Information Sciences Letters
An International Journal

http://dx.doi.org/10.18576/isl/080303

Some Simulation Results on Performance Analysis of VoIP Services in WiMAX Systems

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Received: 21 Mar. 2019, Revised: 22 Apr. 2019, Accepted: 24 Apr. 2019.
Published online:1 Sep. 2019.

Abstract: WiMAX is a wireless networking technology that provides quick access to mobile data and provides communications services. This technology allows server-based coding applications, Voice over Internet Protocol (VOIP) allows. Challenges that threaten the system, reducing the resources that will be send to voice packet loss. In this paper an analytical queuing model with consideration of Markov MMPP traffic model is studied and simulated. We present various performance metrics, such as the average uplink throughput of Wimax, average packet delay and average length of the uplink queue and average numbers of arrived packets.

Keywords: Markov modulated Poisson process (MMPP), performance, uplink, voice over Internet protocol (VoIP), Wimax.

1 Introduction

WiMAX is a wireless networking technology that provides quick access to mobile data and provides communications services. This technology is described in IEEE 802.16e standard and the standard wireless network coding based service that allows voice applications over Internet protocol (VOIP) allows. Other Internet technologies to transmit voice and video over Internet Protocol, IP based on networks and especially the Internet. The communication protocol to replace the old phone technology and technology switched telephone network (PSTN) to be handled [1].

When assessing the impact of WiMAX performance overhead is a signal that the system performance by reducing the sources of data that can be passed down to the next post. Specifically, the signal overhead VOIP service throughput considerably due to the small size of operations and the base station information from the resources allocated to reduce the scheduling users.

Another factor that affects the performance of WiMAX is technology for applications where latency delay, such as video and VOIP is critical because online applications, packets are usually loss as a delay. Therefore, online applications, you must use a much shorter delay of about 60ms or 80ms for example, keep to provide quality VOIP services [3].

1.1 IEEE 802.16e Standard Architecture

Point-to-multipoint technology WiMAX is a wireless network that operates in the range from 2 to 66GHz and can be used in urban networks. With WiMAX subscriber station (S) and Mobile Subscriber Station (SMS) through the air with a base station (BS) communicate [12].
As “Fig.1,” shows, the protocol WiMAX includes both the physical layer and MAC is the MAC layer includes the following three layers of special service MAC Convergence Sublayer (MAC CS), MAC Common Part Sublayer (MAC CPS) and substrate security of substrate privacy [7].
Layer MAC (multiple access) signaling mechanisms and functions transferred by BS and SS quality controls identified and defined [16]. The transition between BS and SS by Time division multiple access (TDMA) or Time division duplexing (TDD) was placed at the MAC layer, the frame is bipartite Uplink and Downlink. Transfer the Downlink (from BS to SS) is relatively easy because the BS is just one transfer in the form of Downlink. Data packets are broadcast to all SS and only one packet is for destined in SS [12].

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In the Downlink, BS can be reached based on traffic demand and without interference MS, the necessary bandwidth allocated to each MS [4].

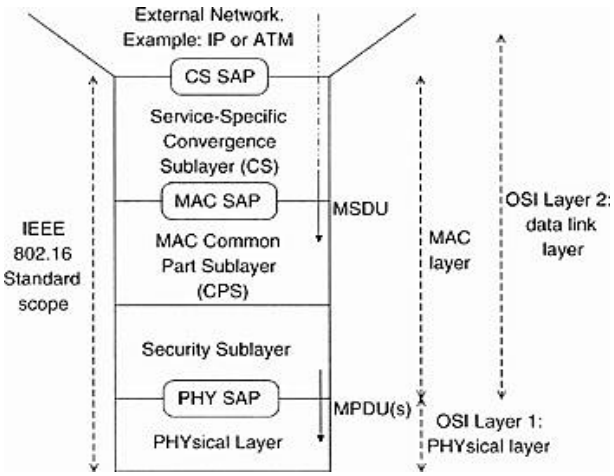

Fig. 1: Wimax standard Protocol.

