Unicast Multimedia Transmission On-board a Business Jet

Keith Chetcuti, Carl J. Debono
Department of Communications & Computer Engineering
Faculty of ICT,
University of Malta, Malta
Tel: (+356) 2340 3011
{keith.chetcuti; cjdebo}@eng.um.edu.mt

Serge Bruillot
Dassault Aviation,
78 Quai Marcel Dassault,
92552 Saint-Cloud,France
Tel: (+33) 1 47113910
Serge.Bruillot@dassault-aviation.com

ABSTRACT

Wireless networks provide the technology that allows the service provisioning of a variety of multimedia streaming applications to mobile stations. In recent years, aircraft manufacturers have been evaluating the possibility of offering this technology on-board their aircrafts. The gained benefits translate into reduced cable complexity, especially in the business jet scenario, where each aircraft has to be customized for the user, and increased passenger satisfaction with upgrades in the infotainment system and services. Wireless networks have bandwidth constraints, which limits the transmitted data rate. To satisfy this limitation, the Joint Video Team (JVT) has developed the H.264/AVC standard, which offers a higher compression ratio for video applications when compared to other coding standards. Different multimedia applications have different Quality of Service (QoS) requirements; for example live conferencing requires an end-to-end delay of less than 200ms while streaming accepts 1s of delay. The radio propagation coverage map of the network on its own is not enough to ensure that the passenger will have adequate QoS. A precise network analysis is therefore required to give an insight on the end-to-end delay, link utilization and throughput performance of the transmission system inside the aircraft.

In this paper, we present the simulation model developed by the University of Malta to assess the QoS of multimedia wireless transmission inside a Dassault Aviation business jet. A Matlab® based discrete-event simulator (DES) was developed to model the network traffic. The model considers the IEEE802.11a standard and uses the basic Distribution Coordination Function (DCF) access scheme. H.264/AVC video coding is employed for the video streaming application. The delay analysis demonstrates that the QoS requirements for streaming applications are attained. Furthermore, the resulting Peak Signal-to-Noise Ratios (PSNR) indicate that satisfactory video quality is experienced by the passengers at different locations inside the aircraft.

Categories and Subject Descriptors

I.6.3 [**Simulation and Modeling**]: Applications.

I.6.8 [**Simulation and Modeling**]: Types of Simulation – *continuous, discrete event, Monte Carlo.*

General Terms

Performance, Design.

Keywords

IEEE802.11a, Multimedia Transmission, QoS, PSNR.

1. INTRODUCTION

Multimedia transmission is very popular with the general public as reported in studies such as [5], which found that 100 million videos were streamed from the You Tube website in the month of July 2006. As wired networks offer higher bandwidth and lower delays, home and office users usually prefer this technology to a wireless system. Nonetheless, in recent years, the use of wireless devices has been on the increase. Various studies [1], [12], evaluated the throughput and the delay performance of wireless systems either via an analytic or a simulation approach. Video streaming is very different from data exchange due to inherent delay constraints – late arrival packets are useless to the video decoder [19]. Such packet delays will introduce the decoder starvation problem.

This work is directed towards the study of the performance of multimedia transmission over area-confined wireless LANs. We have developed a discrete-event simulator (DES), based on the three-phase approach introduced in [23], which models the network traffic of 4 users on-board the business jet. Each user has a different signal-to-noise ratio (SNR), which results in a different bit-error rate (BER) on each channel. The users generate a request for a multimedia stream to be delivered and the server responds by initiating all the traffic. The transmitted stream is encoded using an H.264/AVC coder [11]. The generated RTP packets from the encoder are encapsulated as they pass through different layers of the network protocol stack - UDP, IP, and MAC. The resulting frames are transmitted using the IEEE802.11a [14] standard. Errors are introduced by the channel according on the channel's calculated BER. At the receiving end, the stream is decoded and the resulting Peak Signal-to-Noise Ratio (PSNR) is calculated. Moreover, with regards to the network traffic the link utilization, the delay, and the goodput statistics are recorded.

The work presented here forms part of the integrated project E-Cab (E-enabled Cabin and Associated Logistics for Improved Passenger Services and Operational Efficiency) which is currently being carried out under the EU Sixth Framework Programme Thematic Priority Aeronautics and Space of the European Commission. E-Cab is a process oriented research and technology project that integrates various electronically enabled end-to-end logistic chains. The result will be a paperless information

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management system of the future, for improved passenger comfort, crew convenience and airline and airport efficiency. On-board solutions should provide control over the means by which the passengers can access their connectivity, with options ranging from mobile phones, PDAs, laptop computers, up to new IFE interfaces [9].

The structure of this paper is as follows: Section 2 describes the developed discrete-event simulator model for the network analysis. Section 3 provides details about the simulation process. Section 4 presents the results obtained from the simulation. Conclusions and comments are provided in Section 5.

