Quality of Service based packet scheduling to support video streaming service for Next Generation Wireless Mobile Network

[1] Mr. T. A. Chavan, PhDScholar
[2] Dr. P. Sarasu, Director R&D,
Dept. Of Computer Science & Engineering
Vel Tech Dr. Rr & Dr. Sr Technical
University, Avadi, Chennai, India
[1] e-mail:tachavan@rediffmail.com
[2] e-mail: sarasujivat@gmail.com

Abstract*- Next generation wireless mobile networks is a vast area of research and has many design issues like throughput, delay, packet loss, etc. which deals with data transmission and packet scheduling techniques. Based on the above issues, we focus on data packet delivery based on priority and fairness with minimum delay and jitter. In this proposed paper, we are dealing with packet scheduling of multimedia and non-multimedia data based on priority. According to the application, real-time data packet should be considered as a higher priority, and non-real-time data packet should be considered as a lower priority. Packet scheduler is a decision-making algorithm that selects or drops the packet based on the network load, packet size, bandwidth, and Packet arrival rate, the deadline of packets, quality of the channel, signal to noise ratio (SNR), and type of traffic. Packet scheduling algorithm is the NP-complete algorithm. It becomes very difficult to handle when all packets are coming in with high packet rate, high packet size, and with low bandwidth. Therefore all the packets may not reach the destination or base station. Some of the packets may be dropped due to these mentioned reasons of network characteristics. Many packets scheduling algorithms are generally used to assurance packet data quality of service and transmission rate in wireless mobile Network. In our proposed work longest waiting time first with the fair scheduler and intelligent buffering techniques are used to enhance the quality of service of packet scheduler in Next Generation Mobile Networks. The priority of longest waiting packets is increased for fair scheduling of packets to avoid starvation. Intelligent or dynamic buffering technique is used to reduce the packet loss or to drop of packets during peak time and weak signals of OFDMA channels. In this technique buffer size is calculated dynamically based on network condition and adjusted as per the evaluated value. For buffer size calculation it takes following parameters into consideration 1) packet arrival rate 2) quality of signal and 3) bandwidth. The suggested packet scheduling algorithm constantly updates the control parameter to follow an effective balance between the Quality of Service of video flooding and the network throughput.

Keywords: Mobile networks, packet scheduling, multimedia data, video streaming, Quality of Service.

I. INTRODUCTION

According to the ITU, Next Generation Wireless Mobile Network is a packet based network which can provide various telecommunication and data services. NGN is also called Beyond 3G(B3G) network. The B3G network architecture will progress to accommodate a wider range of users, applications, and economic deployment. NGN also known as beyond 3G is to identify the next step in mobile wireless communications.

Following are the features of next generation wireless mobile networks:

1) The transition towards all IP based Network infrastructure.
2) Support of heterogeneous technologies (e.g.PSTN, Ad-hoc network, LANS, WiMAX, WiFi, etc.)
3) Seamless handoff through both homogenous and dissimilar wireless access technologies.
4) QoS support on the IP layer.
5) Use of policy-based mechanism to determine QoS accounting & billing mechanism for multimedia services.
6) Secure access to multimedia services across different networking environments.
7) Access to multimedia services in hybrid IPV4 or IPV6 based networks.
8) It will offer reliability, availability, security, and performance.

9) It accommodates more users per cell.
10) It supports for backward compatibility with existing wireless standards.

1.1 MULTIMEDIA SERVICES:

NGN will provide convergence of all types of media over IP such as voice, radio, TV and multimedia services. Multimedia is a type of medium which is created by using more than one conventional medium. It is any blend of text, images, audio, animation video delivered and manipulated by electronic means.

ICTs (Information and communication technologies) have expanded the scope of information control and delivery. Through ICT it is now possible to let people experience simultaneously with more than one kind of medium. When an individual is permitted to control multimedia delivery when, what, & how content is presented, it is called interactive multimedia, e.g. Video game.