1.2 Quality of Service in IEEE 802.16e Standard

Quality of Service (Qos) specifies whether a wireless technology can be used successfully with high value services such as voice and video offer or not [7].
Scheduling in the Uplink of four service classes in order to prioritize traffic will use the following order [15]:

- UGS (Unsolicited Grant Service)
- rtPS (real-time Polling Service)
- ertPS (extended real-time Polling Service)
- nrtPS (non-real-time Polling Service)
- BE (Best Effort)

The service flow defines the QoS parameters for the packets that are exchanged on the connection [8]. Each service flow class is associated with corresponding QoS parameters set. These parameters are listed in Table 1. Support for QoS is a fundamental part of the WiMAX MAC-layer design [15].

The higher efficiency, the project would be more complex, so rtPS service class is suitable for image transmission. In WiMAX classification service flow, the priority for each frame in consideration is UGS> ertPS > rtPS > nrtPS and BE (for instance, UGS is at higher priority than BE) [8].

2 System Model

2.1 *System Description*

In this section, a BS Uplink users in a mobile WiMAX system is considered. BS scheduler queue length is limited in two primary queue VoIP list of users is packets and transferred initially on Uplink store and queue second list of users who are imperfect and should be sent back to the store. BS program uses the retransmission queue according to the principle of First-in-First-out the act. When the queue is full, VoIP packets are lost [10].

2.2 User Voice Traffic Source Model

In circuit-switched networks, voice input and output exponential distribution is a Poisson distribution. Traffic engineering circuit-switched networks based on the distribution of Erlang-B. So the sound sources by a queuing system M / M / N / N (input Poisson and exponential service time) where N represents the number of voice channels (According to Table 2, it is thought that the codec G.711 data rate is 64 kbps) [6]. Input rate parameter λ, N number of audio channels and μ the mean holding time (3 min), respectively. Media gate for the audio stream into IP packets before they traverse the Internet and stream restoration after leaving the Internet [10]
At the receiver, after passing the compressed audio stream to reconstruct the media. These activities include the elimination of vibration through a buffer transfer delay, packet decoding audio is compressed. Possibly using the

Table 1: QoS SPECIFICATIONS.

Service flow class	QoS specifications
UGS	support real-time service flows with fixed-size packets generated at periodic intervals with no silence suppression. grant a fixed amount of bandwidth for CBR real-time applications. class is with the highest priority.
rtPS	support real-time service flows with variable-size packets generated at periodic intervals. minimum data rate and acceptable maximum latency for VBR real-time applications. The major difference between rtPS and UGS lies in the periodic request for the bandwidth of rtPS during the service period even when the demand is unchanged.
ertPS	latest one recommended and adopted in the IEEE 802.16e standards. support VBR real-time data services such as VoIP applications with silence suppression and video streaming. The characteristic of this service class is between those of UGS and rtPS.
nrtPS	support non-real-time VBR services which require minimum-data-rate guarantees but can be tolerant to delay. major difference between the two classes is that the bandwidth request of nrtPS can be delivered by a competition cycle mechanism.
BE	with no explicit QoS requirements. A bandwidth request of the BE class can only be sent within the competition cycle.

algorithm and packet loss is destroying the ecosystem. Each packet transmission in tandem with the number of active calls are equal. Gate-media sound source using the following parameters can be expressed, as in

$$U \approx \lambda\mu(1-p_b)\times\frac{PacketSize}{Packetizationdelay}\times\frac{1}{C_{link}} \tag{1}$$

Where ρ_b is the blocking probability and given by the Erlang-B formula. It should be noted that N = 1, and since it is a source of traffic to a traffic source reduction ON / OFF switch to the ON and OFF periods are exponentially [6]. Mean ON time is $\alpha^{-1} = 153Sec$ and mean OFF time is $\beta^{-1} = 280.5Sec$. Extended real-time polling service (ertPS) is assumed to be used for the uplink scheduling, where the BS allocates a fixed amount of bandwidth at periodic intervals during the ON state [16].

The user informs the BS of its transition between the ON state and the OFF state by either using a piggyback request field or sending a code word over a channel quality indicator channel (CQICH) [13].

