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Survey on VoIP transmission in 3GPP standards

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Abstract: This article, presents that how do voice calls deliver for the PSTN/Cellular quality VoIP services over WiMax networks. The good MOS (Mean Opioin score) value is obtained using the OPNET simulator. Video streaming service's provides end to end connectivity in IP based transmission. The main objective of the article is to investigate the voice quality parameters and WiMax network management services during a hierarchical simulation status. VoIP (voice over internet Protocol) is one of the user interfaced application layered protocol, which is used in the proposed work and voice call is established; Initiate; call connect among the number of mobile users. This model provides jitter and delay in voice calls; throughput and interact with mobile users; prevention of miscellaneous node; in peer to peer coverage area. The simulation test bench performances were appropriated by our WAN coverage area.

Keywords: *WiMAX, Streaming, firewall , VoIP, 3GPP*

1. Introduction

In the modern communications world, 4G (Fourth generation) mobile wireless standards' goal is to provide high-speed mobile IP based transmissions, which can support high-quality streaming video and other critical services. LTE and smart phones which are a growing technology is used to store the video and watch the online video. It is one of the application layered IP based technology to carry the voice data throughout the network via the wired/wireless medium. This article, introducing the integrated VoIP over WiMax architecture for broadband mobile user and maintaining the high broadband capability. The goal of the work, outlines the VoIP development scenarios; technical challenges in the design of networks and also Wimax network can offer a comparable solution as PSTN (Public Switch Telephone Networks) voice quality services. In general multimedia voice services have different varieties like, IMS (Internet Networking Multimedia services) based VoIP over networks, Non-IMS based VoIP over networks, design and consideration of Qos and bandwidth based estimation (delay, jitter, MOS, network load, packet loss). The quality of voice calls is measured in terms of mean opinion score value. Many people are made to hear the voice signalling and made to rate the various factor distortions, delay, jitter, noise and MOS. The measurement of MOS values represents the scale factor between 1 to 5. The following standard of MOS values are, below three MOS factor which indicates worst voice clarity, above three

MOS factor which indicates best voice clarity. It is equivalent to the Mobile/PSTN standard value. The voice packets are encoded by G.711 encoder. It has MOS as 4 at 64kbps. The infrastructure less networks commonly known as radio frequency environment in dynamic mobility. VoIP drawbacks of issue state, voice clarity and connectivity in entire the wide area network.

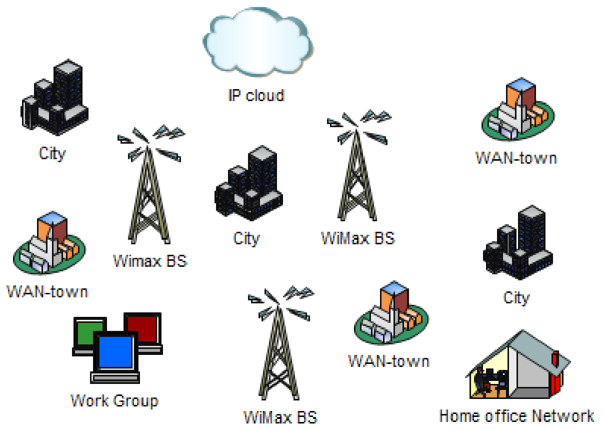

Fig 1: WiMax Integrating VoIP over service in WAN

Figure 1 illustrates VoIP based wide area network utilization of environmental with WiMax architecture. IP cloud is one of the upcoming routers used for VoIP services. It provides unpredictable pass of data traffic on WiMax networks and exchange the data packets from one mobile terminal with

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others. It interconnects the number of cities with WAN environmental area.

VoIP -Quality of service issues & challenges

The main objective of QoS is to maintain the end to end connectivity. Voice packets are connected to the destination a port which reduces the jitter, latency, packet loss and mean opinion value that is environmentally equivalent to cellular coverage area networks. In this article, the above drawbacks are overcome rectified by OPNET scenario.

