Performance Analysis of VoIP Codecs over WiMAX Networks

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Abstract

Real-time services such as VoIP are becoming popular and are major revenue earners for network service providers. These services no longer confined to the wired domain and extended over wireless networks. Although some of the existing wireless technologies can support some low-band width applications, the bandwidth demands of many multimedia applications exceed the capacity of these technologies. The WiMAX promises to be one of the wireless access technologies capable of supporting very high bandwidth applications. However, there are still many challenges that need to be addressed to provide a steady and good quality voice connection over the best-effort WiMAX network. In This paper, we evaluate the performance of different VoIP codecs over the WiMAX network. The network performance metrics such as MOS, packet end-to-end delay, jitter and packet delay variation have been used to evaluate the performance of VoIP codecs. The results showed that, the codec G.723 provides the best results among all discussed Codecs in all presented performance metrics; the lowest delay, the highest MOS and the highest Throughput.

Keywords: *Voice over Internet Protocol (VOIP), VOIP codecs, and Optimized Network Engineering Tools (OPNET), Quality of Service (QoS), worldwide interoperable for Microwave Access (WiMAX).*

1. Introduction

Multimedia applications are gaining much of the user attention with the advent of new broadband technologies [1]. In recent decades, user desires have switched from net surfing and email to multimedia services, such as VoIP & video conferencing and video streaming, etc. To address the specific user needs for rich multimedia applications, the service providers are looking for broadband wireless network. The IEEE 802.16 standard [2] [3] has been designed as an access network to fulfill the user needs of multimedia applications. WMAN provides cost effective infrastructure to service providers and promised QoS to end users without increasing the complexity in the core network as well as at the user side WiMAX is easy to deploy and integrate with the existing IP core network, which acts as a backbone infrastructure. The IP core offers the support of advanced technologies and protocols to WiMAX that fulfills the required Quality of Service (QoS) and security features [4].

In recent years, Voice over Internet Protocol (VoIP) has attracted the attention of the network engineering research and operation communities. The phenomenal growth of VoIP is the result of rapid commercial solutions and network improvements. Other factors include the ongoing decrease in quality differences between existing Public Switched Telephone Networks (PSTN) telephony and VoIP [5] and the increase in bandwidth available to commercial and residential customers over which VoIP may be transported [6]. The QoS parameters closely connected with user's satisfaction with a received speech quality. The user's satisfaction expressed with a subjective listening score that is a result of subjective listening tests. The substitutions to the subjective tests are objective methods. The objective methods assess the speech based on a signal processing without need of real listeners. PESQ method (Perceptual Evaluation of Speech Quality) is mostly used objective method for evaluation of the speech quality [7]. The first version of the IEEE 802.16 standard operates in the 10–66GHz frequency band and requires line of sight (LoS) towers. Later the standard extended its operation through different PHY specification to 2-11GHz frequency band enabling non-line of sight (NLoS) connections, which require techniques that efficiently mitigate the impairment of fading and multipath [8]. It supports both point-to-multipoint (P2MP) and multipoint-to multipoint (mesh) modes. WiMAX will substitute other broadband technologies competing in the same segment and will become an excellent solution for the deployment of the well-known last mile infrastructures in places where it is very difficult to get with other technologies, such as cable or DSL, and where the costs of deployment and maintenance of such technologies would not be profitable. In this way, WiMAX will connect rural areas in developing countries as well as underserved metropolitan areas. It used to deliver backhaul for carrier structures, enterprise campus, and Wi-Fi hotspots. WiMAX offers a good solution for these challenges because it provides a cost-effective, rapidly deployable solution [9]. The purpose of this study was to examine a case of QoS deployment over a WiMAX network and to examine the capability of a WiMAX network to deliver adequate QoS

to voice application. The methodologies taken include creating the WiMAX network using OPNET, carried out extensive simulations to analyze the MOS, packet end-to-end delay, jitter, traffic received, throughput and packet delay variation, deploying the VOIP application, adjusting the variables of VOIP within different codecs. This topic is important to researchers and manufacturers in providing them the necessary background for their works.
The rest of the paper organized as follows. Section (2) gives background of VoIP. Section (3) deals with signaling protocols, Section (4) deals with VOIP codecs, Section (5) deals with the effective of delay and loss for VOIP, Section (6) deals with the Performance Metrics, Section (7) deals with Results and Discussion. Finally, in Section (8) we conclude this paper.