2. Discrete-Event Simulator Model

2.1 Introduction

Simulation for simple systems, where all the processes involved have a fixed time and there are no interconnections between such processes, is not required. However, many systems have inter-dependencies between processes and suffer from variability; hence predicting performance results becomes a difficult task. For such complex systems simulation is a very useful tool. One can even consider implementing the system and then obtain the system results directly. This approach is however, not adequate for all systems. Refer to [25] for a complete discussion of the problems involved when trying to implement the system rather than doing simulations. [25] also compares other techniques that can be used to obtain the system's performance.

The choice of the simulation technique depends on the particular dynamics of the system being analyzed. One of the parameters that needs particular attention is the modeling of the progressing time. The methods suggested by [25] are a time-slicing approach and discrete-event simulation. The main difference between these two methods is the simulation step.

The time-slicing approach adopts a constant time-step, where execution is performed at each step. The most difficult task is to choose the correct time-step. The time-slicing approach is an inefficient simulation technique as for most of the time there is no change in the system states.

Conversely, in discrete-event simulation execution is performed at times when there is a system change, resulting in a more efficient technique. In other words, the system is modeled as a series of events that are computed when a state-change occurs. Various programming techniques have been proposed to carry out discrete-event simulation, among them are the event-based, activity-based, process-based and three-phase approaches [25]. The most popular technique with simulation software is the three-phase approach [25] proposed in [27].

In the three-phase approach, events are organized into two types:

B (bound or booked) events: these are state changes that are scheduled to occur at a point in time. In general, B-events relate to arrivals or the completion of an activity.

C (conditional) events: these are state changes that are dependent on the conditions in the model. In general, C-events relate to the start of some activity.

Once all the events have been identified, the system can be simulated. The flow of the three-phase simulation process is outlined in Figure 1. The initial state is determined by the user, in

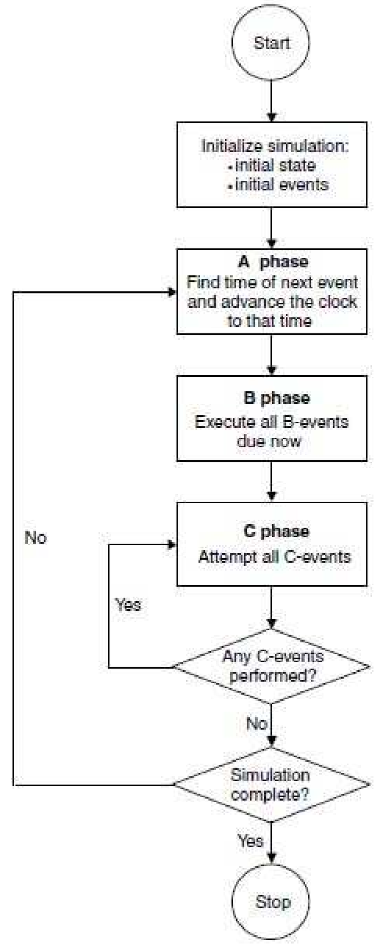

Figure 1. Flowchart of the three-phase approach [25]

order to create a realistic starting condition. The initial B-events are also scheduled. All future scheduled events are recorded and listed in an event list, which holds all the scheduled events.

Following the initialization setup, the simulation moves into the execution of the three phases, which are repeated continuously. In the A-phase, known also as the simulation executive, the time of the next event is determined by inspecting the time of the first event in the list. All B-phase events due at the present clock time are executed. In the C-phase all C-events are attempted and those for which the conditions are met are executed. Since the successful execution of a C-event may signify that another C-event can now be executed, the simulation continues to attempt all C-events until no further events can be executed. The simulation then returns to the A-phase unless the event list queue is empty.

2.2 Network Traffic Analysis

Various simulators that model network traffic have been suggested in literature, amongst them [25], [20], [17]. We decided to opt for a discrete-event simulator. Identification of the B-phase and C-phase events are crucial for accurate modeling, and their implementation necessitates knowledge on the Medium-Access Control (MAC) mechanism.

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The IEEE 802.11 standard incorporates two medium access methods: the Distributed Coordination Function (DCF) method is mandatory while the Point Coordination Function (PCF) which provides Time Bounded Services (TBS) is optional [14]. DCF is an asynchronous data transmission function.

The DCF is based on the Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) protocol. Carrier sensing can be performed on both the physical and MAC layers. On the physical layer, physical carrier sensing is done by detecting any channel activity by other stations [3]. In addition to the physical channel sensing, virtual carrier sensing is achieved by using time fields in the frames, which indicate to other stations the duration of the current transmission. All stations that hear the data or the RTS frame, update their Network Allocation Vector (NAV) field, based on the value of the duration field in the received frame. When performing a system simulation only virtual carrier sensing is possible. To make the simulation execution time faster, a technique known as Lazy MAC State Update, proposed in [16], is integrated in our model, where the NAV is continuously updated only for the stations involved in the transmission. On the other hand, the NAV of the other stations is only updated when one of the stations tries to send some data over the channel.

Two access schemes are used by the DCF; a basic access mode, shown in Figure 2, or a handshaking access method based on RTS/CTS, shown in Figure 3. Refer to [14] for the exact timings of the mechanisms referred to by the labels in the abovementioned figures. The performance of CSMA protocols was reviewed in various research papers, amongst them [2], [6], and [13].

