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Performance Analysis of VoIP Codecs for H323

Senthilkumar L
M.E.Communication Systems
Department of Electronics and Communication Engineering
College of Engineering Guindy, Anna University
Chennai, India

Krishna Chennakesava Rao.M.
M.E.Communication Systems
Department of Electronics and Communication Engineering
College of Engineering Guindy, Anna University
Chennai, India

ABSTRACT
This paper analyzes the performance of G.711, G.723, G.729 and G.726 VoIP Codecs in H323 standard. Recently, VoIP (Voice over Internet Protocol) is a great interesting voice communication over the Internet, with high level quality of service (QoS) along with circuit switch and cellular technologies. VoIP Codecs decides the performance of the VoIP network. Qualnet 5.0 is used for simulation.

Keywords
VoIP, H323, G.711, G.723, G.729 and G.726

1. INTRODUCTION
Multimedia applications are gaining much of the user attention, with the advent of new broadband technologies. In recent decades user desires have switched from net surfing and email to multimedia services, such as VoIP & video conferencing and video streaming, etc. The H323 standard has been designed as the broadband access network to fulfill the user needs of multimedia applications. The public switched telephone network (PSTN) has developed itself to accommodate these requirements. The internet has become a very popular means of communication in a very short period of time. It has established itself as a massive communication infrastructure that provides many services such as electronic mail .In the recent years it has further developed itself into providing Internet Telephony or Voice over internet protocol (VOIP).

A. *VOIP Architecture:*
VoIP is a transmission technology, for delivery of voice communications over IP networks such as, the Internet or other packet-switched networks. Internet telephony refers to communication services such as voice, facsimile, and/or voice-messaging applications that are transported via the Internet, rather than PSTN. VOIP converts the voice signal from telephone into a digital signal that travels over the internet and is converted back at the other end. VoIP allows the users to make a call directly from a computer using a conventional telephone. VoIP systems employ session control protocols to control the set-up and tear-down of calls as well as audio codecs which encode speech allowing transmission over an IP network as digital audio via an audio stream. [1]
VoIP represents the next generation in communication services. By moving voice services to the data network, we eliminate a separate, managed voice infrastructure and dramatically reduce the cost of telephone moves.

B. *VOIP codecs:*
Codec stands for coder-decoder. It converts an audio signal into compressed digital form for the transmission and then back into an uncompressed audio signal for replay. Codecs accomplish the conversion by sampling the audio signal. There are different sampling rates in VoIP depending on the codec being used:[2]
64,000 times per second
32,000 times per second
8,000 times per second

I. **G.711**: It is also known as Pulse code modulation (PCM) of voice frequencies. It is the required standard in many technologies like H.320 and H.323. It can also be used for fax communication over IP networks (as defined in T.38 specification). G.711 uses a sampling rate of 8,000 samples per second, with the tolerance on that rate 50 parts per million (ppm). Non-uniform (logarithmic) quantization with 8 bits is used to represent each sample, resulting in a 64 Kbit/s bit rate.[2]

II. **G.723**: G.723 is a ITU-T standard speech codec with extensions of G.721. It provides the voice quality covering 300 Hz to 3400 Hz by using Adaptive Differential Pulse Code Modulation (ADPCM).

III. **G.726:** G.726 is an ITU-T ADPCM speech codec standard covering the transmission of voice at rates of 16, 24, 32, and 40 Kbit/s. It was introduced to supersede both G.721, which covered ADPCM at 32 Kbit/s, and G.723, which described ADPCM for 24 and 40 Kbit/s. The four bit rates associated with G.726 are often referred by the bit size of a sample, which are 2-bits, 3-bits, 4-bits, and 5-bits respectively. The most commonly used mode is 32 Kbit/s, which doubles the usable network capacity by using half the rate of G.711. It is primarily used on international trunks in the phone network. The principal application of 24 and 16 Kbit/s channels is for overload channels carrying voice in digital circuit multiplication equipment (DCME). The principal application of 40 Kbit/s channels is to carry data modem signals in DCME, especially for modems operating at greater than 4800 Kbit/s.

IV. **G.729:** G.729 is an audio data compression algorithm for voice that compresses digital voice in packets of 10 milliseconds duration. It is officially described as Coding *of speech at 8 Kbit/s using conjugate-structure algebraic code-excited linear prediction (CS-ACELP)*. Because of its low bandwidth requirements, it is widely used in VoIP applications. Standard G.729 operates at a bit rate of 8 Kbit/s, but there are extensions, which provide rates of 6.4 Kbit/s and 11.8 Kbit/s for worse and better speech quality, respectively.

C. Mean Opinion Score (MOS):

In voice and video communication, quality usually dictates whether the experience is a good or bad. MOS is a numerical method of expressing voice and video quality. MOS gives a numerical indication of the perceived quality of the media received after being transmitted and eventually compressed using codecs.

