Analytical Model on Secure Transmission for SIP-Video Call Setup for WiMax Heterogeneous Networks

S. Jacily Jemila[1], S. Chitra[2]

[1,2] Asst. Prof, Department of ECE, Prince Dr.K.Vasudevan college of engineering & Technology, Chennai, Tamil Nadu, India.

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ABSTRACT: *In LTE networks can provide IP based media streaming connectivity storage and control solution where voice, data, video streamed multimedia can be provided to users on an "Anytime, Anywhere" in the world. The aim of the work is to estimate quality of metric values with fixed and mobile WiMAX standards utilized for SIP (Session initialized protocol) video call setup procedure over wireless link scenario directed to mobile devices. We used two video call modes of operation over heterogeneous WiMAX standards. The proposed model application layers in SIP protocol based on multicast mobile agents used in two different modes of video call established. Initiate services through CBR used for Session control overhead packets. Our analytical model provides that the video call set-up performance, jitter and delay in peer to peer networks. Moreover, the call setup performance can be improved significantly using the robust in application link layer such as SIP with a comparison of WiMAX heterogeneous network standards proposed in our paper and also to establish and maintain secure transmission via IP based security services such as ISAKMP protocol in the network layered protocol. The analytical results were validated by our experimental measurement.*

General Terms

Your general terms must be any term which can be used for general classification of the submitted material such as Pattern Recognition, Security, Algorithms et. al.

Keywords: IEEE 802.16. e, ISAKAMP, SIP, Streaming, CBR

INTRODUCTION

In the modern communications world of today, 4G (fourth generation) mobile wireless standards' goal is to provide high-speed mobile phone transmissions, which can support high-quality streaming video and other critical services. Realizing Video streaming over-LTE Viewing live and prerecorded video carried over mobile networks are exploding in popularity with the sharp growth in Smartphones and the advent of Long Term Evolution (LTE) next-generation mobile networks. As a result, LTE carriers and service providers are presented with a compelling revenue generating opportunity – provided they build LTE networks optimized for large-scale video streaming. At first glance LTE, with its far higher bit-rates and IP packet-based architecture, looks ideally suited for delivering mobile video, including High Definition. But wireless network technology alone is insufficient. To meet the overwhelming demand for live and prerecorded mobile video, the transmission capabilities of LTE networks must be matched by equivalent increases in video server output. LTE networks will need to be integrated with server infrastructures with the scaling capacity, transmission efficiency and wireless network smarts to handle millions of concurrent requests for High Definition mobile video streams. This need will be especially acute when large numbers of users request the same video content at the same time as is increasingly common with live video streaming of popular events. LTE's increasing use of unicast networking where a unique stream is transmitted to every user, even if the content is identical compounds the requirement for next-generation server infrastructure Streamonix as a pioneer in streaming High Definition video over 4th Generation and LTE networks, is uniquely qualified to assist clients in assessing, designing and building next-generation video server technologies matched to their LTE networks.

Challenges for Implementing IEEE 802.16.e

In the data transmission from point to multi point transmission need for priority based scheduling algorithm design for the equal importance of transmission of bits then end of transmission in order to improved high throughput/ traffic modelling and also reduces the loss of transmission between wireless links. Here streaming transmission voice is more important than video so requirement of issues continuity of arrival of bitters in collection to archive significant reductancy at the same time acceptable value of tolerable.

Security Securing the voice communication is also a big challenge for VoIP over WiMAX as care has to be taken that it can be eavesdropped with key management IP based protocol used as two phase process. The Double encryption process - X.509 for Authentication and 152-bit AES, 3DES or 56-bit DES for data flow ensure the transmission is secure and eavesdropping is very difficult on the traffic modelling.

Delay Voice transmission over wireless brings along with it a big problem of packet delay or latency. The factors that add to the delay are the Propagation delay, the serialization delay, channel coding delay at the physical layer and the Medium access delay at the MAC layer. Similarly at the

Network layer the Forwarding and the Queuing delays are encountered, and at the application layer Packetization / depacketization, delay, Algorithmic delay and look-ahead delay, decoding delay are inherently caused.

Packet Loss/ Dropped Packets : Packet loss does excessive damage to the voice signal, as retransmission cannot be considered as an option while transmitting voice. Loss of voice frames at unvoiced/voiced transition causes significant degradation of the signal. Advanced error detection and correction algorithms are used to fill the gaps created by the dropped packets. A sample of the speaker's voice is stored and is used to create a new sound from an algorithm which tries to approximate the contents of the dropped packets or lost packets.

