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WIMAX QUALITY MANAGEMENT OF VOIP SERVICE OVER ENTERPRISE WAN ARCHITECTURE FOR STREAMING VIDEO MOBILE APPLIANCES

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ABSTRACT

This research article describes about the voice call transmission among WiMax coverage area. The theme of this work is to obtain better voice clarity than mobile communication networks. The objective of fourth generation networks is to provide larger coverage, low power handling of base stations and good compatibility. Similarly, this work examines the best voice clarity with reliable quality of network management in the LTE streaming coexisting standards. It offers comparative analysis of the two scenarios. They are the different coverage locations (50x50 km and 100 X 100 km), a large number of mobile users, same configuration of connection setup with devices and the diverse interference among users. Finally the high coverage area has achieved the results better than lower coverage region.

KEYWORDS: *Streaming, VoIP, WiMax, Network management, Mean Opinion score, G.711 voice CODEC*.

I. INTRODUCTION

WiMAX is one of the wireless WAN (Wide Area Network) technologies. The WAN is integrating the WiMax and VoIP over services. VoIP is a rapidly growing mobile internet IP based technology. It provides the cheap call setup connection and better coverage than other location based networking technologies. The drawbacks of TDMA technology are i) Multipath distortion ii) every user has a predefined in the time slot iii) reestablishment connection setup. The above drawbacks are rectified by VoIP over IP based data communication networks. Upcoming 4th generation network objectives are, to reduce the complexity, power/energy dissipation levels and improved coverage area. This article suggests that WiMAX could be a perfect proposal for an internet protocol (IP) based on video services. The PSTN is one of the telephone networks which achieve good voice quality (MOS is more than 4) in the second generation.

Nowadays, the world goes behind wireless technology built the quality of voice is very poor due to more packet loss and jitter. In this paper, WiMax scenario is obtained achieving good voice quality which is nearly equal to the voice quality of PSTN/Mobile Networks. The lots of factors are affected in the mobile networks such as network maintenance in the larger coverage area, utilization of bandwidth, installation cost, subscriber billing charges, interferences among mobile users and propagation loss. The above drawbacks are overcome easily by VoIP technology. The VoIP transmission provides high quality of voice calling/called signalling, secure communication through IP devices. The integration of VoIP technology and WiMAX makes a cheap and free voice calls (gtalk, Skype) using the notebook, laptop and phone. While transmitting voice calls in wireless medium, the parameters like high speed delivery, low jitter, reduced end-to-end delay, improving throughput, utilization by users and Mean Opinion Score value are focused mainly. WiMAX based IP technology support high speed internet access, telephone services, voice application, video conferences, video streaming services, etc. VoIP is one of the signalling and speech transmission protocol. In VoIP service, the voice is converted into data packets which is done by voice CODEC encoder and are transmitted to the destination through Internet Protocol (IP). In VoIP is used in one of the best voices CODEC (encoder scheme) is G.711. The some other standard protocols in VoIP are H.323, SIP (Session Initiation Protocol), MGCP (Media Gateway Control Protocol), RTP (Real-time Transport Protocol), SDP (Session Description Protocol) and IAX (Inter-Asterisk eXchange).

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The LAN and WAN network architectures are supported by VoIP services. The Investigation of VoIP testing and installation has been done by LAN and WAN networks to help of gateways, servers and router devices (Cisco, IBM, HP, DELL, Mainframe). The performance analysis of VoIP over services is based on this LAN network that has following considerations, i) speed of delivery ii) the voice quality iii) short delay and iv) jitter and also other LAN architectures (Ethernet, token bus, token ring , FDDI (fibre distributed data interface)) are supported with VoIP services. The Ethernet architecture is one of the popular standards particularly, used as video LAN with wired IP based applications. The simulation parameters are evaluated by following results, i) no short delay, 2) jitter (delay variation), 3) only affected by the collision of packets in topology structure, 4) few errors appear in Ethernet VoIP services. Hence LAN is the recommended standard for VoIP service in order to improve the Qos with security Issues. The WAN gives better VoIP services than a LAN. The major issues focused in WAN networks are: i) bandwidth is limited ii) end to end delay is longer than LAN networks iii) jitter iv) transmission rate is low but VoIP signalling protocol is enough transmission rate so it doesn't require the retransmission in WAN networks. While receiver IP devices are compensating the jitter parameters, the arrival of incoming packets is converted into an analog sound. The receiver IP is simulated the VoIP packets in order to eliminate the holes in between words when the packets are lost during transmission. Finally, such as voice packets are modified by CODEC-compressor scheme. In this simulation time reduces the jitter, packet loss, extra delay occurred between IP devices.