The B-phase and C-phase events are identified from the above mentioned system flow. The following list includes all the identified B-phase events:

- Generate new transmission packet at the server's side
- Generate Slot Time
- Generate DCF-IFS (DIFS) time
- Generate packet arrival time at station's side
- Generate short inter frame space (SIFS) time
- Generate packet arrival time at server's side
- Generate ACK timeout time
- Generate extended inter frame space (EIFS)
- Generate arrival time of 1st bit at server's side.

Another list was generated for all the C-phase events.

The data transmitted on the channel is composed of the RTP packets generated by the H.264/AVC codec. The chosen test video sequence, *Foreman*, has a length of about 10 seconds but we repeat it 6 times such that the DES could simulate the traffic for a 1 minute sequence. The format of the considered video is the Common Intermediate Format (CIF), having a video resolution of 352*288 pixels. Such video resolution is adequate for the receiver stations used in a business jet. Literature suggests a minimum PSNR of 30dB for an adequate video quality [22]. Table 1 summarizes the PSNR values for different compression bit rates for the *Foreman* test sequence using 4 different encoding schemes. The difference between each scheme is the entropy slice. Two of the rows show the PSNR for Context-Adaptive Binary Arithmetic Coding (CABAC) as entropy coding, with a maximum slice length of 200 bytes (CABAC1) and 1000 bytes (CABAC2), respectively. The other two rows evaluate the performance of Context-Adaptive Variable-Length Coding (CAVLC) with maximum permissible slice length of 200 bytes (CAVLC1) and 1000 bytes (CAVLC2), respectively. CABAC typically offers an enhancement in coding efficiency; resulting in savings of 5% to 15% when compared to CAVLC [21]. For this application, bit-rate reduction is important as we need to maximize coding efficiency in order to share the maximum bit rate of 54Mbps between the highest number of possible users. The Maximum Transmission Unit (MTU) for the MAC layer of the IEEE802.11a/g [1] is 2300bytes; this limit goes down to 1500 when the system forms part of an IEEE802.3 infrastructure.

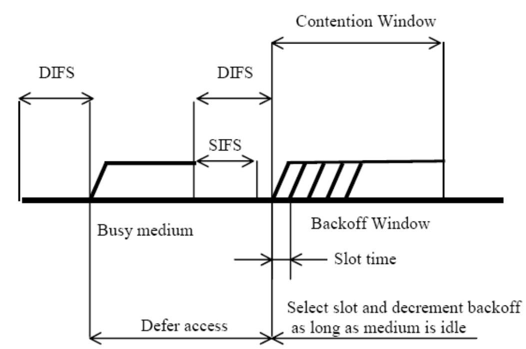

Figure 2. Basic access method [3]

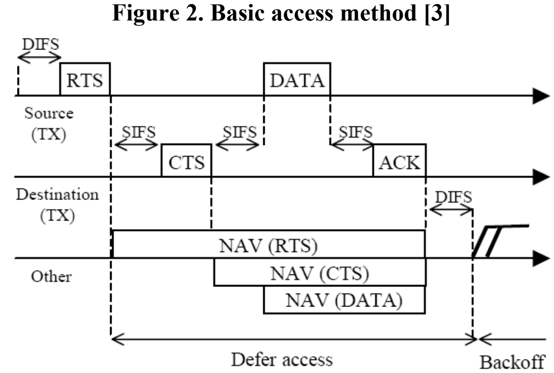

Figure 3. RTS/CTS access scheme [3]

In order to achieve 30dB of PSNR, the minimum bit rate required is 96kbps. Mathematically, PSNR is the logarithmic representation of the mean squared error (MSE) between two sequences [28];

$$MSE = \frac{1}{N_x . N_y} \sum_{x=0}^{N_x-1} \sum_{y=0}^{N_y-1} \left[F(x,y) - R(x,y) \right]^2 \qquad (1)$$

$$PSNR = 10 \log_{10} \frac{255^2}{MSE} \qquad (2)$$

where *F(x,y)* represents the data of the uncompressed video stream, *R(x,y)* represents the data of the compressed video stream, N_x is the number of pixels in the *x* direction, and N_y is the number of pixels in the *y* direction.

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Table 1. PSNR at different bit rates and encoder settings

Sequence Reference	bit rate [kbps]					
	64	96	128	256	512	1024
CABAC1	30.37dB	31.10dB	32.48dB	35.71dB	38.06dB	41.41dB
CAVLC1	29.97dB	30.46dB	31.92dB	35.38dB	37.90dB	41.18dB
CABAC2	30.66dB	31.59dB	32.88dB	35.94dB	38.25dB	41.63dB
CAVLC2	30.22dB	30.89dB	32.37dB	35.62dB	38.03dB	41.38dB

The easy way of calculating the PSNR makes it the most popular objective quality metric for video quality. Despite its popularity, PSNR only has an approximate relationship with the video quality perceived by human observers; simply because it does not take into account what each byte actually represents [29].

