# MAC Protocol Simulator for Speech and Video transmission in Optical Wireless Communications

*Bilal R. Qazi and Jaafar M. H. Elmirghani*
School of Engineering
University of Wales Swansea, Swansea UK

**Abstract-** Optical wireless has brought forward distinct advantages in the field of telecommunication and network systems as compared to radio systems. In this respect, there has been a need to design and evaluate MAC protocols for this particular environment. This paper presents a be-spoke MAC protocol simulator for a MAC protocol that was analytically proposed earlier. Initially, the simulator is run and tested for all audio markovian sources using modified PRMA technique and CSMA/CA for packet transmission and collision avoidance respectively. Then, it is extended to add 1 markovian video source with all other audio sources, at higher transmission rate to see the effect on other audio sources in the system. The system performance is evaluated in terms of throughput, packet access delay and pack dropping probability showing good agreement between simulation and analytic results for audio sources in the system in both cases i.e. 1) when there are only audio sources and 2) when one video source is added in the system.

## 1 INTRODUCTION

Traditionally, radio frequency is used for communication purposes but due to limited bandwidth and broadcast nature, Optical Wireless Communication has come forward to take a noticeable part in this contest more recently. Users now want to utilise fixed network services, such as electronic mail, WWW browsing, file transfers, voice and music broadcasting, and video transmissions with low cost and high bandwidth, while retaining the freedom of mobility [1,2] and QoS [3]. Instead of conventional cellular radio networks specializing in voice transmissions, we have started seeing networks capable of simultaneously providing, for example, video, voice and file transfer services through a common optical wireless interface in which the Infrared (IR) medium is used rather than radio frequency but still protocols above the physical layer are responsible for sharing the IR medium between the users, such protocols are now receiving considerable attention. A be-spoke computer simulator has been designed to verify the analytic results presented in [4] and extended in [5,6]. This paper is an attempt to make programming issues clear including generation of real-time VBR traffic for simulation and the simulator's functionality.

## 2 MAC PROTOCOL

### 2.1 Introduction

The MAC protocol in [4-6] is a slotted modified PRMA where a base station severs a number of optical wireless cells where each cell is 3-5 m in diameter. Time is divided into an uplink and downlink frames with slots that can be reserved with reservation and acknowledgements slots.

### 2.2 Uplink/Downlink frame

The time organisation of Uplink and Downlink frame of the MAC protocol can be described as follows. Each frame is divided in to J channel frames. The uplink frame consists of a reservation (R) slot and N information slots. The downlink frame has an acknowledgement (ACK) slot and N information slots. A Node needs to reserve a slot before it actually transmits the data.

#### 2.2.1 Number of Slots

The MAC protocol and the simulator allocate fixed number of slots to the TUs in each channel frame. The number of time slots N per channel frame (uplink or downlink) during the simulation is given by

$$N = \text{int}\left[\frac{R_C T_{cf}}{2JT_{cf} R_S + H}\right] \tag{1}$$

where int [x] is the largest integer smaller than or equal to x, $R_c$ is the channel bit rate, $R_s$ is the source rate, H is the header of IWC and J is the channel reuse factor. $T_{cf}$ is the channel frame duration and is given $T_{cf}$ = (Tf/J)-(ACK/$R_c$), where $T_f$ is the uplink/downlink frame duration. A channel bit rate of 10 Mb/s was selected (optical wireless dispersion limit in average room) and the audio source data rate was assumed to be 64 Kb/s giving a useable value of N=9, so uplink has 10 slots including 1 R slot. In the simulation environment, we consider 1 cell, 1 channel, 1 BS and M audio nodes in that cell which contend for 9 information slots in the case of all audio sources in the system.

## 3 SIMULATOR

### 3.1 Simulator Environment

The simulator employs event driven simulation and is programmed in Java (2) using object-oriented concepts. Being an event driven simulator, it is very fast and as Java is a machine independent language, the simulator can be run on any operating system or platform/environment. The simulator has one global variable name sysTime which tells how many times uplink frame has gone to the BaseStation and a waiting queue that holds all the nodes which are in ON state and waiting to transmit their packets as soon as a slot becomes available to them. As each node has a waiting packet queue associated to it that is used only by that particular node for its packets. So when the node is in waiting queue its ON packets start going into waiting packet queue from where packet dropping probability (PDP) is calculated. For all speech sources in the system, PDP at which speech quality degradation is indiscernible must be less than 1%.