2. VOIP Overview

VoIP, known as IP Telephony, is the real-time transmission of voice signals using the Internet Protocol (IP) over the public Internet or a private data network [10]. In simpler terms, VoIP converts the voice signal from your telephone into a digital signal that travels over the Internet. One of the most significant advantages of VoIP over a traditional public switched telephone network (PSTN) is that one can make a long distance phone call and bypass the toll charge. This integrated voice/data solution allows large organizations (with the funding to make the transfer from a legacy network to a VoIP network) to carry voice applications over their existing data networks. Not only will this technological advancement have an impact on the large traditional telecommunications industry, it will alter the pricing and cost structures of traditional telephony [11]. Furthermore, when compared with circuit-switched services, IP networks can carry 5 to 10 times the number of voice calls over the same bandwidth.

2.1 How does VoIP work?

VoIP uses Internet Protocol for transmission of voice as packets over IP networks. The process involves digitization of voice, the isolation of unwanted noise signals and then the compression of the voice signal using compression algorithms/codecs. After the compression, the voice packetized to send over an IP network, each packet needs a destination address and sequence number and data for error checking. The signaling protocols are added at this stage to achieve these requirements along with the other call management requirements. When a voice packet arrives at the destination, the sequence number enables the packets to be place in order and then the decompression algorithms are applied to recover the data from the packets. Here the synchronization and delay management needs to be taken care of to make sure that there is proper spacing. Jitter buffer is used to store the packets arriving out of order through different routes, to wait for the packets arriving late.

3. The Signaling Protocols

H.323 and SIP are used to setup the route for the transmission over the IP network; Gateway protocols like media gateway control protocol are used to establish control and status in the media and signaling gateways. Routing (UDP, TCP) and transport protocols (RTP) are used once the route is established for the transport of data stream.

3.1 H.323 Protocols

H.323 is the ITU-T standard for packet based multimedia communication, though originally developed for multimedia conferencing over LAN's it was later modified for VoIP as well. With versions coming out in 1996 and 1998, the standard has faced stiff competition from the other protocol SIP that was specifically designed for VoIP, but is more used because of its wide existence in the already installed networks. The standard is interoperable and has both point-to-point and multipoint capabilities.

3.2 Session Initiation Protocol (SIP)

SIP is the IETF standard for VoIP signaling, it based on the existing protocols like SMTP and HTTP, and uses a text-based syntax that is comparable to HTTP uses in web addresses. A web address is comparable to a telephone number in a SIP network; also, the PSTN phone numbers are also compatible in a SIP network ensuring interfacing with PSTN systems. It provides a mobility function to the users, supports multiple media sessions during a single call hence users can - share a game, use instant message (IM), and talk at the same time, work with most protocols like RTP, Session Description Protocol (SDP), and Session Announcement Protocol (SAP).

3.3 Soft-Switch

It is far less expensive in terms of both purchase and maintenance as compared to the conventional switches used in PSTN networks. The soft-switch coordinates the call control, signaling, and the other features that enable making a call across networks possible. It provides all the essential call control and service logic functions, coordinates routing of signaling messages between networks, it also provides all the administrative functions like billing statistics and providing other value added services to the users. The various components of the Soft-switch are signaling Gateway, media Gateway and application Server

4. VOIP Codec's

A Voice Codec used at the subscriber side to convert the analogue voice waves to digital pulses and vice versa. There are different codecs types based on the selected sampling rate, data rate, and implemented compression algorithm listed in table.1. In order to determine the bandwidth requirements of VoIP connections, we will start by presenting common VoIP Codecs and their associated characteristics.