The H.264/AVC coder has a number of error resilient/error concealment features that can be enabled in order to enhance the end-to-end picture quality [21]. Error resilient features, like data partitioning, flexible macroblock ordering (FMO), and arbitrary slice ordering (ASO), are generally important for low delay applications, such as live video conferencing, at the expense of less compression efficiency. As our study is focused on streamed video, no error resilient features are enabled. The decoder system uses the standard error concealment to make up for lost packets introduced by transmission errors.

3. SIMULATION

3.1 Business Jet Environment

The simulation environment is the aircraft cabin of a Dassault Aviation business jet. The business jet considered is a high speed short to medium haul aircraft. Its seating capacity depends upon the interior design chosen. The aircraft has a maximum cabin height of 1.88m, a maximum cabin width of 2.34m, and cabin length of 7.06m.

In our previous study [8], we examined the channel parameters inside the aircraft. These parameters are characterized by the time dispersion parameters whose main components are: (i) the mean excess delay and (ii) the root mean square (rms) delay spread. These parameters give a clear indication whether the channel will be introducing inter-symbolic interference in an IEEE802.11a system once deployed on-board the Falcon business jet.

The mean excess delay, τ_m, is defined as [24]:

$$\tau_m = \frac{\sum_k \left\{ \left(\tau_k - \tau_1 \right) a_k^2 \right\}}{\sum_k a_k^2} \tag{3}$$

while the rms delay spread, τ_{rms}, is defined as [24]:

$$\tau_{rms} = \left(\frac{\sum_k \left\{ \left(\tau_k - \tau_m - \tau_1 \right)^2 a_k^2 \right\}}{\sum_k a_k^2} \right)^{\frac{1}{2}} \tag{4}$$

where a_k represents the amplitude, τ_k is the time index, and τ_1 is the time index of the first arriving signal at the receiver.

Coherence bandwidth, B_c, is a measure of the flatness of the channel, i.e. the range of frequencies over which two frequency components have a strong potential for amplitude correlation [24]. Coherence bandwidth is a relation derived from the rms delay spread. Two signals with frequency separation less than or equal to B_c are affected similarly by the channel, otherwise the channel will introduce frequency selective fading.

For a frequency correlation function of 0.9 or above, the coherence bandwidth is approximately [24]:

$$B_c \approx \frac{1}{50\sigma_\tau} \tag{5}$$

where σ_τ is the rms delay spread. For a frequency correlation function between 0.5 and 0.9, the coherence bandwidth can be approximated by [24]:

$$B_c \approx \frac{1}{5\sigma_\tau} \tag{6}$$

In this study, four candidate station locations are considered inside the cabin. A static channel is used, i.e. the passengers are seated in their seats and no hostesses are walking up the aisle. Table 2 provides a summary of the time dispersion parameters at 3 of the 4 chosen positions. The missing position was so close to the transmitter that it didn't experience multipath, consequently no time dispersion parameters are available for this station.

The results presented in Table 2 show that the maximum rms delay spread is 3.9ns for the station in position 4. This is much less than the limit of 800ns for the guard interval (GI) given by the IEEE802.11a standard [14]. Therefore, this guarantees that the station will not experience inter-symbolic interference (ISI) or inter-carrier interference (ICI). The coherence bandwidth results show that the channel will only introduce flat fading as the values obtained are much greater than the bandwidth of each subcarrier in the system, i.e. 312.5kHz. Consequently the bit-stream is passed through an Additive White Gaussian Noise (AWGN) channel and no selective fading is considered. The level of noise introduced dependents on the SNR at each receiver location. Table 3 gives the SNR values at these locations. The chosen positions are located as follows: Position 1 is close to Window 7; Position 2 is in the front part of the business jet; Position 3 is close to Window 3; Position 4 is located in the back part of the cabin; while the access point is above Window 8.

Table 2. Time dispersion parameters at the receivers [8]

Time Dispersion Parameters			
	Position 2	Position 3	Position 4
Mean Excess Delay (s)	1.35E-08	1.00E-08	5.40E-09
RMS Delay Spread (s)	2.76E-09	2.29E-09	3.90E-09
Coherence Bandwidth 50% (Hz)	7.24E+07	8.73E+07	5.13E+07
Coherence Bandwidth 90% (Hz)	7.24E+06	8.73E+06	5.13E+06

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Table 3. SNR values at all 4 candidate locations [8]

Station	SNR (dB)
Position 1	27 ± 2
Position 2	18 ± 2
Position 3	21 ± 2
Position 4	14 ± 2

3.2 Simulation Parameters

The only assumption considered is that no hidden stations are present within the network. Hidden stations are stations visible to one or more of the receivers but not visible to the transmitter (in this case the server). A hidden station introduces collisions. As the environment considered was a business jet (relatively small dimensions) and from results obtained in [8], all locations on board the aircraft have sufficient signal strength to ensure adequate communication with the access point, hence the assumption considered represents the true picture.

