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Performance Evaluation of Video Streaming with Background Traffic over IEEE 802.11 WLAN Networks

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ABSTRACT

There is an increasing demand for multimedia streaming applications over WLAN networks. MPEG-4 and H.264 are compression standards targeted at high-quality streamed multimedia services over wireless best-effort IP networks. However, the dynamic nature of wireless networks in terms of fluctuating bandwidth and time-varying delays makes it difficult to provide good quality streaming under such constraints. Multimedia streaming applications are a demanding and challenging service to deliver over wireless networks. There is a trade-off between the capacity of the wireless network and the quality of the multimedia streaming application. In this paper we investigate the effect the background traffic load has on unicast streaming video sessions. We show that above a certain load value, the video streaming session is slowly starved of bandwidth. The load value at which this occurs depends on the characteristics of the background traffic load in terms of packet rates and the number of sources contributing to the load.

Categories and Subject Descriptors

C.2.[**Computer Communications Networks**]: Network Architecture and Design – Wireless communication

General Terms

Measurement, Performance, Experimentation

Keywords

WLAN, Video streaming, MPEG-4 Encoding, Performance Evaluation.

1. INTRODUCTION

In recent years there has been an explosive growth in the use of wireless LANs arising from the advent of the IEEE 802.11b standard. Streaming multimedia over wireless networks is becoming an increasingly important service. These applications impose stringent demands on the network in order to ensure that users enjoy an "acceptable level" of QoS. In wired networks the QoS targets for multimedia applications can be met by over-provisioning. However, such an approach cannot be adopted with wireless networks due to the limited network resources. There are many performance related issues when using wireless networks. The main difficulty is that wireless networks allow for much lower delivery rates than wired networks where typically up to 100Mbps can be supported. For example, a wireless IEEE 802.11b network can support rates up to 11Mbps, whereas using IEEE 802.11g up to 54Mbps can be reached. Yet in practice the effective throughput data rates are approximately half these values. Wireless networks are particularly error-prone and since they use radio waves, the data signals are subject to attenuation with distance and signal interference. In addition, the transmission quality is also affected by contention between users who are attempting to access and transmit data on the shared radio channel. This contention results in users having to wait until their backoff process is complete before they can access the channel. All these factors ultimately affect end-user perceived quality. Support for such traffic with QoS requirements is being addressed by the IEEE 802.11e Task Group. However, IEEE 802.11e is only a QoS enabling mechanism that requires some higher level management functionality in order to deliver QoS guarantees. Typically, some form of radio resource management is required to allocate the available resources among the contending users in accordance with their respective needs and priorities.

In order to address the issue of radio resource management for the provision of QoS guarantees, it is first necessary to understand how multimedia streaming applications behave in IEEE 802.11b networks and how they interact with other applications and traffic sources in the network. In this paper, we evaluate the effect that background traffic load has on the video stream. This paper is structured as follows: Section two gives a brief discussion of video streaming, MPEG-4 encoding, MP4 files and the importance of hint tracks. Hint tracks are required to stream MP4 and .3gp multimedia files and indicate to the server how to packetise and transmit the elementary stream. Then we provide an analysis of the video content used during the experiments. Section three, describes the experimental test bed used for the experiments including the streaming system setup, the traffic generator, and the WLAN probe used to measure the resource usage of the WLAN. Section four describes the experiments conducted and presents the results. We show that as the background load is steadily increased, beyond a certain threshold level, the video client becomes slowly starved of bandwidth until the streaming session can no longer be supported and is finally terminated. However, the

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load value at which this occurs varies with the packet size and number of sources contributing to the load. Finally, we present some conclusions and directions for future work.

2. VIDEO STREAMING

Video streaming is a server/client technology that allows multimedia data to be transmitted and consumed. Streaming applications include e-learning, video conferencing, video on demand etc. The main goal of streaming is that the stream should arrive and play out continuously without interruption. However, this is constrained by fluctuations in network conditions. An adaptive streaming server keeps track of the network conditions and adapts the quality of the stream to minimize interruptions and stalling. Real-time streaming can be delivered by either peer-to- peer (unicast) or broadcast (multicast). There are two types of real-time streaming services [1,2], on-demand or live streaming. In addition to the different types of streaming, there are a large and diverse number of variables that must be taken into consideration when evaluating the performance of such applications. Such variables include:

- The actual content and complexity of the content being streamed which in turn affects the efficiency of the encoder to compress the stream. For example, if two different video clips were encoded using the exact same encoding configuration, they would have very different bit rate variations over time.
- The compression scheme being used, that is, different compression schemes have differing levels of efficiency. For example, a 512kbps MPEG-2 stream will have very different characteristics from a 512kbps MPEG-4 stream.
- The encoding configuration [3]. There could be any number of possible encoding configurations possible such as the error resilience, frame rate, the I-frame rate, the quantization parameter, the target bit rate (if any) supplied and target stream type i.e. VBR, CBR or near CBR.
- If the file to be streamed is .MP4 or .3gp, then a hint track must be prepared that indicates to the server how the content should be streamed.
- The streaming server being used, the rate control adaptation algorithm being used, and the methods of bit rate adaptation used by the server [4,5].

In this paper, we focus on unicast video streaming applications for MPEG-4 video encoded clips with near real-time constraints. In the following sections, we provide background information on MPEG-4 encoding and discuss how hint tracks are used as a mechanism to optimally packetise and deliver the multimedia streams over the network. In addition, Section 2.3 provides an analysis of the encoded video content used during the experiments.

2.1. MPEG-4

MPEG-4 dramatically advances audio and video compression, enabling the distribution of content and services from low bandwidths to high-definition quality across broadcast, broadband, wireless, and packaged media [6]. MPEG-4 decomposes a scene into media objects, each with its own audio and video track that will vary over time. The visual part of a media object is known as a Video Object Planes (VOPs). In this paper we consider only rectangular shaped VOPs that correspond to the entire video image and shall refer to them as frames in the remainder of this paper. In the MPEG-4 standard, there are a number of profiles which determine the required capabilities of the player to decode and play out the content. The purpose of these profiles is that a codec only needs to implement a subset of the MPEG-4 standard whilst maintaining inter-working with other MPEG-4 devices built to the same profiles. The most widely used MPEG-4 visual profiles are the MPEG-4 Simple Profile (SP) and the MPEG-4 Advanced Simple Profile (ASP) and are part of the non-scalable subset of visual profiles. The main difference between MPEG-4 SP and ASP is that SP contains only I and P-frames whereas ASP contains I, P and B-frames.

MP4 files comprise a hierarchy of data structures called atoms and each atom has a header, which includes its size and type [7,8,9]. A parent atom is of type *moov* and contains the following child atoms: *mvhd* (the movie header), a series of *trak* atoms (the media tracks and hint tracks), and a movie user data atom *udta*. A *trak* represents a single independent data stream and an MP4 file may contain any number of video, audio, hint, Binary Format for Scenes (BIFS) or Object Descriptor (OD) tracks. Within an MP4 file, each video and audio track must have its own associated hint track. Hint tracks are used to support streaming by a server and indicate how the server should packetise the data. As with MP4 streaming, .3gp files use the "hint track" mechanism for streaming the content, although in .3gp files the BIFS and OD tracks are optional and can be ignored.

2.2. HINT TRACKS FOR STREAMING

Within an MP4 file, each video and audio track must have its own associated hint track. Hint tracks are used to support streaming by a server and indicate how the server should packetise the data. As with MP4 streaming, .3gp files use the "hint track" mechanism for streaming the content, although in .3gp files the BIFS and OD tracks are optional and can be ignored. Streaming media requires that the media be sent to the client as quickly as possible with strict delay requirements. Hint tracks allow a server to stream media files without requiring the server to understand media types, codecs, or packing. Each track in a media file is sent as a separate stream and the instructions for packetising each stream are contained in a corresponding hint track [10]. Each sample in a hint track tells the server how to optimally packetise a specific amount of media data. The hint track sample contains any data needed to build a packet header of the correct type, and also contains a pointer to the block of media data that belongs in the packet. For each media track to be streamed there must be at least one hint track. It is possible to create multiple hint tracks for any track, each optimised for streaming over different networks. Hint tracks have the same structure as media tracks and are atoms of type *trak*. Hint samples are protocol specific by specifying the protocol to be used and providing the necessary parameters for the server. The *stsd* child atom contains transport-related information about the hint track samples. It specifies the data format (currently only RTP data format is defined), the RTP timescale, the maximum packet size in bytes (MTU). The hint track MTU setting means that the packet size will not exceed in the MTU size.