Jitter: When the transmission times of the arriving packets vary as a result of different queuing times or different routes it is referred to as Jitter. Jitter can be taken care of by using an adaptive jitter buffer which adapts itself according to the delay encountered over the networks, to provide a smooth voice stream at the output.

Strict Priority Scheduler:

The real-time traffic occupies a significant percentage of the available bandwidth and Internet must evolve to support the new applications. Such as VoD (Video on Demand), VoIP (Voice over IP), VTC (Video-Teleconferencing), interactive games, distributed virtual collaboration, remote classrooms, grid computing, etc.,The best effort delivery is unacceptable, since in case of a congestion the Quality of Service (QoS) and Quality of Experience (QoE) declines to an unsatisfactory level. The main contribution of this paper is the strict priority scheduler designed to provide the minimum guaranteed transmission rate for all active flows with the respect to their priorities and to provide a fair share of the additional bandwidth. The scheduler also rejects flows, for which the minimum rate requirements exceed the available bandwidth. The proposed solution is applicable for the WiFi wireless network, to accomplish QoS along the path. The Strict Priority Scheduler is the default scheduling discipline in QualNet. It service the highest priority queue until it is empty, and then moves to the next highest priority queue, and so on. It is possible that if there is enough high priority traffic, the lower priorities could be completely frozen out. We can configure only one node at the first scheduler level as strict priority. If any node or queue above the strict-priority node has packets, it is scheduled next. If multiple queues above the strict-priority node have packets, the HRR algorithm (Hierarchical Round Robin) selects which strict-priority queue is scheduled next. One strict priority traffic-class group is called the auto-strict-priority group. The scheduler nodes and queues in the auto-strict-priority group receives strict-priority scheduling.

ISAKMP Protocol

The ISAKMP is used by AH (Authentication header) and ESP (encapsulated security payload) to establish the security associations needed to accomplish the protocols. However ISAKMP advantages can be exploited by any other security protocol, and in this way it will be possible to avoid the duplicity of single purpose negotiations of security parameters. Current security protocols negotiate its parameters by the exchanging messages. The ISAKMP protocol is divided into two phases. In the first phase the parties establish a ciphering key by a key establishment protocol. Once the key is established, they authenticate one each other and negotiate how to protect the second phase. In the second phase, this is made the negotiation of the security parameters on behalf of any security service. This second phase is protected using the parameters.

RELATED WORKS

There are many papers proposed in VOIP based video streaming technology with H.323/SIP protocols. The authors are mainly concerned with routing protocol based transmission in order to achieve high robustness and capacity of voice transmission over IP networks various methodologies have been implemented and verified they proceed.

[1] Ehsan Haghani et al estimates the quality of metric values for MPEG standard video transmission and anaysised cross layer solution over Wimax scenario [2] Jae-Woo So et al investigated the OFDM (Orthogonal Frequency division multiplexing) system VoIP based up/down link signalling overhead information streaming over IEEE 802.16. e networks. [3] Castro, M.A.V et al investigated video streaming under temperate and tropical propagation conditions based on DVB-S2/RCS and WiMax standards. [4] Shyam Parekh et al examined mobile user provide high speed network connectivity and satisfies different quality of metric values while optimizing resource allocation problem. [5] Xiangyang Li et al proposed the concerns and requirements of VoIP security and it is review of security issues and the defence mechanisms and also focus on attacks and countermeasures unique to VoIP systems [6] Jae-Hyun Kim et al investigate the maximum number of supportable VoIP users for the resource request schemes in terms of the packet-generation-interval in the silent-period, the duration of the silent-period, and the major VoIP speech codes. [7] Oktay, M. Et al examined Earliest deadline first scheduling algorithm commonly used in the VoIP and VoD in wire and wireless application in order to improve the quality of metric values. [8] Gordon –Ross et al analysed two novel interworking architectures to integrate WiMax and third-generation (3G) networks. And also analyse the SIP-based IMS registration and session setup signalling delay for 3G and WiMax networks with specific reference to their interworking architectures. [9] Nilanjan Banerjee et al

examined in the undesirable delay and packet loss coexisting with heterogeneous IP based network and also achieved good quality of services in application layered SIP protocol. [10] M. Atif Qureshi comparative analysis voice traffic pattern of the MAC and transport layer (TCP/UDP) in the Wimax and Wifi architecture