When an end user is presented with inter and intra-linked multimedia content through which he can navigate is called as hypermedia or nonlinear multimedia. A piece of multimedia content is called linear if it has a predefined beginning, source and end which cannot be altered for example watching a movie. We know that to transmit the video of one second, we need 177MBPS bandwidth, which is commonly not available. Therefore, audio-visual streaming service is one of the most important uses of the multimedia service. Video data is becoming one of the most dominant traffic components over the network. It is very difficult to transmit the large amount of video data over the low bandwidth wireless network. To improve the QoS of multimedia or video data, the effective video Compression algorithms are used to overcome the bandwidth limitation. International standards such as moving pictures expert group MPEG-1 MPEG-2[14] MPEG-4[15] AVC H.261 [16], H.263 [17], H.262 and H.264 [18] have been developed to accommodate different needs of ISO/IEC and ITU-T respectively.

In the Internet, multimedia applications can be classified into three major classes: 1) Streaming stereo-audio or video for example movie 2) Streaming live audio or video for example live cricket match and 3) real-time interactive audio or video, for example video conference. For the time-sensitive multimedia application timing consideration hold most significance. If the packet of any video-audio application come across a delay of more than some hundred milliseconds, they are essentially useless. Time-sensitive applications, such as online audio or video areto a good extent loss tolerant, for example, web, Telnet, and file transfer applications. Elastic applications are extremely sensitive to data loss and for them, completeness and integrity of data transferred are invaluably precious [4].

We have organized this paper as follows: we presented WiMAX architecture in section II. Section III describes a review of the work. The projected packet scheduling algorithm is presented in section IV. Results are discussed in section V, and section VI concludes the paper with the future work.

II. WiMAX System Architecture:

In wireless mobile communication systems, wireless broadband WiMAX systems are major technologies in near future. Mobile IP allows data handoff over different sub-networks. IPV6 is the next generation internet protocol. The world is touching toward a convergence of voice, data and audio-visual. This heterogeneity will expect inter-operability and very large data rate. So IEEE802 committee has set up the 802.16 working group in 1999 to develop wireless broadband standards. WiMAX offers wireless data transmission with the help of different transmission modes such as portable, point to multi-point links and fully mobile internet access. This technology provides up to 10 Mbps bandwidth without cables. The 802.16 technology is estimated to provide inexpensive access having ubiquitous broadband access with integrated data and audio-video services. The most important benefit of wireless broadband technology is that the networks can be created in two to three days by developing a small number of base stations at buildings to create high capacity wireless access systems. The wireless network can grow as the demand increases.

The IEEE 802.16 regulates the air-interface and associated functions related with WLL. Three working groups have been formed to produce the following standards:

1) IEEE 802.16.1- Air interface for 10 to 66 GHz.
2) IEEE 802.16.2-coexistence of broadband wireless access systems.
3) IEEE 802.16.3-Air interface licensed frequencies 2 to 11 GHz.
4) Extensive radio spectrum is available in the frequency band from 10 to 66 GHz worldwide. In business scenario, 802.16 can serve as a backbone for 802.11 networks.

IEEE 802.16 standards are related to the air interface between a subscriber's transceiver station and base transceiver station. The 802.16 standards are organized into three layer architecture as shown in the figure 1.

1) **The physical layer:** This layer specifies the frequency band, the modulation schemes, error correction techniques, synchronization among data rate, transmitter and receiver and the multiplexing technology. WiMAX network supports number of transmission types at physical layer such as single carrier (SC) type, OFDM&OFDMA types. In the OFDM system Forward Error Correction (FEC) coding scheme is used to decrease the rate of error in the transmitted data stream. Reed-Solomon concatenated with convolution code is compulsory for all WiMAX networks. The final data block is categorized into various parallel low-speed data blocks and mapped to an individual data sub-carrier. Then modulated with the help of either Phase Shift Keying (PSK) or Quadrature Amplitude Modulation (QAM) such as Binary Phase Shift Keying (BPSK), Quadrature Phase Shift Keying (QPSK), 16 QAM, and 64 QAM. Thus the modulation is the process of translating data blocks into a suitable transmission form over the physical medium. Modulation and coding rate combination of WiMAX OFDMA are shown in Table 1.

