FINAL REPORT:-Investigations on VoIP for a WiMAX Communication Network

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Abstract: **Voice over internet protocol refers to the transmission of telephone traffic over IP-based networks. Voice transmission over internet is also called internet protocol (IP) telephony. IP telephony has gained wide popularity and has become an important services on the internet. The use of IP telephony became real due to the high bandwidth available on the internet and its low implementation cost. Digital form of communication has further led to its vital use. Due to above reasons, voice communications using the IP, is also called Voice over IP or VoIP, has become attractive. The simultaneous transport of voice and data over the internet has already been demonstrated using 3G networks. This paper presents the performance evaluation of a WiMAX network. In this paper only dynamic conditions for the application voice over IP has been considered.**

I. INTRODUCTION

As according to the need of modern telecommunication system Wireless networking has become an essential part the demand of high speed data transfer with high quality is being the leading factor for the evolution of technologies like WiMAX and WLAN and is still Increasing day by day. Therefore, new ways to improve quality and speed of connectivity are being searched for. Moving towards the fourth generation communication networks, integrated networks are coming into operation. In same manner voice over IP is expected to be a low cost communication medium. The voice codecs are big constraints which influence the quality of the voice in a high data rate communication network. Therefore, before real time deployment of VoIP over a network it is essential to evaluate the voice performance over altering networks for various codecs.

VoIP have been widely accepted for its cost effectiveness and easy implementation. A Voice over internet protocol (VoIP) system is divided into three indispensable components, namely 1) codec, 2) packetizer, and 3) playout buffer. Analog voice signals which are to be transmitted compressed , and encoded into digital voice streams by the help of codecs. The output digital voice streams are then packed into constant-bit-rate (CBR) voice packets with the help of the packetizer. A two way conversation is very sensitive to packet delay jitter but could tolerate certain degree of packet loss. Hence a playout buffer must be used at the receiver end to smooth the speech by eliminating the delay jitter. Quality of noise sensitive VoIP is generally measured in terms of jitter, MOS and packet end-to-end delay.

II. SIMULATIONN FINDINGS FOR WIMAX DYNAMIC SCENARIO

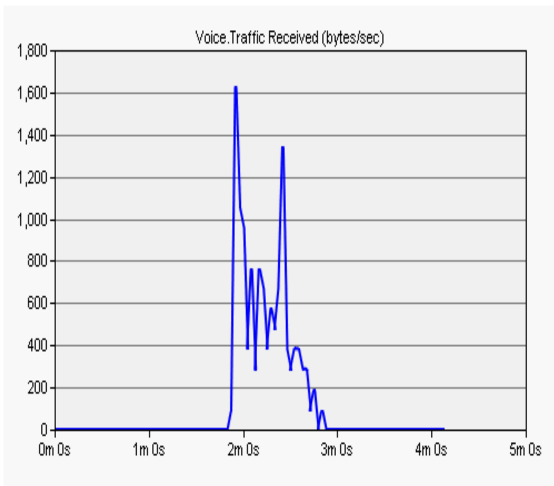

Figure 5.2 Voice Traffic Received in bytes/sec for rtPS Protocol

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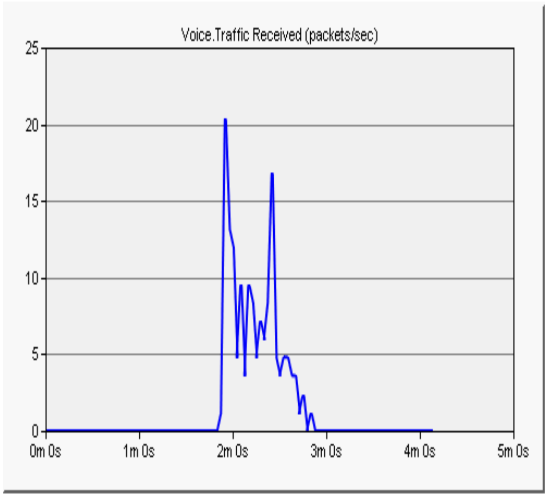

Figure 5.3 Voice Traffic Received in packets/sec for rtPS Protocol

Fig5.2 and 5.3 shows variation of voice received in bytes per second and packets per second fo rtps protocol respectively.from fig 5.2 and 5.3 it has been concluded that maximum value of voice traffic received is approx 1600 bytes per second and20 packet per second.