The BS allocates the uplink resource to the SS scheduled from the uplink queue in accordance with a first-in–first-out (FIFO) policy. The FIFO scheduler is generally selected to provide a constant delay and jitter performance in VoIP service [10].

VoIP traffic N_v requested by the user, the first line is a collection of BS. The primary function of the transmission line can be modeled by Markov Modulated Poisson Process (MMPP) model. The MMPP is a stochastic process in which the intensity of a Poisson process is defined by the states of a Markov chain. This MMPP model is very suitable for formulating the multi-user VoIP traffic, because the MMPP captures the inter frame dependency between consecutive frames [14],[17].

Where **Λ** is the Poisson arrival rate matrix, and **R** is the transition rate matrix in the two-state MMPP. as in

$$R=\begin{bmatrix}-r_1 & r_1\\ r_2 & -r_2\end{bmatrix} \qquad \Lambda=\begin{bmatrix}\lambda_1 & 0\\ 0 & \lambda_2\end{bmatrix} \tag{2}$$

Ways to implement the two-state MMPP model parameters is presented [17]. Average arrival rate (λ_1 and λ_2) and mean residence time is (r_1^{-1} *and* r_2^{-1}) [10]. To calculate the two-state MMPP model parameters are as in [5]

$$\lambda_1 = A\frac{\sum_{j=0}^{M_v} j\pi_j}{\sum_{i=0}^{M_v}\pi_i}$$
$$\lambda_2 = A\frac{\sum_{j=M_v+1}^{N_v} j\pi_j}{\sum_{i=M_v+1}^{N_v}\pi_i} \tag{3}$$

Where A is the emission rate in the ON-State (A = 1/T) and T is a frame duration of voice codec. According to Eq. (2), the parameters to be calculated as is follows

$$\pi_j = \binom{N_v}{j}\rho^j(1-\rho)^{N_v-j}$$
$$\rho = \beta/(\alpha+\beta)$$
$$M_v = \lfloor N_v.\rho \rfloor \tag{4}$$

Transfer rate is calculated as

$$r_1 = \frac{2(\lambda_2-\lambda_{avg})(\lambda_{avg}-\lambda_1)^2}{(\lambda_2-\lambda_1)\lambda_{avg}(IDC(\infty)-1)}$$
$$r_2 = \frac{2(\lambda_2-\lambda_{avg})(\lambda_{avg}-\lambda_1)}{(\lambda_2-\lambda_1)\lambda_{avg}(IDC(\infty)-1)} \tag{5}$$

According to Eq. (5), λ_{avg} to be calculated as

$$\lambda_{avg} = N_v \times A \times \rho \tag{6}$$

If we denote the number of arrivals over a time interval of t by random variable X(t), then the Index of Dispersion for Counts would be defined as the ratio of the variance of X(t) over the mean of X(t). By computing this parameter for different values of time interval t, we will have a curve for Index of Dispersion for Counts, IDC(t) versus t [18]. Although IDC curve is a measure of characterization of the traffic rather than a measure of queueing behavior, it has been shown that this curve has a definite effect on the queueing performance. Any model must have an IDC curve as close as possible to the original traffic, to have the same queueing performance as it [18]. Where rate IDC X (t) variance here is the value ∞. The average arrival rate at the queue is in [9]

$$\rho = s\left(\sum_{k=0}^{N_v.A_{\max}} k \times D(k)\right)1 \tag{7}$$

where each diagonal element of D(k) is the probability of k packets arriving at the BS for the frame duration T_f [10].

where A_{max} is the maximum number of packets that have arrived from a node during the interval $[0, T_f]$, 1 is a column matrix of ones, and Average throughput as the number of sent have been successful state VOIP traffic is equal to as in

$$S = \bar{u}.L/T_f \tag{8}$$

The average number of packets that have been successfully sent to the as [9]

Table 2: Modulation and Coding Schemes.

