Stochastic Modeling and Analysis of Feedback Control on the QoS VoIP Traffic in a single cell IEEE 802.16e Networks

Said El Kafhali, and Mohamed Hanini

Abstract—**One of the major challenges in the next generation networks is maintaining the Quality of Service (QoS) for different users who use the services of such a technology. The problem increases when the flows that traverse the wireless link belong to classes with different requirements of QoS. In this case, the network services face several disadvantages caused by the unreliability of wireless channel and the channel sharing by several users. In this paper, we consider a single-cell IEEE 802.16e environment in which the base station allocates subchannels to the subscriber stations in its coverage area. Therefore, two systems are compared. In the first system, called system without Feedback Control, for each uplink subchannel, a separate single queue is used for buffering the VoIP packets from Subscriber Station (SS) to Base Station (BS). However, in the second system, called system with Feedback Control, an Active Queue Managment (AQM) is used to control the VoIP packets. Two thresholds are used in the queue in an effort to control the arrival rate of the VoIP packets. The VoIP arrivals are modeled by a two-state Markov Modulated Poisson Process (MMPP) process. A queuing analytical model is presented to evaluate the performance of both systems. Numerical results obtained show the positive impact of the AQM added to the second system on the performance parameters of VoIP packets compared to the first system.**

Index Terms—**IEEE 802.16e, Active Queue Managment, VoIP Traffic, Markov Chain, Queuing Theory, Quality of Service, Performance Parameters.**

I. Introduction

NOW there are many wireless communication innovations occurring in telecommunications, it may be a part of small incremental innovations or an important innovation that can lead to technological breakthroughs. Consequently, these innovations led to new concepts network referred to next generation networks.

Generally, a next-generation network is a network based on a new technology that, according to its supporters, will be adopted across the board in the coming years. This term is used frequently in the field of mobile services where it has deployed networks, so called second generation networks (2G) (Global System for Mobile communications (GSM) [1], Code Division Multiple Access (CDMA) [2]), since the late 80s; and, third generation networks (3G) (Universal Mobile Telecommunications System (UMTS) [3], Code Division Multiple Access 2000 (CDMA2000) [4]) and new developed technologies such as WiMAX (Worldwide Interoperability for Microwave Access) [5] and LTE (Long Term Evolution) [6], considered as fourth generation networks (4G).

In this network evolution appeared WiMAX Network [5], which envisioned supporting multiple multimedia services such as internet browsing, voice telephony, interactive gaming, email, video messaging, etc. These services demand different Quality of Service requirements, such as average packet delay, average Packet Drop Rate, bandwidth, throughput, transmission delay, availability, jitter, and minimum throughput requirements [7], [8]. To fit diverse QoS requirements is more difficult in wireless networks as compared to the wired networks. This is due to the capacity of a wireless channel varies randomly with time, the time-varying channel conditions and resource conflict among multiple users. Packet scheduling deals with radio resource allocation and is directly related to the Quality of Service provision to users demanding different applications [9].

To avoid congestion in high-speed networks (such as WiMAX network), due to increased traffic which transits among them, we use buffers to handle the excess of traffic when the debit outstrips the buffering capacity. But the limited space of these queues causes the loss of packets of information over time. The management mechanisms queues have great utilities to avoid buffers congestion. The traditional mechanisms such as, Passive Queue Management (PQM) detects congestion only after a packet has been dropped which can cause problems in the quality of service in the network. The Active Queue Management (AQM) [10] is an efficient tool to avoid saturation of the queue by warning the transmitter that the queue is almost full to reduce its speed before the queue is full. The buffer management mechanisms focus on space management, in the other hand scheduling priorities attempt to guarantee acceptable delay to applications for which it is important that delay is bounded. These mechanisms enable networks to improve the required quality for multimedia application to end users. Hence, in this work we make two mathematical models based on Queueing Theory [11] to evaluate the performance of VoIP traffic in a single cell IEEE 802.16e Networks. In the first model, for each uplink subchannel, a separate single queue is used for buffering the VoIP packets from Subscriber Station (SS) to Base Station (BS). But in the second system, an AQM is used to control the VoIP packets in order to reduce the speed of arrivals VoIP packets and manage the loss of packets before the queue is full. These models are compared in terms of VoIP packets loss probability, mean number of VoIP packets in the system, throughput and average VoIP packet delay.

The next section, Section II reviews the related work and

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introduces the problem statement. In section III, we introduce the QoS of IEEE 802.16 MAC layer. In Section IV, we give a description of proposed system models. The queueing models and performance parameters have been described in Section V. Section VI states numerical results. Finally, section VII is devoted to the conclusion.

II. Related Work

Many researchers have been interested in performance evaluation issues related to VoIP applications over different technologies. The authors in [12] presented and compared two queue management mechanisms with a time and space priority mechanism for an end user in HSDPA. Those mechanisms are used to manage access packets in the queue giving priority to the Real Time packets and avoiding the Non Real Time packet loss. A discrete time Markov chain is formulated by considering MMPP as the traffic source of RT packets. Numerical results obtained show the positive impact of the AQM on the performance parameters of NRT packets. In [13], the authors compared the performance of Mixed Traffic Scheduler algorithms and identify one which suitably trades off RT (Real-Time) as well as nRT (non Real- Time) traffic capacity with user perceived Quality of Service, for any ratio of VoIP and video users in a cell. They proposed two MTS algorithms which treat voice and video traffic. These are: dynamic packet scheduling and a new limited Joint Time Frequency (JTF) scheduling. They compared these algorithms with known ones like Modified Largest Weighted Delay First [14] and SP-Dynamic [15]. These are compared for three different cases when number of video users is very high, when number of Voice over IP users is high, and when the number of video and VoIP users are balanced.