To make the system simulation faster, when a new user (station) requests a multimedia stream from the server; the simulation software retrieves all the RTP packets, generated by the H.264/AVC encoder, of the requested video and encapsulates each packet with the necessary header and/or trailer to form the UDP message, IP datagram, MAC frame and the physical layer bit stream [15], [10], and [14]. This was preferred to encapsulating every packet on its own, mainly because some routines can work on the whole stream together and not sequentially, resulting in a faster simulation time.

B-phase event generation time was taken as a geometrically distributed random variable with a mean of 1ms; at execution time the next packet to be transmitted will be moved to the transmission queue. After performing virtual carrier sensing, contention sequence is simulated (taking into consideration slot time, and contention window) as per IEEE802.11 standard [14]. Finally, the packet is transmitted to its intended recipient with the generated errors based on the discussion above.

At the receiver's side, the packet is de-encapsulated and if correctly received an acknowledge packet (ACK) is sent back. Otherwise, if an error is detected no reply is sent back towards the server. At the server's side, two scenarios are considered:

If timeout elapsed, as a result of lack of successful reception of an ACK frame, the server will retransmit the packet if the maximum number of retransmissions has not been exceeded. A good balance is found to allow up to 5 retransmissions, after which the packet is discarded. For higher threshold values, the number of packets at the transmission queue will increase and the server will saturate resulting in no multimedia transmissions being received by any passenger, else delay times will be so long that decoders at stations will suffer from deadline starvation issues. In conjunction with the retransmission of packets, another adaptive issue is considered – reducing the modulation parameters of the PHY layer. For IEEE802.11a there are 8 transmission bit rates, each having a different set of modulation parameters [14]. The adaptive modulation scheme was suggested and optimized by [4]. This algorithm suggests that two counters are kept one for successful transmission of packets and another for the unsuccessful ones. When timeout is reached the successful counter is reset while the unsuccessful one is incremented, conversely when a packet is successfully transmitted the unsuccessful counter is reset while the successful counter is incremented. Two different thresholds are set for the two counters. On reaching these thresholds, the modulation scheme is increased or decreased depending on which counter reaches the threshold. According to [30], a system where the transmission errors are the consequence of the channel quality rather than collisions on the medium, the best choice for these threshold values are 1 for the unsuccessful counter and 3 for the successful counter.

When an ACK is successfully received, the current time of execution is recorded. In addition, the number of retransmissions will also be recorded. These are used to obtain the link utilization, goodput, and delay of the system. The output stream is also recorded.

3.3 Simulation Assumptions

Following previous discussion on the MAC layer, the MTU can directly transmit any packet having a length equal to or less than 1500 bytes. Depending on the quality of the channel, such long packets might introduce delay problems, as errors will imply retransmission of these long packets. As we already discussed, delay is an important issue to be considered when the data being transmitted is multimedia information. [18] suggests packet lengths of around 200 to 250 bytes as an optimum value for multimedia transmission over wireless mediums. On the other hand, for normal data with packets lengths of around 1000 bytes the basic access scheme can be still used.

The H.264/AVC encoder can be programmed to generate NAL packets with a predetermined length, else we can have longer NAL packets which will be fragmented by the MAC layer into shorter MAC packet data units (MPDU) of the required length. While the latter is the preferred way for data transmission, for H.264/AVC this will negatively affect the error resilience of the system. Figure 4 and Figure 5 highlight these two scenarios. In Figure 4 we have a NAL packet containing a full video frame. This frame was transmitted within 4 different MPDUs, one of which got lost due to errors. The effect at the decoder side is that the full frame needs to be discarded and concealed. Conversely, Figure 5 depicts a situation where the NAL packets are already of the required length, hence there is no need for MAC fragmentation. As before one MPDU got lost due to transmission deficiencies, but this time only the lost slice is concealed.

The simulation duration depends upon the length of the transmitted multimedia file. When the timeline queue is empty, no more events have been scheduled, hence all transmission packet have been successfully/unsuccessfully transmitted.

In our study, we used the RTP packets generated by the encoder. It is a general practice that important packets like I and P frames are considered to be successfully delivered always [26]. The same consideration holds for packets like Sequence Parameter Set (SPS) and Picture Parameter Set (PPS). We transmitted all packets with the same priority, hence any errors introduced by the channel, affects all packets with the same probability.

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At the PHY layer, the MPDU are encapsulated to create the Physical Layer Convergence Protocol (PLCP) Protocol Data Unit (PPDU). The simulation software assumes that all the PPDU are transmitted with the same power level and no synchronization problems are introduced by the medium. Also the channel conditions are assumed to remain constant for the whole duration of the packet.

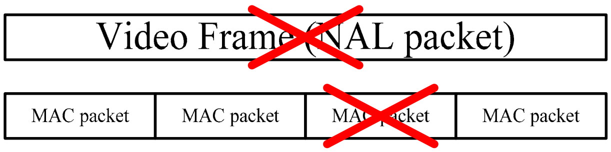

Figure 4. MPDU loss on MAC fragmented Video Frame

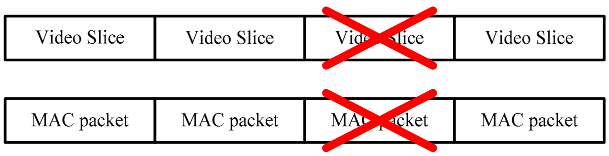

Figure 5. MPDU loss on Video Slice packets

4. RESULTS

The *Foreman* sequence is encoded and packetized before feeding it to the simulator. Table 4 maps a short name to each of the sequences thus generated to help in referencing each sequence.