Hint track settings are required for streaming MP4 and .3gp multimedia files and are particularly important for audio streaming since multiple audio samples can be packetised into one packet. In general most video-frames are quite large and so at most one video frame can be packetised into a single 1024B packet. If the video frame is larger than the packet, several packets are required to send the video frame resulting in a group of packets with a size of the hint track MTU setting and a smaller packet containing the remainder information. In the rest of this paper, we shall analyse the effects the hint track MTU setting has on the bandwidth requirements in the WLAN with the understanding that packets vary significantly in size but

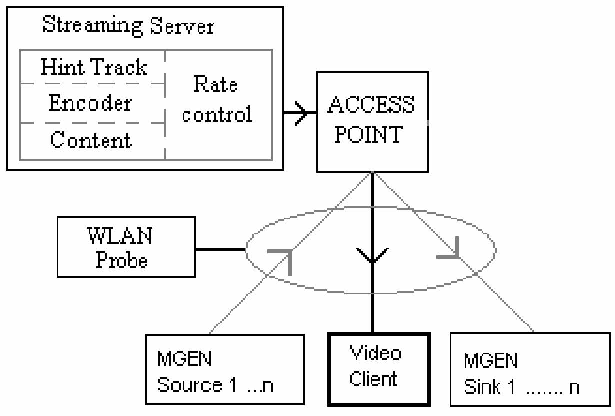


Figure 1. Experimental test bed

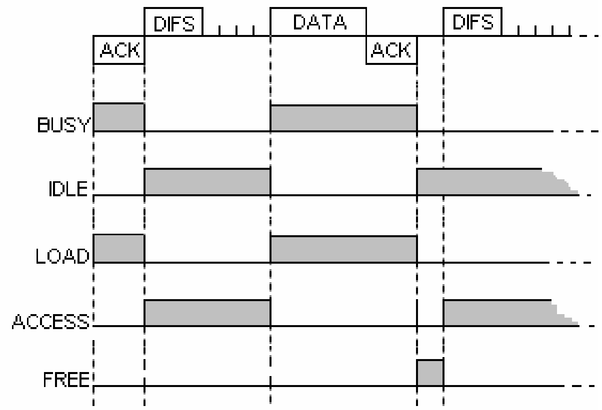


Figure 2. MAC Bandwidth Components

never exceed the hint track MTU setting. The mean packet size for video with a hint track setting of 1024B and 512B is 912B and 468B respectively.

2.3. VIDEO PREPARATION

The video content used in the experiments reported here was encoded using the commercially available X4Live MPEG-4 encoder from Dicas. The video content, 'JR', is a 5 minute extract from the film 'Jurassic Park' with a CIF display size whilst the video clip 'EL' is a 5 minute extract from the animated cartoon, 'The Road to Eldorado'. Animated videos are very challenging for encoders often resulting in very bursty bitrate fluctuations since animations generally consist of line art and as such have greater spatial complexity and detail. Both video clips were encoded using MPEG-4 SP at 25fps and one I-frame every 10 frames. Each clip was then subsequently hinted with an MTU of 1024B and/or 512B using MP4Creator from the MPEG4IP Project [11].

3. EXPERIMENTAL TEST BED

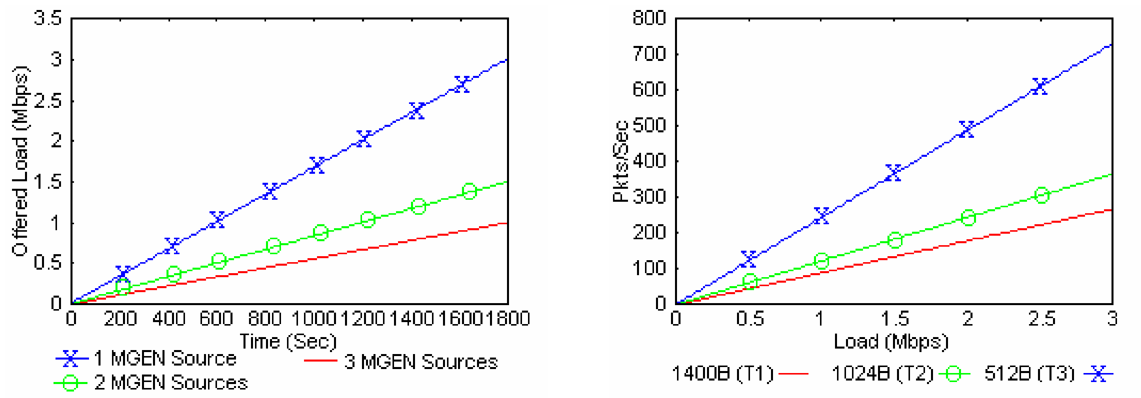
The WLAN test bed shown in Figure 1 consists of a video server on the wired network that is streaming unicast video to a client connected to the WLAN. The video stream is relayed from the wired network to the client via the Access Point (AP). The background traffic is generated by a number of stations on the wireless side using the MGEN traffic generator [12] with all background traffic being relayed through the AP. To measure the resource usage a WLAN probe is used. The next sections describe the streaming system and WLAN probe in more detail. The goal of this work is to investigate the effect the background traffic load has on a unicast video streaming session. The experiment is designed such that the client and server establish a streaming connection and the background traffic load is increased over time. At some background traffic load value, the video stream becomes starved of data and as a result severely affects the video stream. The background traffic load can be generated in many different ways, for example, using different packet sizes will result in different packet rates to generate the same load, the number of stations contributing to the total background load and so on. These aspects have been included in our investigation and are described in more detail in Section 3.3.