The performance of WiMAX and UMTS for VoIP traffic is evaluated by Jadhav *et al.* [16] using the OPNET simulator. This evaluation is restricted to a low number of simultaneous users. The main focus is the comparison between WiMAX and UMTS, leaving the comparison between different QoS service classes. They analyzed several performance metrics such as jitter, delay and packet loss, but also the Quality of Experience (QoE) perceived by the end user, through the Mean Opinion Score (MOS). This study showed that WiMAX is better than UMTS to support VoIP. The work in [17] analyzes the efficiency of resource utilization and VoIP capacity in IEEE 802.16e. It is stated that ertPS service introduced by the IEEE 802.16e standard is more appropriate than UGS and rtPS for VoIP services with variable data rate and silence suppression. Authors in [18] analyzed the QoS of VoIP services using three different traffic models: the cross-layer analytical model, M/G/1(m) model, and the IEEE 802.16e/m scheduling approach to evaluate the blocking probability and throughput of the VoIP services of the cognitive radio. By comparing the VoIP performance under the cross-layer analytical model, M/G/1(m), and the IEEE802.16e/m traffic, they have proved that the relative performance of the three traffic models is sensitive to the queue length. In [19], the performance of VoIP application is analyzed for a cognitive radio system using two state MMPP model. Various numerical and simulation parameters, such as packet dropping probability and VoIP capacity are demonstrated. These results conclude that the VoIP capacity is determined by bottleneck-link, which can be different according to system parameters. In [20], the authors proposed a Discrete-Time Markov Chain (DTMC) framework based on a MMPP traffic model to analyze the performance of VoIP traffic. Through the DTMC based on MMPP, they analyzed and demonstrated various performance parameters, such as average packet dropping probability, average throughput, and average queue length. In the analytical model presented in this paper, the signaling overhead is considered in the evaluation of the performance of VoIP services in the IEEE 802.16e OFDMA system. The authors in [21], evaluated the performance of WiMAX for VoIP by varying number of nodes failure. The performance is analyzed by using OPNET Modeller tool. The performance of VoIP is compared in terms of end-to-end delay, throughput, and traffic sent. The results showed that with increase in nodes failure end-to-end delay increases and throughput and traffic Sent decreases. Feng *et al.* [22] proposed a POMDP (Partially Observable Markov Decision Process) based Sleep Window Determination (PSWD) approach for improving the performance of sleep mode operation of IEEE 802.16m. Simulations results show that the proposed PSWD scheme outperforms conventional IEEE 802.16e and IEEE 802.16m corresponding to various traffic demands and satisfies respective delay constraints at the same time. A quality of service mechanism based on concepts of fuzzy logic for scheduling different traffic classes in WiMAX networks is introduced in [23]. The proposed mechanism works out new weight value for different queues adaptively by exploring three input parameters at simulation time: amount of real time and non real time traffic in queues, throughput requirement for non-real time flows, and latency requirement of real time data. The proposed framework simplifies fair allocation of resources to real as well as non real time in queues of SS together with latency and throughput requirements.

Dai *et al.* [24] proposed a simple enhancement of the bandwidth request messages in IEEE 802.16 for supporting voice traffic. They put forwarding resource allocation and scheduling schemes for use under real-time traffic conditions. The obtained results show that the proposed bandwidth request and scheduling systems achieve remarkably lower packet loss probability than the standard IEEE 802.16 bandwidth request with round robin scheduling. Capacity of VoIP traffic in a CRN with imperfect spectrum sensing is studied in [25]. They modelled the VoIP traffic as MMPP; channel as two state Markov chain. However, the authors have concentrated more on finding the minimum target-detection and false alarm probabilities. They demonstrated various analytical and simulation parameters, such as the average throughput, packet dropping probability, and VoIP capacity. In [26], the authors compared QoS, throughput and mean waiting time of traffic of IEEE 802.16e and IEEE 802.11 WLAN to observe the impact of fading on the networks. The performance parameters of IEEE 802.11 WLAN are evaluated employing Giuseppe Bianchi state transition chain. A mathematical model of VoIP application over wireless channel under IEEE 802.16e WLAN is analyzed under Rayleigh and Nakagami-m fading cases with the help of MMPP and DTMC model. Due to immunity of fading, the performance of video and data integrated traffic of wired LAN is better than both of IEEE 802.16e and IEEE 802.11. In [27], the authors proposed an M/M/n/n+K performance model of the voice application of IEEE 802.16e under femto cellular network, the QoS is measured based on throughput, mean queue length and blocking probability. The steady probability states of the proposed model are compared with the existing model of WiMAX under Rayleigh fading environment. The profile of blocking probability, normalized throughput and mean queue length are shown against the length of packet. Authors in [28] proposed and developed enhanced WiMAX uplink traffic scheduling algorithm. The performance parameters are average throughput, average delay, missed deadline ratio and average queue size utilization ratio. The results acquired from the performance analysis have proven the importance of a check-and-balance system achieved across all computed performance metrics. The authors in [29] proposed a graded priority-based call admission control algorithm for LTE and WiMAX networks. In this algorithm, the number of permissible connections shall be segregated into two classes and the new connection shall be graded, prioritized, and admitted based on the device requesting the connection. Simulation results show that graded priority-based admission control algorithm improves a higher connection admission rate for select users compared to non-priority users.

In this paper, an analytical model is presented to evaluate the performance of VoIP packets in a single cell IEEE 802.16e networks. A Continuous Time Markov Chain (CTMC) is formulated by considering two-state MMPP as the traffic source of VoIP packets. An AQM is used to enhance the performance of VoIP packets. Our main objective is to compare two queue management mechanisms so as to show the utility of the AQM mechanisms to enhance the QoS for an end user in a single cell IEEE 802.16e networks.

III. Quality of Service support in IEEE 802.16e

In any mobile broadband network, users generate different types of applications, including voice, video, email, ftp, and browsing content. Each type comes with its own requirements for QoS. From the end users point of view, the QoS of the service that the user has requested is perceived by the users experience in relation to a particular application. From the technical point of view, this duration results from a complex interaction of factors like throughput, packet delay, and residual bit error ratio. Similarly, the quality of a Voice over IP (VoIP) call is perceived by the end users in terms of delay and voice quality.