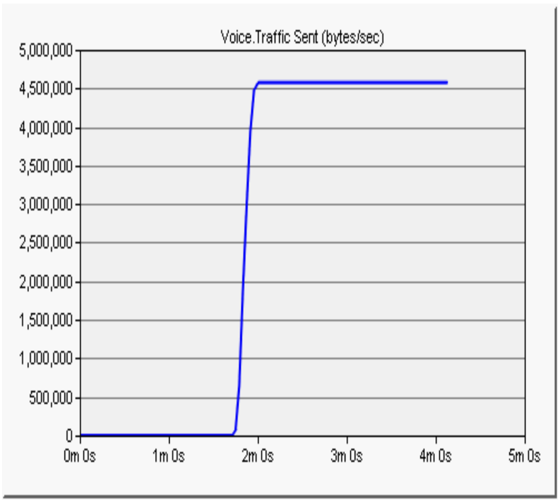

Figure 5.4 Voice Traffic Transmitted in bytes/sec for rtPS Protocol

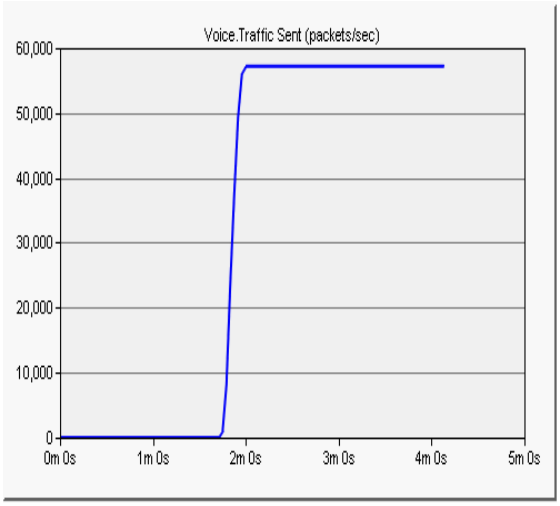

Figure 5.5 Voice Traffic Transmitted in packects/sec for rtPS Protocol

Fig 5.4 and 5.5 shows value of voice traffic sent in bytes /second and packets per second.From fig 5.4 and 5.5 it has been concluded that max value of voice transmited is approx 45000 byte per second and 58000 packet per second respectively.

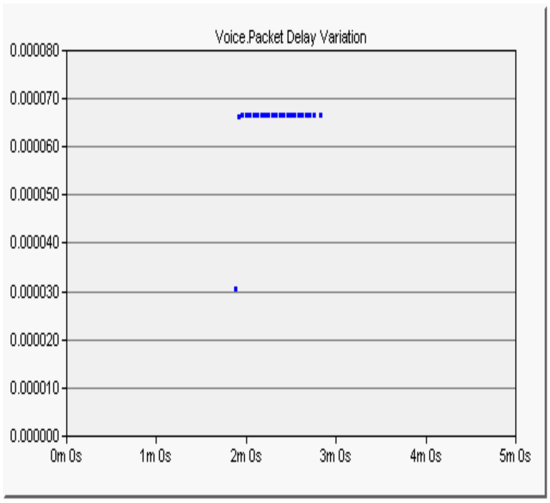

Figure 5.6 Variation of Voice Packet Delay for rtPS Protocol

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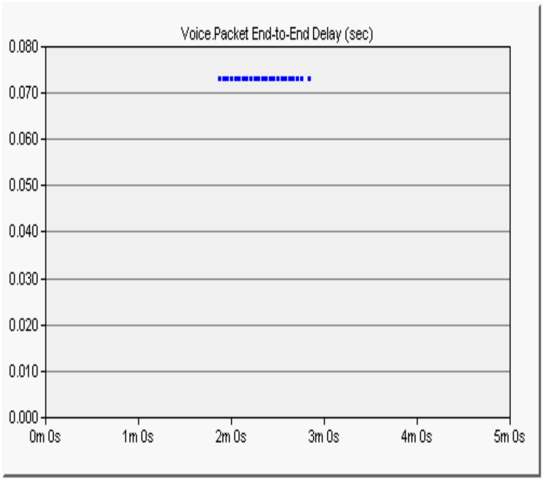

