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Effectively Deployment of VOIP over WiMAX Networks

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ABSTRACT: VoIP application of WiMAX is a widely deployed technique and has a high effect in 4th generation networks field. Significant benefit of this technique is use of available infrastructure in the internet connection form. There are various QoS classes for VoIP application in IEEE 802.16 systems, i.e. unsolicited poll grant, nrtPS, rtPS, BE and Ertps. The important VoIP issue is quality of service. For sending better quality in voice data some factors i.e. jitter, delay, packet delivery ratio, throughput and mean opinion score (MOS) require to be enhanced so as to manage this application performance in WiMAX. The introduced idea need no changes of the IEEE 802.16 system framework and we have examined and talked about the performance of the quality of service classes proposed in the IEEE 802.16 system and compared it with introduced technique. The technique introduced in the thesis takes the parameters of Peak Sustained Traffic Rate and Orthogonal Frequency Division Multiplexing (OFDM) frame duration to enhance the QoS for VoIP. The VoIP simulation has been performed with Peak Sustained Traffic Rate OFDM frame duration with these values of parameter performance metrics delay and jitter have been enhanced which in turn has improved the performance of available Extended Real Time Polling Service.

KEYWORDS:-VoIP, rtPS, MOS, QoS, nrtPS, Ertps, OFDM

I. INTRODUCTION

The developing requirement for higher transmission speeds and large capacity to achieve for data intensive multimedia in combination with real time applications, the wireless network have observed an explosive development in last some years [1-2]. WiMAX stands for Worldwide Inter- Portability for Microwave Access can be a communication technique for easily providing high speed data rates to huge geographical region utilizing orthogonal frequency division multiplexing(OFDM) from Base Station(BS) to Subscriber Station(SS) which mitigates noise, interference and multipath impacts[3-4]. The WiMAX network is an integration of SS and BS. Here the packets are transmitted from source node to target node after adopting various modulation, scheduling method and routing mechanism. In compliance with IEEE 802.16 the highest range of WiMAX network is 50 km from the Base Station, where the duty of the BS of offering the air interface to the master station with supplementary services that may be part of the Base station are traffic classification, micro mobility management functions, QoS policy enforcement, tunnel establishment, key management, multicast group management, DHCP proxy and session management. The recipient and antenna could be a small box or personnel computer Memory Card International Association (PCMCIA) card or a laptop [5-6]. With the 4G technology introduction in WiMAX providing metropolitan area network facilities that can utilize one or more Base station and every BS offer the service to the subscribers up to 50 km radius for distributing broadband wireless data over wide geographical region. WiMAX provide high speed, inexpensive, flexible and last mile facilities with performance same as those of wire line infrastructure DSL, T1, cable modem based links, optical fiber or copperware with a no. of QoS requirements. WiMAX offer broad area coverage and QoS abilities for applications ranging from real – time delay-sensitive Voice Over Internet Protocol(VoIP) to non-real time downloads and real time streaming video, assuring that users achieve the performance they require for all kinds of communication with a bandwidth support of up to 10 MHz [7-8]. Depending on IEEE 802.16 standards, WiMAX offer up to 30 miles broadband access to mobile subscribers having a telecommunication etiquette providing full access to mobile internet across countries and cities with a broad range of devices. WiMAX technique is providing very high speed broadband access to mobile internet. Normally 10MHz with the TDD mechanism offers 3:1 up and down link ratio. The WiMAX

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technology architecture depend on MAC layer which is a connection oriented layer. Through MAC layer a subscriber can perform several functions i.e. several types of applications involving multimedia and voice can be utilized. it also supports best attempts for data traffic as real time, bit, traffic flaws etc. the objective of design WiMAX technique is to provide huge no. of subscriber with a variety of connection per terminal[9-10]. This paper is categorized into five sections. Some introductory application and features area of WiMAX technique are shown in Section 1 adopted by Section 2 that shows the QoS requirement and IEEE 802.16 QoS scheduling techniques of WiMAX technology, introduced methods and flow chart shown in section 3 and in section 4 simulations results are explained and at last, the conclusion is drawn in Section 5.