SIMULATION RESULTS:

We use a Qualnet simulator as our performance analysis platforms various evaluation parameters include the time between 1st and last packet, no of packets, Average packet size, and throughput of the simulated scenario. The simulation parameters are summarized in table1. We designed the infrastructure networks Setup containing 12 no of nodes. We are compiling the all nodes with video traffic in VoIP transmission between the source and destination nodes. Our work is investigating voice quality of metric values of a fixed WiMax and mobility WiMax providers in different coverage area, enterprise warehouse and wireless security surveillance under the mobility nodes scenario.

Table 1. Parameters for Simulation Evaluation

Parameters	IEEE 802.16	IEEE 802.16.e
Data rate	52 Mbps	52 Mbps
No of Nodes	12	12
Application	SIP	SIP
Routing Protocol	Bellmen ford	DSR
Traffic Type	CBR (video)	CBR (video)
Running Time	300s	320s
File Name	Terminal alias address file (.endpoint)	Terminal alias address file (.endpoint)
Simulation area	900×900 m^2	1000×1000 m^2

The qualities of metrics are analyzed in the video established between two different VoIP users in fixed and mobile WiMAX standards. Video traffic applies between 2/3 and 5/6 respectively. The two applications layered protocols are SIP and H.323 protocol respectively. The transport layer is generating traffic in RTP/RTCP protocol services. We analyzed average jitter, average delay, CBR in client and server nodes are taken into SIP application layered protocols.

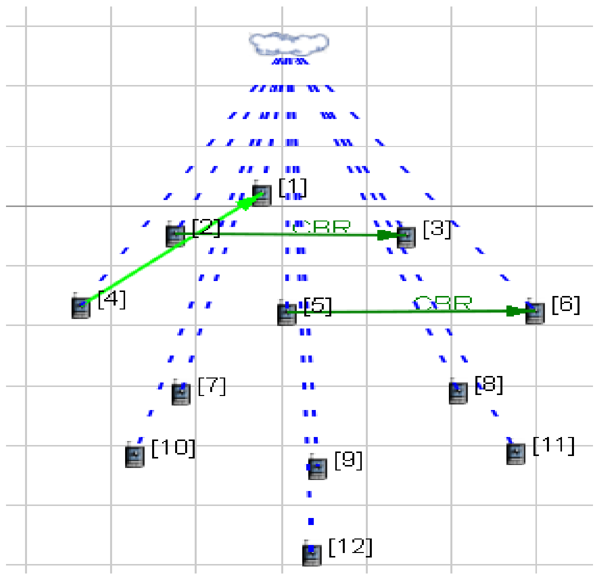

Fig 1: Snapshot of IEEE 802.116 for SIP Protocol for VoIP transmission in Qualnet simulator

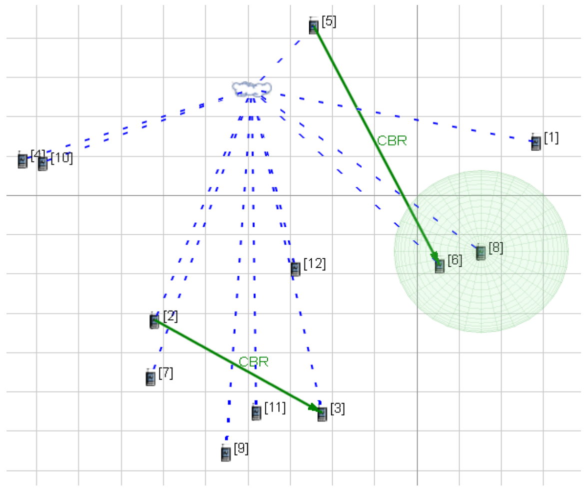

Fig 2: Snapshot of IEEE 802.116. e for SIP Protocol for VoIP transmission in Qualnet simulator