Among these codecs, G.711, G.729 and G. 723 are the most widely used codec's in existing networks. The G.711 codec currently used in a wide range of applications, its voice-sampling rate is 8 kHz and each sample encoded with 8 bits resulting in a constant 64 kbps bit rate and offering a very good voice quality. Samples are packed into frames every 10 ms. In order to further reduce the bit rate generated by this codec without damaging the voice quality in packet based multimedia applications such as VoIP [12]. The G.729 codec also generates speech frames every 10 ms containing 80 voice samples (collected at a sampling rate of 8000 samples per second) [13].

However, it requires a 5 ms look-ahead delay before producing any new frame. Annex A of the standard (G.729 a) has the same main features of G.729 but is considered as a reduced complexity version; developed for multimedia simultaneous voice and data applications. In addition, a silence suppression scheme developed as Annex B of the standard (G.729 b). This version defines a voice activity detector (VAD) and comfort noise generator (CNG) for G.729 in order to reduce the transmission rate during silence periods.

Table.1 Audio Codec's

ITU no.	Rate (CBR) [Kbit/s]	algorithm	Frame length
G.711	64	Puls CodeModulation	0.125 ms
G.723.1	6.4/5.3	Multipulse maximum likelihood quantization/algebraic-code-excited linear prediction	30 ms
G.726	40/32/24/16	Adaptive differential PCM	0.125 ms
G.728	16	Low-delay code-excited linear prediction	2.5 ms
G.729	8	Conjugate-structure algebraic-code-excited linear prediction	10 ms

The G.723 CODEC has 2-bit rates associated with it, 5.3 and 6.3 Kbps (G.723r53 and G.723r63, respectively). The higher bit rate has greater quality. The lower bit rate provides fair quality and provides system designers with additional flexibility. G.723 CODECs require moderate computation complexity and introduces a relatively high delay [14].

5. Effective of Delay and Loss for VOIP

5.1 Effect of Delay

The end-to-end delay (consisting of codec delay, network delay, and play-out delay) and total loss probability (consisting of loss in the network and play-out loss at the receiver’s decoder buffer) affect the VoIP call quality, i.e., the R-score. Effect of Delay in a VoIP system, the total mouth-to-ear delay is composed of three components: codec delay (D_{codec}), play-out delay ($D_{play\text{-}out}$), and network delay ($D_{network}$).

Codec delay represents the algorithmic and packetization delay associated with the codec and varies from codec to codec. For example, the G729.a codec introduces a delay 25 ms delay. Play-out delay is the delay associated with the receiver side buffer required to smooth out the delay for the arriving packet streams.

Network delay is the one-way transit delay across the IP transport network from one gateway to another. Thus the total delay is:

$$D = D_{codec} + D_{playout} + D_{network}$$

The delay impairment Id depends on the one-way mouth-to-ear delay encountered by the VoIP streams. This mouth-to-ear delay, denoted by I_d, determines interactivity of voice communication. Its impact on voice quality depends on a critical time value of 177.3 ms [15], which is the total delay budget (one-way mouth-to-ear delay) for VoIP streams.

5.2 Effect of Loss

VoIP call quality is also dependent on the loss impairment, recall that I_e represents the effect of packet loss rate; I_e accounts for impairments caused by both the network’s and the receiver’s play-out losses [15]. Different codecs, with their unique encoding/decoding algorithms and their packet loss concealment techniques yield different values for I_e, the used E-model; proposed in [16], which relates I_e to the overall packet loss rate:

$$I_e = \gamma_1 + \gamma_2 \ln(1 + \gamma_3 e)$$

where γ_1 is a constant that determines voice quality impairment caused by encoding, and γ_2 and γ_3, describe the impact of loss on perceived voice quality for a given codec. Note that e includes both network losses and play-out buffer losses, which modeled as,

$$e = e_{network} + (1 - e_{network}) \times e_{playout}$$

where $e_{network}$ is the loss probability due to the loss in the network and e play-out is loss probability due to the played-out loss at the receiver side.

