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Performance Analysis of various Codecs Schemes of VOIP over WiMAX

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***Abstract*— Real-time services i.e. VoIP are becoming popular and are high profit earners for network service suppliers. These services no longer limited to the wired domain and available across wireless networks. Although some of the available wireless technologies can provide support to some low-band width applications, the bandwidth needs of several multimedia applications increase the capability of these technologies. The WiMAX predicts to be one of the wireless access techniques able of supporting very high bandwidth applications. However, still there are various challenges that require to be covered to offer a slow and good quality voice link over the best-attempt WiMAX network. In This paper, we measure the performance of several VoIP codecs over the WiMAX network. The network performance metrics i.e. packet end-to-end delay, MOS, packet delay variation and jitter have been employed to measure the performance of VoIP codecs. The results indicated that, the codec G.723 offers the best results among all discussed Codecs in all introduced performance metrics; the highest MOS, the lowest delay 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).

I. INTRODUCTION

Recently, user hopes have shifted from email and net browsing to multimedia services i.e. video conferencing & VoIP and video streaming etc. To cover the particular user requirements for rich multimedia applications, the service suppliers are seeing for broadband wireless network. The IEEE 802.16 standard [2] [3] has been planned as an access network to satisfy the user requirements of multimedia applications. WMAN offers cost effecient infrastructure to service suppliers and predicted QoS to end users without enhancing the complexity in the core network as well as at the user hand WiMAX is easy to combine and deploy with the available IP core network, which behaves as a backbone infrastructure. The IP core provides the support of advanced protocols and techniques to WiMAX that satisfy the needed security features [4] and Quality of Service (QoS).

In recent time, Voice over Internet Protocol (VoIP) has withdrawn the attention of the network engineering research and operation communities. The phenomenal development of VoIP is the result of quick network improvements and commercial solutions. Other factors involve the ongoing reduction in quality differences among available Public Switched Telephone Networks (PSTN) telephony and VoIP [5] and the increase in bandwidth existing to residential and commercial customers over which VoIP may be transmitted [6]. The aim of this study was to analyze a case of QoS deployment over a WiMAX network and to analyze the ability of a WiMAX network to provide sufficient QoS to voice application. The techniques considered include generating the WiMAX network employing OPNET, packet end-to end delay, jitter, carried out wide simulations to examine the MOS, traffic obtained, throughput and packet delay variation, deploying the VOIP application, setting the variables of VOIP within various codecs. This topic is significant to manufacturers and researchers in offering them the essential background for their works.

II. SIMULATION ENVIRONMENT

Table 1: Simulation Parameters for WiMAX

Cell Radius	30km
No. of Base Stations	7
No. of Subscriber Stations per BS	10
Speed of the mobile nodes	50, 100, 150 m/s
Simulation time	600 sec
Base Station Model	wimax_bs_ethernet4_slip4_router
Subscriber Station Model	wimax_ss_wkstn
IP Backbone Model	ip32_cloud
Voice Server Model	ppp_server
Link Model (ASN - Backbone)	PPP_SONET_OC12
Physical Layer Model	OFDMA 20Mhz

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MAC Protocol	IEEE 802.16e
Multipath Channel Model	ITU Vehicular A
Traffic Type of Service	Interactive Voice and Data
Scheduling Type	ertPS, nrtPS
Application	IP telephony
Voice Codec (with and without silence suppression)	G 711, G.729, G.723, G.726,G.728
Inter repetition time	Constant 200

A Voice Codec employed at the user side to convert the analogue voice waves into digital pulses and vice versa. There are various codecs types based on the chosen data rate, sampling rate described in table.2.

Table 2: Various Codecs

Codec	Coding rate
G.711	64
G.722	48
G.723	5.3 AND 6.4
G.726	16, 24
G.727	16 TO 40
G.728	16
G.729	8

III. Environmental Set of VoIP over WIMAX

To examine VoIP in a network, it is essential to study real life scenarios. Thus, OPNET 16.0.A is selected as the simulation tool as it will mirror the real deployment of the WiMAX network.

3.1 Scenario 1 – Static Nodes

Figure 3.1 depicts the simulation setup utilized for WiMAX network. Employing the OPNET Wireless Deployment Wizard a 7 cells WiMAX network, with several user stations in the base station range is deployed. The base station is linked to the core network by a server backbone through an IP backbone.