Figure 5.7 Variation of Voice Packet End-to-End Delay for rtPS Protocol

Fig 5.6 shows variation of voice packet delay for rtps protocol.the variation in voice packet delay is around 0.000068 seconds.also voice end to end packet delay is around 0.072 seconds as shown in fig 5.7.

III. SIMULATION FINDINGS FOR WIMAX STATIC SCENARIO

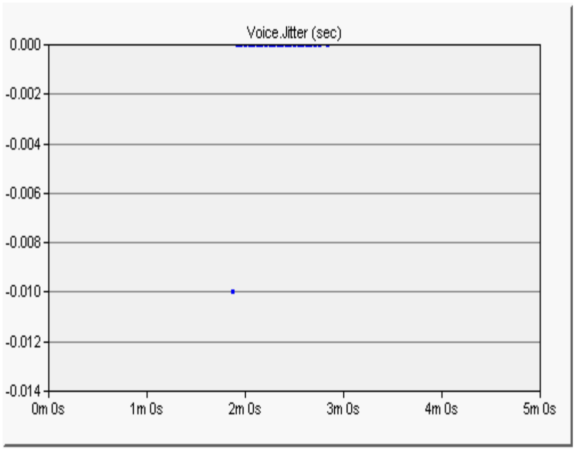

Figure 5.8 Variation of Voice Jitter for rtPS Protocol

Fig 5.8 shows value of voice jitter in seconds.fig shows that voice jitter remains almost zero during simulation session and maximum value of voice jitter has been noticed at -0.010 seconds for rtps protocol.

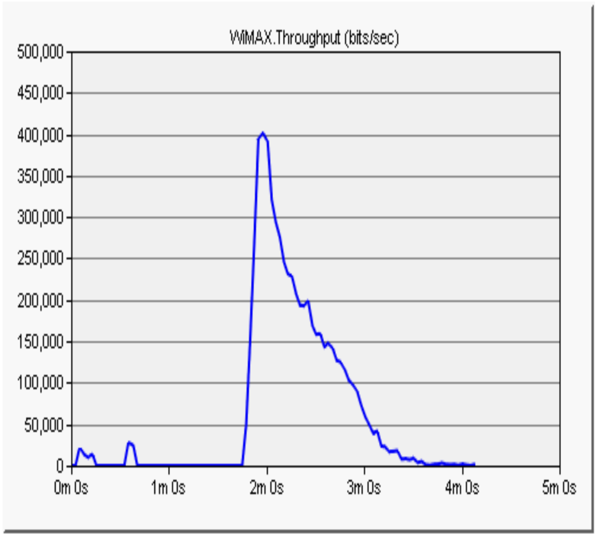

Figure 5.12 Variation of WiMAX Throughput in bits/sec for rtPS Protocol

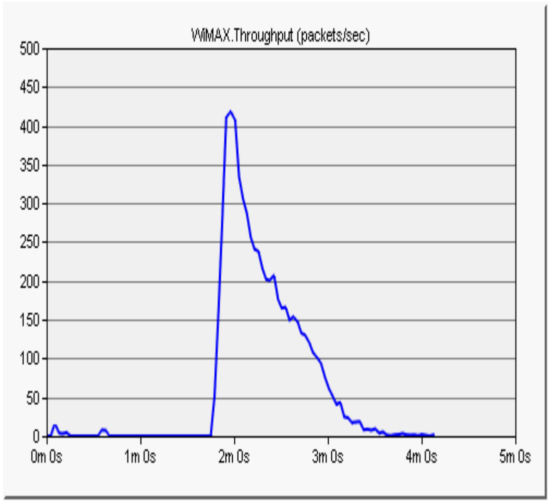

Figure 5.13 Variation of WiMAX Throughput in packets/sec for rtPS Protocol

Fig 5.12 and fig 5.13 shows variation in throughput in bts per second and packets per second. for fig 5.12 and 5.13 it has been conluded that maximum value of throughput comes out to be 4*10 5 bits per second and 420 packets per second respectively for rtps protocol.