The IEEE 802.16e Media Access Control (MAC) layer provides differential Quality of service (QoS) for various classes of services, which are Unsolicited Grant Service (UGS), Extended Real-Time Polling Service (ertPS), Real-Time Polling Service (rtPS), Non-real-time Polling Service (nrtPS), and Best Effort (BE) [30], [31]. The QoS parameters and the supporting application types associated with each classes of the IEEE 802.16e are presented as follows [32], [33]:

- Unsolicited Grant Service (UGS): Supports the real-time constant bit rate (CBR) applications such as T1/E1 and VoIP that generate fixed-size packets at periodic intervals. Unsolicited applications grants are allocated to eliminate the overhead and latency of the request/grant process. VoIP without silence suppression is an example of application that is cartegorized as UGS.
- real-time Polling Service (rtPS): Supports the real-time services that generate variable-size packets on a periodic basis, such as MPEG (Motion Pictures Experts Group) video. An MPEG video categorised as rtPS traffic, needs variable bandwidth at periodic intervals of time to avoid jitter while viewing the video [34]. In this service, the Base Station (BS) provides unicast polling opportunities for the Mobile Station (MS) to request bandwidth.
- extended real-time Polling Service (ertPS): Combines features from UGS and rtPS service classes, supports real-time service flows that generate variable-sized data packets at periodic intervals. VoIP with silence suppression is an example traffic categorised as ertPS.
- non-real-time Polling Service (nrtPS): Is the best appropriate for the delay tolerant applications. As in rtPS, dedicated periodic slots are used for the bandwidth request opportunity, but with much longer periods [35]. In nrtPS, it is allowable to have unicast polling opportunities, but the average duration between two such opportunities is in the order of a few seconds, which is large compared to rtPS. All the MSs which belong to the group can also request resources during the contention-based polling opportunity, which can often result in collisions and additional attempts.
- Best Effort (BE): Provides very little QoS support and is applicable only for services that do not have strict QoS requirements. It is for the traffic with no minimum level of service requirements such as Web browsing or email. Like in nrtPS, contention slots are used for bandwidth request opportunities as long as there is space available.

When a user generates packets, these packets are placed in the queue designated for the type of packets. If a connection does not exist for the service class, the MS sends a request to establish a connection for the data. The connection request contains the QoS parameters that the connection expects. When the BS receives the connection request, it checks if it can service the QoS requirement of the connection. If the connection is accepted, the BS should be able to serve the connection for its quality of service requires [29].

IV. System model

A. Model description

We consider a system model based on a mobile TDD-OFDMA system. Our system is composed on an UpLink (UL) VoIP transmission from a Subscriber Station (SS) to Base Station (BS) in an OFDMA access mode using single carrier air-interface based WiMAX system. Each subscriber station serves multiple users.The base station may allocate different number of subchannels to different SS. For example, a SS with higher priority could allocate more number of subchannels. The TDD-based WiMAX system is operated on a frame basis, where each frame consists of a downlink DL subframe and an UL subframe. The DL subframe consists of a preamble, a frame control header (FCH), data bursts, a DL-MAP message, and an UL-MAP message. By broadcasting a MAP message, the BS indicates the location, size, and encoding of data bursts [35]. The duration of a frame is denoted by T_f.

B. System model without Feedback Control

In this system (Figure 1), for each uplink subchannel, a separate single queue is used for buffering the VoIP packets from SS to BS. The queueing discipline is First-In First-Out (FIFO).The size of this queue is finite (i.e.,some packets will be dropped if the queue is full upon their arrivals).

C. System model with Feedback Control

In this system (as shown in Figure 2), we add two thresholds L_1 and L_2 ($L_1 < L_2 < N$ such that $L_2 = N-L_1$) in the queue in order to control the arrival rate of the VoIP packets [12], [36]. When the number of VoIP packets in the buffer is less than the minimum threshold (L_1), there is no dropping and the source operates normally. If the number of packets exceeds the maximum threshold (L_2), then the excess packets will be dropped. However, we assume that the source stops sending packets when a full buffer is detected (L_2 packets). Packet transmission can commence after the next departure. In this way the packet loss due to buffer overflow might be avoided. If the number of VoIP packets in the system falls between the L_1 and L_2, then the arrival rate of VoIP packets is reduced. As in [37], the key idea for the controller is to calculate the mean queue length and the mean arrival rate over each frame time duration T_f. The mean arrival rate of VoIP packets is measured by counting the number of VoIP arrivals within each frame time and dividing it over the frame time length. These measurements are used to calculate the new position of the threshold L_2 for the next frame time duration T_{f+1} in order to maintain the mean queue length at the required value.

D. Channel model

We consider a Nakagami-m channel model in which the channel quality is determined by instantaneous received Signal-to-Noise Ratio (SNR) γ in each time slot [38]. With adaptive modulation, the SNR at the receiver is divided into $N+1$ non-overlapping intervals (i.e., $N = 7$ in the IEEE 802.16 standard) by threshold $\Gamma_n (n \in 0, 1, ..., N)$ where $\Gamma_0 < \Gamma_1 < ... < \Gamma_N = \infty$. The channel is said to be in state n (i.e., rate ID_nwill be used) if $\Gamma_n < \gamma < \Gamma_{n+1}$. To avoid possible transmission error, no packet is transmitted when $\gamma < \Gamma_0$. Note that, these thresholds correspond to the required SNR specified for VoIP traffic as shown in Table I [39].

With Nakagami-m fading, the probability of using ID_n (i.e. $P_\gamma(n)$) is given by:

$$P_\gamma(n) = \frac{\Gamma(m, \frac{m\Gamma_n}{\overline{\gamma}}) - \Gamma(m, \frac{m\Gamma_{n+1}}{\overline{\gamma}})}{\Gamma_m} \quad (1)$$

where $\overline{\gamma}$ is the average SNR, m is the Nakagami fading parameter ($m \geq 0.5$), $\Gamma(m)$ is the Gamma function which equals $\Gamma(m) = \int_0^\infty t^{m-1} exp(-t)\,\mathrm{d}t$, and $\Gamma(m, \gamma)$ is the complementary incomplete Gamma function which equals $\Gamma(m, \gamma) = \int_\gamma^\infty t^{m-1} exp(-t)\,\mathrm{d}t$.

The Wireless channel is described by an $M/M/1/N$ finite state Markov chain taking the discrete AMC into consideration [40]. The state transition probability of the Modulation and Coding Scheme (MCS) level during the frame duration is given for $j \in \{0, ..., N\}$ by [41]:

$$P_t(i,j) = \begin{cases} \frac{N_{i+1}T_f}{P_\gamma(i)},\ if\ j = i+1, \\ \frac{N_i T_f}{P_\gamma(i)},\ if\ j = i-1, \\ 1 - P_t(i,i+1) - P_t(i,i-1),\ if\ j = i, \\ 0, \quad otherwise. \end{cases} \quad (2)$$

where i is the MCS level in the current frame and j is the MCS level in the next frame. The level crossing rate, N_i, is defined as [42]:

$$N_i = \sqrt{2\pi \frac{m\gamma_i}{\overline{\gamma}}} \frac{f_d}{\Gamma(m)} \left(\frac{m\gamma_i}{\overline{\gamma}}\right)^{m-1} exp\left(-\frac{m\gamma_i}{\overline{\gamma}}\right) \quad (3)$$

where f_d is the maximum Doppler shift given in hertz.