A. Latency

Latency is one of the end to end delay in network environmental area. The voice packets are transmitted between the one device to another. It consists of the three primary factors: network delay, play out delay and CODEC delay. The network delay referred as voice packets transverse from end to end point. It is commonly affected by network congestion, routing challenges, geographical specific end to end points. In the play out delay is caused by voice packets coming up in the jitter buffer. At the receiver end of digital stream packets, CODEC decoded packets from digital back to analog. However, the internet and network traffic arrival rate in general is not constant. Once the voice packet may take 100 ms to arrive and the next voice packet may take 120 ms. This variance is known as jitter. The jitter is no more than simulated delay built into the CODEC in order to compensate for the possible delay in packets. However, an increase in the size of the jitter buffer also increases the end to end delay. CODEC delay is introduced by encoding and decoding scheme process. CODEC algorithm used to digitize analog signal which introduces the certain the CODEC delay. In such a delay depends on hardware/software and CODEC in use.

B. Packet loss

VoIP over IP network requires the voice conservation packetized. Packet loss occurred more than one packet during transmission loss. Packet loss also occurred due to network congestion or connectivity error between end points. Wimax forum application recommended packet loss in VoIP service more than 1%.

C. Jitter

Voice packets travel from the source to destination via different routers over the networks, voice packets may arrive out of duration or variation of latency between devices. In order to maintain a high QoS for the end-user, such a delay variation has been implemented in different VoIP test benches. WiMax VoIP forum recommended that the standard in between 20 ms to 50 ms is preferred.

D. WiMax Qos challenges

VoIP service not only depends on the quality of speech but also on the network quality management services. In Wimax subscribes station support Qos scheduling services that are optimized for VoIP services are unsolicited grand services (UGS), Extended real time service (ertPS) and real time services (rtPS). Wimax integration of the IP based VoIP services, Virtual Local area networks, metropolitan area networks… etc. Such a core network provides flexible communication services. The CODEC scheme is to determine the required bandwidth per call. G.711 used for a wide range of application and Its sampling rate is 8kHz and each sample is encoded with 8 bits constantly and bit rate is 64 kbps. It provides good voice quality. Finally, achieved the quality of metric in network management service delay, jitter, packet loss, throughput and network loaded condition.

2. Related Works

Chien-Ming Chou et al [1] proposed quality of metric in mobile adhoc networks. Vehicle to vehicle (V2V) and vehicle to infrastructure (V2I) are a different mobility mode of communication in the vehicular network. The above authors are investigated static environment. And also WiMax coverage area and latency provides larger than WiFi. Mohamed. A. Mohamed et al [2] investigated low bandwidth over wireless technologies. The authors focused the wireless access technologies skilled high bandwidth features. We analysed QoS on the long distance data communication between two locations under VoIP over WiMax. Iwan Adhicandra et al examined [3] five different data delivery service classes that can be used in order to obtain the quality of services (QoS) requirements of different VoIP applications, such as FTP (File transfer protocol), Web access, video conferences, …. Etc. And also investigated two different transport layer VoIP traffic, i.e ertPS or UGS. Rakesh Kumar Jha et al [4] proposed location based WiMax network for IKE (Internet Key Exchange) under in terms of traffic security with the help of gateway security (GSE). We investigated the performance of IKE attack in packet CS and ATM CS networks. The author has investigated the securities sublayer exchange between the MAC layer and PHY layer. Charles Shen et al investigated [5] TLS (transport layer security) for SIP (session initialized protocol) servers with evaluating the cost of the TLS experimentally using a test bed in open SSL, and Linux running on Intel based server. K. Salah et al proposed [6] two type of traffic (via fixed and empirical video packet sizes) and provide realistic simulation of a real-life network environment.Ravi Shankar Ramakrishnan et al examined [7] variety type of voice,data and video integrate onto a single IP. Which reduces cost and increasing mobility functionality. And analysed VoIP packet loss, jitter and delay in single IP scenario. Mohd Nazri Ismail et al proposed [8] (VoIP) service in the campus environment network. The authors investigated for quality of voice prediction such as i) accuracy of MOS between automated system and human perception ii) different types of CODEC performance measurement via human perception using MOS technique.

