Video Quality Prediction in the Presence of MAC Contention and Wireless Channel Error

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Abstract

In order to provide adequate QoS for both multimedia and data traffic in a wireless network, it is necessary to develop models which can be used to predict the quality of both multimedia delivery (measured by decoded video quality) and data traffic (measured by throughput) in a wireless environment. This paper proposes an analytical model to predict the quality of video, expressed in terms of mean square error of the received video frames, in an IEEE 802.11e wireless network. The proposed model takes into account contention at the MAC layer, wireless channel error, queueing at the MAC layer, parameters of different 802.11e access categories, and video characteristics of different H.264 data partitions. To the best of the authors' knowledge, this is the first model that takes these network and video characteristics into consideration to predict video quality in an IEEE 802.11e network. The proposed model consists of two components. The first component predicts the packet loss rate of each H.264 data partition by using a multi-dimensional discrete-time Markov chain coupled to a M/G/1 queue. The second component uses these packet loss rates and the video characteristics to predict the MSE of each received video frames. We verify the accuracy of our analytical model by using discrete event simulation and real H.264 coded video sequences.

1 Introduction

In the past decade, Wireless Local Area Networks (WLANs) has been considered as a successful technology in telecommunication industry by offering the users convenience orientation towards network setup, network connection and user mobility. As of the WLANs being the mean access scheme for wireless and mobile internet users, there is a massively increasing demand for Quality-of-Service (QoS) support in various types of applications especially real-time multimedia. Providing QoS within wireless network is a mandatory criteria to promote the current state of wireless network to a new generation of wireless multimedia communication. Due to the fact that many advanced multimedia applications has been emerging rapidly starting from video conference, video on demand to real-time video broadcasting and these multimedia applications require different QoS supported in order to fully operated under error-prone wireless network. However, the IEEE 802.11 standard WLANs itself cannot satisfy the requirements of QoS support from these devastating number of newly developed applications. Hence, the IEEE task group has announced a new standard namely, 802.11e that is aimed to provide a traffic-differentiated services at a wide range of user-prioirtiy levels in WLANs. Unfortunately, multimedia applications e.g. video streaming have high dependencies with stringent delay constraint and wireless channel is time varying. Therefore, it is often difficult to foresee the impact of channel conditions towards the quality of multimedia applications and provide an appropriate QoS adaptively right after the change in channel medium.

Although QoS-supported 802.11e for WLANs has been proposed for several years, the impact of its parameters and the operational characteristics have not yet throughly investigated for multimedia applications. To satisfy the users, the delay-sensitive multimedia traffic must be delivered at its highest quality with smallest delay as possible. However, the existing wireless network has limited channel capacity and is time-varying. In fact, wireless communications have been seen challenging due to two fundamental problems which are phenomenon of fading and cross signal interference experienced in a realistic wireless environment. As wireless channel is a shared medium and wireless networks do not have an isolated point-to-point link as in wirelined, the signal strengths can be varied by many factors. On the other hand, video applications is delay-sensitive, long range dependencies and extremely bursty. To provide a stable wireless connection for this bandwidth-intense traffic, it is necessary to develop a system where the performance of both video applications and data traffic can be predicted such that the transmission strategies can be dynamically adapt according to the change of channel condition.

It is seen that the problem of providing suitable QoS in WLANs not only resided in wireless network where fading and interference problems arise but also in the nature of traffic sources. An analytical approach for the evaluation of the WLAN's QoS performance can be an important tool for resource allocation and packet scheduling adaptation algorithm. Without a doubt, provisioning the network performance is an economically viable strategy due to the declining cost of network bandwidth, maximizing throughput efficiency and optimal QoS support. However, developing a complete theoretical performance model is still a current on-going research due to many technical issues. The challenges encountered in modeling the IEEE 802.11e contention-based channel access mechanism are abided to the characteristics of traffic sources e.g. voice, video and ftp, link layer resources management, limited bandwidth and time-varying natured of channel condition.

The main contribution of this paper is to propose an analytical model to predict the fine-grained quality of H.264 [1] videos in an IEEE 802.11e network. The proposed model takes into account contention at the MAC layer, packet loss due to wireless channel, queueing at the MAC layer, parameters of different 802.11e access categories (ACs), as well as the video characteristics of different H.264 data partitions (DPs). The output of the model is the quality of *each* video frame measured in terms of mean square error (MSE). To the best of the authors' knowledge, this is the first model that takes these network and video characteristics into consideration to predict the video quality (on a per-frame basis) in an IEEE 802.11e network.

Our proposed work is different from other people's works in which the video's dependency, unequal error protection (UEP) techniques, error concealment algorithms have been put into consideration along with our developed multi-dimensional discrete-time Markov chain (DTMC) channel throughput estimation model for 802.11e EDCA MAC protocol. Both output video quality and data throughput of each AC can be estimated from the proposed video prediction system. Hence, we can investigate the degradation of the video quality due to error propagation and error concealment at the receiver without employing feedback channel.

The rest of this paper is organized as follows. Section 2 reviews the relevant works in the literature. The overview of channel access mechanism 802.11e EDCA is given in Section 3 whereas Section 4 describes the proposed video quality prediction model. In Section 5, we verify the accuracy of our analytical model by using discrete event simulation and real H.264 coded video sequence. Finally, we conclude the paper in Section 6.

2 Related Works

Much research effort has been spent during the past decade on estimating and interpreting the performance of WLANs and the QoS provided by them. In [2], the author analyzes the behavior of 802.11 protocol assuming that the channel condition is ideal and has finite number of terminals. Their model is able to evaluate the saturation throughput of the WLANs that employs the primary medium access control of 802.11 called distributed coordination function (DCF). Both channel access schemes i.e. basic access and RTS/CTS access mechanisms were included into consideration for performance analysis of the model. The paper claimed that the proposed model is very accurate in predicting the system throughput. An analytical model for finite load is proposed in [3] based on [2]'s work. The article proposed to apply M/G/1 queue model to represent the buffer condition in 802.11 MAC layer as an attempt to enable their model to operate with non-saturated sources. The authors in [3] also acknowledge the solution for fairness problem under multi-rate environments based on their proposed model. The estimate model was compared with the experimental results collected from their 802.11b hardware testbed. In [4] - [5], finite-load models of the 802.11a/b/g

[1] H.264/AVC is the latest video coding stanadard of the ITU Video Coding Expert Group and ISO/IEC Moving Picture Expert Group. It offers a higher compression ratio than earlier standards [1].

DCF are proposed. However, these works do not take into account the traffic differentiated service offered by the recent 802.11e standard.

As QoS being addressed in many literatures and the release of new standard 802.11e [6], the current research studies have been focusing on QoS enhancement using both theoretical and empirical methods. In [7], a survey of potential and future research for QoS-MAC has been done and discussed. Handling time-varying network conditions, adapting to varying application profiles and managing link layer resources are three main technical challenges that are identified in their work. The authors also make a point that doubling the contention window (CW) in an effort to reduce the transmission collisions between stations can in turn increasing unnecessary channel access delay in case of retransmissions are caused by poor channel condition, not by collisions. In addition, the paper highlights some works that attempt to cope this unstable channel condition problem; for example, channel-dependent packet-level tuning (ChaPLeT), adaptive EDCF and adaptive fair EDCF. All of these works share the same common point to reduce the impact of link errors by improving the service differentiation thus the transmission policy is adapted according to the network status. The experimental results in [8] show that EDCA is better than DCF in term of channel access, despite the fact that it also results in higher chance of inter-station collisions. Furthermore, they conclude that $AIFS$ is a superior differentiation mechanism to CW in term of performance since CW will trade off the service differentiation with throughput degradation while $AIFS$ does not.