### 3.2 Traffic Generation

In a simulation environment, traffic generation plays a key role in evaluating the performance of the system correctly. One of the most well known approaches to generate a VBR traffic source is the two state markov ON/OFF model. This source will either be in an OFF state transmitting at zero bit rate, or in an ON state transmitting at its peak rate. All the OFF durations (gaps) and ON durations (messages) are exponentially distributed [7]. With the mean OFF duration (t2), which is the interarrival time of messages, and the mean ON duration (t1), which is the message length or service time, λ and μ can be found i.e. λ=1/t2 or 1/Vt2 and μ = 1/t1 or 1/Vt1 where λ is the interarrival rate between two messages or can be called as a gap and μ is the service rate for the message. A uniform distribution U (0,1) is generated using a pseudo-random number generator and exponential variates for gaps and messages are computed as

$$X_1 = -\log(1-u)/\lambda \quad (2) \quad \text{and} \quad X_2 = -\log(1-u)/\mu \quad (3)$$ (where u is a random number 0.00<= u >1.00)

Packet duration, Tp, is calculated as

$$T_p = (((R_c * T_{cf}) - H * N)/(N * R_s)) \qquad (4)$$

Variates $X_1$ and $X_2$ are divided by $T_p$ to get the number of OFF and ON packets in those spurts. '-1's are written as number of OFF packets and '1's as number of ON packets into the traffic file in a way that ON packets are followed by OFF and this process is repeated for plenty of spurts, i.e. 4000 in our simulation, to make a traffic file for 1 source. Now the complete process is repeated to make M traffic files for M sources. Figure 1 illustrates an example of 1 data file of a source where an OFF spurt comes first, then an ON spurt and then OFF and ON spurts again respectively until OFF and ON packets are written for the total number of spurts in the file.

| | | | | | | | | | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| -1 | -1 | -1 | -1 | -1 | -1 | 1 | 1 | 1 | -1 | -1 | -1 | -1 | -1 | -1 | 1 | 1 | 1 | . | . |

**Figure 1: Traffic/Data file for an source ('-1' represents a OFF packet and '1' represents an ON packet )**

### 3.3 Simulator's functioanlity

Packet Reservation Multiple Access (PRMA) [7] was used for the packet transmission from nodes to the BS in the mathematical model [4]. Considering the complexity of the system in terms of programming and other conceptual issues, PRMA technique has been modified to make it easier to program, more logical and reliable, and error-free by taking the advantage of customized CSMA/CA [8] technique to avoid collisions when nodes try to reserve a slot. The MAC protocol and simulator use fixed slot allocation for audio and video sources. Table 1 shows the system parameters used during the simulation.

| Variable | Notation | Value |
|---|---|---|
| Channel bit rate | $R_c$ | 10 Mb/s |
| Speech peak bit rate | $R_s$ | 64 kb/s |
| Video peak bit rate (coded) | $R_v$ | 320 kb/s |
| Uplink/downlink frame duration | $T_f$ | 3.1 ms |
| R slot duration | R | 280 bits |
| Downlink timing signal | $T_d$ | 1/10 of a slot |
| Speech mean ON duration | $t_1$ | 1 s |
| Speech mean OFF duration | $t_2$ | 1.35 s |
| Speech maximum time delay | $D_{max}$ | 20 ms |
| Video mean ON duration | Vt1 | 33 ms |
| Video mean OFF duration | Vt2 | 67 ms |
| Video maximum time delay | VDmax | 150 ms |
| Channel reuse factor | J | 7 |
| Header of an IWC | H | 70 bits |
| Size of an IWC | S | 53 bytes |