MOS is expressed in one number, from 1 to 5, 1 being the worst and 5 the best. Taken in whole numbers, the numbers are quite easy to grade.

. 5 - Perfect. Like face-to-face conversation or radio reception.

4 - Fair. Imperfections can be perceived, but sound still clear.

3 - Annoying.

2 - Very annoying. Nearly impossible to communicate.

1 - Impossible to communicate

The values do not need to be whole numbers. Certain thresholds and limits are often expressed in decimal values from this MOS spectrum. For instance, a value of 4.0 to 4.5 is referred to as toll-quality and causes complete satisfaction. This is the normal value of PSTN and many VoIP services aim at it, often with success. Values dropping below 3.5 are termed unacceptable by many users.

MOS is used to assess the work of codecs, which compress audio and video to save on bandwidth utilization but with a certain amount of drop in quality. MOS tests are then made for codecs in a certain environment.

D. H323

H.323 is an ITU-T standard that defines the protocols to provide audio-visual communication sessions on any packet network. It addresses multimedia communications over ISDN, the PSTN or SS7, and 3G mobile networks. It also addresses call signaling and control, multimedia transport and control, and bandwidth control for point-to-point and multi-point conferences. It is widely implemented by voice and videoconferencing equipment manufacturers. It is used in various Internet real-time applications such as GnuGK and NetMeeting and is widely deployed worldwide by service providers and enterprises for both voice and video services over IP networks.

2. SIMULATION

Qualnet 5.0 is used for simulation which is network software from scalable networks. 23 nodes and one router is used for simulation. Four VoIP applications are used.

E. Scenario:

Node 1 is the call initializer and node 20 is call receiver. between the 1 and 20 four VoIP applications established. H323 VoIP signaling protocol is used for communication. Each VoIP application is uses different voice codecs i.e., VoIP, H323, G.711, G.723, G.729 and G.726 .

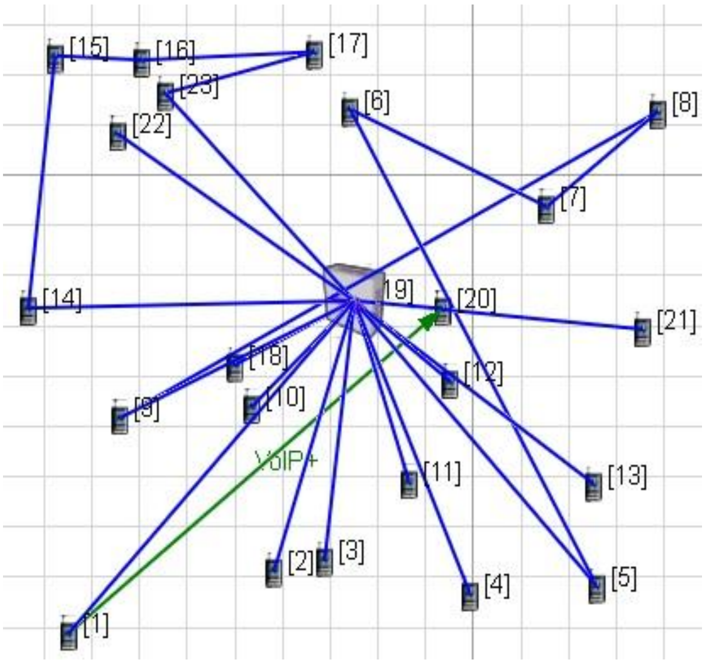

Fig 1 : VoIP scenario

F. Parameter Set:

Table 1: Simulation Parameter Set of VoIP scenario

Number of Nodes	23
Application Layer	VoIP(H323)
Physical Layer	802.3
Mac Layer	802.3
Average Talk Time	10sec

G. Results:

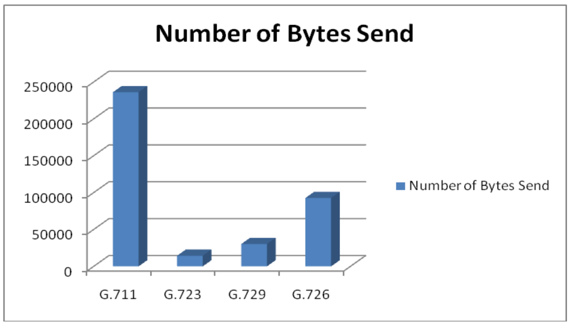

Fig 2:Number of bytes send

Table 2:Number of bytes Send:

VoIP Codecs	Number of Bytes Send
G.711	236160
G.723	14444
G.729	30340
G.726	92800

G.711 codec has the highest metric as it has the low compression rate. Table 2 and figure 2 show that G.723 good compression rate. G.729 has medium rate.