Table1. Modulation and coding rate combination of WiMAX OFDMA (Source wireless network 2011)

Modulation	Overall coding rate	CC code rate	RS code rate
QPSK	1/2	2/3	(24,18,3)
QPSK	3/4	5/6	(30,26,2)
16QAM	1/2	2/3	(48,36,6)
16QAM	3/4	5/6	(60,52,3)
64QAM	2/3	3/4	(81,72,4)
64QAM	3/4	5/6	(90,82,4)

2) **The MAC (Media Access Control) layer:** The MAC layer is responsible for the transmission of data in frames, and it controls the access to a shared wireless medium through media access controls (MAC) layers. The MAC protocol defines how and when a base station or a subscriber station may initiate transmission on the channel.
3) **Convergence layer:** The convergence layer above the MAC layer provides functions specific to the services provided to upper layers. The IEEE 802.16.1bearer service consists of digital audio-video multicast, digital telephony, Asynchronous Transfer Mode, Internet access, wireless trunks in frame relay, and telephone networks.

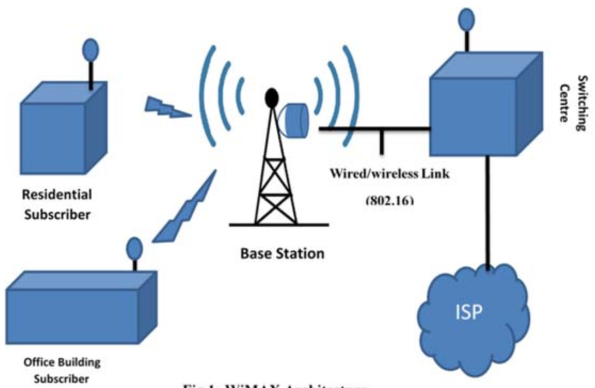

Fig 1: WiMAX Architecture

1) A subscriber sends wireless traffic at a speed ranging from 2 Mbps to 155 Mbps bits/secfrom a fixed antenna on a building.
2) The base station gets transmissions from several sites and drives these traffic through wired or wireless links to a packet switching center using the standard 802.16.
3) The switching-center directs traffic to an ISP, or the PSTN (Public Switched Telephone Network).

In the MAC layer of WiMAX, the resource allocation is dynamically managed by the base station.
WiMAX network can control the modulation and coding scheme (MCS) of each mobile subscriber according to the wireless channel status to improve resource utilization. The Base station collects the

channel quality indication (CQI) of all subscribers, and then decides on an MCS for each user using the own algorithm. This is critical to improving the overall performance of the WiMAX network. Hence, it is an important issue that how many slots are allocated to each user based on the determined MCS. This means that admission of call and packet scheduling techniques are required to enhance the QoS of NGN networks.

III. REVIEW

To support constant video streaming service with high network utilization, packet scheduling, and call admission control algorithms are required. CAC and packet scheduling algorithm have a strong dependency on each other. Packet scheduling technique decides which packet will be scheduled next as per max SNR and PF (Proportional Fair) [11, 12]. Max SNR is based on the best wireless channel condition to select the subscriber to maximize network throughput. The PF method considers long-term average channel condition to maximize the long-term throughput. However these methods do not guarantee QoS.

The research paper [3,9] considers the state of the wireless link between the users and analyze the tolerable queuing delay at the base station. For QoS support in this paper, exponential function maintains queuing delay below the predefined maximum delay.

The paper [22] adaptively controls maximum tolerable delay in which network resources are not allocated to time services even if their delay is less than the maximum delay.

The research paper [23] allocates resources to non-real time service if the current delay is less than deadlines of real time services. The Call admission control algorithms decide whether the new call should be accepted or rejected.

The research paper [6, 7,8, 21]avoids the degradation of the QoS of low priority sessions using adaptive admission controller based on types of service. In these papers, the call request is classified into new or Handoff call or real-time or non-real-time traffic to assign the priorities.

To improve the QoS and network throughput, we need to consider the admission of call & packet scheduling algorithm in the presence of various traffic characteristics and the time-varying wireless channel characteristics [5,13].

In the paper [10], users are accepted based on the queuing model that determines the connection acceptance probability.

The paper [1, 2] presented a joint packet scheduling and call admission control algorithm based on a statistical approach to handling both real-time and non-real-time traffic. This combined algorithm performs their functions by using estimated remaining data volume of the video buffer. In this paper, we propose packet scheduling algorithms for constant video streaming service with high network efficiency over the next generation wireless mobile network (WiMAX network).