**Table 1: System parameters**

PRMA and CSMA/CA are adapted in such a way that if a node cannot find empty slot, that node will not send the request to the BS and keep on waiting in the waiting queue and its packet will start going into the corresponding waiting packet queue for that node. When a packet reaches its threshold, Dmax for audio sources and VDmax for video sources, it is dropped for QoS. A node, which is ready to transmit data i.e. it is in ON state, will sense the uplink frame and if there is any availability of slots, the node requests a reservation by transmitting its source address through the R slot to the BS. Nodes also inform the BS through the R slot if they indent to have packets for the next frame. This improves utilisation during source's OFF periods. If there are more nodes waiting in the waiting queue than the free slots in the uplink frame, then the request will be sent according to FIFO in the waiting queue. On receiving the request through R slot, BS will broadcast the downlink frame with the appropriate node address in the specific slots. Due to this enhanced feature, nodes will know exactly which uplink slot is their's. Nodes, then, start to transmit in their own time slots of uplink frame by means of short downlink timing signals ($T_d$). Considering audio sources in PRMA, an audio node try to reserve a slot as soon as a ONspurt starts and releases the slot as soon as a OFFspurt comes and this process continues until all the ONspurts are finished whereas a video source retains its slot(s) until all of its packets are finished for the whole simulation. An audio node will have 1 slot at a time to transmit but for the video source we will discuss it in the next sub section.

### 3.4 Performance Analysis

As mentioned earlier, each audio source reserves one slot per uplink frame but for a VBR video source multiple slots may be reserved, we evaluate the Delay and PDP against number of slots available to that VBR video source. Figures 2 and 3 represent the delay and PDP(%) graphs for the video source respectively. When less than 4 slots were given to the VBR video source, both delay and PDP(%) were very high. Considering this and the other audio sources in the system, 4 slots have been assigned to the VBR video source transmitting at 320 kb/s as other audio sources transmit at 64 kb/s.

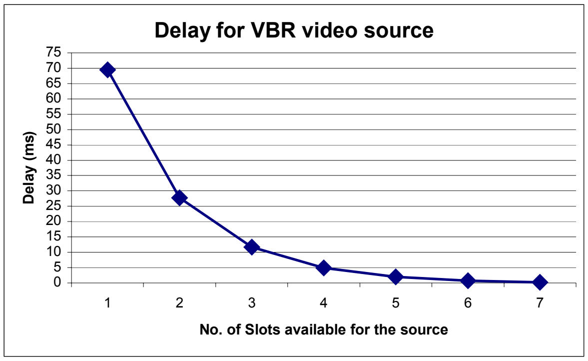


**Figure 2:** Delay (ms) of the video source Vs. no. of slots available

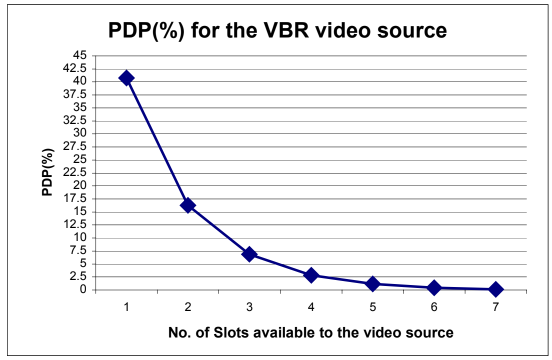


**Figure 3:** PDP (%) of the video source Vs. no. of slots available

System performance in terms of Average Access Delay, PDP and throughput are calculated with respect to audio sources in both cases as mentioned earlier i.e. 1) when there is no video and 2) when there is 1 video in the system.

### 3.4.1 Average Access Delay

A node waiting in the global waiting queue has 6.2 ms added to the packet delay for every uplink frame if it does not get a slot in that frame. So the delay can be calculated as

$$AverageAccessDelay = (totDelay/(totSpurts * totalNodes)) \quad (5)$$

Average access delay increases as the number of voice users M increases which can be seen in Figure 4. Looking at the curve where zero video sources are in the system, when M exceeds 12, average access delay increases rapidly. As delay increases PDP increases because voice packets have to wait more than the time limit delay Dmax. Similarly when there is one video source in the system, if M surpasses 6, average access delay increases dramatically and the effect on the system can also be seen from figure 6.