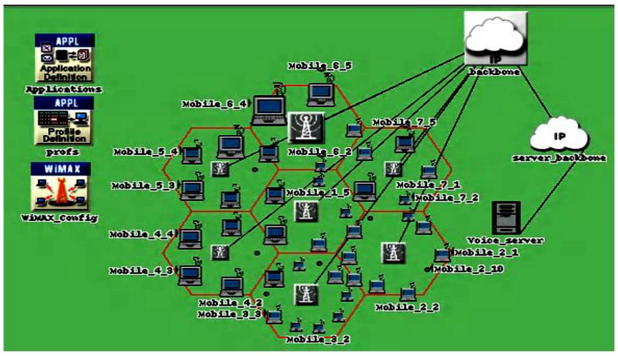

Figure 3.1 Network Model for WiMAX in OPNET

The server backbone is further linked to the voice server which is set up as the SIP server. The base station, Server Backbone, IP Backbone, and the Voice Server together introduce the service supplier company network. The cell radius is adjusted to be 10 km. The Base Station transmission power is adjusted 100 W and the same for subscriber station is adjusted to 0.100 W. The number of users in cells 2 and 4 are 10 and Voice over IP calls are established between them in mesh employing SIP server. With the simulation setup as specified before, the voice codecs being utilized for the Voice over IP calls are changed and the corresponding changes in MOS (Mean Optimal Score), voice jitter, and Packet End-to-end delay are observed.

A. Average Jitter

Figure 5.2 depicts the change in jitter for the WiMAX network without utilizing silence suppression for several codecs. Observed voice quality is best if the jitter is zero. As depicted in the figure, mean voice jitter is nearly 0 for the voice codecs G 723.1 with both 5.3Kbps and 6.3Kbps and G 726 with 32Kbps meaning very good quality of voice while all other codecs indicates some deviation.

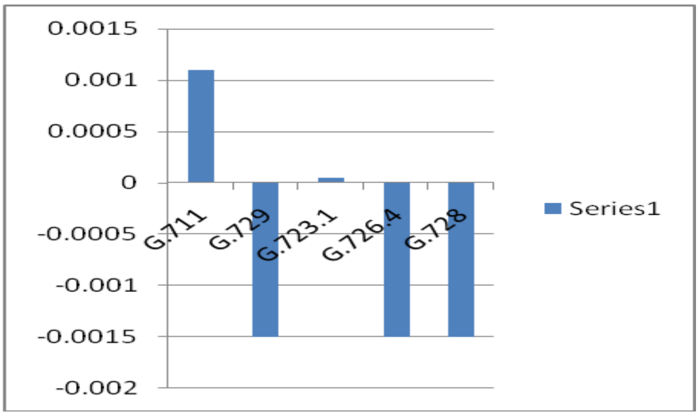

Figure 3.2 Average voice jitter without Silence Suppression

A positive jitter of 0.000000926827 seconds is displayed by G 711 while all others display negative jitter of about 0.000001841621 seconds. Since, the bit rate of G 723.1 is 6.4 or 5.4 Kbps; it results in production of small packets. As described in [15] modem and fax signals cannot be conveyed by G 723.1 and thus it can be utilized only for narrow band communications. Same as G 711, G 726 has its roots in the PSTN network. Thus, it is required to offer users with good quality of voice. It is mainly used for international trunks to preserve bandwidth. Dissimilar to G 711, G 726 utilizes 32Kbps to offer approx. the same quality of voice. This is because 32 Kbps is its real standard [34]. To increase the number of users supported count, silence suppression mechanism is significant. Silence suppression keeps the packetization of the silence length and thus preserve bandwidth. But utilization of silence suppression technique enhances the positive jitter substantially compared to the other voice codecs. This is described in Fig 3.3. Thus, G. 726 cannot be employed in cases where silence suppression mechanism is utilized regardless of its performance in cases where silence suppression has not been employed.