An investigation into protocol performance for IEEE 802.11e with QoS support using theoretical-derived model has been done in [9]. The analytical model is extended from Bianchi's DCF model [2] to represent the deferring mechanism as well as virtual collision handler (VCH) implemented in new standard 802.11e based on discrete time Markov chain (DTMC) model. Although most of fundamental medium access mechanisms are captured in this work, many impractical assumptions were made in order to simplify their proposed model e.g. ideal channel condition and saturated throughput. Neglecting the time-varying channel condition and traffics' properties can lead to a theoretical-like estimation model that is undesirable in real-world application. Still this work shows many significant contributions and provides us a basis for our remarkable QoS-MAC's performance analysis model. By considering channel condition, non-saturated traffic arrival and video characteristics, we proposed an analytical model that estimate channel utilization and predict decoded video performance based on multi-dimensional DTMC model in which ECDA channel access mechanism, M/G/1 queueing systems are deployed under error-prone wireless network. While [10] has proposed a model with finite packet arrival rate for 802.11e, this model assumes that traffic has a constant bit rate and it therefore does not capture the variable bit rate nature of video traffic. A key distinction between our work and all these previous works is that we have developed an analytical model for 802.11e that captures the MAC contention, wireless channel error and the nature of video traffic (e.g. finite arrival rate, variable bit rate and video characteristics of H.264).

With regard to multimedia applications, there is an extensive literature on studying the performance of video delivery in an IEEE 802.11e networks, see e.g. [11] - [12]. The scope of these studies include the effect of the parameters of different ACs on the quality of video delivery. A key distinction of these works, as compared to ours, is that these works are based either on simulation or testbed experiments, while this paper proposes an analytical model to predict the MSE of each H.264 video frame and PSNR of the entire video sequence in an IEEE 802.11e network.

3 IEEE 802.11e Overview

In this section, we first provide a brief summary of the standard IEEE 802.11e QoS enhancements. This new standard extends the coordination functions from legacy 802.11 MAC i.e. distributed coordination function (DCF) and point coordination function (PCF) by devising a new coordination function called hybrid coordination function (HCF). Two medium access modes are defined in HCF which are enhanced distributed channel access (EDCA) and HCF controlled channel access (HCCA). Our focus will be on EDCA channel access mode that is a QoS support extension of DCF.

3.1 Distributed Coordination Function (DCF)

Generally, to gain access to the wireless medium the transmitting station need to verify that there is no other activities on a shared transmission medium. This access method is called carrier sense multiple access with collision avoidance (CSMA/CA). Build on top of this access method is DCF which is a basic MAC channel access mechanism for wireless medium defined in IEEE 802.11 WLANs. In DCF, a station wishing to transmit has to first listen to the channel for a predetermined amount of time called DCF inter-frame space ($DIFS$) to monitor for any transmission on the channel. If the channel is idle then the station can initiate the transmission. This can be done after a DIFS and a backoff counter randomly select between zero and CW. Otherwise, channel is busy then the station must defer its transmission. It is also worth to note that collision can be incurred if more than one station transmit the data frame simultaneously. In this case all the stations involve in the event must stop their ongoing transmission and backoff procedure for retransmission is provoked.

Initially, CW will be set to CW_{min} and will be doubled every time the retransmission is acquired. This will continue until reaching either maximum value for CW (CW_{max}) or maximum retries limit (M). After a successful transmission attempt, the CW is reset to CW_{min} and the backoff procedure is performed on the station before it can transmit another data frame. Basically, backoff counter is an integer multiple of time slots and is decremented every time that the station senses the channel is idle after DIFS duration. The station can start the transmission when the backoff counter reaches zero.

3.2 Enhanced Distributed Channel Access (EDCA)

The standard defines an additional mechanism to enhance QoS support that lacks in DCF mechanism. In addition, EDCA is designed to offer service differentiation to the incoming traffics by providing 8 levels of user priorities and 4 classes of traffic queue called access categories (ACs). Every AC of a station operate on a first-in-first-out (FIFO) basis and behave according to the DCF access mechanism where each AC behaves like an individual entity. As explained in more detail in [6], the traffic streams are mapped into four different ACs which associate the characteristics of each traffic stream with the level of QoS required by that application. Table 3.1 shows the index mapping of four ACs and the traffic stream supported by them. To transmit a packet, each station must contend to gain access to the medium as in the case of DCF; however instead of waiting for fixed DIFS period, EDCA prefers to use arbitration inter-frame space ($AIFS$) in order to strengthen its adaptability.

Table 3.1: AC mapping

Index	**AC**	**Description**
00	AC_BE	Best effort
01	AC_BK	Background
10	AC_VI	Video
11	AC_VO	Voice

In EDCA, two parameters are employed to differentiate the traffic class which are $AIFS$ and CW. High priority queues such as AC_VI and AC_VO will be assigned with a smaller CW and $AIFS$ so that the channel access probability is higher than the lower priority queues. Although the backoff procedure in EDCA does the same as in DCF, there is a slightly difference in the countdown process. Technically, the backoff counter will be frozen when the channel is sensed to be busy and will be resumed once the channel is idle for DIFS period. The backoff counter in EDCA is designed to reduce the channel access delay by decrementing the backoff counter during the last slot of the $AIFS$. The default values of $AIFSN$, CW and transmission opportunity limit (TXOP Limit) for each AC can be found in Table 3.2.

Table 3.2: Default EDCA parameter values

AC	**AIFSN**	**CWmin**	**CWmax**	**TXOP Limit**	
				FHSS	DSSS
AC_BK	7	$CWmin$	$CWmax$	0	0
AC_BE	3	$CWmin$	$CWmax$	0	0
AC_VI	2	$(CWmin+1)/2-1$	$CWmin$	6.016 ms	3.008 ms
AC_VO	2	$(CWmin+1)/4-1$	$(CWmin+1)/2-1$	3.264 ms	1.504 ms

Another accompaniment that has been added into this standard is virtual collision handler (VCH). VCH is a crucial component that will administer the channel access mechanism internally and bias high priority AC to hold possession of the medium when there are two or more ACs accessing the channel simultaneously. Although VCH is functioned to solve the internal collision incurred among the ACs within a station, the external collision caused by multiple-accessing to the channel of nearby stations still remains. Furthermore, granting high-priority ACs to transmit a data frame over the low-priority ACs may bring forth a new drawback of the algorithm as lower priority ACs will be suffered from starvation as internal collision increases.

4 Proposed Video Quality Prediction Model

Our proposed video quality prediction framework is illustrated in Fig. 4.1. There are two components in this framework: the *802.11e MAC channel block* and the *H.264 video distortion block*. The 802.11e MAC channel block assumes the scenario of an IEEE 802.11e WLAN with N wireless stations (STA). Typically, the 802.11e MAC layer of each STA has 4 traffic-differentiated queues called ACs, denoted by $AC_0, ..., AC_3$. We further assume the identical data and video traffic presented in each STA of the 802.11e WLAN in our scenario.

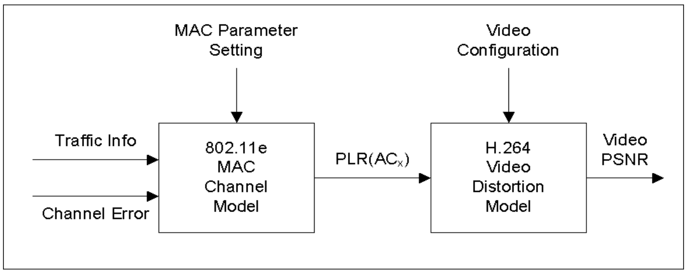

Figure 4.1: Overview of proposed video prediction system based on multi-dimensional DTMC QoS-MAC model.

