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Performance Analysis of G.711 Codec in VoIP Using UDP

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Abstract:
The converged IP network is used to incorporate voice, data and video on the same network. The idea of converged broadband network with PSTN is for increases reliability, speed, mobility and reducing the cost of the network. In this the VoIP Codec's are used to convert analog signals into digital and send it in packet format over the internet. In this paper, we will discuss and analyze the issues of VoIP Codec's which degrades the QoSof VoIP issues like latency, jitter, and packet loss.

Keywords: *VoIP, G.711 CODEC, QoS, wave file*

1. Introduction

Nowadays there is growing trend to using a real time communication using internet protocol (IP) voice over internet protocol (VoIP) is used for IPtelephoney.VoIP allows to a user to transfer a voice using internet. In VoIP data is digitally sent using IP instead of analog telephone lines. Today there are lots of VoIP applications we use as a part of our daily life like Skype, Viber, What's app, Hike and Yahoo messenger. All these applications provide us free calls, massaging and sharing and downloading pictures and videos and the good quality of service. In VoIP analog signal is converted into digital signals and transmitted it in the packet stream to a receiver. At the receiver end the digital signals are converted into analog signals using Public Switched Telephone Network, thus it is important that at the both transmitter and receiver must support the same codec for encoding and decoding.

In VoIP quality of voice is more important so our main aim is to find the solution of the problems which affects the QOS of VoIP like delay, jitter, latency, packet loss.

The simple VoIP network is shown in following fig.1

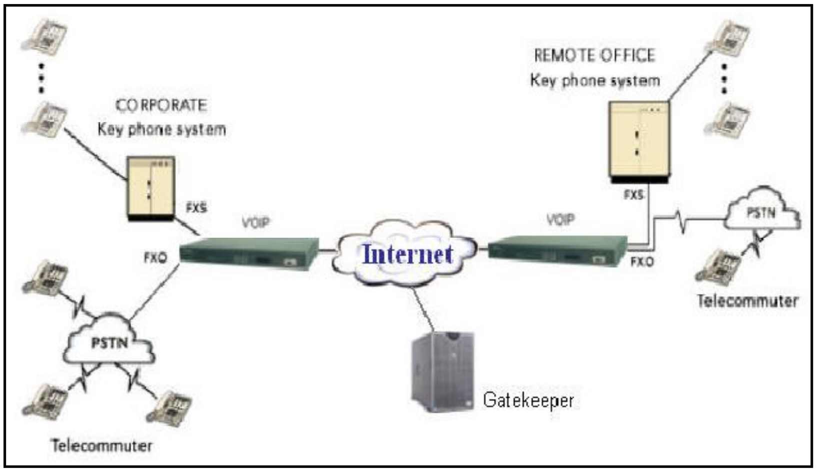

Figure 1: Simple VoIP Network [1]

IP networks are the most bandwidth-efficient and the communication channels have very limited bandwidth. The PSTNs use the INTERNATIONAL TELECOMMUNICATION UNION (ITU) Coding schemes [2].

2. VoIP Codecs

A codec which means coding and decoding, which converts an analog signals into compressed form of digital at transmitter side and at the receiver side, it converts that signal at uncompressed analog signal for playing the audio file at receiver [3].

There are some codec's which are commonly used in VoIP for applications are G.711, G.728, G.723, G.722, G.726 and G.729, etc.This takes the bandwidth 64kbps, 16kbps, 6.3kbps, 48kbps, 24kbps and 8kbps respectively [3].

In our project we first implement the G.711 codec which uses PCM coding. The G.711 codec Samples the analog signals at the sampling rate of 8,000Hz or 8 kHz and encodes the one sample at 8 bit and taking the bandwidth of 64kbps [2].

In our project, we then implement a G.711 codec and check their performance like latency, jitter and packet delay which cause their QOS. For that we mainly consist three parts such as an audio wave file, G.711 Codec and a protocol for transmission.

IUT-T Codec	Algorithm	Codec Delay (ms)	Bit Rate (kbps)	Sampling Rate (kHz)	Comments
G.711	Pulse-code modulation (PCM)	0.375	64	8	Deliver precise speech transmission. Very low processor requirements.

Table 1: Features of the most common codec G.711

3. Protocol

As we know the RTP supports the transfer of the real time media like audio, video and other multimedia services over the packet switched network. This protocol is used by the both H.323 and the SIP [4]. This protocol is used to transmit time sensitive data. The RTP is work in the transport and network layers. RTP protocol is run in unison with user Datagram Protocol (UDP)[3]. The RTP message is encapsulated in a UDP datagram that is further encapsulated in an IP datagram for transmission.