Voice over IP (VoIP) facilities have been importantly obtaining prominence across the last few years due to a no. of impressive benefits over their conventional circuit-switched counterparts involving but not restricted to high bandwidth efficiency, low cost, and reliability of utilizing several compression techniques. At the same time, the usage of wireless networks has also increased extremely over the past years. Wireless LAN (WLAN) solutions, integrated with better physical layer techniques, now promise high data rates of greater than 100 Mbps i.e. IEEE 802.11n.In this context, current attempts have concentrated on marrying the potential advantages of VoIP and WLANs to offer wireless telephone facilities. A natural question that then arises is *how well does VoIP perform over WLAN environments*? The answer of this question is counter-intuitive. Even though the WLANs boast very high data rates and a ganeral VoIP call carries only 128 Kbps of bidirectional data (utilizing G.711 voice codec), the no. of VoIP calls survived by these networks is terribly low. An IEEE 802.11b network for example can survive only 5 VoIP calls even at the greatest data rate of 11 Mbps, as we will present later. VoIP traffic in WLANs is featured by its small frame sizes, and IEEE 802.11 MAC is known widely for very worst performance for small frame sizes. For small frames the overheads at the various layers of the network stack themselves introduce an important burden. Addition to this IEEE 802.11 MAC protocol has other distributed impacts that further decrease the VoIP call capacity.The contributions of this paper are hence twofold: We first offer testbed results for the real VoIP call capacity in an IEEE 802.11b network; we then do mathematical analysis for the call capacity in an IEEE 802.11 network to define the worst performance realized, and determine three dominant components that can be exploited to enhance the call capacity. We then show three algorithms that enhance the features of the three components, and thus result to an increment in the VoIP call capacity of IEEE 802.11 WLANs.

II. QOS AND IEEE 802.16 QOS SCHEDULING SCHEME

QoS is a wide and unbound term that refers to the “collective effect of service,” as obtained by the subscriber. QoS more hardly means meeting specific necessity basically, throughput, packet delivery ratio, packet error rate, network load, jitter, and delay related with a provided application. WiMAX networks must support a diversity of applications, i.e. voice, video, data and multimedia and each of these has various traffic patterns and QoS requirements [20]. The QoS is granted based on application and service type under consideration. For instance, a subscriber forwarding an email requires no real-time data stream like another subscriber having a Voice over IP (VoIP) application. To offer the service parameters respectively, the traffic management is essential. There are four significant service classes such as UGS, rtPS, BE, nrtPS but there is a fifth type QoS service class which is introduced in 802.16e standard i.e.: extended real-time Polling Service (ertPS). These services are assigned priority in decreasing order. Within all these services resources classes are assigned to maintain and satisfy the QoS of higher priority facilities. Generally, IEEE 802.16 has five QoS classes [7].

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Table 1 widely classifies several service classes described in WiMAX and its applications.

Table I. QoS Classes in WiMAX

Service classes	Description	Applications
Unsolicited Grant service(UGS)	For constant Bit rate and delay dependent applications	VOIP
Real Time Polling Service (rtPS)	For variable rate and delay dependent applications	Streaming audio , video
Extended Real time Service (ertPS)	For variable rate and delay dependent applications	VOIP and Silence Suppression
Non real time polling service (nrtPS)	Variable and non real time applications	FTP
Best Effort (BE)	Best effort	Email , Web ,Traffic

UGS: Unsolicited Grant Service is design to support real time service flow which produces the static size data packet in a periodic way. In this algorithms BS allocate static size grant to the user station. The grants allocate are generally of two type such as grant size and grant period. When voice session is started then these values are conciliated. These grants are enough for forwarding data packets.
Advantages of UGS**:** this service decrease uplink access delay and the MAC overhead which are caused when SS make request to the Base station for bandwidth request to forward the voice data packets.
Disadvantages of UGS**:**UGS allocate static size grant for forwarding voice data packets but voice subscriber do not always have voice data packet to forward because they have silence period and it cause a waste of uplink resources.
rtPS: Real time polling service are planned to support real time facilities which normally creates variable size data packets in a periodic way. BS allocate uplink resources to the Subscriber Station when voice session is started then these values are conciliated.
Advantages**:** In this algorithm the Subscriber Station request the Base station for bandwidth of appropriate size grant so that the rtPS can transport data more effectively in comparison of UGS algorithms.
Disadvantages: Because the Subscriber station always built a request for Bandwidth to the Base Station which in turn can cause more uplink access delay and MAC overhead in comparison of UGS algorithms.
ertPS: Extended Real Time Polling Services algorithm is introduced for eliminating the drawbacks of both rtPS and UGS algorithms. The UGS technique permits BS to allocate fixed size grants to voice subscribers which yields to wastage of uplink resources during the period of silent when voice subscribers do not have any data to forward. Meanwhile rtPS although meant for variable size data packets consumes more time in polling procedure and also account for MAC overhead.