Finally, reliability and accessibility performance compared to WAN. Z. Bojovic et al [9] investigated VoIP quality performance measured by SIP traffic. Comparative performance analysis of G.723, G.729 and G.711 CODEC finally, the authors concluded by G.711 CODEC that provides better voice quality with no CODEC delay in transmission over VoIP networks. S. Alshomran et al investigated [10] quality of voice such as delay, jitter, packet loss, MOS such a result indicate a significant impact on VoIP performance in the WiMAX networks.

3. Simulation Results

We use an OPNET simulator as our performance analysis platform with various investigations of voice quality parameters such as Mean Opinion Score, packet end to end delay (Sec), the traffic received (bits/Sec), traffic sent (bits/Sec). Similarly WiMAX evaluation parameters are, delay, jitter, network load, and throughput. The above to select parameters are investigated by 100 X 100 kilometres. The following devices are used in the specified coverage area: Cisco 7200 series routers, WiMAX Base station (BS) routers, fixed WiMAX nodes, Bay Switch Accelar, Nortel Firewall, Ethernet/WiMax supported server and cloud-IP (serial 32 line interfacing). The simulation parameters are summarized in table1. We design the video conferencing service in WiMAXarchitecture setup which contains three base stations with firewall protection to server. The authors investigated the Voice quality and network performance management in the WiMAX architecture in wide area network coverage area. In this scenario model describes upcoming new router, cloud IP, switches. The IP cloud is used for 32 line serial interfacing at selectable data rate through VoIP based on transmission. It is used for fixed time routing; arrival of packets delivery by queuing theory. Protection of voice packets by multi homed-server firewall node. It supports IP based gateway NT_contivity. It acts as a gateway, fixed amount of time to route each packet.

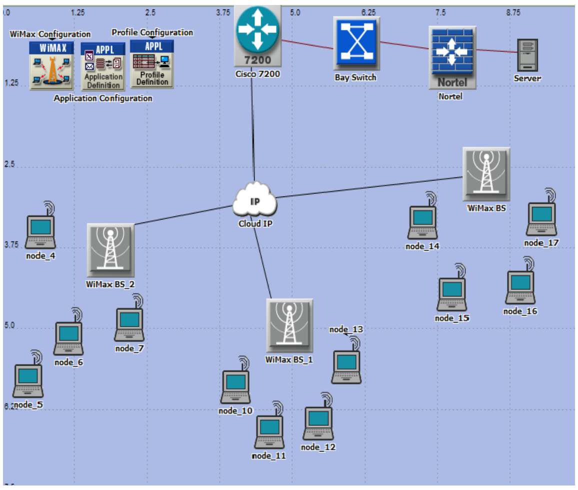

Fig 2 Snapshot of WiMAX for video conferencing via VoIP in OPNET simulator

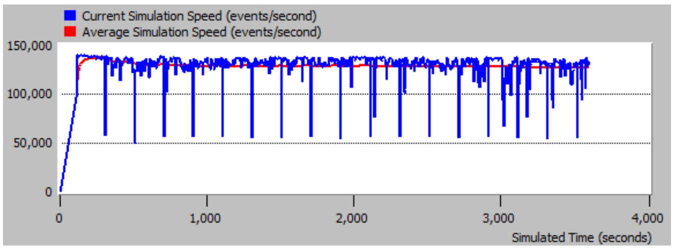

Fig 3 : comparison of the current/average simulation speed with simulation time

Figure 2 and 3 illustrates a snapshot for Wimax scenario and the speed of execution with simulation time. The Simulation time for this Scenario model is one hour with the usage of 1500 MB memory size.

*A. **Tabulation for simulation parameters***

Parameters	***Metric***
Simulation time	3600 Sec
Simulation memory usage	1500 MB
Router specification	Cisco 7206 CS_7206_6s_a2_ae8_f4_tr4_slip16
Wimax Router	wimax_bs_ethernet4_slip4_router
Wimax nodes	12 workstations
IP protocol	IP_32_Cloud (32 series)
Connecting components Switch Connecting link Security Server	Bay switch 1500 accelerator 1000 base T (Ethernet) Firewall (Nortel) Ethernet server
WiMax BS Antenna gain	15 dB I
Maximum Transmission Power	0.5 w
Physical profile	OFDMA (20 GHz)
Voice encoder	H.323 (G.711
Simulation area	100 x 100 Km