Our system structure is like

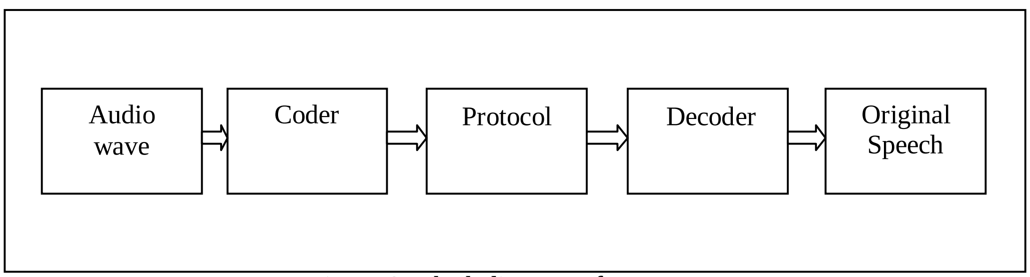

Figure 2: Block diagram of our system

In these the speech signals are converted into wave file and these files are encoded firstly to transmit towards the transmission protocol then at receiver side decode the signal in speech form.

4. Results

The performance of G.711 codec has been analyzed by implementing the VoIP codec in MATLAB using the UDP protocol. The original signal is shown in fig.3. The fig.4 shows the error in the original signal. Jitter is shown in fig.5. Fig.6 shows the latency in the transmitted signal. In dotted point form. Delay is shown in fig.7. All these parameters affect the QoS of VoIP networks