III. ANALYSIS OF VOIP CALL CAPACITY

A. Testbed Implementation

1) Experimental Setup: We carried out a testbed to study the 802.11b network call capacity. The experimental parameters of the testbed is presented in table II.

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Table II: Simulation Parameters for WiMAX

Cell Radius	30km
No. of Base Stations	5
No. of Subscriber Stations per BS	20
Speed of the mobile nodes	50, 100, 150 m/s
Simulation time	1800 sec
Base Station Model	wimax_bs_ethernet4_slip4_router
Subscriber Station Model	wimax_ss_wkstn
IP Backbone Model	Ip64
Voice Server Model	WiMAX_server
Link Model (ASN - Backbone)	PPP_SONET_OC14
Physical Layer Model	OFDMA
MAC Protocol	IEEE 802.16e
Multipath Channel Model	ITU Vehicular A
Traffic Type of Service	IP Telephony
Scheduling Type	ertPS, nrtPS
Application	HTTP
Voice Codec (with and without silence suppression)	G 711, G.729, G.723, G.726,G.728
Inter repetition time	Constant 200

To compete several calls in the wireless domain we utilize three wireless interface cards in a single machine and operate three virtual machines on the physical machine, each related with a various wireless card. Two such machines have three wireless cards, respectively. The laptop linked to the AP is utilized as a real VoIP phone, and a VoIP call utilizing the KPhone [1] and SIP Express Router [2] is established between this laptop and another laptop linked to the wired domain. Other calls are competed utilizing bidirectional constant bit rate (CBR) traffic. The frame size of every UDP packet created by Iperf is adjusted to be 92 bytes (involving 12 byte RTP header) and the data rate is adjusted to be 73.6 Kbps.

IV. ALGORITHM DESIGN

A. Motivation and Overview

We can imagine 5 different techniques to enhance the call capacity by leveraging various components in the equation:
ACK Aggregation (AA): ACK aggregation refers to forwarding a single ACK for a block of n frames. The results of ACK in the *decrement of* TACK.
Frame Aggregation (FA): Frame aggregation means fusing numerous frames targeted to the same end subscriber into a single large frame.
Link Adaptation (LA): Link adaptation means altering the transmission rate for the data frames. The 802.11b standard mentions 4 different data rates that can be utilized. The prevailing channel situations influence the data rate choice.
Time Saving (TS): Time saving refers to *decreasing* TDIFS waiting time between two successive frames.

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Header Compression (HC): Header compression refers to decreasing the several headers size like the UDP/RTP/IP headers utilizing the mechanisms either introduced in literature or otherwise. This scheme also has restricted power in enhancing the call capacity in tune of only 0.1 calls.