MCS level, m	Modulation	Coding Rate	Slots for a VoIP UDP transmission, ι_m	α_i
1	QPSK	1/2	36	40.9811
2	QPSK	1/8	24	30.1983
3	QPSK	1/4	12	21.6450
4	QPSK	1/2	6	16.0658
5	QPSK	3/4	4	15.2871
6	16-QAM	1/2	3	12.8709
7	16-QAM	3/4	2	11.9477

$$\bar{u} = \sum_{mi=1}^{M} u_i = \sum_{i=1}^{M} \alpha_i \bar{x} p_\gamma(i) \quad (9)$$

The Average number of packets sent from the queue VOIP scheduled early in the frame, is as follows

$$\bar{x} = \sum_{b=b_{\max}}^{b_{\max}} \sum_{k=0}^{k} \min(k,b)\pi(k) p_s(b) \quad (10)$$

Where K is the maximum size of the queue and b_{max} is the maximum number of packets scheduled from the initial transmission queue and P_s is probability that the BS maximally schedules b packets from the initial transmission queue. The probability of *k* packets waiting in queue is found as[5]

$$\pi(k) = \sum_{i=1}^{2(N+1)-1} \pi(2k(N+1)+i) \quad (11)$$

2.3 Adaptive Modulation and Coding

In WiMAX standard, adaptive modulation and coding (AMC) technique is used to enhance the system performance in varying channel condition. To aid the AMC process the SS feedbacks the channel state information (CSI) to the BS [2]. After receiving the CSI, the BS decides the appropriate modulation and coding scheme (MCS) for that SS, i.e., based on CSI it selects one of the modulation schemes among 64-QAM, 16QAM and QPSK modulation. The SS uses this MCS to transmit its data burst. This work makes an attempt to utilize channel information to design the proposed CAC and scheduling algorithms [2],[11].
Moreover, depending on the different radio conditions, the modulated packets are transmitted either without repetition or with two, four, or six repetitions. The possible MCSs are summarized in Table 2 [9].
The G.723.1 codec generates a 24-B encoded voice frame every T = 20 ms, and therefore, the size of the VoIP protocol Data Unit (PDU) at the MAC layer is subsequently 33 B, including a 6-B generic MAC header[8].

The VoIP PDU employs an adaptive modulation and coding (AMC) scheme at the PHY layer [14]. The VoIP PDU is assumed not to be fragmented, where the parameter lm is the size of the VoIP PDU, which is modulated with the m MCS level after encoding and repetition in units of slots [11].

2.4 Analysis of Packet Delay

D is Average packet delay, the mean waiting time in the queue W_q and the average time in air T_{tr} can be expressed as in

$$D = W_q + T_{tr} \quad (11)$$

Average waiting times for primary can be expressed by Little low theorem [9], as is follows

$$W_q = \frac{L_q}{\lambda_e} \quad (12)$$

And average length of the initial transmission queue is as follows

$$L_q = \sum_{K=0}^{K} k\pi(k) \quad (13)$$

Here λ_e arrival rate of packets in the sent queue of the frame length is equal to the average number of packets have been scheduled, namely$\lambda_e = \bar{x}$ and T_{tr} is equal T_f [14].

3 Simulation and Results

We assume that the total number of MCS levels available for VoIP packet transmissions in uplink is M=7 and the other parameters used in our simulation are as follows:
L_q = 33 bytes, T_f = 5 ms, ρ = 4 frames, K = 100.
Fig.2, shows, the average uplink throughput of a mobile WiMAX system which the average increases linearly. As with the average queue length, the critical number of active voice users affects the throughput. When average uplink throughput equal is 1000, with increases number of users values equal is 1000.
Fig.3, shows the average packet delay as a function of active voice users. The average packet delay maintains a low value because the BS immediately transmits a packet whenever a packet arrives. The average packet delay increases steeply after a certain number of active voice users because the bandwidth requests overwhelm the uplink capacity and consequently prolong the average waiting time in the queue. The average packet delay approximates to a certain value as the offered load increases because the packets that overwhelm the uplink capacity are eventually dropped as a result of the limited queue size.

Fig.4, shows the average length of the uplink queue as a function of the number of active voice users in the uplink. The average queue length linearly increases, but it increases steeply for a certain number of active voice users N, which we call the critical number of active voice users. The steep rise occurs as the bandwidth requests overwhelm the uplink capacity. The value of N increases as the number of packets transmitted with a low MCS level increases.