E. Arrival VoIP traffic model

VoIP traffic has low data rates (in the order of tens of Kbit/s) and exhibits low burntness. Because these stringent requirements and particular characteristics, VoIP traffic should be treated differently than other traffic in the network. VoIP traffic at source level is characterized by an active period followed by inactive period. During the active period, the source sends packets at regular intervals. In WiMAX networks, the VoIP packets are assumed to be scheduled from the uplink queue in accordance with the FIFO policy for every frame [43]. Each VoIP packet uses the AMC scheme at the physical layer. The uplink MCS levels are determined by the BS in relation to the quality of the signal received from each SS. In [44], the arrivals process of the VoIP traffic has been modeled as an exponentially distributed on-off model with a mean on-time of $\alpha^{-1} = 352\,ms$ and a mean off-time of $\beta^{-1} = 650\,ms$. We use the two-state MMPP to model the aggregate VoIP traffic requested frome active voice users, as shown in Figure 2. The MMPP processes are very suitable for formulating the multiuser VoIP traffic and capturing the interframe dependency between consecutive frames [45]. The two-state MMPP are characterized by the arrival Poisson rates and the transition rates between them. The probability that the status of the users is inactive (= off) in the simple on-off model can be obtained by $p_{off} = \beta^{-1}/(\alpha^{-1} + \beta^{-1})$, and $p_{on} = 1 - p_{off}$. The transition rate matrix (R) and the Poisson arrival rate matrix (A) of the two-state MMPP process can be expressed as follows:

$$R = \begin{pmatrix} -\delta_0 & \delta_0 \\ \delta_1 & -\delta_1 \end{pmatrix},\ A = \begin{pmatrix} \lambda_0 & 0 \\ 0 & \lambda_1 \end{pmatrix} \quad (4)$$

The average arrival rate for the VoIP traffic at the queue during the frame duration T_f, denoted λ, is given by:

$$\lambda = p_0\lambda_0 + p_1\lambda_1 = \frac{\delta_1\lambda_0 + \delta_0\lambda_1}{\delta_0 + \delta_1} \quad (5)$$

where p_m is the probability that the process is in phase $m, (m = 0, 1)$.

For simplicity, we use the Index of Dispersion for Counts (IDC) matching technique to determine the four parameters of the two-state MMPP as follows [46]:

$$\lambda_0 = A\frac{\sum\limits_{j=0}^{M_v} j\pi_j}{\sum\limits_{i=0}^{M_v} \pi_i},\ \lambda_1 = A\frac{\sum\limits_{j=M_v+1}^{N_v} j\pi_j}{\sum\limits_{i=M_v+1}^{N_v} \pi_i} \quad (6)$$

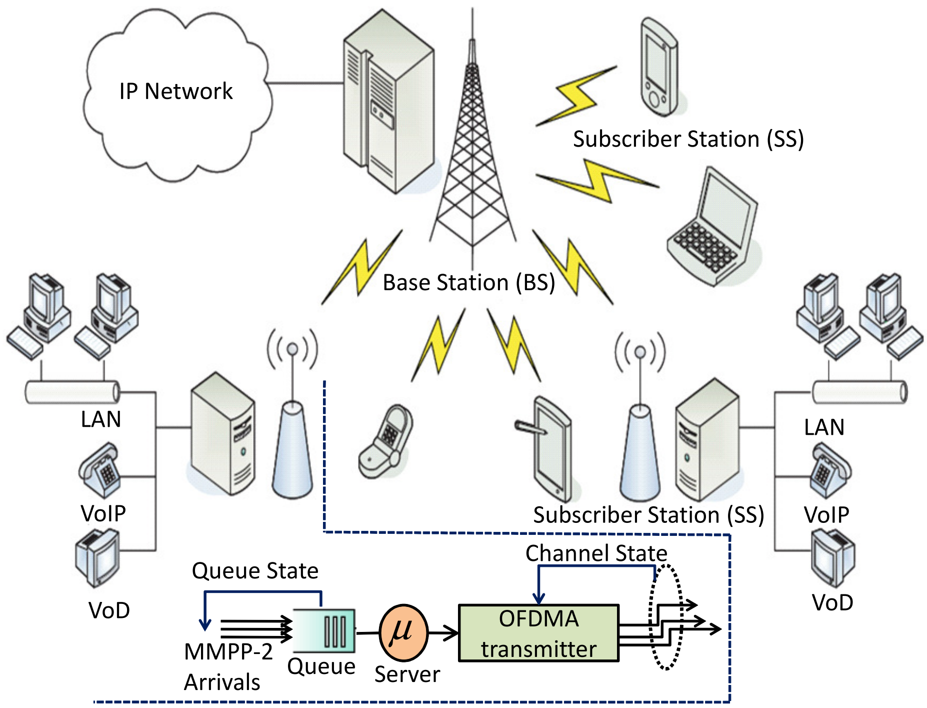

Fig. 1. System model without Feedback Control.

TABLE I
MODULATION AND CODING SCHEMES FOR VOIP TRAFFIC

Rate ID	Modulation Level (Coding)	Information Bits/Symbol	Required SNR (db)
0	BPSK (1/2)	0.5	6.4
1	QPSK (1/2)	1	9.4
2	QPSK (3/4)	1.5	11.2
3	16QAM (1/2)	2	16.4
4	16QAM (3/4)	3	18.2
5	64QAM (2/3)	4	22.7
6	64QAM (3/4)	4.5	24.4

where $\pi_j = \binom{N_v}{j} p_{on}^j (1-p_{on})^{N_v - j}$, $M_v = \lfloor N_v.p_{on} \rfloor$, and A, which is the emission rate in the active state, equals $1/T_f$. The transition rates are given as follows:

$$\delta_0 = \frac{2(\lambda_1 - \lambda_{avg})(\lambda_{avg} - \lambda_0)^2}{(\lambda_1 - \lambda_0)\lambda_{avg}(IDC(\infty) - 1)} \tag{7}$$

$$\delta_1 = \frac{2(\lambda_1 - \lambda_{avg})^2(\lambda_{avg} - \lambda_0)}{(\lambda_1 - \lambda_0)\lambda_{avg}(IDC(\infty) - 1)} \tag{8}$$

where $\lambda_{avg} = N_v.A.p_{on}$ and $IDC(\infty)$ is given as follows [47].

$$IDC(\infty) = 1 + \frac{2(\lambda_0 - \lambda_1)^2 \delta_0 \delta_1}{(\delta_0 + \delta_1)^2(\lambda_0\delta_1 - \lambda_1\delta_1)} \tag{9}$$

V. QUEUEING MODEL AND PERFORMANCE PARAMETERS

A. System without Feedback Control

To model this system, we use a queueing system $MMPP-2/M/1/N$ characterized by a two states MMPP arrival process with parameters λ_0, λ_1, δ_0, and δ_1, and exponentially distributed service time with parameter μ. The inter-arrival times are exponential and all these variables are mutually independent between them, the corresponding state transition diagram is shown in Figure 3.