The right-hand-side of the proposed video prediction system in Fig. 4.1 is the H.264 video distortion model. A key feature of H.264 is that it encodes videos into different data partitions (generally denoted by DP-A, DP-B and DP-C) where each data partition (DP) contains packets of different type of information, e.g. motion vectors, intra-coded blocks, inter-coded blocks etc. Our model assumes that all data frames from a DP are mapped to the same AC in 802.11e, in other words, two data frames from a DP will *not* be mapped to different ACs in 802.11e. (For example, we may have all DP-A packets mapped to AC_3 and all DP-B packets mapped to AC_2. As another example, it is also possible to map both DP-A and DP-B to the same AC.) This is sensible because all packets within a DP have the same importance, therefore they should be given the same level of service. An implication of this assumption is that the packet loss rate (PLR) of a DP is equal to the PLR of the AC to which this DP is mapped. The H.264 video distortion block, proposed by us earlier in [13], requires the PLR of each DP and the characteristics of the video, in order to compute the decoded video's peak signal-to-noise ratio (PSNR) or MSE on a per-frame basis. By simply rearranging different DP into different AC; for instance, DP-A packets into AC_3 queue and DP-B into AC_2 queue, we can eventually match the PLR of DPs with PLR of ACs. Note that AC_0 will have constant traffic bitrate and act as a packet-loss-driven agent.

The 802.11e MAC channel block is our main contribution in this article. It is contructed based on a multi-dimensional discrete-time Markov chain (DTMC) coupled with M/G/1 queueing system. Furthermore, the model takes traffic information (e.g. the *finite* arrival rate of different DPs, the packet size of different DPs), PLR due to wireless channel and MAC parameter setting of all the ACs as the inputs; and yields the throughput and PLR of each AC as the outputs. The proposed DTMC, which follows the operation of IEEE EDCA channel access mechanism, will be thoroughly described in this section. Note that a key difference between our 802.11e MAC channel block and earlier works in modelling the 802.11e MAC layer is that our model considers a *finite* packet arrival rate (rather than saturated traffic) as well as packet loss due to wireless channel (in addition to packet loss due to packet collision). In addition, our model also considers the effect of queueing at each AC and the variable bit rate nature of video traffic, therefore the need to couple our model to an M/G/1 queue.

Note that our proposed video quality prediction model makes two ideal assumptions: (1) The packet arrival to each AC is Poisson distributed. (2) The H.264 video distortion block [13] proposed by us earlier assumes that the packet loss of each DP is Bernoulli distributed. In spite of these ideal assumptions, simulation results in Section 5 show that the our model gives good prediction on video quality.

We will briefly discuss the mechanism of 802.11e EDCA and how our model operate in high level perspective. EDCA starts by inspecting the arrival of packet to the MAC layer. The arrived packet is then classified and sorted into the particular AC queue that the packet is mapped to. The packet at the head of each AC queue will be transmitted to destination node unless the MAC layer senses that the channel is occupied by other STA or other AC queues in the same station. After a destination node has successfully received a transmitted frame, an acknowledgement (Ack) frame will be sent back as a reply. Note that failure of frame transmission due to busy channel, channel error or frame collision will trigger the exponential backoff routine to defer any AC queue from transmission. In the following subsection, all seven states of the DTMC process are introduced and then we show how to derive system equations of the proposed model. Eventually an analysis of the throughput and packet loss rate or each AC is drawn from the system equations.

4.1 Multi-dimensional DTMC Model

We assume that the time is discrete and each time slot unit is equal to *aSlotTime* interval described in the standard [6]. Each time slot may trigger an event that causes the DTMC model to move from one state to another state. Note that each AC of a station performs the same set of operations following the 802.11e standard except that different system parameters are used in different ACs, therefore it is logical to construct a model for one AC and employ the same model for the other ACs with a change of parameter values.

According to the IEEE 802.11e standard [6], the $AIFS$ and CW are the two parameters that provide service differentiation in EDCA and can be calculated using the following equations:

$$\begin{aligned}
AIFS &= SIFS + AIFSN \times aSlotTime && (4.1)\\
CW_{i+1} &= 2CW_i + 1, \quad 0 \le i \le M && (4.2)\\
CW_0 &= CW_{min} && (4.3)\\
CW_i &\le Cw_{max} && (4.4)\\
\therefore CW_i &= 2^i CW_{min} + 2^i - 1, \quad CW_{min} \le CW_i \le CW_{max} && (4.5)
\end{aligned}$$

where $SIFS$ is a multiple of *aSlottime* defined in DCF, $AIFSN$ is deferring number of $AIFS$ (must be an integer greater than 1 for stations and more than

0 for APs) and CW_i is the CW at the i^{th} retransmission attempt with maximum M transmission retries.

From our perspective, the EDCA channel access procedure can be described by using 7 primary states: idle (I), AIFS (AI), backoff (B), transmission (T), collision (C), error (E) and receive state (R). As the future state of the EDCA can be predicted based solely on the current state without the need to consider the previous states, that means the access mechanism has the Markov property and can be modelled using a DTMC. Based on our observation and assumptions, a multi-dimensional DTMC model with finite states can be constructed.

Fig. 4.2 illustrates our proposed DTMC model for EDCA channel access mechanism and Fig. 4.3 shows the details of the backoff procedure within the dashed-line box in Fig. 4.2. Let $S_{i,j,k}$ be the state S (where S is one of the 7 primary states, i.e. S can take on the value of I, AI etc.) in the i^{th} retransmission attempt, j^{th} backoff counter (for B state only) and the k^{th} countdown time for the activity that the station has carried out in state S. We also use $P\{X|Y\}$ to denote the transition probability from state Y to state X. The seven states in this DTMC are described as follows.

1) Idle state (I): This state describes the EDCA when there is no frame to be transmitted. It will remain in the I state as long as there is no reception of data frame from the upper protocol layer. This can be modelled as:

$$P\{I|I\} = 1 - P_a \tag{4.6}$$

where P_a is frame arrival probability. If a new frame arrives during the I state and the channel is not busy (which occurs with a probability $1 - P_b$), then a transmission event is triggered and caused the process to move to T state:

$$P\{T_{0,\lceil Ts \rceil}|I\} = P_a(1 - P_b) \tag{4.7}$$

On the other hand, if the channel is sensed to be busy during the arrival of a new frame then EDCA invokes the backoff procedure, hence

$$P\{B_{0,j,0}|I\} = \frac{P_a P_b}{CW_0 + 1}, \quad 0 \le j \le CW_0 \tag{4.8}$$

where j is a number randomly chosen between 0 and CW_0 (equivalent to CW_{min} due to the operation of backoff mechanism.

2) AIFS state ($AI_{i,k}$): An AC will start a backoff procedure (B state) to iniate a transmission after the channel is idle for AIFS period (refers by subscript A in Fig. 4.2). This is true for any i^{th} retransmission in which M is the maximum retries so we can write:

$$P\{B_{i,j,0}|AI_{i,0}\} = \frac{1 - P_b}{CW_i + 1}, \quad 0 \le i \le M, \;\; 0 \le j \le CW_i \tag{4.9}$$

Unless the channel is busy before AIFS period is ended then the AC will defer and start the countdown once more:

$$\begin{aligned} P\{AI_{i,A}|AI_{i,k}\} &= P_b, \quad 0 \le i \le M, \;\; 0 \le k \le A && (4.10)\\ P\{AI_{i,k}|AI_{i,k+1}\} &= 1 - P_b, \quad 0 \le i \le M, \;\; 0 \le k \le A-1 && (4.11) \end{aligned}$$

3) Backoff state ($B_{i,j,k}$): The backoff routine starts by choosing a random integer number in the $[0, CW_i]$ range where CW_i is the CW of the current