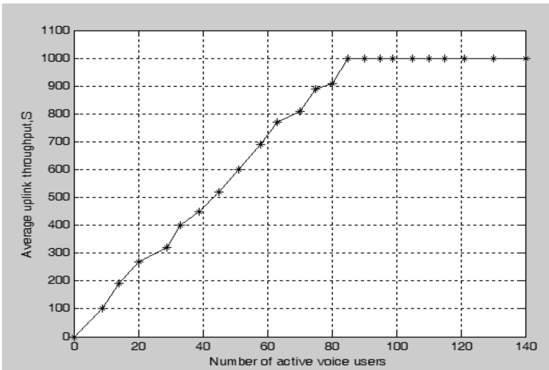

Fig. 2: the average uplink throughput of a mobile WiMAX system.

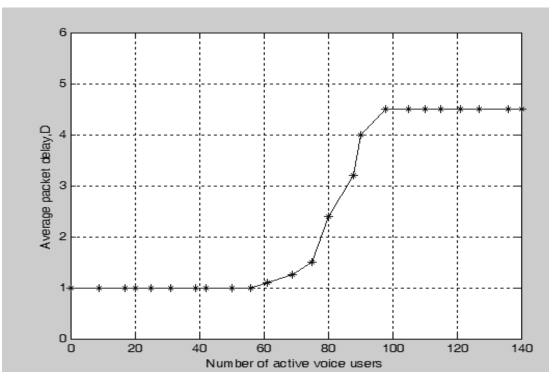

Fig. 3: The average packet delay.

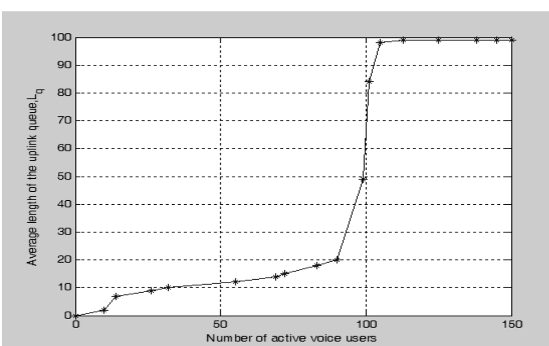

Fig. 4: The average length of the uplink queue.

Fig.5, shows the average numbers of arrived packets. In accordance with the increment of the number of VoIP users, the number of arrived packets grows linearly. On the other hand, the number of serviced packets increases against the increment of VoIP users up to a certain number. We here define this certain number as a threshold number. If the number of VoIP users becomes larger than the threshold number, the number of serviced packets does not increase further.

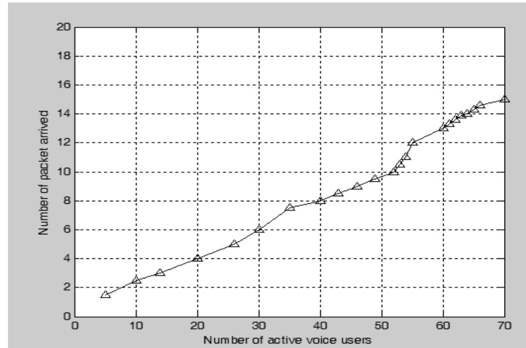

Fig. 5: The average numbers of arrived packets

4 Conclusions

This paper describes the development of an analytical model for evaluating the performance of VoIP services in a mobile WiMAX system. By using this MCS-level set transition matrix and MMPP traffic model, we designed the uplink VoIP system. We demonstrated various results, such as the average number of serviced and arrived packet, the average uplink throughput of a mobile WiMAX system, the average packet delay and the average length of the uplink queue. From the numerical results, the average packet delay as a function of active voice users. The average packet delay maintains a low value because the BS immediately transmits a packet whenever a packet arrives. The average packet delay increases steeply after a certain number of active voice users because the bandwidth requests overwhelm the uplink capacity and consequently prolong the average waiting time in the queue.

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