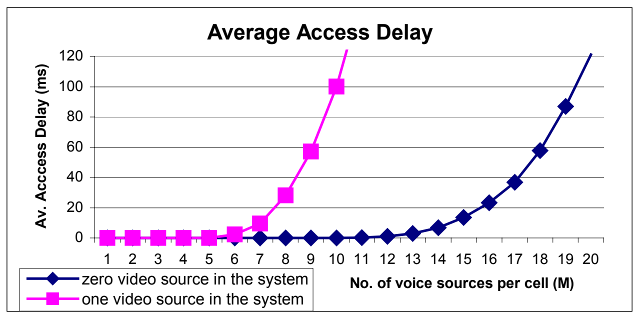


**Figure 4: Average Access Delay Vs. no. of voice sources per cell**

### 3.4.2 Packet Dropping Probability

Consider an example, an audio source n1 is waiting to transmit its packets, initially the first packet goes into the waiting packet queue as 0.0 ms because it has not waited at all. When the uplink frame comes next time after 6.2 ms, first packet's waiting time becomes 6.2 ms and second packet goes as 0.0 ms. When the first packet's waiting time reaches 20 ms it is dropped and so on until the waiting packets find the slot before their maximum limit.

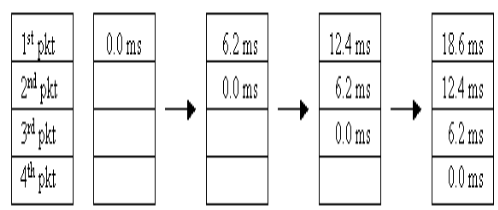


**Figure 5: Waiting packet queue for node n1**

So PDP can be computed as follows

$$PDP(\%) = totPktsDropped * 100 / totONPkts \quad (6)$$

Figure 6 depicts the PDP(%) of the system versus the number of voice sources (M) for various number of video sources in the same cell. To avoid quality degradation a voice packet is dropped when it reaches its maximum waiting time limit Dmax. If there is no video source in the system, for PDP = 1 % the system is capable of supporting upto 15 users simultaneously. QoS of the system is not satisfied if M exceeds 15. Adding 1 video source in the system, the PDP (%) is much higher like average access delay than the first case when there were no video sources in the system.

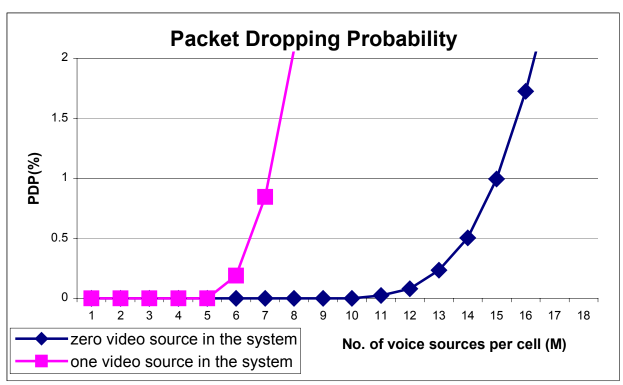


**Figure 6: PDP (%) Vs. no. of voice sources per cell**

### 3.4.3 System Throughput

In simple words, system throughput defines how well the channel (uplink frame) has been used. Here in this paper, throughput is calculated by considering overhead (R slot and downlink timing signal ($T_d$)). System throughput is computed as

$$\text{Throughput}\,(\eta) = (\text{totFilledSlots}/(\text{sysTime} * \text{totSlotsPerUplinkFrame})) \tag{7}$$

$$\text{totSlotsPerUplinkFrame} = 9 \text{ Information Slots} + 1 \text{ R slot} + 1 \text{ slot for } T_d = 11 \text{ slots} \tag{8}$$

Here, sysTime is a global variable that specifies how many times uplink frame has gone to the BS. If we were not taking overhead 11 would have been 9 and would result in higher throughput due to no overhead. But 9 Information slots are used for all audio sources without any video in the system. If we consider a video taking 4 slots as mentioned earlier, only 5 information slots will be left for other audio sources, and then totSlotsPerUplinkFrame will turn out to be 7 and similarly totFilledSlots is the slots filled by audio sources not by a video source.

Figure 7 represents the system throughput ($\eta$) versus the number of voice sources per cell. For both curves throughput gradually increases until it reaches its saturation point. In the case of zero video source in the system, it reaches 0.8 whereas with one video, it saturates at 0.7 as lesser slots are available to the voice sources. When the throughput reaches it saturation point, it remains constant even though M keeps increasing which indicates good system stability. As mentioned earlier, in both cases overhead is included for calculating the system throughput.