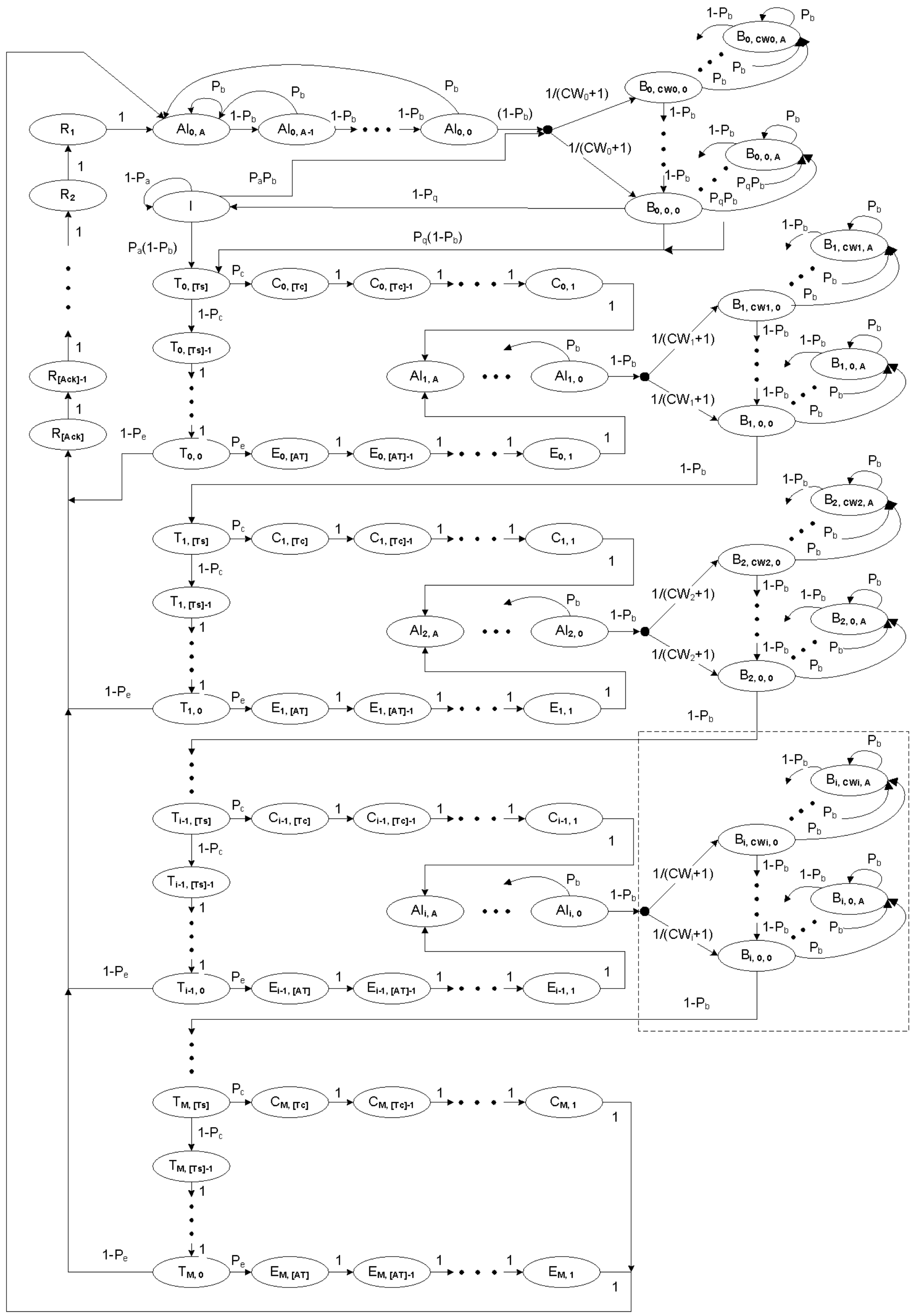

Figure 4.2: Multi-dimensional DTMC model for EDCA under finite-load and channel error.

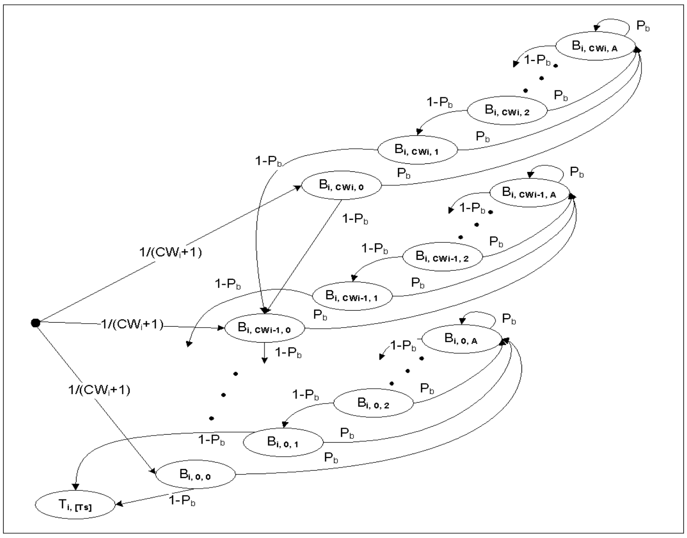

Figure 4.3: detailed backoff stage of the proposed DTMC model.

retransmission that can be computed using Eq. 4.5. This random number is called *backoff counter* and will decrement for each time slot after AIFS period when the channel is sensed to be idle:

$$P\{B_{i,j,k}|B_{i,j,k+1}\} = 1 - P_b, \quad 0 \le i \le M, \ 0 \le j \le CW_i, \ 1 \le k \le A \quad (4.12)$$
$$P\{B_{i,j,0}|B_{i,j+1,1}\} = 1 - P_b, \quad 0 \le i \le M, \ 0 \le j \le CW_i - 1 \quad (4.13)$$

If the channel is busy during AIFS period, the AIFS counter (subscript k) will be reset to A but the backoff counter (subcript j) will remain the same:

$$P\{B_{i,j,A}|B_{i,j,k}\} = P_b, \quad 0 \le i \le M, \ 0 \le j \le CW_i, \ 1 \le k \le A \quad (4.14)$$
$$P\{B_{i,j,A}|B_{i,j,0}\} = P_b, \quad 0 \le i \le M, \ 1 \le j \le CW_i \quad (4.15)$$

The backoff state will be carried out when any of the following events occurs: a.) the medium is busy during the arrival of new frame in idle state b.) after each successful transmission c.) after the transmission is failed due to the channel error or collision. At the end of the backoff, if the system determines that there is no frames waiting in the queue, then it moves to idle state:

$$P\{I|B_{0,0,k}\} = 1 - P_q, \quad 0 \le k \le 1 \quad (4.16)$$

where P_q is the probability that the queue for that AC is empty; otherwise, it triggers the transmission process if the channel sensed to be idle:

$$P\{T_{0,\lceil Ts\rceil}|B_{0,0,k}\} = P_q(1 - P_b), \quad 0 \le k \le 1 \quad (4.17)$$
$$P\{T_{i,\lceil Ts\rceil}|B_{i,0,k}\} = 1 - P_b, \quad 1 \le i \le M, \ 0 \le k \le 1 \quad (4.18)$$

4) Transmission state ($T_{i,k}$): The station that has accessed to the channel can start its frame transmission, this transmission duration is denoted by T_s in this paper. T_s will decrement for every time slot that has passed and eventually will reach 0 when the entire frame has been transmitted.

$$P\{T_{i,j}|T_{i,j+1}\} = 1, \quad 0 \le i \le M, \ 0 \le j \le \lceil T_s\rceil - 2 \quad (4.19)$$

During the transmission process, there are three possible events that can occur, which are collision (internal & external), channel error and successful transmission. Each of these events leads to different state of our DTMC model. If there is no collision, the station continues its frame transmission:

$$P\{T_{i,\lceil Ts\rceil-1}|T_{i,\lceil Ts\rceil}\} = 1 - P_c, \quad 0 \leq i \leq M \tag{4.20}$$

where $P_{c,n}$ is probability of frame collision at queue AC_n. Otherwise, the AC moves to collision state where the AC need to infer the collision for T_c duration, hence:

$$P\{C_{i,\lceil Tc\rceil}|T_{i,\lceil Ts\rceil}\} = P_c, \quad 0 \leq i \leq M \tag{4.21}$$