As the channel conditions are static, only two sequences are required to evaluate the end-to-end delay and to analyze how this will be affected by the packet lengths. The chosen sequences are Sequence1 and Sequence3. Figure 6 represents the end-to-end delay experienced by each packet for the two different scenarios. The maximum delay experienced for Sequence1 and Sequence3 is 1.83ms and 163ms respectively. Both values are below the limit of 1s allowed for streaming applications and are also lower than the limit of 200ms for live conferencing. Figure 7 summarizes the number of retransmissions experienced by the same two sequences. For Sequence1, 2774 packets out of the 6125 packets or 45.29% are successfully transmitted from the server to Station 1 at the first attempt. This drops to 1243 for Station 4, i.e. 20.29%. For Sequence3 the successful transmission rate on the first attempt is 38.19% for Station 1 and 18.39% for Station 4. Hence, the difference in end-to-end delay is the result of the number of packet retransmissions. This mainly occurs because the system reduces the modulation scheme each time a packet is not successfully transmitted, resulting in longer propagation times.

The QoS is inversely proportional to the packet error rate (PER), which is directly affected by the length of the packets and the channel's SNR. PER results are also obtained from Figure 7. For Sequence1 a total of 38 packets (9 (0.15%) for Station 2 and 29 (0.49%) for Station 4) are lost during transmission, hence they never reach the destination. The total number of unsuccessful packet transmissions for Sequence2 increases to 279 (59 (2.75%) for Station 2, 2 (0.09%) for Station 3, and 218 (10.18% for Station 4). Refer to Table 5 for a summary for all 4 stations (S1, S2, S3 and S4) unsuccessful packet rate (UPR) and the mean number of retransmissions (MRTS).

Table 4. Name to sequence mapping

Name	Packet length	Bit rate [kbps]	Entropy Coding	PSNR [dB]
Sequence1	200	96	CABAC	31.10
Sequence2	200	96	CAVLC	30.46
Sequence3	1000	96	CABAC	31.59
Sequence4	1000	96	CAVLC	30.89

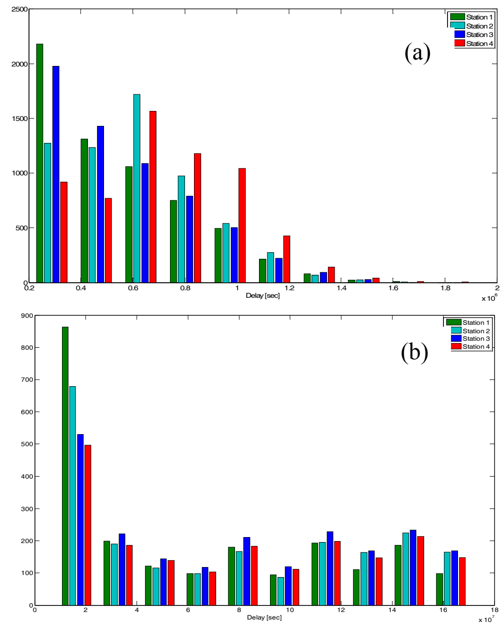

Figure 6. Packets End-to-End Delay for all 4 stations for (a) Sequence1; and (b) Sequnece3

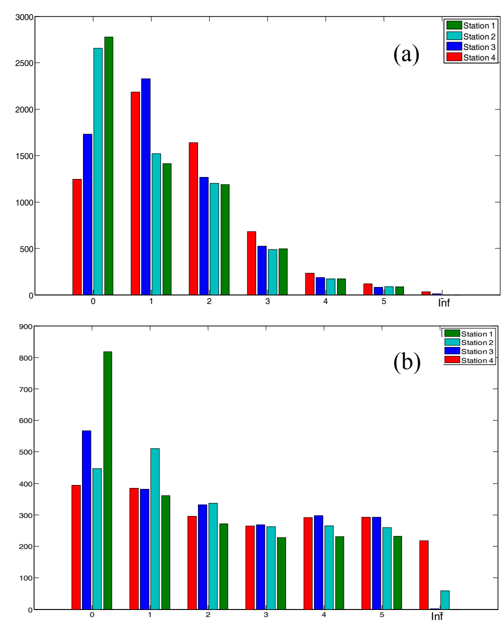

Figure 7. Number of retransmissions for (a) Sequence1 and (b) Sequence2 for all 4 stations

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Furthermore, we decode the received packet sequence and subsequently compute the resulting PSNR for all the 4 sequences at the 4 distinct receiver stations. As some of the sequences have suffered major packet losses, the standard error concealment algorithm in the JM Software V14.2 wasn't able to recover all the frames, resulting in dropped frames, and consequently the received video sequences are shorter than the original. Thus, the PSNR cannot be evaluated for such sequences. Table 6 lists the PSNR values for the fully decoded sequences. As expected, the stations with good channel conditions have PSNR values with good correlation to the full sequence. Station 1 has successfully received all 4 sequences with PSNR values equal to the transmitted sequence. Station 3 has successfully received both Sequence1 and Sequence2 but dropped frames were present for the other 2 sequences. Station 2 and station 4 have adverse channel conditions, resulting in a reduction in the PSNR values even for the sequences that have not experienced any loss of frames. Station 4 cannot decode the packets received for Sequence4 as it lost both the PPS and the SPS.