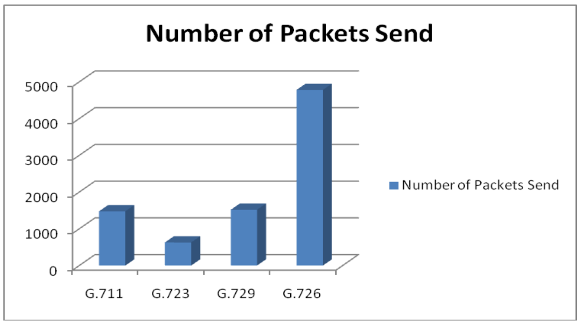

Fig 3: Number of packets send

Table 3: Number of packets send

VoIP Codecs	Number of Packets Send	Bytes per Packets
G.711	1476	160
G.723	628	23
G.729	1517	20
G.726	4781	20

Number of Bytes per packet is high in G.711. So in G.711 packet loss is a crucial factor which affects the VoIP performance. G.729 and G.726 has less Bytes per Packet value.

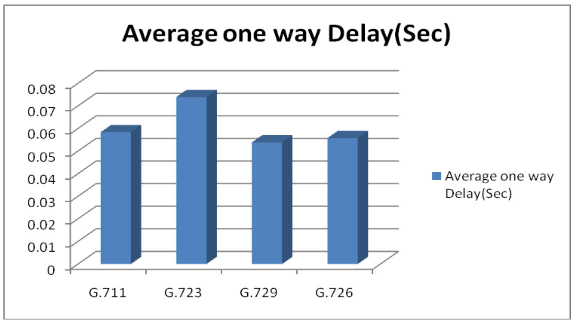

Fig 4: Average one Way Delay

Table 4: Average one Way Delay

VoIP Codecs	Average one way Delay(Sec)
G.711	0.0583512
G.723	0.0739344
G.729	0.0538376
G.726	0.0557722

Figure 4 and table 4 show that, G.723 has the highest Average one way delay so it will affect the performance the VoIP networks. G.729 has low delay so it offers good performance.

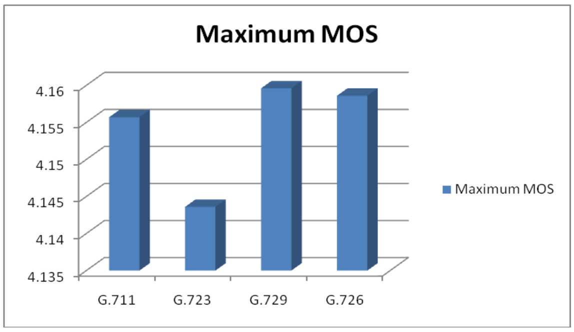

Fig 5: Maximum MOS

Table 5: Maximum MOS

VoIP Codecs	Maximum MOS
G.711	4.1557
G.723	4.1436
G.729	4.1596
G.726	4.1586

Figure 5 and table 5 ensure that, G.729 has the highest MOS of 4.1596 which can offer a good performance in VoIP communication. G.723 provides less MOS among the four codecs.

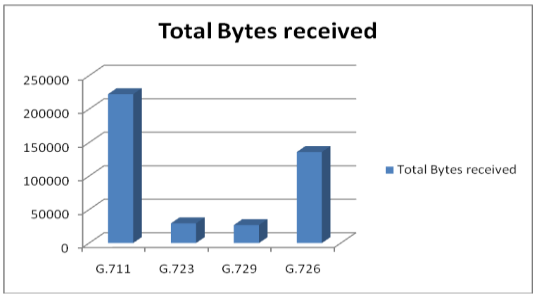

Fig 6: Total Bytes received

Table 6: Total Bytes received

VoIP Codecs	Total Bytes received
G.711	222240
G.723	29417
G.729	26860
G.726	135920

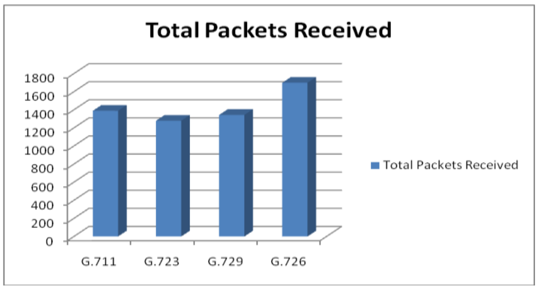

Fig 7: Total packets received

Table7: Total Bytes received

VoIP Codecs	Total Packets Received
G.711	1389
G.723	1279
G.729	1343
G.726	1699

Table 6 and 7 show the total bytes and packets received respectively for various codec protocols.

3. CONCLUSION

Performance of VoIP codecs have been analyzed in H323 network. Compression rate of G.711 is very less as compared to G.723, G.729 and G.726 but it has good MOS. The results show that G.729 has highest MOS and comparatively low one way Delay. Bytes per packet are high in G.711. So in G.711, packet loss highly affects the VoIP performance. G.726 comes next to G.729 which provides moderate performance. G.723 gives average performance (MOS = 4.1436).

4. FUTURE WORK

VoIP network can be analyzed in WiMAX with different routing protocols. Security can also be analyzed with different threats.

5. REFERENCES

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