The frame being transmitted might be in error or the acknowledgement frame (Ack) is lost due to channel error which causes the AC to shift to error state.

$$P\{E_{i,\lceil AT\rceil}|T_{i,0}\} = P_e, \quad 0 \leq i \leq M \tag{4.22}$$

If no error during the transmission, the AC will waits for the Ack frame from the destination node.

$$P\{R_{\lceil Ack\rceil}|T_{i,0}\} = 1 - P_e, \quad 0 \leq i \leq M \tag{4.23}$$

5) Collision state ($C_{i,k}$): The Virtual Collision Handler (VCH) within a station handles the internal collision where the AC with the highest priority wins the contention and the other ACs need to initiate the backoff. Although the colliding ACs from either internal or external collision behaves the same way i.e. invoke backoff routine, the external collision is a more difficult problem to deal with since all the stations involve in the collision need to defer whereas the internal collision will always has one AC to occupy the medium. When an AC is in collision state, it needs to wait for a collision time interval called T_c:

$$P\{C_{i,k}|C_{i,k+1}\} = 1, \quad 0 \leq i \leq M, \;\; 1 \leq k \leq \lceil T_c\rceil - 1 \tag{4.24}$$

After collision, an AC defer its transmission for an AIFS period thus:

$$\begin{aligned} P\{AI_{i+1,A}|C_{i,1}\} &= 1, \quad 0 \leq i \leq M-1 & (4.25)\\ P\{AI_{0,A}|C_{M,1}\} &= 1 & (4.26) \end{aligned}$$

6) Error state ($E_{i,k}$): In this state, we introduce P_e as a channel error probability. The data frame transmission is failed if either the data frame or the ACK frame is in error. For both cases, the station waits for the ACK reply from the destination that would not be arrived due to the transmission error. The maximum waiting time is called $ACK_{timeout}$ and is referred in our context by the notation AT:

$$P\{E_{i,k}|E_{i,k+1}\} = 1, \quad 0 \leq i \leq M, \;\; 1 \leq k \leq \lceil AT\rceil - 1 \tag{4.27}$$

After $ACK_{timeout}$, an AC will defer from transmission for AIFS period thus move from E state to AI state:

$$\begin{aligned} P\{AI_{i+1,A}|E_{i,1}\} &= 1, \quad 0 \leq i \leq M-1 & (4.28)\\ P\{AI_{0,A}|E_{M,1}\} &= 1 & (4.29) \end{aligned}$$

7) Receive state (R_k): When the data frame has been successfully transmitted, the station need to wait for the reply from the destination. As soon as the data frame reach the destination address without channel errors, the acknowledgement frame will be transmitted back to the source in order to complete the transmission routine. The time taking to receive the acknowledgement frame is denoted by Ack, hence:

$$P\{R_k|R_{k+1}\} = 1, \quad 1 \leq k \leq \lceil Ack \rceil - 1 \tag{4.30}$$

If the transmission is successful, the process will move from receive state to AIFS and backoff state to initiate another transmission so:

$$P\{AI_{0,A}|R_1\} = 1 \tag{4.31}$$

4.2 System Analysis

In order to solve the steady state probability for the DTMC, we express all unknown quantities in term of one variable, which is $B_{0,0,0}$ in our case. Since the sum of all steady state probabilities in DTMC is 1, we write the normalization equation as follows:

$$\begin{aligned} 1 \quad &= \quad I + \sum_{i=0}^{M}\sum_{j=0}^{CW_i}\sum_{k=0}^{A} B_{i,j,k} + \sum_{i=0}^{M}\sum_{k=0}^{A} A_{i,k} + \sum_{k=1}^{Ack} R_k \\ & \quad + \sum_{i=0}^{M}\sum_{k=1}^{Tc} C_{i,k} + \sum_{i=0}^{M}\sum_{k=1}^{AT} E_{i,k} + \sum_{i=0}^{M}\sum_{k=0}^{\lceil T_s \rceil} T_{i,k} \end{aligned} \tag{4.32}$$

After substituting all the probabilities in the normalization equation, we have

$$\begin{aligned} (B_{0,0,0})^{-1} &= \frac{P_e(1-P_{c,n})(1-P_b)\lceil AT \rceil \sum_{i=0}^{M} X^i}{1-P_qP_b} + \frac{(1-P_b)L}{1-P_qP_b} \\ &+ \frac{(1-(1-P_{c,n})\lceil T_s \rceil)(1-P_b)\sum_{i=0}^{M} X^i}{1-P_qP_b} + \frac{(J+1)\sum_{i=0}^{M} X^i}{1-P_qP_b} \\ &+ \frac{\lceil Ack \rceil (1-P_b)(1-X^{M+1})}{1-P_qP_b} + \frac{P_{c,n}(1-P_b)\lceil T_c \rceil \sum_{i=0}^{M} X^i}{1-P_qP_b} \\ &+ \frac{P_qP_b(1-P_b)J}{1-P_qP_b} + \frac{P_b(1-P_b)JL}{1-P_qP_b} + \frac{K+P_bJK}{CW_0+1} \end{aligned} \tag{4.33}$$

where

$$L = \sum_{i=1}^{M} \frac{X^i}{W_i+1} \sum_{j=0}^{W_i} (W_i - j + 1), \qquad J = \sum_{k=1}^{A} (1-P_b)^{-k}$$

$$K = \sum_{j=1}^{W_i} (W_i - j + 1), \qquad X = P_{c,n} + P_e - P_{c,n}P_e$$

All states of the DTMC model can be calculated if P_a, P_b, $P_{c,n}$, P_e and P_q are known since $T_s, T_c, AT, Ack, M, A, N$ are given parameters for a given

AC. We derive the expressions of these probabilities. We assume that the frame arrival has exponential distribution thus

$$P_a = 1 - e^{-\lambda \times aSlotTime} \tag{4.34}$$

where λ is the frame arrival rate (in frames/s) to the particular AC. $P_{ac,n}$ and $P_{oc,n}$ are introduced in order to determine P_b and $P_{c,n}$. $P_{ac,n}$ represents the probability that AC_n accesses the channel to initiate the transmissiion to other STA where $P_{oc,n}$ is the probability of channel being occupied by an AC_n of a STA. Hence,

$$P_{ac,n} = \sum_{i=0}^{M} T_{i,\lceil T_s \rceil} \tag{4.35}$$

$$P_{oc,n} = \sum_{k=1}^{\lceil Ack \rceil} R_k + \sum_{i=0}^{M} \left(\sum_{k=0}^{\lceil Tc \rceil} C_{i,k} + \sum_{k=0}^{\lceil AT \rceil} E_{i,k} + \sum_{k=0}^{\lceil T_s \rceil} T_{i,k} \right) \tag{4.36}$$

In the previous equations we consider the case where the probabilities associated with an AC, however we also need to express these probabilities in term of a station's viewpoint which can be computed as follows:

$$P_{ac} = 1 - \prod_{n=0}^{3} (1 - P_{ac,n}) \tag{4.37}$$

$$P_{oc} = 1 - \prod_{n=0}^{3} (1 - P_{oc,n}) \tag{4.38}$$

As the collision can occur either internally (collide with other ACs) or externally (collide with other STAs), so $P_{c,n}$ (the collision probability of AC_n) is given by

$$P_{c,n} = 1 - (1 - P_{ac})^{N-1} \prod_{n'>n} (1 - P_{ac,n'}) \tag{4.39}$$

The channel busy probability P_b is computed as the probability that at least one of the stations occupy the channel whereas P_e can be directly obtained from channel error.

$$P_b = 1 - (1 - P_{oc})^N \tag{4.40}$$

$$P_e = PER \tag{4.41}$$

where PER is packet error rate calculated from channel error. P_q is the probability that an AC queue is non-empty and will be calculated according to M/G/1 queueing theory. Assuming the frame arrival rate λ has exponential distribution whereas the average service time for each frame W_s has a general distribution, we have:

$$P_q = \lambda W_s \tag{4.42}$$

The state of the queue has to be taken into consideration when calculating the average service time. For the empty queue, the frame will be immediately served by the queue upon its arrival and thus the starting state for empty-queue is I state. On the other hand, the frame requires one post-backoff following its previous transmission when the frame is fetched from the queue that is the starting state is $AI_{0,A}$. So we have:

$$W_s = (1 - P_q)D_I + P_q D_{AI} \tag{4.43}$$

where D_I and D_{AI} are the expected slot count from state I to R_1 and state $AI_{0,A}$ to R_1 respectively. To obtain these slot counts, we apply the Chapman-Kolmogorov equation [14] to find the *n-step transition probabilities* starting from I or $AI_{0,A}$ to reach R_1. We computed D_I and D_{AI} using MATLAB's *fsolve* function.