II. 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 sub layer 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. This 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.

III. SIMULATION RESULTS

The platform for our performance analysis is OPNET scenario. In this paper, the WiMAX is analysed under two different coverage areas i.e. 50 × 50 Km and 100 × 100 Km. Figure 1 shows a snapshot of the WiMAX scenario consists of three sub nets. In 50x50km, the subnets look very close to one another, which indicate that the users are nearer to each other. [Voice signals to make interference each other] Similarly, figure 2 shows a snapshot of the Wimax scenario in the selected coverage region is 100x100km. The mobile users are located far away from each other. Then these subnets are combined with cloud IP which is then connected to the WiMAX server via Cisco 7200 series routers, Bay Network Accelar 1050 switch and Nortel firewall. In this article, authors are presenting the two parameters. They are voice quality (traffic packets sent/ received, jitter, end to end delay and mean opinion score value) and Wimax network quality management (throughput, Delay, average delay). The voice call initializes the connecting setup to the IP destination nodes through the entire WAN networks. Table –I description about the selection of simulation parameters with specification of desired value used in OPNET scenario.

Table –I Simulation Parameters

Parameters	***Specification of parameter***
Max no of subscribes nodes	100
Number of subnet	3
Transmits Power (W)	0.5
Received power tolerance	-90 to 60 dbi
Transmission scheme	OFDMA
Transmission frequency	20 MHz
Modulation	QPSK 3/2
Block time interval (Seconds)	3
Connecting retries	16
Antenna Gain	15 dbi
Application encoder	G.711
Router	7200 CISCO
Firewall	Nortan Firewall
Switch	Bay Network Accelar 1050
Workstation BS	Wimax_bs_ethernet4_slip_router
Traffic type	VoIP
Simulation Time	1 hour
Simulation Area	100 X 100 kilometres
Simulation Events	68,341,981

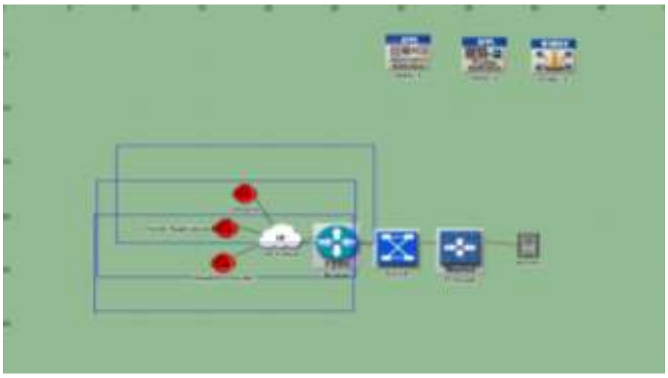

Fig 1: WiMax scenario for less number of mobile users with high interference to every user in 50x50 WANcoverage area

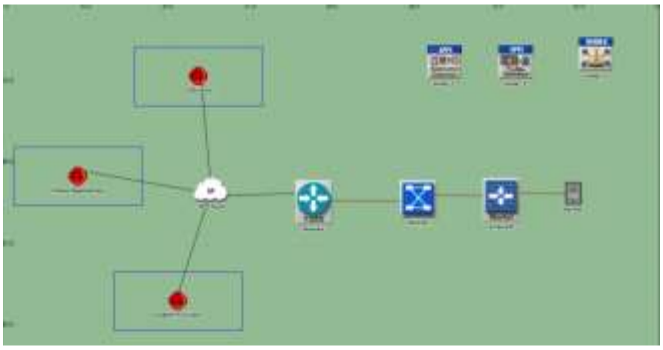

Fig 2: WiMax scenario for large number of mobile users with 100x100 km WAN coverage area