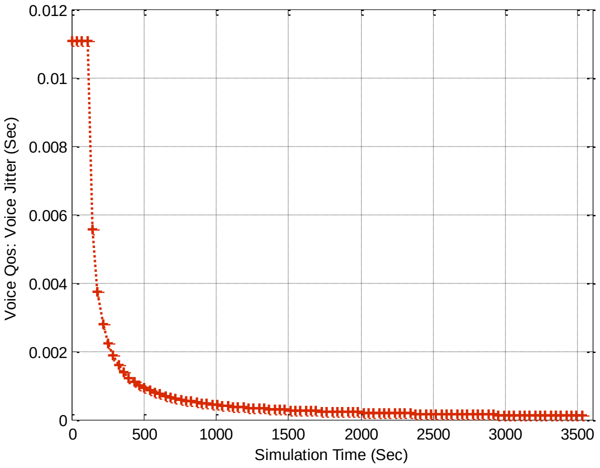

Fig 4 : Voice Qos : Voice quality of jitter corresponding simulation time using regressive analysis

Figure 4 illustrates the quality of voice jitter (average delay). The voice packets are transmitted in the wide area coverage network which reduces the jitter almost to zero with the help of Cisco router (7200) and bay network accelar switches (1500). The video packet variation of jitter caused by queuing, contention, serialization of packets throughout whole scenario.

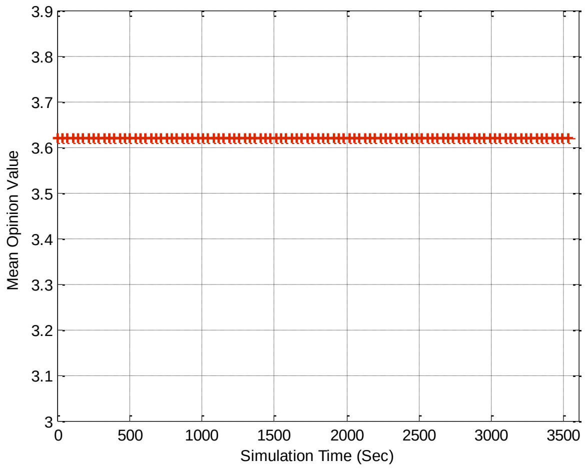

Fig 5 : Voice Qos : Voice quality of Average MOS value corresponding simulation time using regressive analysis

Figure 5 illustrates the mean opinion score. It represents the number of voice calls and its quality of clarity at the receive end. We obtain the average value is 3.675. It is represented by the annoying factor; applicable to mobile conversation standard.

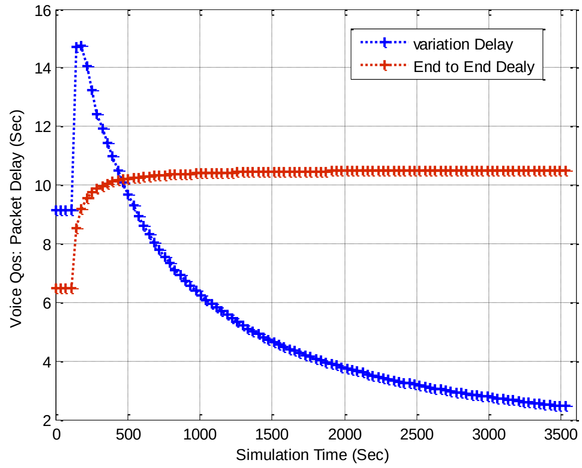

Fig 6: Voice Qos : Voice quality of packet delay corresponding simulation time using regressive analysis

Figure 6 describes the end to end delay factor and variation delay of the simulation model. The delay factor depends on processing delays and propagation delay. Packet variation delay always decreases the exponential factor. The end to end delay is obtained the constantly in WiMax simulation. Authors are getting the zero jitter and reduction of delay in the WAN network coverage area.