4.3 Performance Analysis

We describe the derivation of network throughput and PLR of each AC_n in this section. Technically, the throughput S_n of an arbitrary AC_n is computed as the ratio of expected useful data received by the destination over the transmission time. It is important to note that the throughput also depends on the transmission mode called access mode. In fact, two types of access modes are available which are RTS/CST and basic mode. The parameters associate with these access mode are listed in Table 5.1. Based on these parameters, we can write the throughput S_n of a given AC_n as

$$S_n = \frac{P_{s,n} E\left[P\right]}{T_{s,n}} \tag{4.44}$$

where $E\left[P\right]$ is the expected payload size, $P_{s,n}$ is the conditional successful transmission probability for AC_n and $T_{s,n}$ is the average time for one frame in AC_n to be delivered and are given by:

$$P_{s,n} = \frac{\left(\sum_{k=1}^{\lceil Ack \rceil} R_k\right)\left(\sum_{i=0}^{M}\sum_{k=0}^{\lceil T_s \rceil} T_{i,k}\right)}{\sum_{k=1}^{Ack} R_k + \sum_{i=0}^{M}\left(\sum_{k=0}^{\lceil Tc \rceil} C_{i,k} + \sum_{k=0}^{\lceil AT \rceil} E_{i,k}\right)} \tag{4.45}$$

$$T_{s,n} = max[\lambda^{-1}, W_s] \tag{4.46}$$

Note that the average time to transmit one frame is bounded by either the frame rate or the service time.

Since a successful transmission only occurs when the frame is neither corrupted by wireless channel or lost due to frame collision, hence, the PLR of the AC_n is:

$$\begin{aligned} PLR(AC_n) &= 1 - (1 - P_{c,n})(1 - P_e) \\ &= P_{c,n} + P_e - P_{c,n}P_e \end{aligned} \tag{4.47}$$

4.4 Model Verification

This section describes how we verify the correctness of the proposed analytical model. As can be seen from Fig. 4.4, the system consists of three parts: traffic source, traffic destination and performance estimator, which is defined as the combination of the proposed DTMC model and our previous work, video distortion model [13].

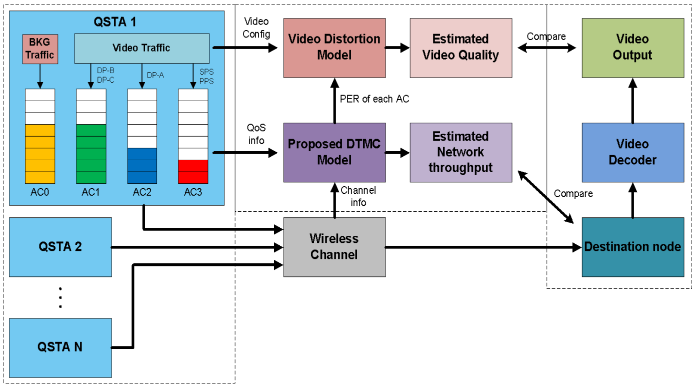

Figure 4.4: Proposed model verification Scheme.

On the left side of the diagram is the traffic source module. This component is composed of N QoS-STAs (QSTAs) in which each station generates two types of traffics namely, background traffic and video traffic. Background traffic has constant bitrate and is injected into AC_0 queue for the purpose of creating packet collisions, whereas video traffic is chosen to test QoS provided by EDCA. We employ the H.264's data partitioning to encode the video sequences in the simulation. Moreover, the three AC queues are used for transmitting video traffic and the DPs are mapped to the ACs as follows: data partion A (DP-A), which contains the header information and is crucial for decoding the video frame, will go to AC_3 which has highest priority, DP-B and DP-C packets will be loaded to AC_2 and AC_1 respectively. The traffics will undergo the wireless channel and compete with other traffics coming from the other $N-1$ stations and eventually reaching the destination node where the throughput will be measured and the video will be decoded in order to compare with our proposed model. Lastly, the proposed DTMC model will estimate the throuhput performance of each queue (AC) based on QoS and channel information acquired from Qualnet's MAC and Physical layer at the source traffic's side respectively. The video distortion model takes the per-AC PLR from the DTMC model as an input in conjunction with the encoding configuration to calculate the quality of video perceived by end-user.

v

5 Experimental Results

In this section we evaluate the accuracy of our video prediction system where estimated channel utilization, PLR and decoded video's PSNR from our proposed analytical model will be compared to the simulation results generated from Qualnet [15]. To simplify our model, we also assume that the transmission occurs in an one-hop wireless network where all the wireless stations share the same medium and will compete to transmit video bitstream and background traffic to one sink node. The simulation environment is configured as follows: standard 802.11b is employed for physical layer whereas MAC layer adopts EDCA mechanism from standard 802.11e. Video and background traffic are

Table 5.1: Summary of Simulation Parameters

P(ayload)	1000 bytes
H(eader)	PHY+MAC+IP+UDP
PHY (including preamble)	192 bits
MAC (including CRC bits)	272 bits
IP	160 bits
UDP	64 bits
PHY STD	IEEE 802.11b
MAC STD	IEEE 802.11e EDCA
MAX Data Rate	2 Mbps
Time Slot	20 μs
SIFS	10 μs
M	[7 7 7 7]
AIFSN	[7 3 2 2]
CWmin	[31 31 31 31]
CWmax	[1023 1023 1023 1023]
$CTS_{timeout}$	DIFS+CTS
$ACK_{timeout}$	DIFS+ACK
Data Frame	PHY + MAC + IP + UDP + P
RTS Frame	PHY + 160 bits
CTS Frame	PHY + 112 bits
ACK Frame	PHY + 112 bits
T_s (RTS/CTS)	RTS + SIFS + CTS + SIFS + H + P
T_c (RTS/CTS)	RTS + SIFS + $CTS_{timeout}$
AT (RTS/CTS)	SIFS + $ACK_{timeout}$
Ack (RTS/CTS)	SIFS + ACK
T_s (basic)	H + P
T_c (basic)	H + P + SIFS + $ACK_{timeout}$
AT (basic)	SIFS + $ACK_{timeout}$
Ack (basic)	SIFS + ACK

generated at the application layer of each source node. The video sequence will be encoded first and then the trace file for the encoded video bitstream is created to be used in the simulation. The information conveyed by the trace file are RTP sequence, RTP timestamp, packet type, packet size, encoding time and priority level. In the experiment, three QCIF size test video sequences *news*, *foreman* and *football* are selected to represent slow, medium and fast inter-frame movement pattern. We encode 200 frames of *news* and *foreman* and 100 frames of *football* in I-P-P coding mode with 15% intra-rate using JM13.0 [1] [16].