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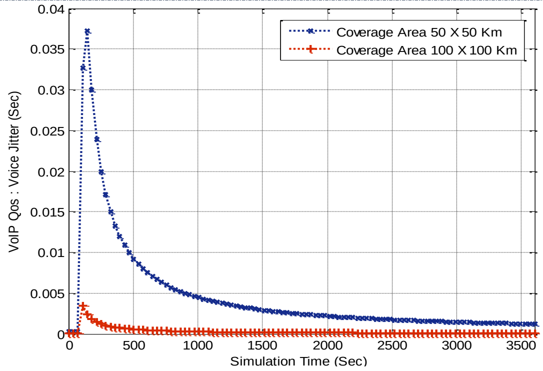

Fig 3: VoIP QoS : Voice jitter (Sec) in coexisting of different coverage area in WAN

From 3 shows Jitter defines the delay in the arrival of packets at the receiver side which is due to the network congestion, change in route (transmission path) and timing drift. In the above figure, the jitter is more about a 50km coverage area as there are more network congestion due to the smaller coverage area and closely located users. In 100km coverage area, the users are far away from each other and so the traffic is less which in turn reduces the network congestion. The maximum jitter in 100km area is 0.004 which is very negligible.

From the voice and video communication, the quality is described in terms of mathematical value. It is called as Mean Opinion Score (MOS) value. It ranges from 1 to 5. The value below 2.5 which indicates the worst quality and above 3 specifies the good quality of voice or video. The MOS can calculate from the average of voice call received by mobile users. Figure 4 shows the best speech quality in the 100km coverage area. Since the transmission period of packets between source and destination user is negligible the encoded delay is also less than 10ms. Hence voice packets must be clear and under acceptable standards.

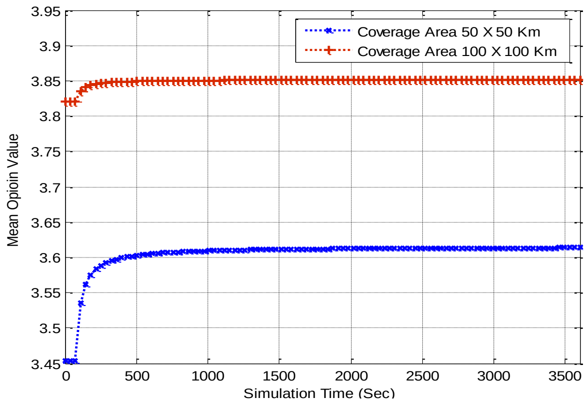

Fig 4: VoIP QoS : Voice Mean Opinion Score in coexisting of different coverage area in WAN

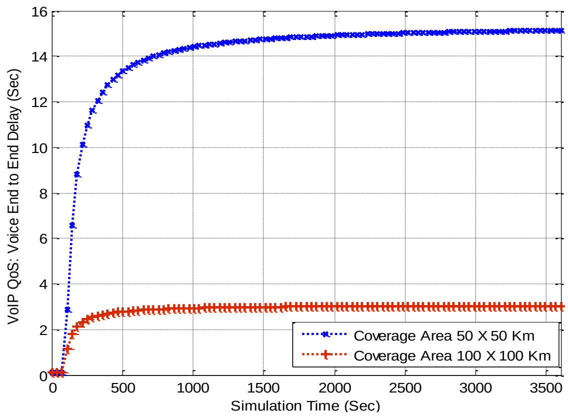

Fig 5: VoIP QoS : Voice End to End delay (Sec) in coexisting of the different coverage area in a WAN

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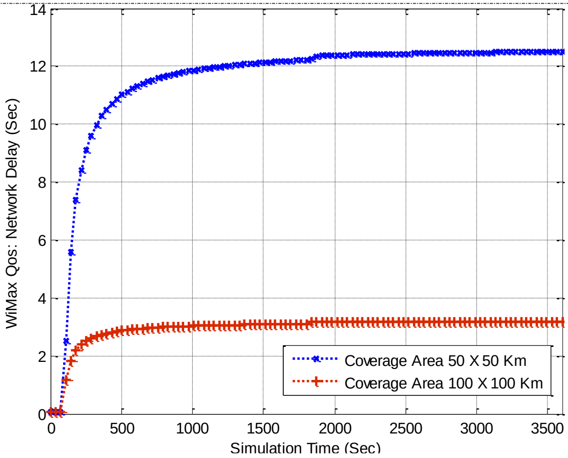