6. Performance Metrics

In our simulations, we use the following metrics to evaluate the performance of WiMAX network in terms of end-to-end QoS for VoIP.

6.1 Mean Opinion Score (MOS)

MOS provides a numerical measure of the quality of human speech in voice telecommunications, with value ranging from 1 to 5; where 1 is the worst quality and 5 is the best quality. MOS calculated using a non-linear mapping from R-factor as in [17]:

$$MOS = 1 + 0.035 \times R + 7 \times 10^{-6}[R(R-60)(100-R)] \quad (1)$$

where $R = 100 - I_s - I_e - I_d + A$; Is: the effect of impairments that occur with the voice signal; I_e: the impairments caused by different types of losses occurred due to codec's and network, and Id: represents the impairment caused by delay particularly mouth-to-ear delay. Using the default setting for Is and A Eq. (1) can be:

$$R = 94.2 - I_e - I_d$$

6.2 Packet End-to-End Delay (D_{e2e})

The total voice packet delay; De2e calculated using the formula:

$$D_{e2e} = D_n + D_e + D_d + D_c + D_{de} \quad (2)$$

where D_n, D_e, D_d, D_c and D_{de} represent the network, encoding, decoding, compression and decompression delay, respectively.

6.3 Jitter

Jitter defined as the signed maximum difference in one-way delay of the packets overa particular time interval. Let t(i) and t'(i) be the timetransmitted at the transmitter and the time received at thereceiver, respectively.

$$Jitter = \max_{1 \le i \le n} \{[t'(n) - t'(n-1)] - [t(n) - t(n-1)]\}$$

6.4 Packet Delay Variation (PDV)

PDV is the variance of the packet delay:

$$PDV = \frac{\sum_{i=1}^{n}([t'(n) - t(n)] - \mu)^2}{n}$$

where; µ is the average delay of n selected packets.

7. Results and Discussion

This implementation was performed using OPNET 14.5 on an Intel Core I 7; 1.7 GHz/2MB Cache processor sing Windows-7 64-bit operating system. In our OPNET modeler implementation of VOIP over WiMAX, there are some parameters specified as IEEE802.16 parameters and some specified as Voice parameter's with different codecs listed in Table.2 for different scenarios. The simulation duration is 15 min.

7.1 MOS

Figures1and 2 plots the average MOS for different scenarios Variables, which lists in table (2). A major observation is that, the average MOS of the codecs G.723 is the highest one, it is equal 2.5 but the average MOS of others codecs equal 1, so G.723 is better than other codecs

7.2 Packet end-to-end delay

Packet end-to-end delay is one of the most important performances. Metric in VoIP, Figures 3and 4 show the average packet end-to-end delay, we see that the average end-to-end delay in codec G.723 equal 0.16 sec but in others codecs it varying from 30 to 37 sec.

7.3 Jitter

According to equation (3), the jitter value can be negative which means that the time difference between the packets at the destination is less than that at the source. Figures 5and 6 the jitter

7.4 Packet delay variation (PDV)

Packet delay variation plays a crucial role in the network performance degradation and affects the user-perceptual quality. Higher packet delay variation; results in congestion of the packets, which can results in the network overhead. Figures 7and8 plot packet delay variation for different scenarios.

8. Conclusions

Next generation networks with multiple technologies offer different multimedia services to the user. It also provides the luxury of utilizing the best available technology for the required service to a user, companies and business organizations .In this study; we have conducted extensive simulation study to evaluate the performance of WiMAX for supporting VoIP traffic. We have analyzed several important critical parameters such as MOS, end-to-end delay, jitter and packet delay variation. Simulation results show that the G. 723 is better than codecs G. 711, G. 726, G. 728 and G.729 because it has lower delay and higher MOS, traffic received and throughput. Future work includes the combination between more than one codec to achieve a VOIP application using different networks.