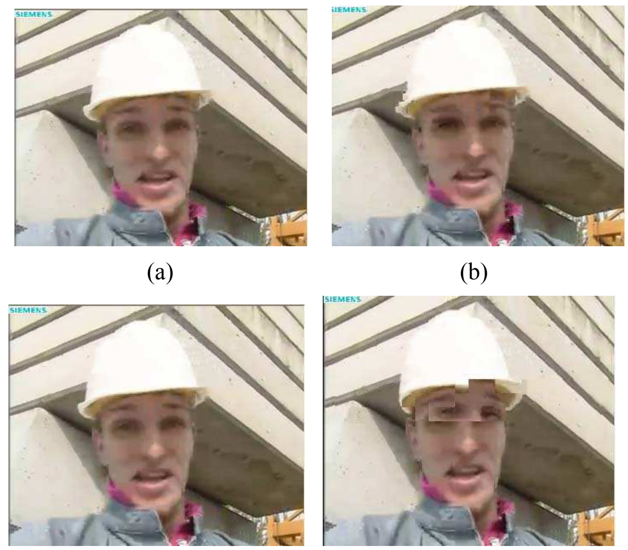
Figure 8 displays a frame of the *Foreman* sequence (Sequence1) as it appears on each of the four stations. It is evident that the video sequence received by Station 4 has suffered the most errors as there are some clearly visible artifacts in the picture. Packet losses will not only affect the current frame, but depending on the frame type, could affect previous and future frames, through error propagation. The artifacts present on the face for the frame on Station 2 are the result of a packet loss in frame 735.

Table 5. Retransmissions and Failure rates of all 4 stations

		Sequence 1	Sequence 2	Sequence 3	Sequence 4
S1	UPR	0	0	0	0
	MRTS	1.04	1.06	1.71	1.67
S2	UPR	9 (0.15%)	3 (0.05%)	59 (2.75%)	49 (2.60%)
	MRTS	1.25	1.26	2.19	2.15
S3	UPR	0	0	2 (0.09%)	2 (0.11%)
	MRTS	1.06	1.07	2.11	2.08
S4	UPR	29 (0.47%)	3 (0.52%)	218 (10.18%)	186 (10.10%)
	MRTS	1.5	1.53	2.67	2.64

Table 6. PSNR at the 4 receiver stations

Station	S1	S2	S3	S4
Sequence1	31.10dB	28.61dB	31.10dB	27.35dB
Sequence2	30.46dB	30.32dB	30.46dB	28.93dB
Sequence3	31.59dB			
Sequence4	30.89dB			



(a) (b) (c) (d)

Figure 8. Frame740 from *Foreman* Sequence1 as experienced by (a) Station 1, (b) Station 2, (c) Station 3, (d) Station 4.

5. CONCLUSIONS

The results presented in this paper show that an adequate quality video streaming service is possible on board the business jet if we limit the length of the packets being transmitted on the channel. Furthermore, delay results suggest that live conference services are possible, but further study needs to be performed in order to model the bi-directional video packet transmission system and verify that the experienced delay will still remain below the 200ms limit.

As expected, PER is found to be higher in transmission using longer packets when compared to short packet transmissions. Station 1 is not affected by the packet's length as it is close to the transmitter. On the other hand, Station 4 has the worst channel conditions, and therefore its PER is higher resulting in PSNR values that are below the 30dB target even for the short packet sequences. To mitigate this, we suggest that packet size should be shorter than 200bytes. Moreover, intelligent techniques can be applied to reduce retransmission rate and maximize throughput.

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7. REFERENCES

[1] Bianchi G. 2000. Performance Analysis of the IEEE802.11 Distributed Coordination Function. In IEEE Journal on Selected Ares in Communications vol. 18, no. 3, (March 2000), 535 – 547.

[2] Bianchi G., Fratta L., and Oliveri M. 1996. Performance Evaluation and Enhancement of the CSMA/CA MAC

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http://dx.doi.org/10.4108/ICST.SIMUTOOLS2009.5655

Protocol for 802.11 Wireless LANs. In Proceedings of PIMRC (1996).

[3] Chatzimisios P., Vitsas V. and Boucouvalas A. C. 2002. Throughput and delay analysis of IEEE 802.11 protocol. Proc. of the 5th IEEE International Workshop on Networked Appliances, (Oct. 30-31, 2002), 168-174.