Fig 6: WiMax QoS : Network delay (Sec) in coexisting of the different coverage area in a WAN

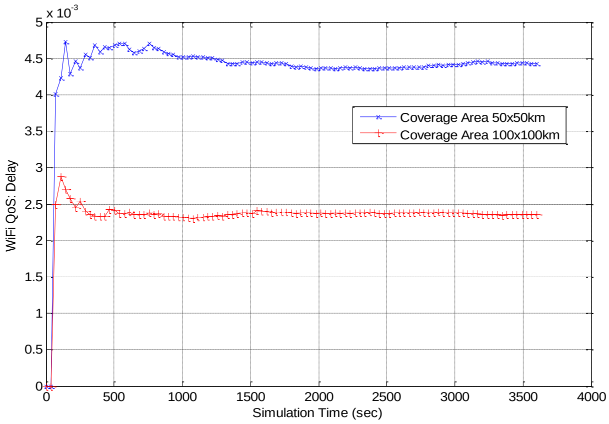

Fig 7: WiFi QoS : Network delay (Sec) in coexisting of the different coverage area in a WAN

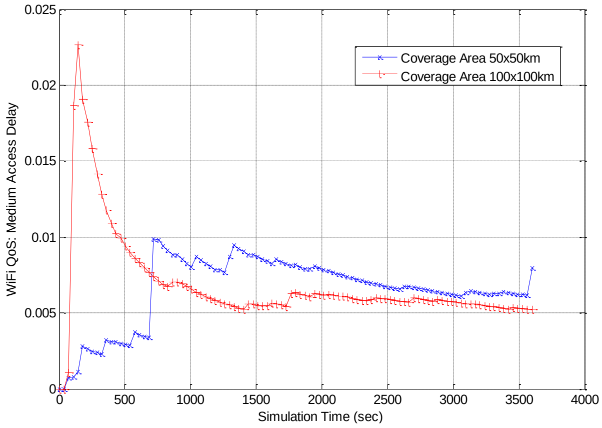

Fig 8: WiFi QoS : Medium Access delay (Sec) in coexisting of the different coverage area in a WAN

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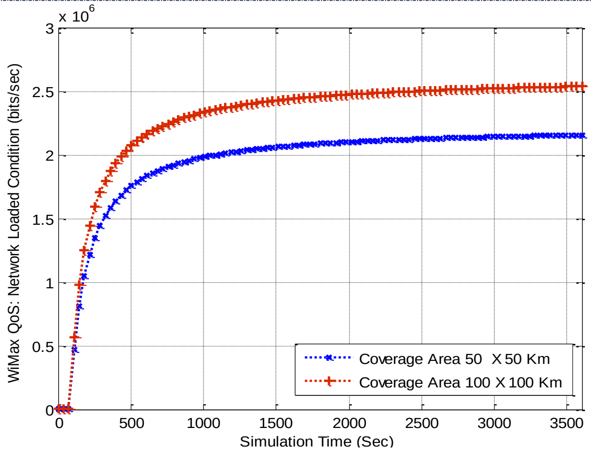

Fig 9: WiMax QoS : Network Loaded Condition (bits/Sec) in coexisting of the different coverage area in a WAN

Since the smaller coverage has many disadvantages like low SNR, interference, traffic and congestion, the packets taken to the destination takes a long time. Even the larger coverage area takes long time to transmit the voice data as the area is wide. But the usage of router and switch make the transmission in the 100x100km coverage area more effectively. Figure 5, 6 and 7 portray the minimum delay for 100x100km coverage area.

Figure 8 shows the medium access delay for WiFi where an initial 100x100km coverage area has more delay but it gradually decreases and becomes lesser than 50x50km. It is because of the configuration of the router which has the ability to reduce the delay.