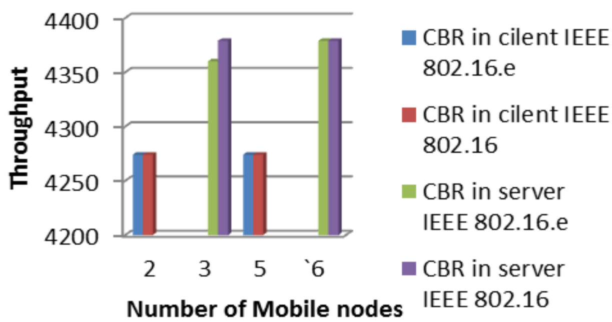

Fig 3: Number of mobility nodes corresponding with throughput in client and server nodes

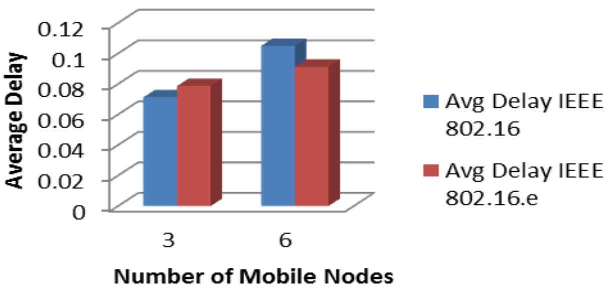

Fig 4: Number of mobility nodes corresponding with an average delay in the mobile nodes

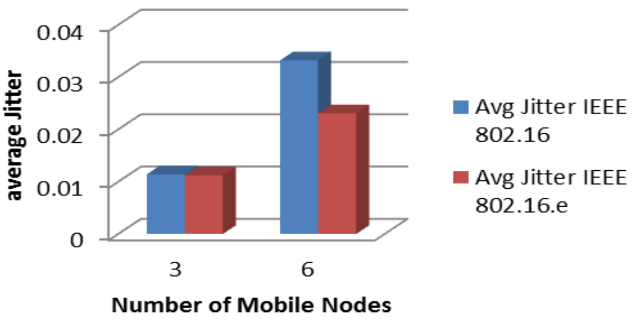

Fig 5: Number of mobility nodes corresponding with an average Jitter in the mobile nodes

The above figure 3-5 show the video transmission packets occurred in mobile nodes in order to archived better quality of metric values (QOM) than fixed wimax scenario. The QoM value can be following parameters as Voice traffic establishment in the source nodes 2 and 5 respectively, similarly VoIP receiver in the destination nodes 3 and 6 respectively. The above two scenarios are comparing the performance analyse in the Voice transition time period (established in source and destination nodes), average jitter , the average delay in between the traffic path between source and destination nodes.

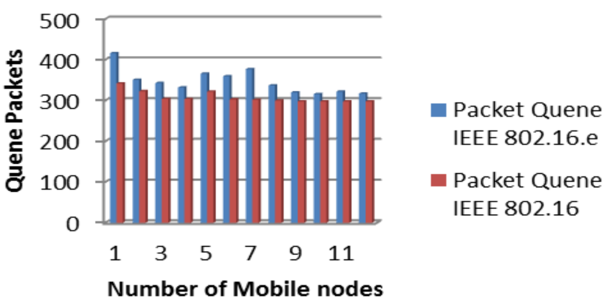

Fig 6: Number of mobility nodes corresponding with quene packets (Strict Priority Scheduler)

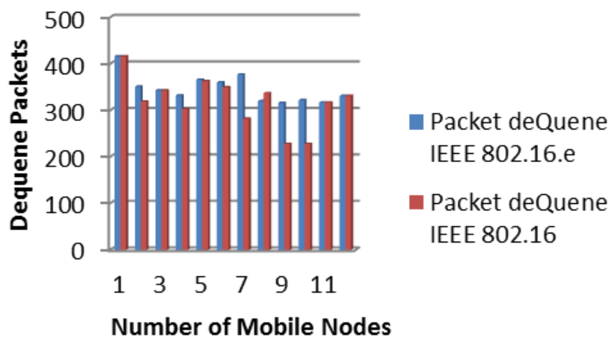

Fig 7: Number of mobility nodes corresponding with de-queue packets (Strict Priority Scheduler)