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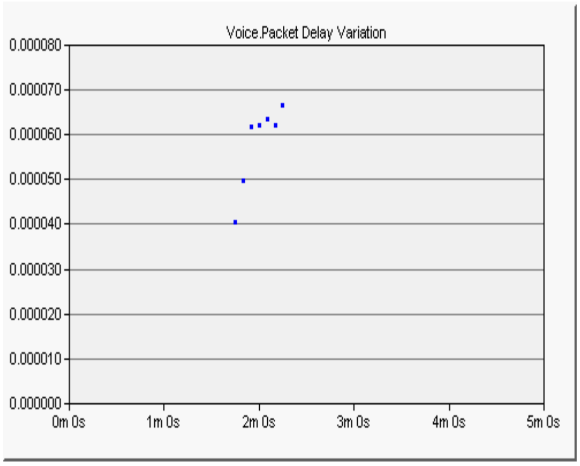

Figure 5.14 Variation of Voice Packet Delay for UGS Protocol

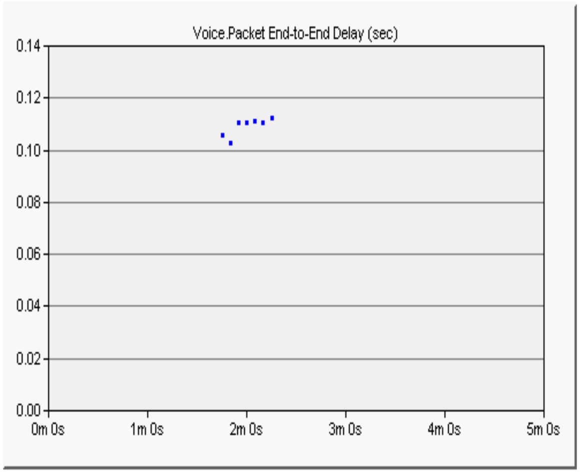

Figure 5.15 Variation of Voice Packet End-to-End Delay for UGS Protocol

Fig 5.14 shows variation in voice packet delay for ugs protocol.the voice packet delay is not constant as per in case of rtps protocol and the value of maximum value of packet delay variation comes out to be 0.000070.also value of end to end packet delay as shown in fig 5.15 is not uniform for ugs protocol and is maximum value is around 0.11 seconds.hence in terms of variation in packet delay and end to end delay rtps protocol sows better performance as compared to ugs protocol.

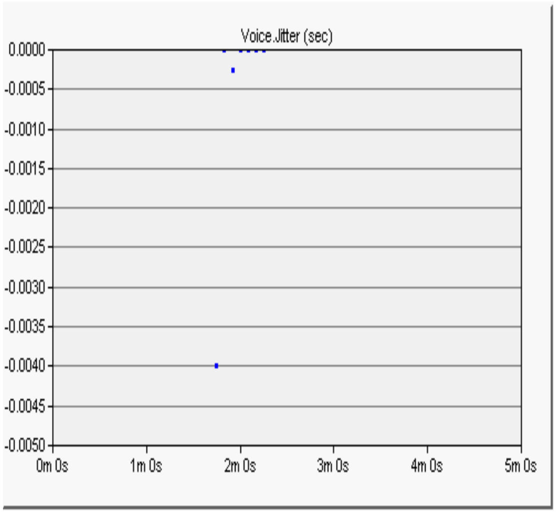

Figure 5.16 Variation of Voice Jitter for UGS Protocol

The maximum value of voice jitter has been calculated as -0.0040 seconds for ugs protocol as shown in fig 5.16.hence in terms of voice jitter ugs protocol shows better performance over rtps protocol.