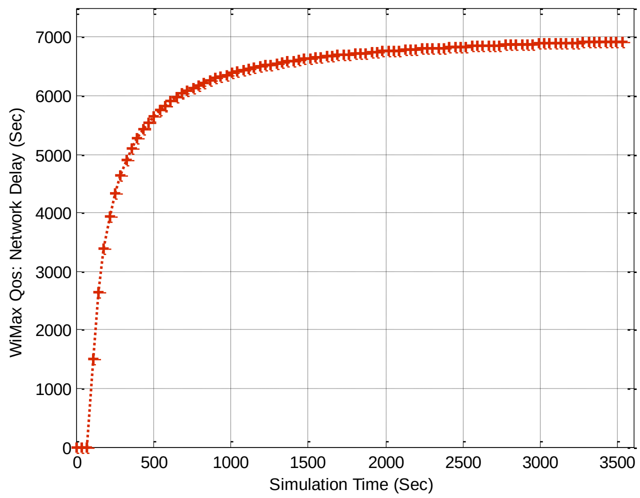

Fig 7: Voice Qos : WiMax quality of network delay corresponding simulation time using regressive analysis

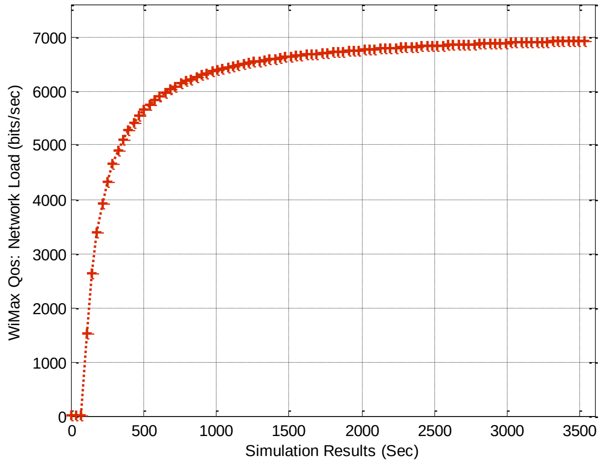

Fig 8: Voice Qos : WiMax quality of Network Loaded condition simulation time using regressive analysis

Figure 7 describes the design of Wimax Networks and its performance. The voice data transmitting from one end point to another endpoint. We reported maximum network delay for designing the network such a delay is provided growing and kept constant. Figure 8 shows the loaded network condition of WiMax model. In maximum number of users representing the WiMax coverage area in order to reduce delay and the load parameters kept the constant.

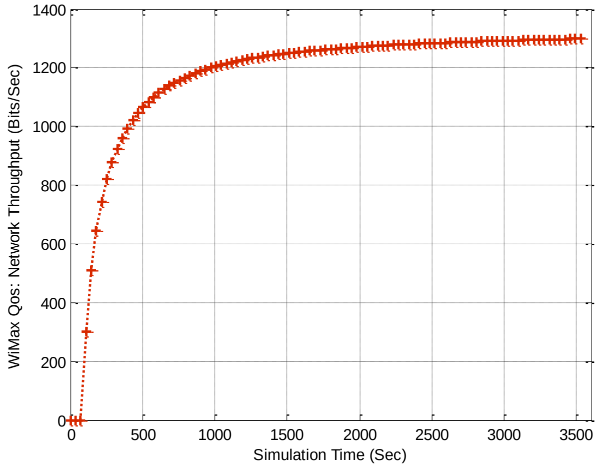

Fig 9: Voice Qos : WiMax quality of throughput simulation time using regressive analysis

Figure 9 describes the throughput in the Wimax scenario. In this article, main goal of scenario improved efficiency of WiMax quality management. Throughput quality parameter is considered for the received traffic packets, jitter, end to end delay and network loaded condition. Finally, raised throughput depends on receiving packets by using the regression function.

4. Conclusion

In this article, authors investigated the voice quality parameters and with integrating the network management. The OPNET simulation of VoIP transmission during 10ms in jitter is occurring zero. And same time, mean opinion value nearly four. It's a good voice clarity and equivalent to face to face conversation of PSTN/Mobile communication network. These experimental results were examined in the realtime measurement. In this future work, extended the Wimax integrating with UMTS, GPRS mobile networks for video conferencing, video online gaming and Streaming applications. And also we will investigate the vehicular mobile ad hoc network environment. Finally, we will start the simulation scenario in the controller area network environment in order to obtain the same voice quality and network performance then the work will extend the broadcast coverage area.

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Competing Interests

The authors declare that they have no competing interests.

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