Table.2 The VOIP codecs parameters

Scenario No	Codecs type
1	G.711
2	G.723.1
3	G.723.1
4	G.726
5	G.726
6	G.726
7	G.726
8	G.728
9	G.729

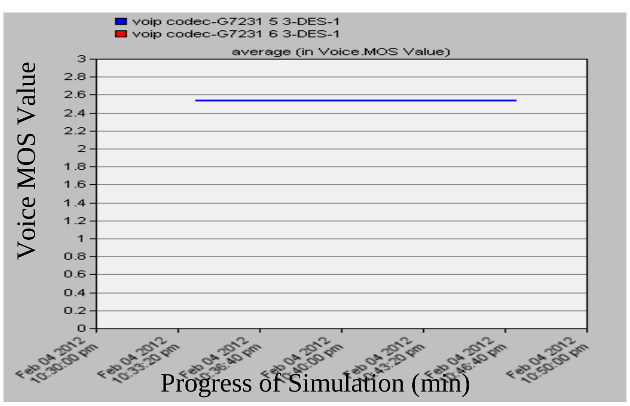

Figure 1 (Average MOS G.723)

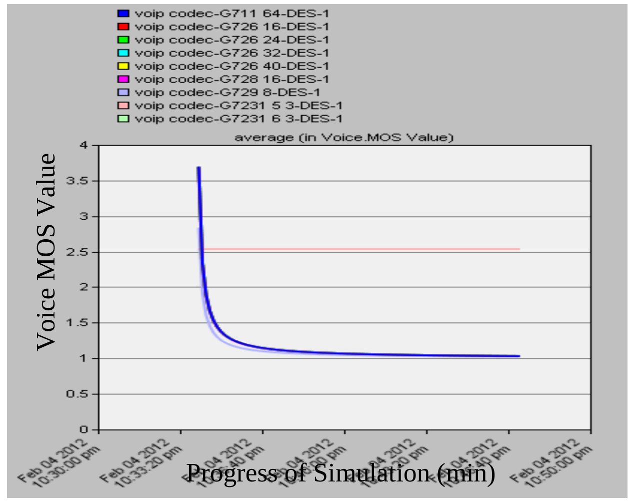

Figure 2 (Average MOS all codecs)

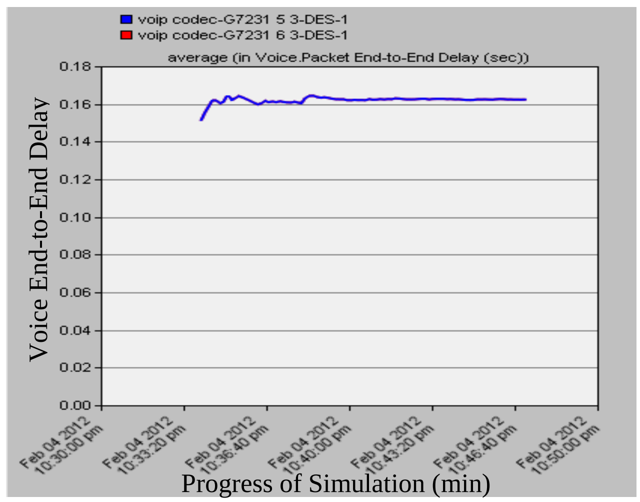

Figure 3 (Average MOS all codecs)