The state of the system is described by the two dimensional process $Y_t = (S_t, X_t)$, where S_t is the state (phase) of an irreducible Continuous Time Markov Chain (CTMC), X_t is the number of VoIP packets in the queue at the end of every frame. Thus, the state space of the system is given by:

$$E = \{(s,x)/s \in \{0,1\}, 0 \le x \le N\} \tag{10}$$

From the Figure 3 we observe that if lexicographic ordering of the states is used, then the infinitesimal generator $2N \times 2N$ matrix is $Q = [Q_{i,j}]$, where i and j are two-dimensional vectors, is given by [48]:

$$Q = \begin{pmatrix} D_0 & A_1 & 0 & 0 & 0 & 0 & \dots & 0 & 0 & 0 \\ A_2 & A_0 & A_1 & 0 & 0 & 0 & \dots & 0 & 0 & 0 \\ 0 & A_2 & A_0 & A_1 & 0 & 0 & \dots & 0 & 0 & 0 \\ 0 & 0 & A_2 & A_0 & A_1 & 0 & \dots & 0 & 0 & 0 \\ \vdots & \vdots & \vdots & \vdots & \vdots & \vdots & \vdots & \vdots & \vdots & \vdots \\ 0 & 0 & 0 & 0 & 0 & \dots & 0 & A_2 & A_0 & A_1 \\ 0 & 0 & 0 & 0 & 0 & 0 & \dots & 0 & A_2 & D_1 \end{pmatrix} \tag{11}$$

where

$$D_0 = \begin{pmatrix} -(\lambda_0 + \delta_0) & \delta_0 \\ \delta_1 & -(\lambda_1 + \delta_1) \end{pmatrix} \tag{12}$$

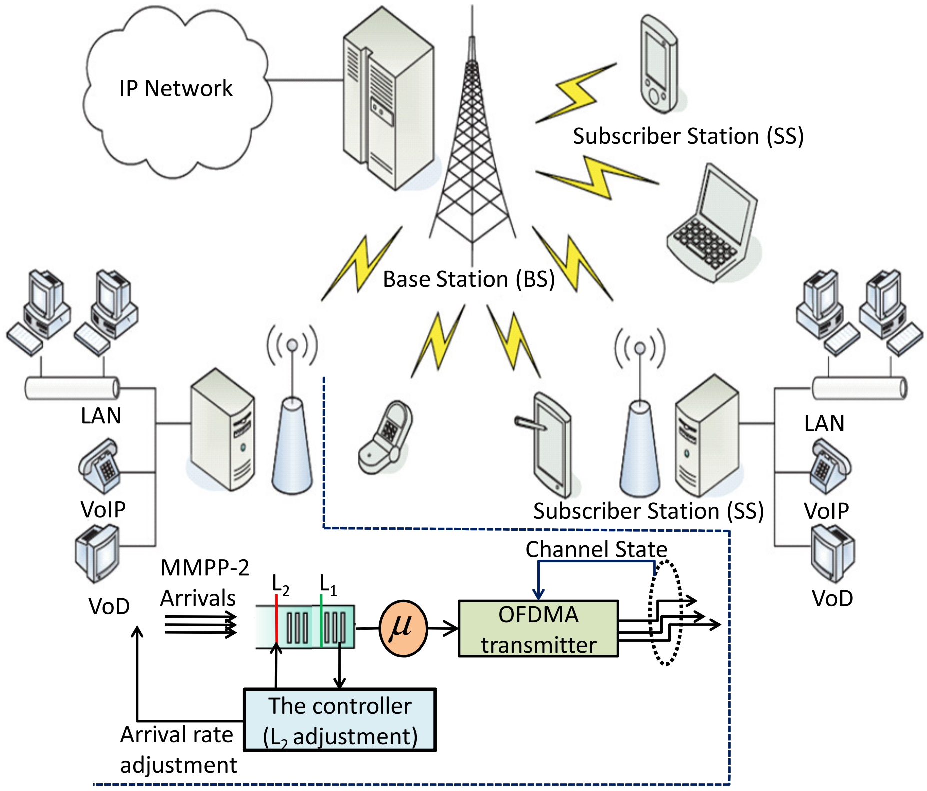

Fig. 2. System model with Feedback Control

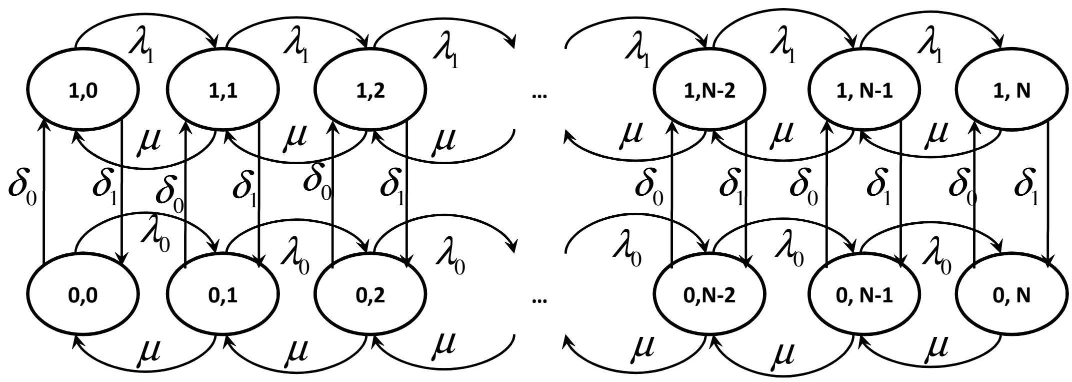

Fig. 3. State transition diagram for system model without Feedback Control

$$D_1 = \begin{pmatrix} -(\delta_0+\mu) & \delta_0 \\ \delta_1 & -(\delta_1+\mu) \end{pmatrix} \tag{13}$$