The distinctive feature of the suggested algorithm is that packet scheduling algorithm calculates the network throughput, delay, and packet loss based on control parameters. It is continuously adjusting and checking to provide the good quality of video for the subscriber and to optimize the network throughput, delay, and packet loss.

IV. PROPOSED WORK

The main objective of the projected work is to improve the QoS of video streaming services and the network utilization over WiMAX network. The main use of the proposed system is a video on demand streaming service which is the example of real-time polling service class [19]. According to the IEEE 802.16 standard, it uses two OFDM symbols spread over 24 subcarriers called Physical Resource Blocks (PRBs).The base station can classify incoming flows according to their scheduling service classes and stores them at each buffer after admitting the packet of the rtps service class. Then it is determined their transmission priority with the help of packet scheduling algorithm of the rtps service class. To present our work, we assume that WiMAX standard does not specify how to operate packet scheduling and in WiMAX a fixed number of slots are dedicated to the rtps traffic. Fig2 shows the Architecture of Packet scheduling Algorithm.

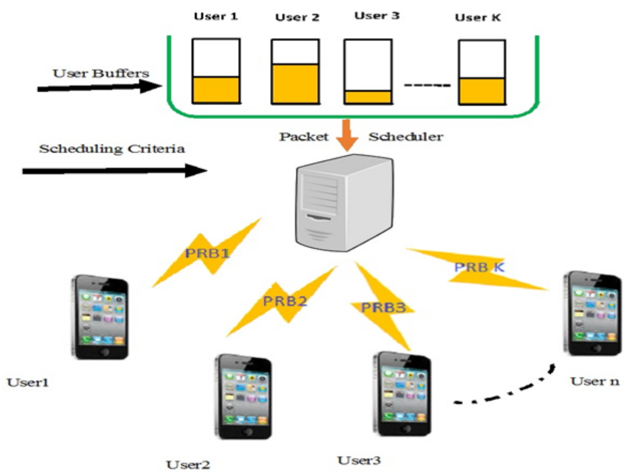

Fig2: Architecture of Packet scheduling Algorithm

The proposed packet scheduling algorithm selects the subscriber u_i from admitted subscribers with the maximum value of the following equation as the next in-service subscriber.

$$i^* = \arg max\, W_i.\begin{Bmatrix} \lambda.buffer_i + \\ (1-\lambda).BW(mcs_i(t)) \end{Bmatrix},$$

Whereas W_i is the largest weighted time first of user i, the N_{ss} is the total users in the cell and the $buffer_i$ (t) (the buffer urgency) is the buffer occupancy of the i^{th} user at time t and BW ($mcs_i(t)$) is a bandwidth of i^{th} user at time 't' when $mcs_i(t)$ is chosen where mcs is the modulation and coding scheme. This is based on the CQI which relates to the signal to interference and noise(S/I) ratio in WiMAX network, Word error indication (WEI), and Received signal strength indication (RSSI). So size of buffered data is not a proper measure for stable video streaming quality. The number of video pictures stored at the buffer is a more reasonable measure. The buffering urgency called intelligent buffering technique can be stated as

$$buffer_i(t) = \max((T_{TH} - f_i(t), 0),$$

Here T_{Th} is the threshold value for smooth video streaming over a time-varying wireless network, and fi (t) is the number of video frames or pictures kept in the buffer at time t. When the number of buffered video pictures is larger than T_{Th}, a user starts video playing.

The λ can be adaptively controlled and the packet scheduling algorithm can work together. If λ increases, the urgency of buffering becomes more heavily weighted than that of the wireless channel condition, and vice versa.

The transmittable data is denoted by $R_{i^*}^{sch}(\tau, \lambda)$ for the next WiMAX frame, this can be calculated by

$$R_{i^*}^{sch}(\tau, \lambda) = BW\big(mcs_{i^*}(t)\big)\,.n_{slot},$$

Here n_{slot} is the number of slots allocated to the i^*th user for next WiMAX frame. λ plays an important role in a weighting factor between the buffering urgency and the network utilization. If λ decreases then the average $BW_{avg}^{avail}(t, \lambda)$ increases.