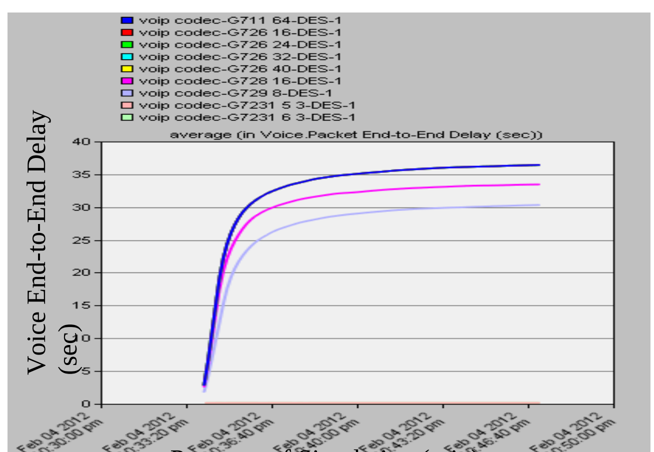

Figure 4 (Average End-to-end delay all codecs)

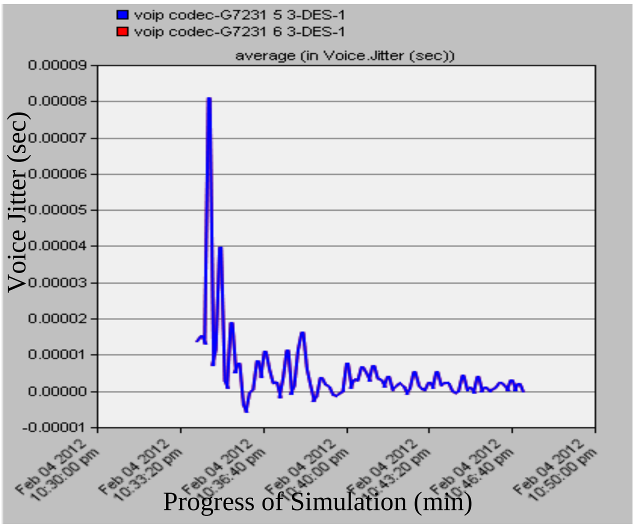

Figure 5 (Average jitter G723)

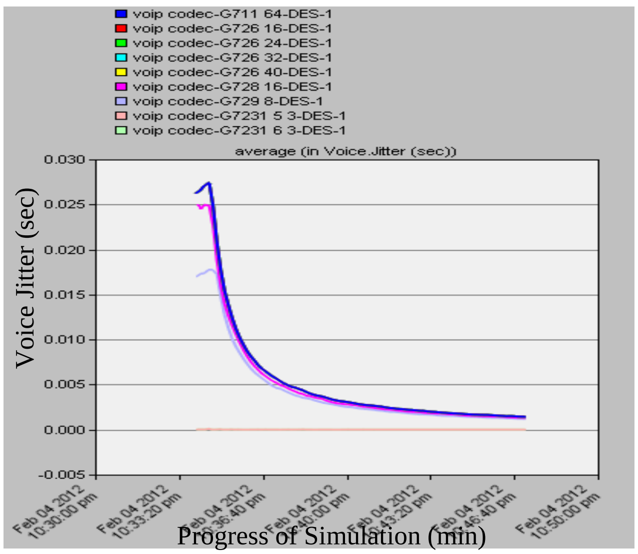

Figure 6 (Average jitter all codecs)

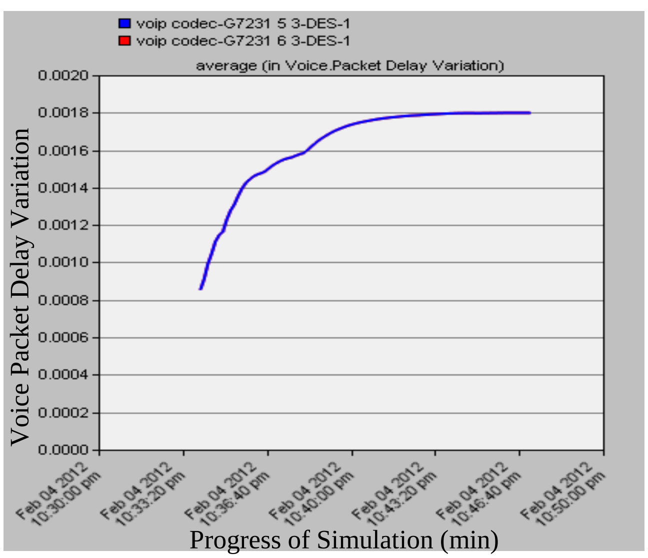

Figure 7 (Average PDV G723)