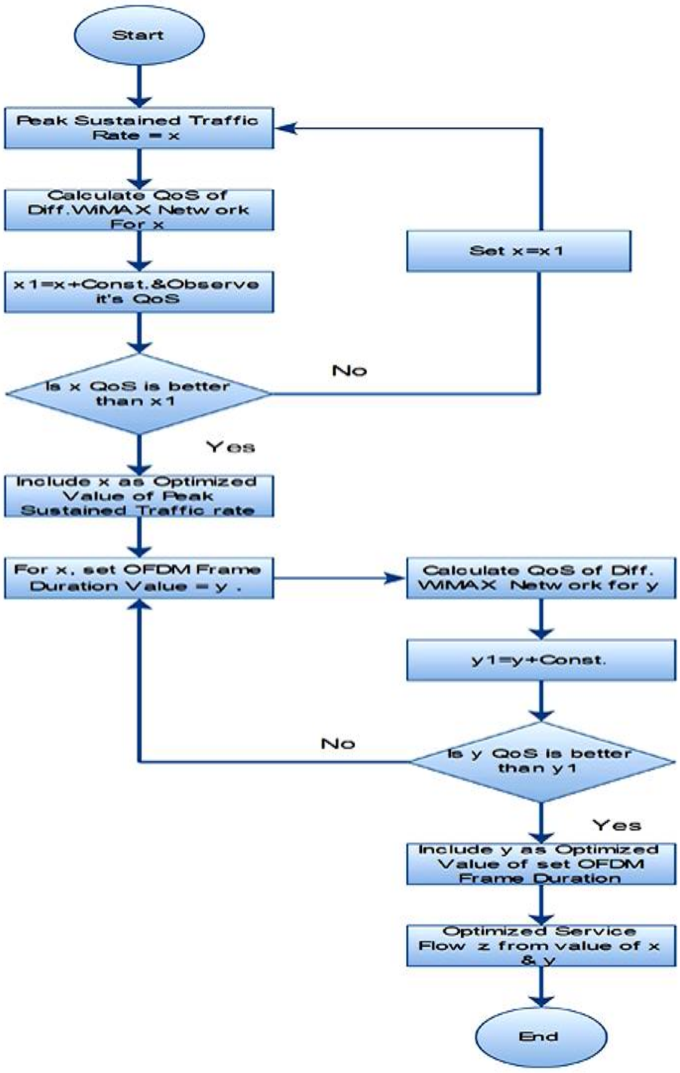

Figure 1: Proposed Algorithm

B)ACK Aggregation (AA)

Adaptive ACK Aggregation Algorithm: Depending on the above results that there are cross over points we can imagine of an algorithm that alters the block size adaptively. We model this technique as a 2.5 layer solution in between the *interface queue and* MAC layer. We consider that there is no preset block size and we forward a block ACK request from the source node once all frames in the existed block are forwarded. Upon obtaining the block ACK request, the target node replies with a block ACK consisting the needed information. The source then initiates a new block and retransfers the needed frames of the prior block and proceeds with newer frames. Due to the existence of a block ACK request we can change the *block* size at will. We implement a simple adaptive technique

Where we increase the size of block upon obtaining a block ACK with all successes and decrease the size on obtaining a block ACK with even a single data loss. The explained algorithm is presented in Figure 3. The adaptive algorithm selects the right block size based on the no. of losses in the current block. Hence the adaptive algorithm should provide

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a performance that is the best of both worlds such as it should have a good delay feature as well as high saturated capability.

B)Frame Aggregation (FA)

2) Enhanced Piggybacking: We now describe the algorithm in Figure 4 utilized for frame aggregation. The core concept behind the algorithm is that we should aggregate frames only when needed. Particularly, we aggregate only those frames that

V. SIMULATION SCENARIO AND RESULTS

3.1 Performance Results

After a no. of simulations, the following results were collected. Depending upon these results, a explained analysis is shown. QoS on WiMAX network is examined through simulation metrics. The metrics which we consider are below:-

- **Delay:** Delay or latency could be described as the time consumed by the packet to arrive from source to the destination node. The results are presented graphically as:

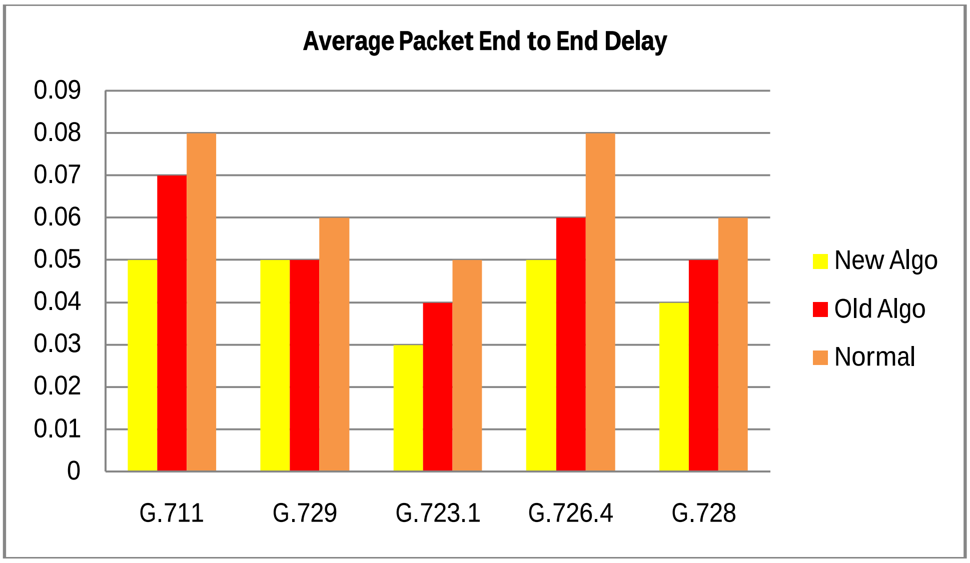

Fig.2 End-to-End Delay Vs Transmission time

- **Jitter: jitter or** packet delay variance could be explained as the variation in delay or packet delay variation. The jitter value is computed from the end to end delay. It is the variation in the time between packets arriving. The results are represented graphically as:

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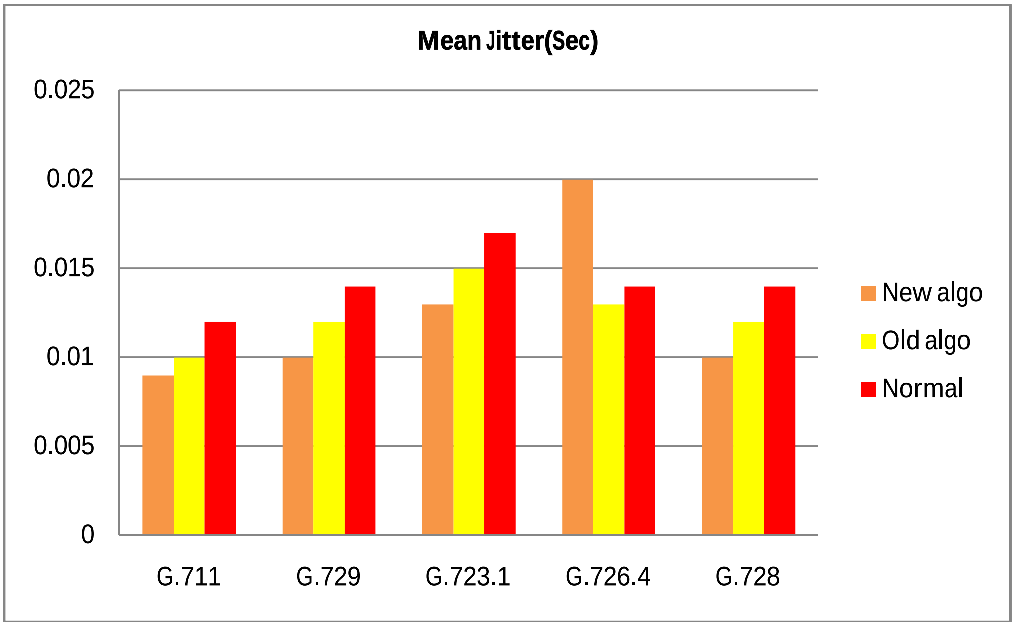

Fig.3 WiMAX Voice Jitter Vs Transmission Time

VI. CONCLUSION

Measurement of QoS is severe for any WiMAX network. The WiMAX network must integrate a no. of quality of service parameters involving low jitter, delay, network load, high packet delivery ratio and throughput. Today in broadband wireless access the observation is that as adoption increase, so does the requirement for ensuring a good QoS. The QoS issues, there have become a severe area of concern for provider of broadband wireless access resources and their users too. In this paper we have carried out general simulation to measure the WiMAX performance for supporting VoIP traffic we have examined various significant severe parameters i.e. jitter and end-to end delivery. Simulation results represent that we have achieved low jitter, delay and network load.

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