The proposed packet scheduling algorithm controls packet arrival rate λ based on the detected average buffer freezing rates(buffer size) of subscribers and total target buffer freezing rate, i.e.

$$\lambda = \frac{1}{1 + \left(\frac{BF_{target}}{BF_{avg}}\right)^k}$$

$$BF_{avg} = \frac{\sum_{i=1}^{Nss} BF_i . u_i}{\sum_{i=1}^{Nss} u_i} \quad ,$$

$$BF_i = \frac{P_i^{frozen}}{P_{win}}$$

Where BF_{target}=pre-determined target Buffer freezing rate, P_{win} =number of pictures in a sliding window, N_{SS}=size of sliding window, and Pi^{frozen} =the number of frozen pictures of the i^{th} subscriber in the window and k is constant.

V. Simulation Results and Discussion:

For the simulation of our proposed work NS-2[20] is used for WiMAX network. Initially, 29 subscribers are equallyspread in the cell and moving by the Tow Ray Ground model. And, a streaming service request from 29 subscribers is generated every 2second after the start of service. The simulation parameters are set as shown in Table 1.

table1simulation parameters for ns2

Parameter	***Value***
Channel bandwidth (Mbps)	100
Simulation Time (Sec)	100
Cell Radius (Meter)	300
Speed (m/s)	4
Interval (Sec)	0.1
Frame Structure	OFDM
Maximum Delay (Sec)	10
Video Frame Size (bytes)	500
Scheduling Flow / Traffic Type	rtps

Actually, Proportional Fair(PF),Round Robin(RR), MLWDF, and EXP/PF algorithms show a similar results.On the other hand, the projected algorithm selects next in service user by considering the intelligent video buffering urgency and the wireless signal status. Packet delays of video flows for rtps traffic, reported in fig.3 demonstrates that for video flows, in case of existing algorithms the packet delay increases as compared to our proposed largest weighted time first algorithm with intelligent buffering technique. The averagethroughput for video flows of rtps calls are shown in fig.4. The average throughput improves with our proposed method as compared to existing packet scheduling algorithms. The Packet Delivery Ratio (PDR) experienced by video flows are demonstrated in fig.5, illustrates that PDR rises in our proposed method which is improved one as compared to existing algorithms.