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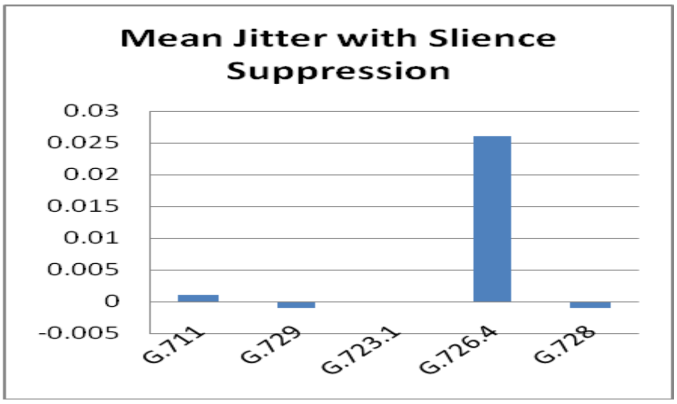

Figure 3.3 Average voice jitter using Silence Suppression

A. Average Packet End to End Delay

Packet networks works on packet switching principles; thus voice in an WiMAX or IP network would be transported to the destination node as a set of packets where every one might adopt various routes, thus reach at the destination with various delays. Factors influencing packet end to end delay involve Packetization Delay, Look Ahead Delay, Network Delay, Serialization Delay, etc. Figure 3.4 and 3.5 indicates, the Packet End-to-End delay for the voice codec G 723.1 is the highest regardless of silence suppression. This is due; G 723.1 utilizes coding rate of 6.3 Kbps or 5.3 Kbps which results in the constructing of packets of larger count and smaller size. Now with increasing number of packets in the network, the congestion in the network increases. Congestion directly influences the network packet delay and thereby results in increased packet end to end delay.

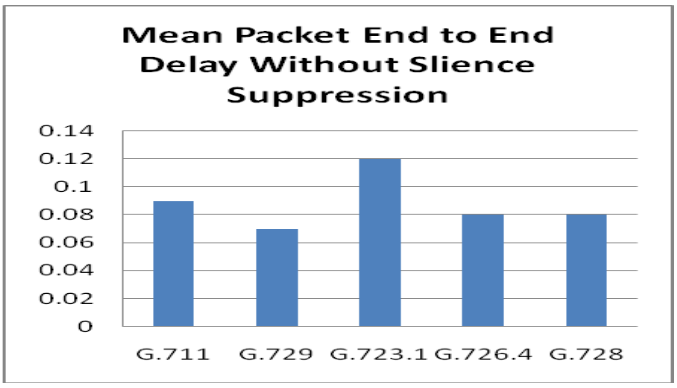

Figure 3.4 Average packet end to end delay without Silence Suppression

Also the Look Ahead delay of G 723.1 voice codec is 7.5 msec [35] whereas the same for G 729A is 5 msec and other assumed voice codec's is 0 msec. On the other hand the serialization delay of G 726 32Kbps is rather high.

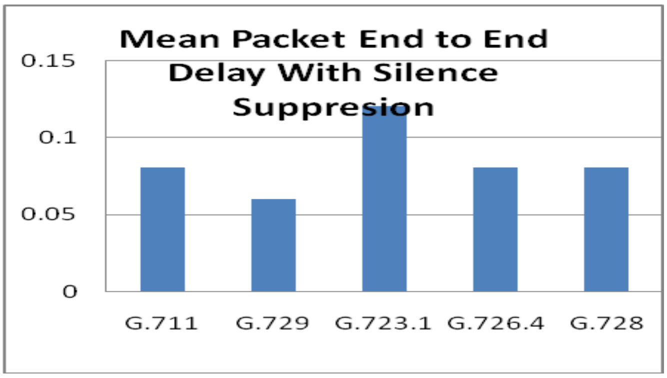

Figure 3.5 Average packet end to end delay using Silence Suppression

The serialization delay is inversely proportional to the link speed and is directly proportional to the payload size. Since the test network has same link topology, the link speed is same for all codecs. Thus, based on the payload size, the variation in the serialization delay influences the total packet end to end delay importantly.

B. Average MOS

The Mean Optimal Score (MOS) as displayed in Fig 3.6 and 3.7 is not dependent of Silence Suppression. MOS depends on number of packets lost. G 723.1 is a low bit rate codec which creates packets of size 6.3 Kbps or 5.3 Kbps. This results in network congestion and thus packet loss. Thus, the MOS value for the voice codec G 723.1 is rather low. Voice having MOS of 3 can be assumed to be of considerable quality. Thus, all other codec's considerable with respect to their MOS value.