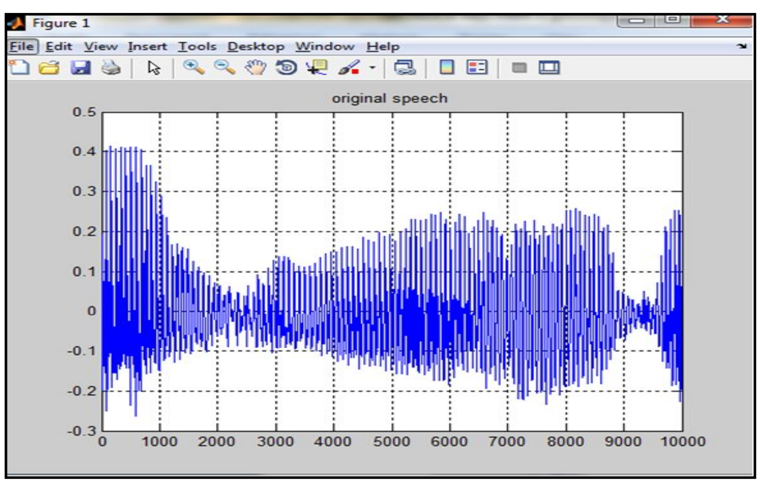

Figure 3: Original Speech

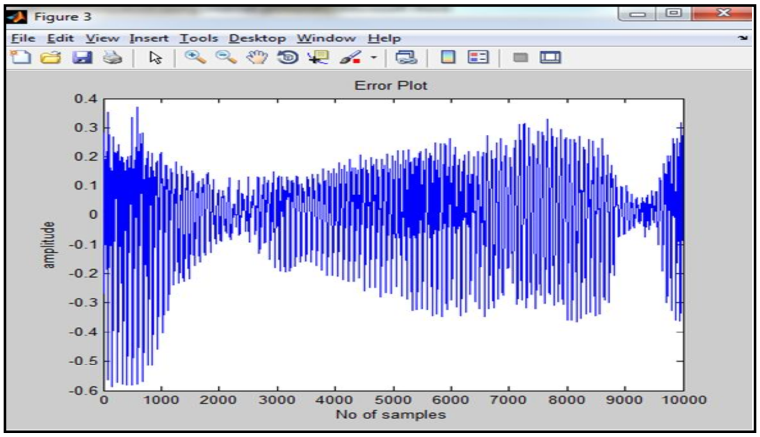

Figure 4: Error Plot in Original Speech

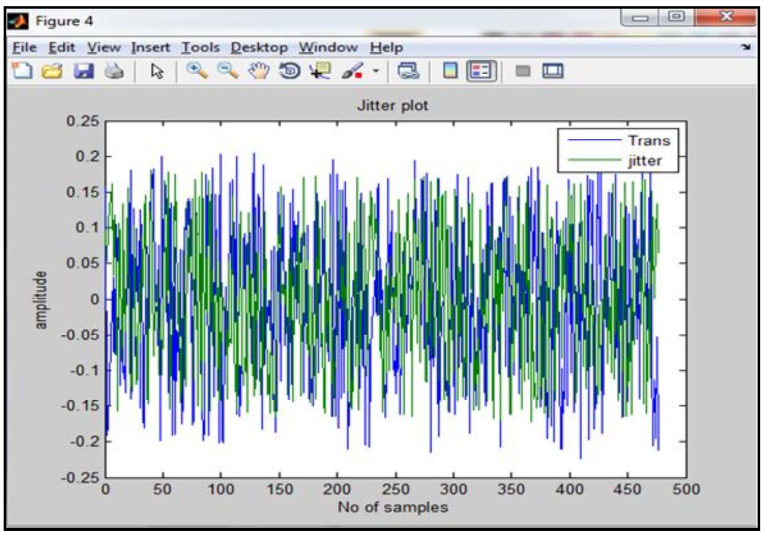

Figure 5: Jitter Plot Original Speech

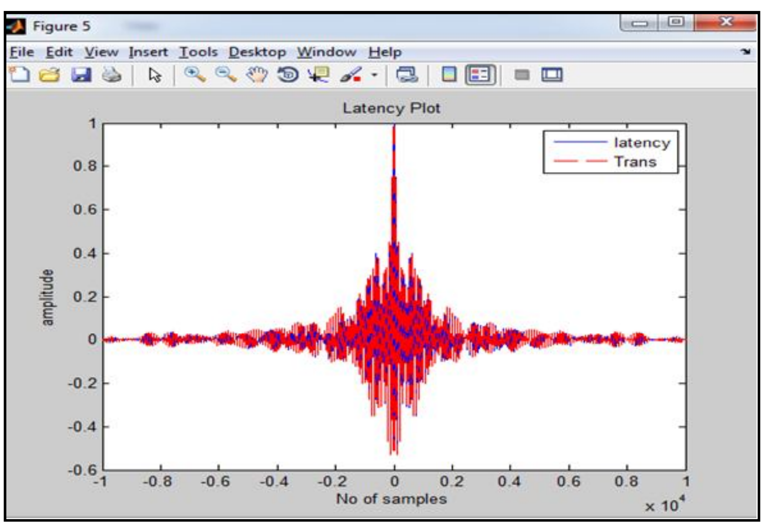

Figure 6: Latency Plot in original speech

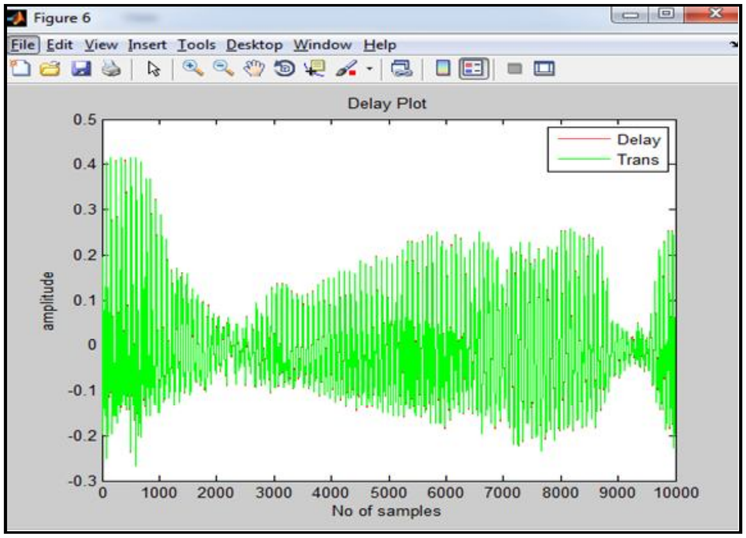

Figure 7: Delay Plot of Original Speech

5. Conclusion

In our project, we developed our VoIP software, which is used for voice communication between two hosts. In our present paper, we implement and analyzed the G.711 codec using the UDP protocol. This is a common narrow band codec. This codec consumes more bandwidth, but has the same quality as landline. The G.711 Codec is widely used in VoIP because it is an open source codec and its results are suitable for landline telephoney. It can be analyzed from the result that G.711 is an ideal solution for PSTN networks with PCM scheme.

6. References

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