We conducted the experiments using the default parameter setting in Qualnet as shown in Table 5.1 and 1000 bytes fixed packet size with RTS/CTS access mode is used for background traffic. In fact, the bitrate of background traffic will be set depending on the purpose of the experiment. If the expected throughput or PLR are our focus, the background traffic is set with varied bitrates to enable us to observe the variation of the channel throughput and PLR caused by traffic congestion. On the other hand, the background traffic is set to

[1] JM is the reference software codec that implementing the H.264/AVC video compression standard.

fixed bitrate if we want to see the results of other aspects e.g. how throughput changes according to different number of wireless stations or expected throughput of each AC queue with diffferent PLRs. In each test, the experiment is run 1000 times with random nodes' position then the average channel throughput is calculated and compared with the estimated throughput from our proposed DTMC model. Similarly, the coded video data is decoded and presented in term of MSE to compare with the pre-determined outcome from the video distortion model.

5.1 Verifying the DTMC model

We first verify that our DTMC is able to correctly predict the throughput and packet loss rate of each AC accurately under different traffic conditions. These comparison are shown in Fig. 5.1-5.4. In Fig. 5.1 the mean channel throughput of each AC of the wireless stations is plotted against background traffic rate. It can be seen that the model gives a very good approximation of the throughput experienced by the destination node. The queue AC_0, where the background traffic is injected, has a throughput which initially increases with the background traffic rate but is later capped at 0.7 Mbps even background traffic rate continues to increase. On the other hand, the queue AC_2 , which carries the video partition DP-B, has an almost constant throughput of 0.12 Mbps despite changes in background traffic rate. According to Table 5.2, the average bitrate of *news* video sequence used in this experiment for queue AC_3 to AC_1 are 63.10, 124.76 and 62.25 Kbps and thus the saturated net throughput for one wireless station is roughly 1 Mbps (0.7 Mbps from background and 0.25 Mbps from video bitstream).

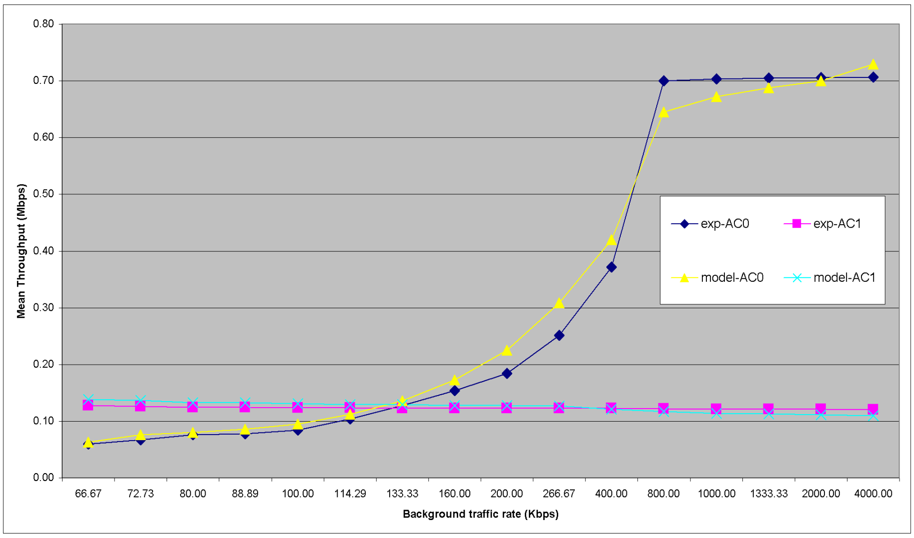

Figure 5.1: Mean throughput of each station achieved by two priority queues AC_0 and AC_2 plotted against background traffic rate with 4 QSTAs simulated

The PLR for each AC queue predicted by the model is compared to the simulation result in Fig. 5.2. The x-axis indicates the bitrate of background

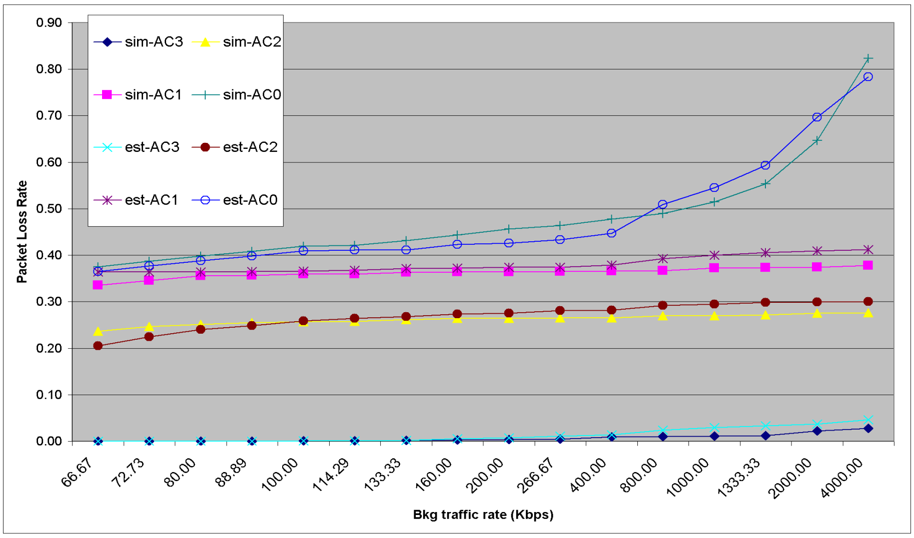

Figure 5.2: Packet loss rate of AC_0, AC_1, AC_2 and AC_3 queues under different background traffic rate with 4 QSTAs simulated.

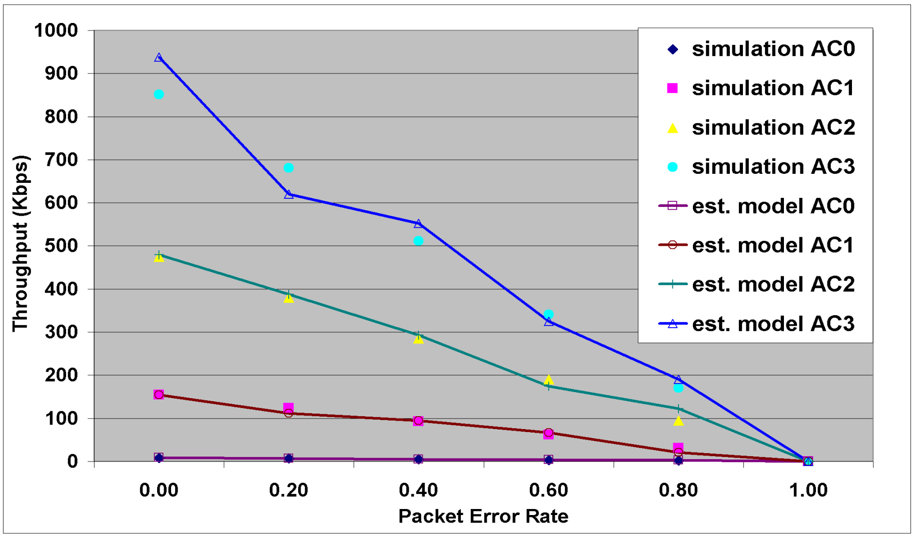

Figure 5.3: Mean throughput of all four priority queues AC_0-AC_3 in regards to packet loss rate. 1Mbps data rate is injected into all AC queues.

traffic injected into queue AC_0, and the y-axis represents the average PLR predicted by the model and obtained from the simulation. All queues has rather constant PLR regardless of the bitrate of background traffic rate except for queue AC_0 where the PLR increases rapidly after a certain threshold.