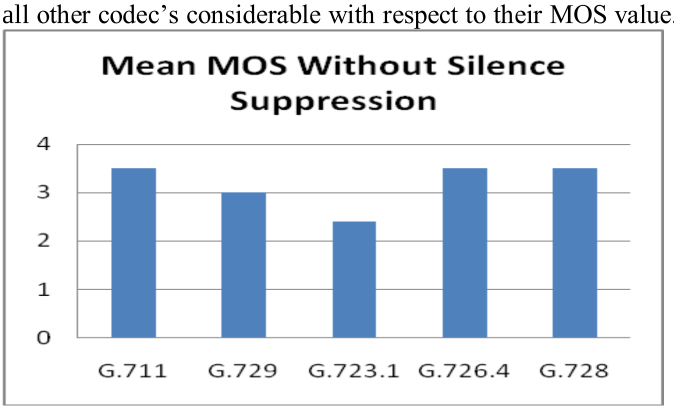

Figure 3.6 Average MOS without Silence Suppression

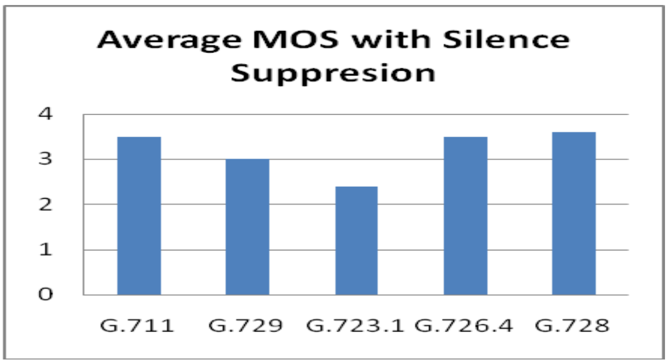

Figure 3.7 Average MOS with Silence Suppression

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3.2 Scenario 2 – Mobile Nodes

Fig 3.8 describes the simulation setup utilized for WiMAX network. Employing the Wireless Deployment Wizard of OPNET a 7 cells WiMAX network, with various subscriber stations in the base station range is deployed. The base station is linked to the core network by a server backbone through an IP backbone. The server backbone is further linked to the voice server which is set up as the SIP server. The base station, Server Backbone, IP Backbone, Voice Server together present the service supplier company network. Generic Routing Encapsulation (GRE) tunnel is configured between the base stations and the ASN gateway to handle the mobility of the mobile nodes. The cell radius is adjusted to be 30 km. The mobile nodes are set up to move at a rate of 50 km/hr, 100 km/hr and 150 km/hr. The Base Station transmission power is adjusted to 0.100 W and the same for subscriber station is adjusted to 0.5W based on [35]. Voice over IP calls are configured between them utilizing SIP and are displayed by the blue lines in the Fig 3.7. With the simulation configuration as specified before, the voice codecs being utilized for the Voice over IP calls are changed and the corresponding variation in Packet End-to-end delay and voice jitter are observed. Because of the restrictions of OPNET, Mean Optimal Score (MOS) could not be calculated. Also only some of the voice codecs could be simulated. G 723.1 with 5.3 kbps, G 726 with 32 Kbps and G 728 with 16 Kbps could only simulated.

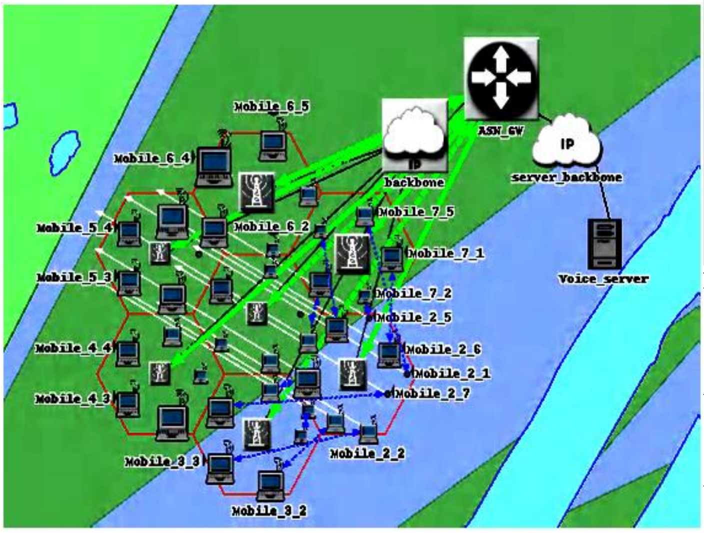

Figure 3.8 Network Model for WiMAX

Average Jitter

G 711 codec has packet rate of 64 Kbps which is quite large in comparison of G 723.1 which is nearly 5.3 Kbps. Therefore, G711 has fewer number of packets in comparison of G 723.1 for fixed amount of voice. For larger packets the overhead because of header is small [38].