$$A_0 = \begin{pmatrix} -(\lambda_0+\delta_0+\mu) & \delta_0 \\ \delta_1 & -(\lambda_1+\delta_1+\mu) \end{pmatrix} \tag{14}$$

$$A_1 = \begin{pmatrix} \lambda_0 & 0 \\ 0 & \lambda_1 \end{pmatrix} \tag{15}$$

$$A_2 = \begin{pmatrix} \mu & 0 \\ 0 & \mu \end{pmatrix} \tag{16}$$

The steady state probability π of the system is expressed as follows:

$$\pi = [\pi_{0,0}, \pi_{1,0}, \pi_{0,1}, \pi_{1,1}, ..., \pi_{0,N-1}, \pi_{1,N-1}, \pi_{0,N}, \pi_{1,N}] \tag{17}$$

where $\pi_{s,x}$ values can be obtain by solving the following finite set of steady-state equations [49]:

$$\pi Q = 0, \ \sum_{s=0}^{1}\sum_{x=0}^{N} \pi_{s,x} = 1 \tag{18}$$

Using stochastic balance equations, we obtain the steady-state probability as :

$$\begin{cases} \pi_{0,0} = \frac{\pi_{0,1}\mu+\pi_{1,0}\delta_1}{\lambda_0+\delta_0} \\ \pi_{1,0} = \frac{\pi_{1,1}\mu+\pi_{0,0}\delta_0}{\lambda_1+\delta_1} \\ \pi_{0,i} = \frac{\pi_{0,i+1}\mu+\pi_{1,i}\delta_1+\pi_{0,i-1}\lambda_0}{\lambda_0+\delta_0+\mu}, \quad i=1,...,N-1 \\ \pi_{1,i} = \frac{\pi_{1,i+1}\mu+\pi_{0,i}\delta_0+\pi_{1,i-1}\lambda_1}{\lambda_1+\delta_1+\mu}, \quad i=1,...,N-1 \\ \pi_{0,N} = \frac{\pi_{0,N-1}\lambda_0+\pi_{1,N}\delta_1}{\mu+\delta_0} \\ \pi_{1,N} = \frac{\pi_{1,N-1}\lambda_1+\pi_{0,N}\delta_0}{\mu+\delta_1} \end{cases} \tag{19}$$

We then calculate the performance parameters as follows. First, to obtain the loss probability P_{loss} we again notice that $\pi_N = \pi_{0,N} + \pi_{1,N}$ is the proportion of time that the buffer is full. The proportion of VoIP packets that are lost is therefore, the ratio of the number of VoIP packets arrive during the frame time duration T_f that the buffer is full to the total number of VoIP packets that arrive. Therefore, the loss probability can be obtained as follows:

$$P_{loss} = \frac{\lambda_0 \pi_{0,N} + \lambda_1 \pi_{1,N}}{\lambda} \tag{20}$$

We compute $E[N]$, the mean number of current VoIP packets in the system as:

$$E[N] = \sum_{s=0}^{1} \sum_{x=0}^{N} x \pi_{s,x} \tag{21}$$

The throughput measures the number of packets transmitted in one frame. It can be obtained from:

$$\phi = \lambda(1 - P_{loss}) \tag{22}$$

Finally, the average VoIP packet delay is defined as the number of frames that a packet waits in the buffer (queue) since its arrival before it is transmitted. From Littles law, we can obtain the average delay as follows:

$$D = \frac{E[N]}{\lambda(1 - P_{loss})} \tag{23}$$

B. System with Feedback Control

The queueing model used in this system can be considered as a modification of a $MMPP - 2/M/1/N$ queue. The state transition diagram for the proposed system is shown in Figure 4. The first threshold L_1 is fixed and the second threshold L_2 can be adjusted to any position in the queue.

Let k be the total number of VoIP packets in the queue at time t.

- If $0 \le k < L_1$, then the arrival rate of the VoIP packets is λ_0 in state 0 (MMPP is in state 0) and λ_1 in state 1 (MMPP is in state 1).
- If $L_1 \le k < L_2$, then the arrival rate of the VoIP packets is reduced to $\lambda_{0L_1} = \lambda_0/2$ in state 0 (MMPP is in state 0) and $\lambda_{1L_1} = \lambda_1/2$ in state 1 (MMPP is in state 1).
- If $k \ge L_2$, then no VoIP packets arrives in the queue.

For this system model, the state of the system is described at time $t (t \ge 0)$ by the stochastic process $Z_t = (U_t, V_t)$, where U_t is the phase of the MMPP and V_t is the number of the VoIP packets in the queue at time t. The state space of Z_t is given by:

$$F = \{(u, v)/u \in \{0, 1\}, 0 \le v \le L_2\} \tag{24}$$

Due to the use of two thresholds in this system model, the queue needs to be considered in two parts in order to calculate the steady-state probabilities. Using the same analysis, the steady-state probabilities can be expressed as a solution of the balance equations as follows:

$$\begin{cases}
\pi_{0,0} = \frac{\pi_{0,1}\mu + \pi_{1,0}\delta_1}{\lambda_0 + \delta_0} \\
\pi_{1,0} = \frac{\pi_{1,1}\mu + \pi_{0,0}\delta_0}{\lambda_1 + \delta_1} \\
\pi_{0,i} = \frac{\pi_{0,i+1}\mu + \pi_{1,i}\delta_1 + \pi_{0,i-1}\lambda_0}{\lambda_0 + \delta_0 + \mu}, \;\; i = 1, ..., L_1 - 1 \\
\pi_{1,i} = \frac{\pi_{1,i+1}\mu + \pi_{0,i}\delta_0 + \pi_{1,i-1}\lambda_1}{\lambda_1 + \delta_1 + \mu}, \;\; i = 1, ..., L_1 - 1 \\
\pi_{0,L_1} = \frac{\pi_{0,L_1-1}\lambda_0 + \pi_{1,L_1}\delta_1 + \pi_{0,L_1+1}\mu}{\lambda_{0L_1} + \delta_0 + \mu} \\
\pi_{1,L_1} = \frac{\pi_{1,L_1-1}\lambda_1 + \pi_{0,L_1}\delta_0 + \pi_{1,L_1+1}\mu}{\lambda_{1L_1} + \delta_1 + \mu} \\
\pi_{0,k} = \frac{\pi_{0,k+1}\mu + \pi_{1,k}\delta_1 + \pi_{0,k-1}\lambda_0}{\lambda_{0L_1} + \delta_0 + \mu}, \;\; k = L_1 + 1, ..., L_2 - 1 \\
\pi_{1,k} = \frac{\pi_{1,k+1}\mu + \pi_{0,k}\delta_0 + \pi_{1,k-1}\lambda_{1L_1}}{\lambda_{1L_1} + \delta_1 + \mu}, \;\; k = L_1 + 1, ..., L_2 - 1 \\
\pi_{0,L_2} = \frac{\pi_{0,L_2-1}\lambda_{0L_1} + \pi_{1,L_2}\delta_1}{\mu + \delta_0} \\
\pi_{1,L_2} = \frac{\pi_{1,L_2-1}\lambda_{1L_1} + \pi_{0,L_2}\delta_0}{\mu + \delta_1}
\end{cases} \tag{25}$$

The performance parameters are derived from the steady-state probabilities as follows. The loss probability of VoIP packets P_{CF}^{loss} is defined as the flow rate of a VoIP packet lost on the total flow rate and is given by the following formula:

$$P_{CF}^{loss} = \frac{\lambda_{0L_1}\pi_{0,L_2} + \lambda_{1L_1}\pi_{1,L_2}}{\lambda_{eff}} \tag{26}$$

Where λ_{eff} is the effective arrival rate of VoIP packets in this system. It is computed by the following formula:

$$\lambda_{eff} = \sum_{s=0}^{1} \lambda_s \sum_{x=0}^{L_1} x\pi_{s,x} + \sum_{s=0}^{1} \frac{\lambda_s}{2} \sum_{n=L_1+1}^{L_2} n\pi_{s,n} \tag{27}$$

We can compute $E_{CF}[N]$, the mean total number of VoIP packets in the system is given by:

$$E_{CF}[N] = \sum_{s=0}^{1} \sum_{x=0}^{L_1} x\pi_{s,x} + \sum_{s=0}^{1} \sum_{n=L_1+1}^{L_2} n\pi_{s,n} \tag{28}$$

The throughput can be obtained from:

$$\phi_{CF} = \lambda_{eff}(1 - P_{CF}^{loss}) \tag{29}$$

Finaly, from Little's law formula, we can obtain the average delay of VoIP packets in the system as follows:

$$D_{CF} = \frac{E_{CF}[N]}{\lambda_{eff}(1 - P_{CF}^{loss})} \tag{30}$$

VI. NUMERICAL RESULTS AND ANALYSIS

In this section we present the numerical results of both systems. We use the Matlab software to numerically solve and evaluate various performance parameters. The performance parameters are measured respectively under different traffic intensities with channel SNR in the range of rate *ID = 0*, and under different channel qualities with constant traffic intensity.

According to the data reported in this work, the maximum data rate capacity of the WiMAX channel was $20Mbps$. Average SNR on each subchannel is $5dB$. The carrier frequency was $2.6GHz$ with a bandwidth equal to $12MHz$, number of sub-carriers was 2048 and the modulation Level was $QPSK(1/2)$. The VoIP traffic can be modeled by a two-state MMPP model, namely transition rate δ_0, from state 0 to state 1, is 0.17, while the reciprocal transition rate δ_1, from state 1 to state 0, is 0.08. The VoIP arrival rates λ_0

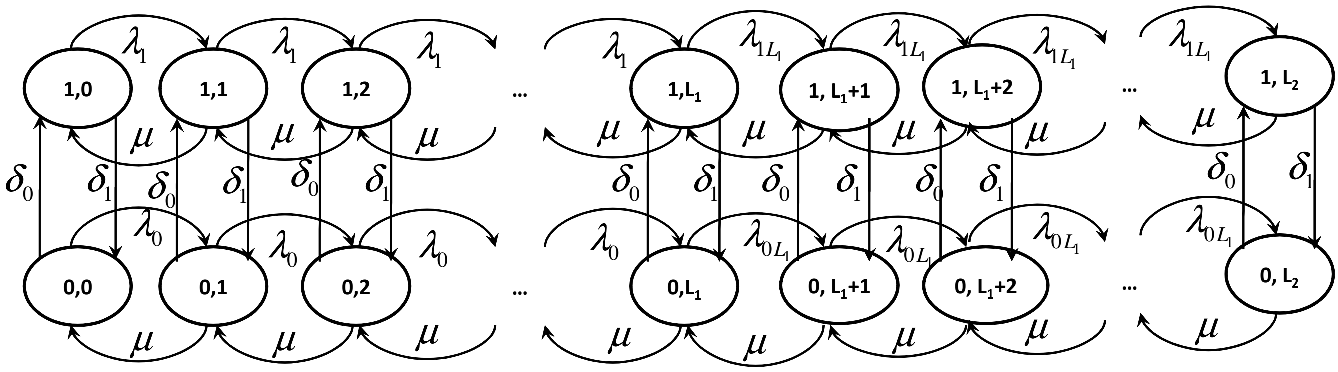

Fig. 4. State transition diagram for system model with Feedback Control

and λ_1, associated to state 0 and 1, are equal, respectively, to 22.1 and 7.16. Bandwidth demand for VoIP application was uniformly distributed in the range of 128 to $512Kbps$. The queue size is 150 VoIP packets (i.e. $N = 150$).

For system with Feedback Control, the value of the threshold L_1 is fixed to 50 VoIP packets (i.e. $L_1 = 50$). Note that, we vary some of these parameters based on the evaluation scenarios whereas the others remain fixed.

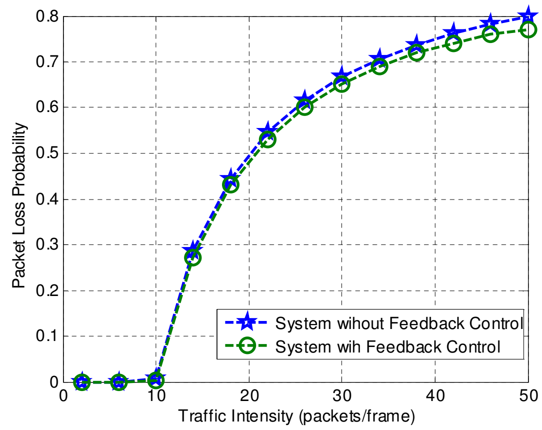

Fig. 5. Packet Loss Probability under Traffic Intensity.