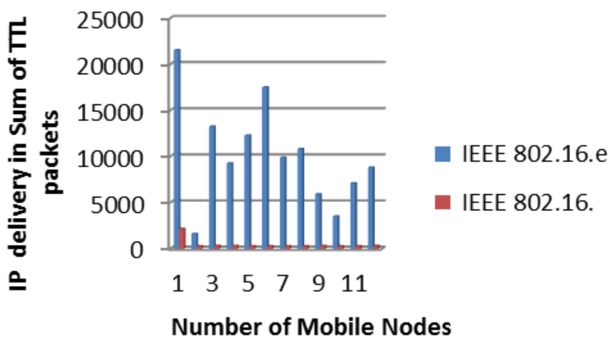

Fig 8: Number of nodes corresponding to IP Delivers TTL sum of packets sent in ISAKMP protocol

In the above two figure 6-7 represented as the real time packet transmission over the scheduling round robin algorithm. The video information is converted into packets the then they will arrive the probability of queuing theory. It depends on the total no of establishments of packets and arrival of packets between client and server nodes. In order to obtain maximum guaranteed transmission rates for all active flows with the respect to their priorities and to provide a fair share of the additional bandwidth.

In the above figure 8 shows the number of mobile nodes represented as initialized cipher key phase factor with user specified delay after phase one completed. It is also possible to start phase two in authentication when some data packet comes at ISAKMP server and it doesn't find any IPSec SA for that packet's source and destination networks. The ISAKMP protocol for creating cookies, generating keys and nonce is being simulated by some simple stub functions. The severs nodes established Security Associations (SA) in the wireless links are bidirectional, that is same SA is used for both inbound and outbound packets.

CONCLUSION

We experimentally investigated application layered protocols to the quality of metric values of SIP protocol video streaming transmission over peer to peer link in wireless scenario. Video transmission through CBR from source to destination node in the overall scenario investigate throughput, average delay, jitter, SIP analyzed video traffic from source to destination node. The SIP protocol issued new method should be an efficient, fair, high throughput, bandwidth guarantee and that will enhance performance of video transmission over mobile WiMAX.

REFERENCES

[1] Ehsan Haghani, Shyam Parekh, Doru Calin “A Quality-Driven Cross-Layer Solution for MPEG Video Streaming Over WiMAX Networks” IEEE Transactions On Multimedia, October 2009, Vol. 11, NO. 6, PP 1140-1148.

[2] Jae-Woo So, “Performance Analysis of VoIP Services in the IEEE 802.16e OFDMA System With In band signaling” IEEE Transactions On Vehicular Technology, May 2008,Vol. 57, No. 3, PP 1876 -1886.

[3] Castro,M.A.V , Fernandez.D.P “VoIP Cross Layer load Control for Hybrid Satellite WiMAX Networks” June 2008, Issue 3,PP 32-39.

[4] Shyam Parekh doru calim, Eunyoung Kim ; Ansari, N “A Quality driven cross layer solution for MPEG video streaming over WiMAX networks” 2009,Vol 11, issue 6 IEEE transaction on multimedia communication PP1140-1147.

[5] Xiangyang Li, Jinhua Guo “Seurity challenge and defense in VoIP infrastructures” 2007, IEEE transaction on systems management and cybernetics part-C application & reviews, Vol 37, Issue 6 PP 1152-1162.

[6] Jae-Hyun Kim “Application aware design to enhance system efficiency for VoIP services in BWA Networks” 2011,IEEE transaction on multimedia Vol 13, Issue 1 PP 143-154.

[7] Oktay, Mantar, H.A. “A real time scheduling architecture for IEEE 802.16 –Wimax systems ” Jan 2011, IEEE international conferences, PP 189-194.

[8]Gordon –Ross, Munir A “SIP based IMS signalling Analysis for WiMAX -3 G internetworking architecture” May 2010, IEEE transaction on mobile computing, Vol 9 issues 5 PP 733-750.

[9] Nilanjan Banerjee, Arup Acharya, Sajal K. Das “Seamless SIP-Based Mobility for Multimedia Applications” IEEE Transaction networks March 2006, Vol 20,Issue 2,PP 6-13.

[10] M. Atif Qureshi , Arjumand Younus,Muhammad Saeed, Farhan Ahmed Sidiqui, Nasir Touheed “Comparative Study of VoIP over WiMAX and WiFi” International Journal of Computer Science Issues, May 2011,Vol. 8, Issue 3, PP 433-437.

[11] Chatterjee, M, Ganguly, S. “Improving quality of VoIP streams over WiMAX” IEEE Transaction On Computers**,** 2008,Vol 57 ,Issue 2,PP 145-156.