We illustrate the acuracy of our proposed DTMC model to estimate expected throughput achieved by different priority queues at various packet loss rates (due to wireless channel) in Fig. 5.3. These experimental results are simulated by injecting 1 Mbps traffic source into all four ACs and transmitting the frames

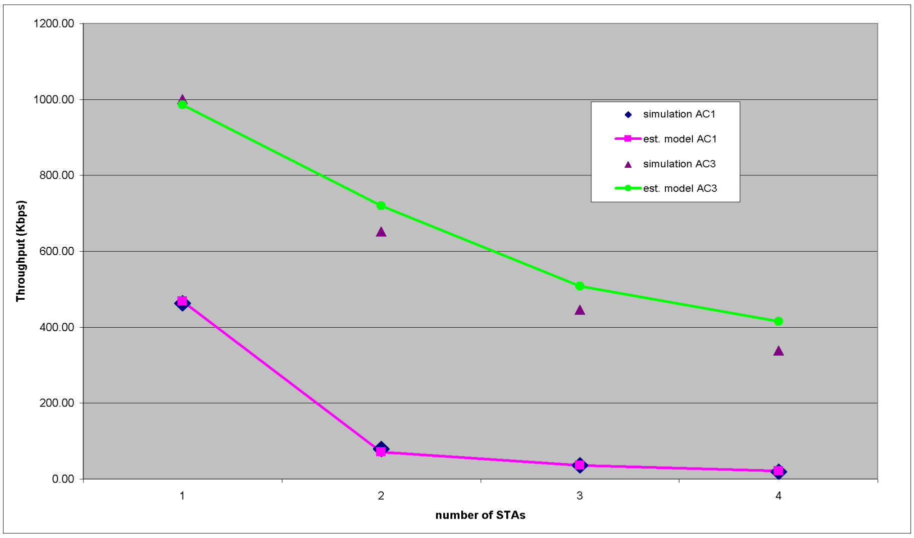

Figure 5.4: Mean throughput of AC_3 and AC_1 vs. number of stations with 1 Mbps data rate is injected to all AC queues.

under MAC contention and wireless channel error. It can again be seen that the DTMC model produces accurate prediction of throughput for each AC.

Fig. 5.4 shows, for two different priority queues (AC_3&AC_1), how the throughput varies with the number of contending stations, each with 1 Mbps data rate in all AC queues. It can again be seen that the prediction by the DTMC model is rather accurate.

5.2 Verifying the prediction of video quality

The aim of this section is to verify the performance of the video quality prediction model depicted in Fig. 4.1. The prediction is carried out in two steps, according to the two blocks in the figure. In the first step, the arrival and packet size characteristics of the video sequences (see the top part of Table 5.2), as well as those of the background traffic, are input into the 802.11e MAC channel model to obtain the PLR of each AC for each video sequence. The predicted PLRs, for each video sequence, are then input into the H.264 video distortion model to obtain the average PSNR of the entire video as well as the MSE of each video frame.

Table 5.2 compares the PLR for each DP for each video sequence estimated by the model ('**est**') against that obtained from simulation ('**sim**'). 'Δ' is the difference of '**sim**' and '**est**'). It can be seen that the PLR prediction is rather accurate for all DPs for all the three video sequences. Table 5.2 also compares the estimated average PSNR of three video sequences against that obtained from simulation. The prediction results are pretty accurate, especially for the video sequences *news* and *foreman*. However, the error in estimating the average PSNR for *football* is slightly worse than the other two. This is due to the higher standard deviation of the packet size of *football* which results in a poorer approximation in the M/G/1 queueing system and this in turn affects the ac-

Table 5.2: Summary of video information and prediction results

video sequence		news (200 frames)			foreman (200 frames)			football (100 frames)		
Average packet size (bytes)	DP-A	264.22			643.77			1455.36		
	DP-B	522.43			431.97			522.92		
	DP-C	260.67			463.57			1710.61		
	all	349.10			513.11			1229.63		
Standard deviation (packet size)	DP-A	78.37			152.62			432.44		
	DP-B	77.24			56.79			169.38		
	DP-C	135.62			130.52			646.24		
	all	158.71			152.38			686.79		
Average bitrate (Kbits/s) @30Hz	DP-A	63.10			153.74			345.79		
	DP-B	124.76			103.16			124.55		
	DP-C	62.25			116.70			406.44		
	all	250.10			367.59			876.48		
PLR		sim	est	Δ	sim	est	Δ	sim	est	Δ
	DP-A	0.0102	0.0244	0.0142	0.0129	0.0543	0.0414	0.0236	0.1159	0.0923
	DP-B	0.2696	0.2922	0.0226	0.3075	0.2799	-0.0276	0.3911	0.3404	-0.0507
	DP-C	0.3665	0.3925	0.0260	0.4403	0.4744	0.0341	0.5074	0.6445	0.1371
Average PSNR(dB)		sim	est	orig	sim	est	orig	sim	est	orig
		33.89	33.42	35.69	31.46	30.19	35.29	25.87	22.61	34.17

curacy in predicting the PLR for each AC. Finally, Fig. 5.5 shows the MSE for each video frame of *news*. It shows that the predicted MSE is very close to that given by the simulation. The results for per-frame MSE for *foreman* and *football* are similar but they cannot be included here due to a lack of space.

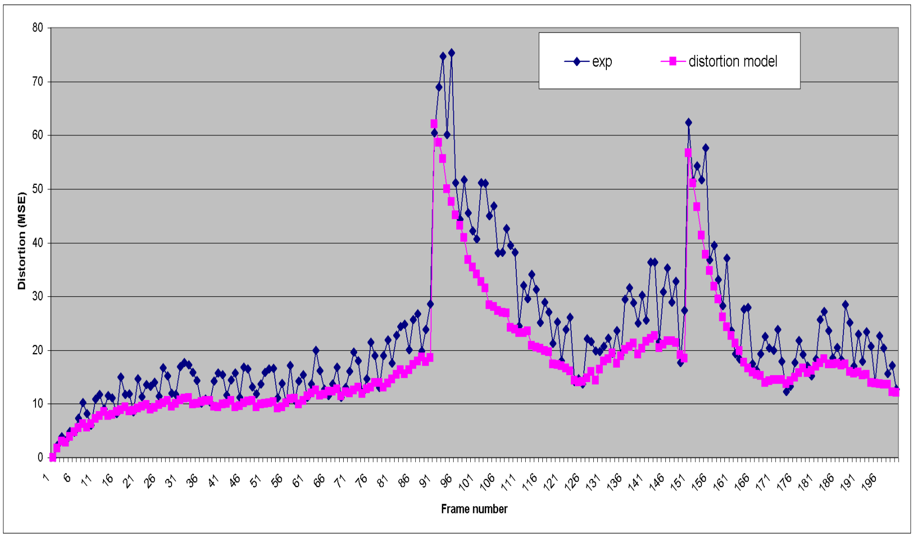

Figure 5.5: Video distortion based on MSE for "news" video sequence with I-P-P format and 15% intra-rate. 800 Kpbs background traffic with 4 QSTAs simulated.

6 Conclusion

In this paper, we propose a model to predict quality of wireless video transmission under IEEE 802.11e EDCA standard. The proposed algorithm is superior to the previous work in several aspects. Firstly, the model is able to handle four priority queues along with the VCH and service differentiation mechanisms. Secondly, we incorporate the M/G/1 queueing system in our algorithm allowing the model to capture the properties of both finite-load and saturation conditions. Thirdly, the model also takes into account the error-prone characteristics of wireless channel in the form of packet loss rate (PLR). Therefore, assisting us to acheive the analytical model that close to the real world scenario. Finally, by incorporating this QoS-MAC channel model with H.264 video distortion model, we can also estimate the quality of decoded video experienced by end-user. The performance of the proposed analytical model is observed through the comparison with the simulation results. According to the experimental results, frame arrival rate has significant impact on throughput performance. In fact, the higher the frame arrival rate, the higher chance of internal collision to be incurred and hence the lower priority queues have much higher PLR than the higher priority queues. Although it can be seen that increasing number of wireless stations leads to more external collisions, the impact is effected in station-level not in queue-level thus all ACs suffer equally from external collisions. Similarly, the packet loss due to channel errors is considered to have the same characteristic as external collision as it does not aware of the priority of the queues and views a station as a single entity so every AC encounters the same effect.

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