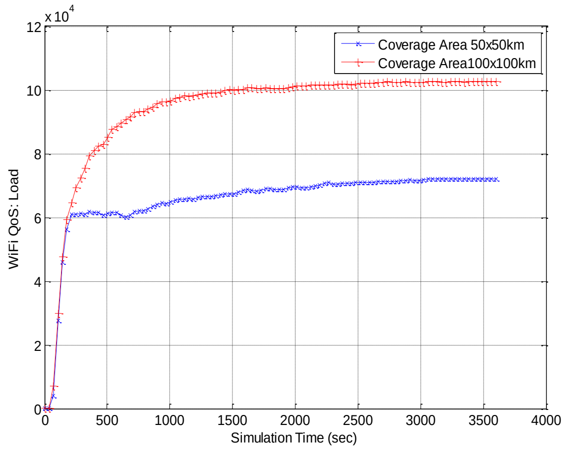

Fig 10: WiFi QoS : Network Loaded Condition (bits/Sec) in coexisting of the different coverage area in a WAN

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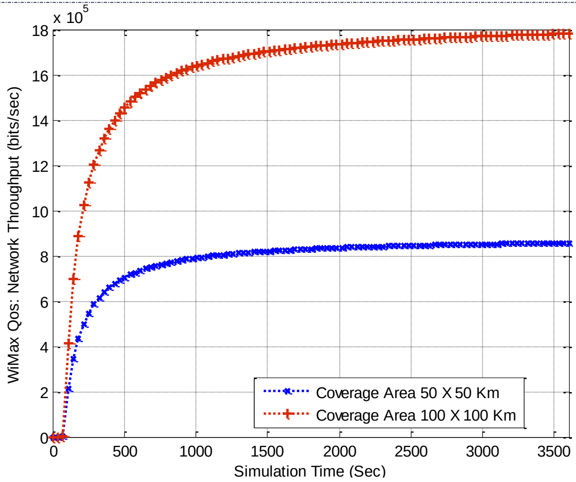

Fig 11: WiMax QoS : Network Throughput (bits/Sec) in coexisting of the different coverage area in a WAN

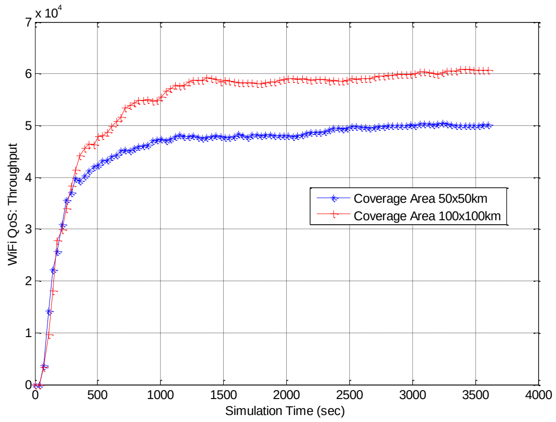

Fig 12: WiFi QoS : Network Throughput (bits/Sec in coexisting of the different coverage area in a WAN

Figure 9 and 10 shows the load for WiMAX and WiFi network respectively. It infers that the load is higher in the 50x50km coverage area than the 100x100km coverage area, since the smaller the coverage area has more interference which in turn reduces the network connectivity. As the users are far apart from each other it has negligible amounts of interference or network congestion. Figure 11 and 12 show the throughput for WiMax and WiFi. The 50x50km coverage area has more delay, jitter and so on. These pitfalls automatically diminish its performance in VoIP. The main challenge and issues are faced in 100x100 km coverage area. They are maximum mobility users, propagation delay, average delay, end to end delay and throughput. The above diagram describes the same concept where the 100x100km coverage area has more throughput than 50x50km coverage area.

IV. CONCLUSION

In this work, the authors examined the voice quality and its clarity identical to PSTN/mobile network voice call quality management and used for mobile appliances. In OPNET WiMax simulation considers the two varieties of issues accounted. They are: 1) low coverage with interference mobile users 2) larger coverage area with far above the ground mobile users. Wimax merging of VoIP in order to get the best voice clarity and same time created quality of the face to face conversation of mobile networks. This quality of results was investigated in the real-time Wimax coverage environmental area. In this future, WiMax network will be integrated with UMTS, GPRS, GSM mobile network VoIP over service for video streaming, video conferences and online gaming applications. Finally, the work will start the wireless control area networks region. Then this work proposes extended for broadcast mobile region..

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