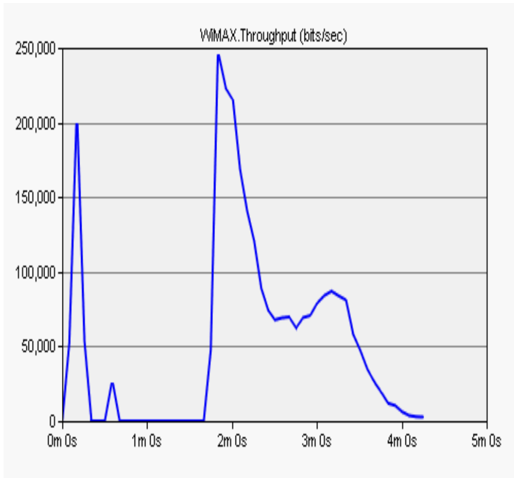

Figure 5.18 Variation of WiMAX Throughput in bits/sec for UGS Protocol

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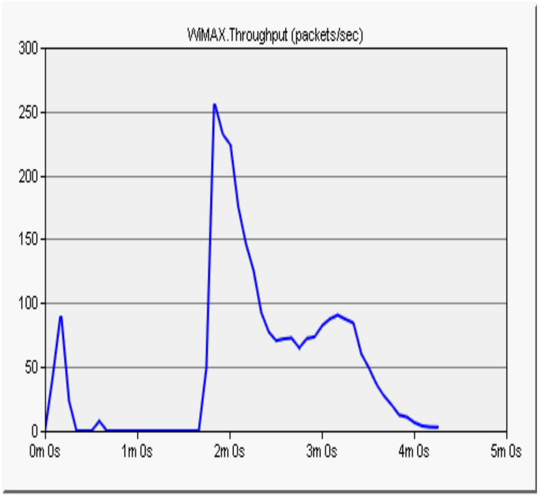

Figure 5.19 Variation of WiMAX Throughput in packets/sec for UGS Protocol

Fig 5.18 and 5.19 shows throughput in bits per second and packets per second for ugs protocol.AS compared to rtps protocol throughput of ugs protocol is very low and its maximum value is around 2.5 *10 5 bits per second and 250 packet per second. hence throughput enhance in case of ugs protocol and perform much better as compared to rtps protocol.

IV. CONCLUSION

In this thesis, a WiMAX static and mobile network has been developed for 19 base stations and each base station has 30 subscriber stations. WiMAX network has been configured for static and mobile environment and for voice transmission. The network performance has been compared for UGS and ertPS protocols using QoS parameters like Jitter, Throughput and end to end packet delay. Both the protocols show their superiority for different parameters and selection of protocol can be done on the basis of system requirements

From the results, it has been concluded that voice packet delay is almost same in case of static and mobile environment for ertPS protocol and mobile environment for UGS protocol. But for static environment voice packet delay is very high for UGS protocol. Hence from simulation results it has been conclude that UGS protocol is not suitable for static environment in terms of voice packet delay.

From simulation results it has also been concluded that in terms of voice jitter, UGS protocol performs better in static environment and ertPS protocol performs better in mobile environment.

Results also show that performance of UGS protocols is not suitable for both static and mobile networks in terms of WiMAX delay. Also ertPS protocol shows better performance in mobile WiMAX network as compared to static WiMAX network

Throughput analysis shows that performance of UGS protocol is better in both static and mobile WiMAX networks. Results also show that in terms of throughput ertPS protocol shows worst performance for static WiMAX network.

V. FUTURE WORK

In the present work the WiMAX network has been evaluated for UGS and ERTPS protocol. But future work can be concentrated for other protocols like BE and RTPS,. Only one CODEC G.711 with silence suppression is used but in future work performance for different codec and different modulation format can be compared for given protocol.

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