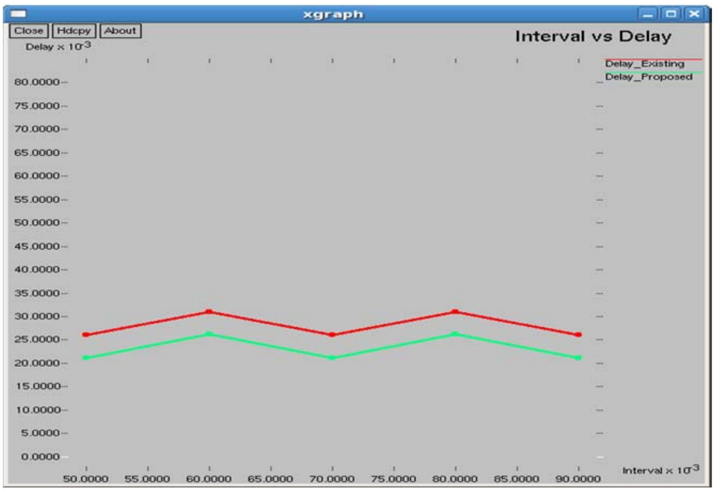

Fig. 3: Interval vs Delay

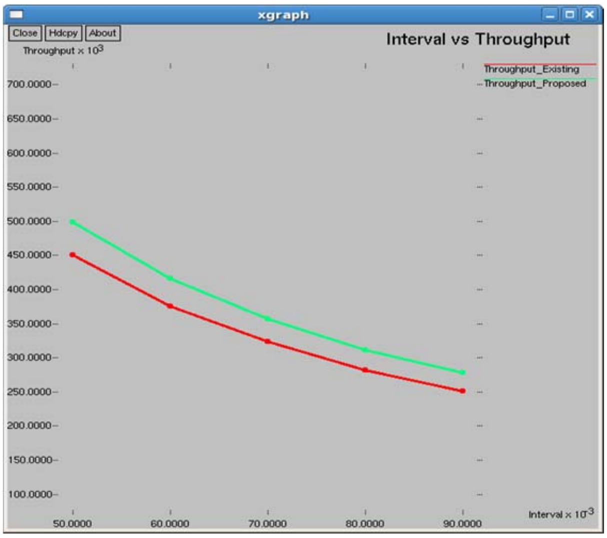

Fig.4: Interval vs Throughput

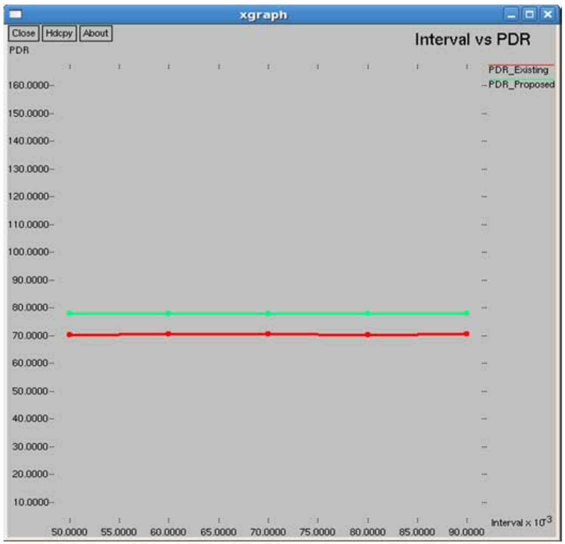

Fig. 5: Interval vs PDR

VI. CONCLUSIONS AND FUTURE WORK:

From the results, we can conclude that the largest weighted time first scheduler is the best when talking about the fairness of the users no matter how many users we have in the cell. Developing efficient algorithms for WiMAX OFDMA in next generation wireless systems is an interesting path of research. However, it is also a big challenge for a simulation tool which can combine the complexity of the physical layer and the MAC layer, in investigating resource allocation and scheduling of these systems. NS3 may be suitable for higher layer investigation of this topic. When the users speed increases the performance of these scheduling algorithms drops remarkably. Our proposed scheduling model for next generation wireless mobile network shows much better performance than other existing scheduling algorithms regarding throughput, maximum delay, and packet delivery ratio of user packets.

It is of particular interest to assess the performance of other scheduling schemes such as Proportional Fair, round robin, and EXP/PF algorithms in 3GPP LTE using a cross-layer approach. However, we state that the proposed model can be extended for live video streaming with real time traffic estimation and monitoring scheme for ad-hoc network.

ACKNOWLEDGMENT

I would like to take this opportunity to express my profound gratitude and deep regard to my supervisor Prof. Dr. P. Sarasu for her exemplary guidanc, valuable feedback and constant encouragement throughout the duration of this research article. Working under her is an extremely knowledgeable exprience for me. I would like to express my sincere thanks to Prof. Dr. P. Anandhakumar, Professor and Head, Computer Technology, MIT Campus,Anna University, Chennai for his constant support, guidanc, valuable suggestions and encouragement for this research article. I would also like to express my gratitude towards Mr. V. S. Bidve and Mr. D. A. Mesram for their kind cooperation and support which helped me in completion of this research article.

REFERENCES

[1] Wan Kim • Hwangjun Song, “QoS-aware joint working packet scheduling and call admission control for video streaming service over WiMAX network”, Wireless Network (2011) 17:1083–1094

[2] Reza Malekian, Abdul Hanam Abdullah,”Novel packet queuing algorithm on packet delivery in mobile internet protocol version 6 networks”, Appl. Maths. Inf. Sci. 7 No. 3 881-887 (2013).

[3] H. A. M. Ramli, R. Basukala, K. Sandrasegaran, and R. Patachaianand,"Performance of well-known packet scheduling algorithms in the downlink 3GPP LTE system," in Communications (MICC), IEEE 9th Malaysia International Conference , Dec. 2009, pp. 815-820.

[4] Lyyapillai AmbikaVelayudhan Pillai Sadasivam and Perumal Eswaran,” An effective queuing architecture for elastic and inelastic traffic with different dropping precedence in MANET”, Eurosip journal on wireless communication and networking 2014.