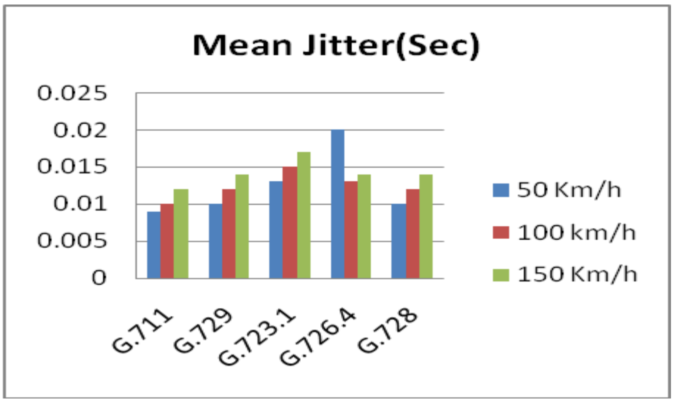

Figure 3.9 Mean Jitter without Silence Suppression

As a result, for scenario 1, the delay for G 711 is lesser in comparison of G 723.1. furthermore, in scenario 1, the nodes are static thus the packets trace almost the same path and reach in order more or less but wireless networks, have a underlying property of neglecting any packets containing one or more wrong bit.

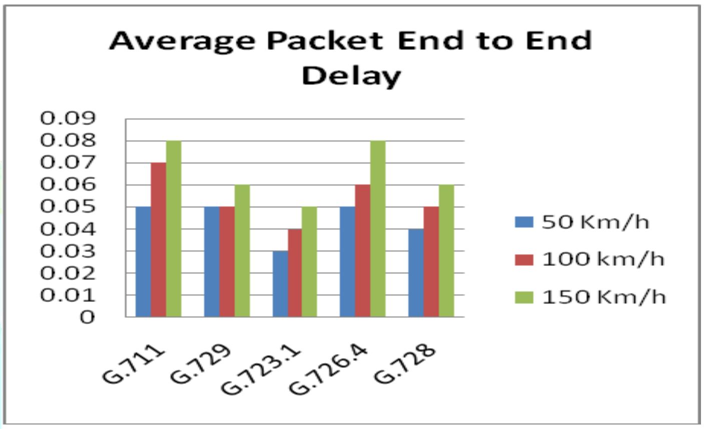

Figure 3.10 Mean Packet end to end delay without Silence Suppression

So the possibility of discarding G723.1 packets are less and the jitter is also least for this codec. Still in the mobile scenario, scenario 2, the possibility of packet drop is more and any packet drop in the physical layer is translated as delay by the upper layers because of the concealing property of the link or MAC layer protocols [38]. Since, the packet size of G 711 is larger; it has higher possibility of suffering from packet drop as compared to G 723.1 thus having higher delay in comparison of G 723.1. This is displayed in Figure 3.9. As the subscriber stations become mobile, their supporting base stations changes rapidly because of handover and the mobile nodes get disseminated between various base stations based upon their trajectory. Thus, the packets may trace different path and reach out-of-order. Since, the number of packets for G 723.1 is much more as compared to G 711, this problem is more dangerous as compared to G 711. As a result it suffers from highest jitter.

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CONCLUSION

The primary objective of this work is to noted the variation of packet end to end delay and jitter in a WiMAX network under several mobility scenarios. This work indicates the variation of the specified parameters with respect to a static network having all fixed nodes and with respect to a mobile network with changing speed of 50km/hr, 100km/hr and 150km/hr. Mean packet end to end delay is more in case of fixed nodes as compared to mobile nodes. Figures display the variation of jitter in the above specified networks. As Figure 3.3 displays, the jitter in a fixed network increases considerably when silence suppression is enabled in the network and it increases further when mobility is presented in the network. This is because as a node becomes mobile it undergoes handover. This generates inconsistency in the order of the packets delivered thus jitter. Figures 3.4 and 3.5 displays the variation of packet end to end delay in the above specified WiMAX network. it is realized that the packet end to end delay reduces in the entire network when silence suppression is enabled. This is because of the decrement in the queuing delay. As the number of packets reduces with silence suppression, the number of packets queued in the intermediary nodes reduces. This decreases the queuing delay. The packet end to end delay reduces considerably when mobility is presented in the network. With mobility, the mobile nodes get scattered under various base stations and this eliminates the bottle neck of one base station in case of static nodes and thus reduces the delay.

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