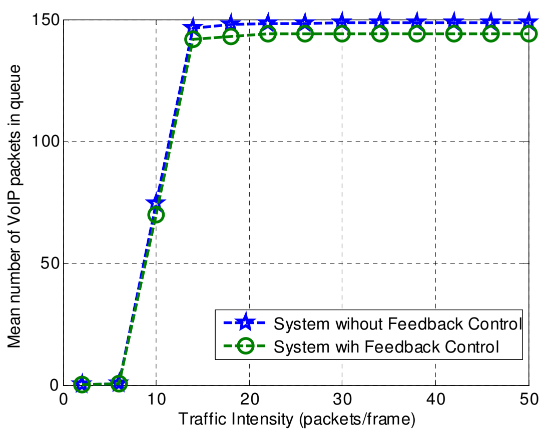

Fig. 6. Mean Number of VoIP Packet in queue under Traffic Intensity.

The performance parameters under different traffic intensity are shown in Figures 5, 6, 7 and 8 for packet loss probability, mean number of VoIP packets in queue, average VoIP packet delay, and throughput, respectively. These performance parameters are significantly impacted by traffic intensity. When the traffic intensity varies and shows that the second system is more effective, especially when traffic intensity is higher. We remark that with lower traffic intensity, both systems have the same performance. Indeed, with lower values the control has no effect as the number of packets in the queue does not exceed the first threshold.

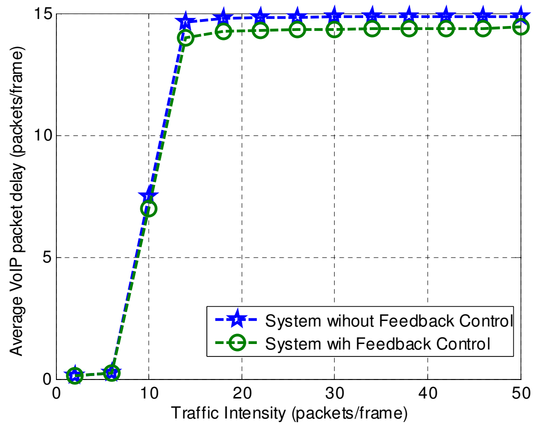

Fig. 7. Average VoIP Packet Delay under Traffic Intensity.

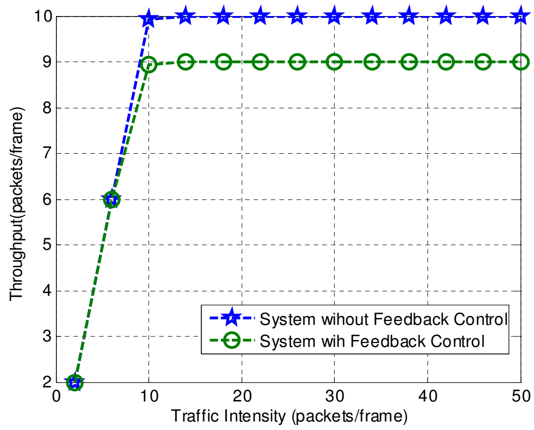

Fig. 8. Throughput under Traffic Intensity.

Variations in packet loss probability and average packet delay under different channel qualities are shown in Figures 9 and 10. We remark that when the SNR increases, the packets

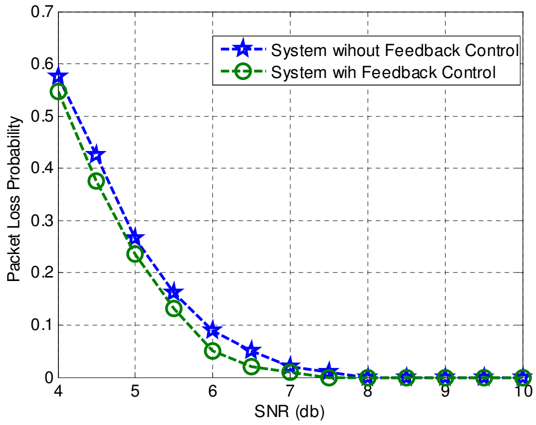

Fig. 9. Packet Loss Probability under different channel qualities.

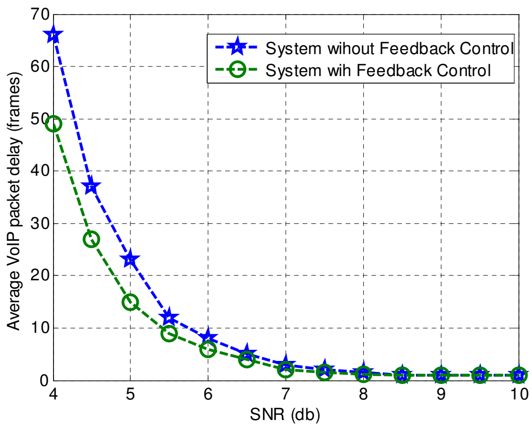

Fig. 10. Packet Loss Probability under different channel qualities.

loss probability and the average delay of the VoIP packets are lower in the second system than in the first system and when SNR is lower, the second system is clearly more effective.

We remark that the second system where the system is combined with an AQM achieves a gain on the loss probability and average delay of VoIP packets. The results show that when the SNR level is high, the mechanism with control and the mechanism without control have the same behavior as the quality of channel is good and all packets can be processed.

VII. Conclusions

In this paper, we have analyzed and compared the performance of VoIP traffic in a single cell IEEE 802.16e using two system models, called system without Feedback Control model, and system with Feedback Control model. The performance parameters of these system models are analyzed in terms of packet loss probability, mean number of VoIP packets in queue, average VoIP packet delay, and throughput. We have considered a WiMAX system model in which a base station serves multiple subscriber stations and each of the subscriber stations is allocated with a certain number of subchannels by the base station. Mathematical tools are used in this study, we have used two-state MMPP processes to model the arrival of VoIP packets in the system, and performance parameters are analytically deducted. By comparing the VoIP performance parameters under the system without Feedback Control model, and the system with Feedback Control model, we showed that the second system model outperforms than the first one.

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