[5] J.J. Jaramillo, R. Srikant, Optimal scheduling for fair resource allocation in ad hoc networks with elastic and inelastic traffic. IEEE/ACM Trans. Networking 19, 1124–1136 (2011)

[6] B. Liu, H. Tian, and L. Xu, "An efficient downlink packet scheduling algorithm for real time traffics in LTE systems," IEEE 10th Consumer Communications and Networking Conference (CCNC), Jan. 2013, pp. 364-369.

[7] F. Capozzi, G. Piro, L.A. Grieco, G. Boggia, P. Camarda, "Downlink Packet Scheduling in LTE Cellular Networks: Key Design Issues and a Survey," IEEE Communications Surveys & Tutorials, Second Quarter 2013, Vol. 15(2), pp.678-700.

[8] R. Malekian, A. Abdullah, N. Ye, ”Novel Packet Queues Algorithm on packet delivery in mobile Internet protocol version & Networks”, Applied Mathematics Information Science Volume No. 3PP 881-887(2013)

[9] R. Basukala, H. A. M. Ramli, and K. Sandrasegaran, "Performance analysis of EXP/PF and M-LWDF in downlink 3GPP LTE system," 1st AH-ICI on internet, Nov. 2009, pp. 1-5.

[10] Niyato, D., & Hossain, E. (2005). Connection admission control algorithms for OFDM wireless networks. St. Louis, USA: IEEEGLOBECOM

[11] H. Al-Jaradat, K. Sandrasegaran, "On the Performance of PF, MLWDF and EXP/PF algorithms in LTE," International Journal of Computers & Technology, Vol. 8, No. 1, June 2013, pp. 698-706.

[12] Basukala, R. Mohd Ramli, H.A. Sandrasegaran, K.,”erformance analysis of EXP/PF and M-LWDF in downlink 3GPP LTE system”, Internet, 2009. AH-ICI 2009. First Asian Himalayas International Conference.

[13] Niyato, D., & Hossain, E. (2005). “Call admission control for QoS provisioning in 4G wireless networks: Issues and approaches”, IEEE Network, 19(5), 5–11.

[14] ISO/IEC 13818 (MPEG-2). (1994, November). Generic coding of moving pictures and associated audio information.

[15] ISO/IEC JTC 1/SC 29/WG 11 N4030. (2001, March). Overview of the MPEG-4 standard.

[16] ITU-T Recommendation H.261. (1993, March). Video codec for audio visual services at px64 kbits/sec.

[17] ITU-T. (1998, February) Recommendation H.263 version 2 Video coding for low bit-rate communication.

[18] ITU-T. (2007, November) Recommendation H.264 version 3.Advanced video coding for generic audiovisual services.

[19] Song, H. (2004). A metadata-based video-on-demand transmission scheme over bandwidth renegotiating network. IEICE Transactions on Communications, E87-B(5), 1373–1381.

[20] The network simulator ns2 2.34 VERSION

[21] Sohn, K. H., Kim, J., Kim, T. S., Young, Y., & Seo, J. (2006), “Adaptive admission control algorithm for multiuser OFDMA wireless networks”, Barcelona, Spain: IEEE INFOCOM.

[22] Kang, C. G., Kim, T. W., & Kim, J. H. (2008). Adaptive delay threshold-based priority queuing scheme with opportunistic packet scheduling for integrated service in mobile broadband wireless access systems. IEEE Communications Letters, 12(4),241–243.

[23] Ryu, S., Ryu, B., Seo, H., & Shin, M. (2005, May). Urgency and efficiency based packet scheduling algorithm for OFDMA wireless system”, IEEE International Conference on Communications. Seoul, Korean.

AUTHOR PROFILE

[1]Mr. T. A. Chavan MTECH Computer Engg, PhD pursuing at VelTech Dr. RR Dr. SR Technical University, Avadi, Chennai working as asst. Professor, at Smt. Kashibai Navale College of Engg Pune having 15+ years of teaching exprience in engg college.

[2]Prof. Dr. P. Sarasu working as R & D Director at VelTech Dr. RR Dr. SR Technical University, Avadi, Chennai having 20 + years of teaching exprience in engg college.