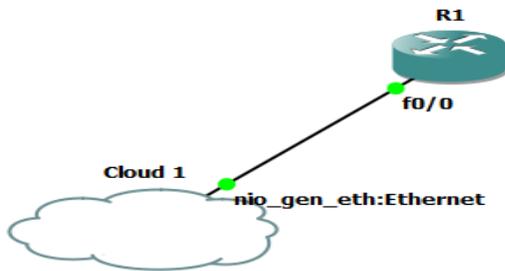


Figure 1: VoIP Connection on LAN using GNS-3

#### Steps to connect VoIP on WAN

A Router with two interfaces (namely f0/0 and f0/1) along with 2 clouds is configured for practical implementation on WAN (Wide Area Network) environment. VMware Net1 is the slot to be set for 1st Cloud, and registration of IP phone is done virtually using VMware and IP address of VMware's adapter is set accordingly and for the 2<sup>nd</sup> Cloud, Ethernet is set as the slot and accordingly IP address of Ethernet Adapter is set.

All the steps required for VoIP connection on LAN has to be applied for WAN connection too. Along with that, dial-peer voice connection for VoIP needs to be assigned where according to the e-phone-numbers assigned destination pattern is set for each of them. Session target happens to be the IP address of the other LAN to which the communication has to be made.

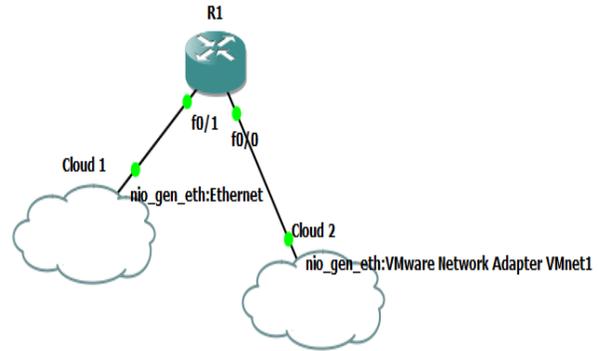


Figure 2: VoIP Connection on WAN using GNS-3

#### VOICE COMPRESSION USING CODEC

After successful implementation and registration of e-phones over the VoIP network one may wish to know the bandwidth at which the communication is established. Henceforth, this arouses the curiosity of compression of voice over the above mentioned platforms for VoIP.

Codec G729 working on CS-ACELP algorithm has been devised to compress the bandwidth and in turn the voice at the communication has been established. Codec is used for coding and decoding stream of audio. It works on the core principle of quantization, sampling, encoding, decoding etc. facilitated through the complex algorithm named CS-ACELP. Codec G729 has been used to compresses voice from 64Kbps to 8Kbps (81Kbps to 24Kbps including the 16Kbps header size practically).

#### Working of CS-ACLEP Algorithm for Voice Compression

CS-ACELP algorithm basically performs encoding at the sender side and decoding at the receiver side. The fundamental principle is that when we talk the analog voice is converted into the digital voice and then transmitted and processed for compression and then at the receiver side the digital voice is again converted to the analog form and received at the other end of the communication setup.

#### CS-ACELP Encoding

CS-ACELP uses analysis by synthesis technique on speech frames to compress the voice. A 140Hz high pass filter is used to process the input analog signal; next to extract Linear Prediction Coefficients (LPCs) tenth order Linear prediction analysis is performed using Levinson-Durbin recursion Algorithm using autocorrelation method. Using Vector Quantization method LPCs are converted to Line Spectral Pairs (LSPs). Using excitation signal weighted mean square error (WMSE) is minimized between synthesized and original signal and this in turn is achieved using a perceptual weighing filter. Using open and closed loop pitch analysis pitch delay is calculated and analyzed. Algebraic structure forms the backbone of the fixed-codebook which is a deterministic

codebook where codebook index is used to determine the excitation vector.

useful purposes. The compressed voice can be further subjected to analysis and experimentation.

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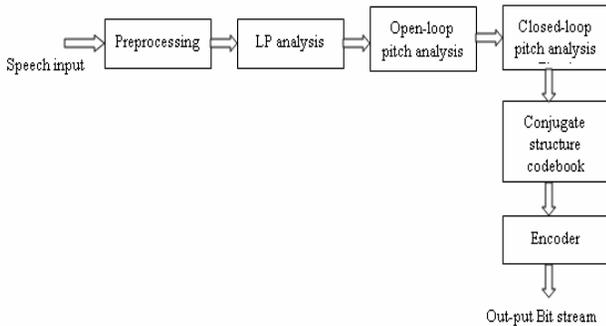


Figure 3: CS-ACELP Encoding [3]

## CS-ACELP Decoding

Decoding phase consists of three major stages namely:- Extracting speech parameters from the received speech bit stream. Next step is the combination of excitation reconstruction and synthesis filtering in the speech reconstruction stage. Next step is post processing into which Nyquist equation plays a major role. Human ears can hear between 20Hz to 20,000Hz but human voice frequencies can vary between 300Hz to 3,500Hz. Then using Nyquist Equation  $2 \times 3500\text{Hz} = 7000\text{Hz}$  happens to be the peak frequency at which the human voice can be recorded. To avoid any kind of conversion errors 8000Hz is chosen for practical purposes. At a sampling rate of 8000Hz the task of digitizing human voice at eight bit numbers for each sample leads to the bandwidth =  $8 \times 8000\text{Hz} = 64\text{Kbps}$ . Codec G729 based on CS-ACELP chooses best 1000 samples out of these available 8000 samples with 8-bit number for each sample and compresses voice at a bandwidth =  $8 \times 1000\text{Hz} = 8\text{Kbps}$  without data or packet loss thus retaining the quality of voice over VoIP platform. Codec G729 offers 3.93 out of 5 as MOS (Mean Opinion score) which is considered as efficient and reliable.

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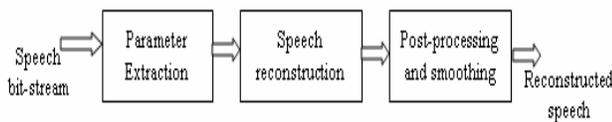


Figure 4: CS-ACELP Decoding [3]

## CONCLUSION

Through this paper a technique has been devised to compress voice and in turn bandwidth on the above crafted platform for VoIP implementation using Codec G729 which in turn is based on CS-ACELP algorithm. VoIP platform design will efficiently enhance the voice quality and facilitate easy and facile way to connect users over the network without adding much to the cost factor. Presently, the implementation encoding and decoding of CS-ACELP algorithm is demonstrated without highlighting much of the complicated intricacies involved. In future these can be exploited for other