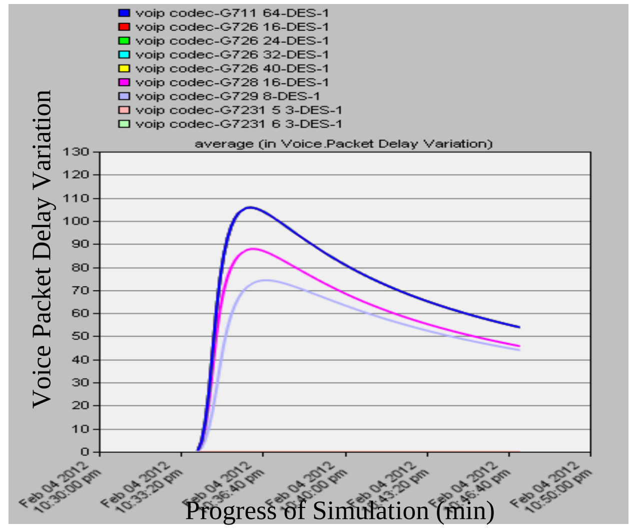

Figure 8 (Average PDV all codecs)

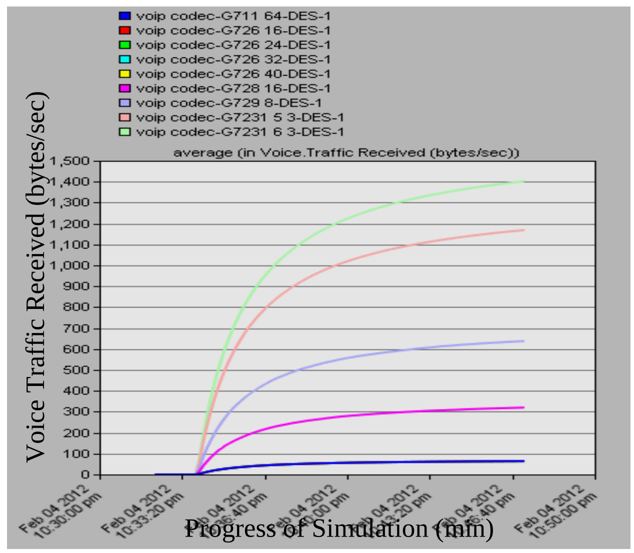

Figure 9 (Average Traffic received all codecs)

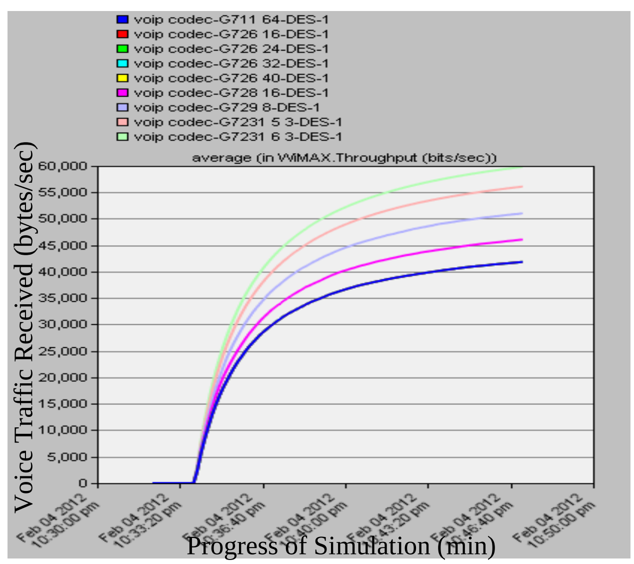

Figure 10 (Average Throughput all codecs)

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