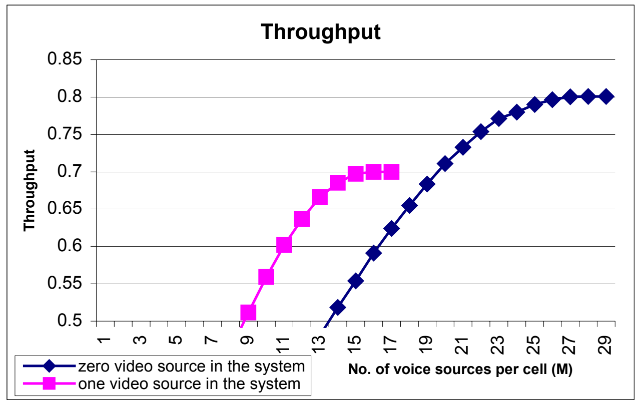


**Figure 7: Throughput ($\eta$) Vs. no. of voice sources**

## 4 CONCLUSIONS

An event driven MAC protocol simulator for indoor optical wireless LANs using IR technology has been proposed and run and tested for all audio sources and adding 1 video source in the system. Programming issues relating to simulator have been made clear, such as simulation environment, variables, buffers/queues, traffic generation and simulator's functionality. System performance has been evaluated over a range of system parameters in terms of average access delay, PDP, and system throughput for all audio sources in two cases. Simulator's results are in good agreement with the theoretical results showed in [4,5].

## 5 FUTURE WORK

Markovian models following exponential or Poisson distribution exhibit Short Range-Dependence that does not show the burstiness nature of a VBR video traffic which displays self-similarity as LAN traffic [9]. A VBR self-similar video source can be generated as a superposition of multiple Pareto sources [10]. We have started looking at the procedure to generate self-similar traffic for a video source and evaluating the performance of the system.

## REFERENCES:

1. Rui T. Valadas, António R. Tavares, and A. M. de Oliveira Duarte, "The Infrared Physical Layer of the IEEE 802.11 Standard for Wireless Local Area Networks", IEEE Communications Magazine • December 1998.
2. Richard O'Donnell, "Prolog to Mobility Management in Next-Generation Wireless Systems", PROCEEDINGS OF THE IEEE, VOL. 87, NO. 8, AUGUST 1999.
3. Huang-Lang Tzeng, Chienhua Chen, "Performance comparison of two MAC protocols for wireless ATM networks". Information Networking, 2001. Proceedings. Pages: 805 - 810
4. Theodorou P, Elmirghani J M H, and Cryan R A, "ATM Infrared Wireless LANs: A Proposed Architecture", IEEE Communications magazine Dec. pp: 118-123 1998.
5. *Bilal R. Qazi, Yamini Harikrishnan, Syed B. Ahmed, Ian Wells and Jaafar M. H. Elmirghani*, "Performance Analysis of a MAC Protocol for Speech transmission in Optical Wireless Communications", London Communication Symposium 2003, pp 61-64
6. Bilal R. Qazi, Yamini Harikrishnan, Syed B. Ahmed, and Jaafar M. H. Elmirghani, "Design and Simulation of a MAC Protocol for Optical Wireless Communications" PostGraduate Networking Conference PGNet 2003.
7. David J. Goodman *et a*l., "Packet Reservation Multiple Access for Local Wireless Communications", IEEE Trans Commun, vol 37, no.8, pp 885-890, 1989.
8. T. S. Rapport, "Wireless Communication", Prentice Hall PTR 2nd edition 2002.
9. Will E. Leland, Murad S. Taqqu, Walter Willinger, Daniel V. Wilson, "On the Self-Similar Nature of Ethernet Traffic", SIGCOMM 1993
10. Walter Willinger, Murad S. Taqqu, Robert Sherman, Daniel V. Wilson, "Self-Similarity Through High-Variability: Statistical Analysis of Ethernet LAN Traffic at the Source Level", IEEE/ACM Transactions on Networking, Vol. 5, No. 1, February 1997, pp: 71-86