[4] Chevillat P., Jelitto J., Barreto A. N., and Truong H.L. 2003. A Dynamic Link Adaptation Algorithm for IEEE 802.11a Wireless LANs. In Proceedings to ICC'03, vol. 2, (Anchorage, AK, May 11 -15) 1141 – 1145.

[5] comScore website: http://www.comscore.com

[6] Crow B. P., Widjaja I., Kim J. G., and Sakai P. T. IEEE (1997) 802.11 Wireless Local Area Networks. In IEEE Communication Magazine, 1997.

[7] Dassault Aviation website: http://www.dassault falcon.com

[8] Debono C.J., Chetcuti K., and Bruillot S. 2009. 802.11a Channel Parameters Characterization on board a Business Jet. In Proceedings of IEEE Aerospace 2009 (Big Sky, Montana, March 7 – 14).

[9] E-Cab consortium website http://www.e-cab.org

[10] Gast M. 2002. 802.11 Wireless Networks – The Definitive Guide.

[11] H.264/AVC Software Coordination, "JM Software," ver. 14.2, [Online], Available: http://iphone.hhi.de/suehring/tml.

[12] Hadzi-Velkov, and Z., Spasenovski, B. 2003. Saturation Throughput-Delay Analysis of IEEE802.11 DCF in Fading Channel. In Proceedings of the IEEE International Conference on Communications, (May 11-15, 2003), 121-126.

[13] Ho T., and Chen K. 1996. Performance Analysis of IEEE 802.11 CSMA/CA Medium Access Control Protocol. In Proceedings of PIMRC'1996 (Taipei, Taiwan), 407-411.1996.

[14] IEEE Standard for Information technology — Telecommunications and information exchange between systems — Local and metropolitan area networks — Specific requirements Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications.

[15] Kozierok C. M. 2005. The TCP/IP Guide – Version 3.

[16] Lee Y.J., and Riley G. F. 2005. Efficient Simulation of Wireless Networks using Lazy MAC State Update. In Proceedings of the Workshop on Principles of Advanced and Distributed Simulation (Monterey, California, June 1-3, 2005).

[17] Liu, Y., Presti, F. L., Misra, V., Towsley, D., and Gu, Y. 2003. Fluid models and solutions for large-scale ip networks. In Proceedings of ACM SIGMETRICS'03 (San Diego, Calif.). ACM, New York.

[18] Magis Networks, "IEEE 802.11e/a Throughput Analysis, E10282," Magis Networks Inc, Tech. Rep., 2003, http://www.magisnetworks.com.

[19] Majumdar A. Sachs D.G., Kozintsev I. V., Ramchandran K. and Yeung M.M. 2002. Multicast and Unicast Real-Time Video Streaming Over Wireless LANs. In IEEE Transactions on Circuits and Systems for Video Technology, vol. 12, no. 6 (June 2002), 524 – 534.

[20] Noble B.D., Satyanarayanan M., Nguyen G. T., and Katz R. H. 1997. Trace-Based Mobile Network Emulation. In Proceedings of ACM SIGCOMM'97 (Cannes, France). ACM, New York.

[21] Ostermann J., Bormans J., List P., Marpe D. Narroschke M.,Pereira F., Stockhammer T. and Wedi T. 2004. Video coding with H.264/AVC: tools, performance, and complexity. IEEE Circuits and Systems Magazine, vol. 4, no. 1, (1st quarter 2004), 7-28.

[22] Park Y., Seok Y., Kwon T., ChoiY., and Bonni J.M. 2005. A Rate-Adaptive Multimedia Multicasting Mechanism in Multi-Rate 802.11 Wireless LAN. In Adaptation Techniques in Wireless Multimedia Networks. Nova Publishers.

[23] Pidd, M. 1998. Computer Simulation in Management Science, 4th edn. Wiley.

[24] Rappaport T.S. 2002. Wireless Communications Principles & Practice. Prentice-Hall.

[25] Robinson S. 2004. Simulation: The Practice of Model Development and Use. Wiley.

[26] Seeling P., Reisslein M., and Kulapala B. 2004. Network Performance Evaluation Using Frame Size and Quality Traces of Single-Layer and Two-Layer Video: A Tutorial. In IEEE Communications Surveys & Tutorials (3rd quarter 2004).

[27] Tocher, K.D. 1963. The Art of Simulation. The English Universities Press.

[28] Van der Auwera G., David P.T., and Reisslein M. 2008. Traffic and Quality Characterization of Single-Layer Video Streams Encoded with H.264/AVC Advanced Video Coding Standard and Scalable Video Coding Extension. In IEEE Transactions on Broadcasting vol. 54, no. 3, September 2008.

[29] Winkler S., Mohandas P. 2008. The Evolution of Video Quality Measurement: From PSNR to Hybrid Metrics. In IEEE Transactions on Broadcasting vol. 54, no. 3, September 2008.

[30] Yomo H., Nguyen C.H., Kyritsi P., Nguyen T.D., Chakraborty S. S., and Prasad R. 2005. PHY and MAC Performance Evaluation of IEEE 802.11a WLAN over Fading Channels. In IETE Journal of research vol. 51, no. 1, 83 – 94.

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