3.1. STREAMING SYSTEM

There are two open-source streaming servers available, Helix from Real [13,14] and Darwin Streaming Server (DSS) from Apple [15,16]. In this work, we have chosen DSS to be the streaming server for our experiments since it is a typical streaming system that does not employ sophisticated adaptation techniques. DSS is an open-source, standards-based streaming server that is compliant to MPEG-4 standard profiles, ISMA streaming standards and all IETF protocols. The DSS streaming server system is a client-server architecture where both client and server consist of the RTP/UDP/IP stack with RTCP/UDP/IP to relay feedback messages between the client and server. The server is configured with an RTSP timeout of 180sec and RTP timeout of 120sec. The client can be any QuickTime Player or any player that is capable of playing out ISMA compliant MPEG-4 or .3pg content. The client connects to and interacts with the server via RTSP to establish a unicast video streaming session. In addition, RTSP can be used by the client as network remote control to fast-forward/rewind/skip to any location in a pre-encoded video clip with a 3second pre-buffering delay.

3.2. WLAN PROBE

At the wireless side, a WLAN resource monitoring application reported in [20, 21] was used to measure and record the resource utilisation of the video streams. This application non-intrusively monitors and records the busy and idle intervals on the wireless medium and by analysing the temporal characteristics of these intervals infers the resource usage on a per station basis. The WLAN resource utilisation is characterised in terms of MAC bandwidth components that are related to the line rate (Figure 2). Specifically, three MAC bandwidth components are defined: A load bandwidth (BW_{LOAD}) associated with the transport of the traffic stream and is related to the throughput, an access bandwidth requirement (BW_{ACCESS}) that represents the "cost" of accessing the wireless medium, and a free bandwidth (BW_{FREE}). An access efficiency may be defined as the ratio of the BW_{LOAD} to the BW_{ACCESS} and gives an indication of how efficiently a station accesses the medium. The intervals during which the medium is busy correspond to the intervals during which frames are being transmitted on the medium (i.e. data and management frames) and is associated with the transport of the traffic load. The busy bandwidth (BW_{BUSY}) is the portion of the transmission rate used for the transport of the total traffic load and is the sum of the BW_{LOAD} overall stations. Similarly, when the medium is not busy, it is said to be idle. The idle bandwidth (BW_{IDLE})



(a) **(b)**

Figure 3. (a) Offered uplink load for each background traffic station over time (b) Background traffic source packet rates with varying packet sizes

Table 1: Characteristics of UL Background Traffic

Number of Contributing Sources	Test Name	Background Packet Size (B)	Max Pkts/Sec	Max Offered Load Per Source (Mbps)	Total Load (Mbps)
C1	T1	1400	269	3.02	3.02
	T2	1024	368	3.02	3.02
	T3	512	736	3.02	3.02
C2	T1	1400	134	1.51	3.02
	T2	1024	184	1.51	3.02
	T3	512	368	1.51	3.02
C3	T1	1400	89	1.01	3.02
	T2	1024	122	1.01	3.02
	T3	512	245	1.01	3.02

represents the portion of the transmission rate that is idle and may be used by any station to win access opportunities for its load. The sum of BW_{BUSY} and BW_{IDLE} must equal the line rate i.e. 11Mbps in IEEE 802.11b. This technique has been shown to be particularly effective in characterising WLAN resource utilisation in a manner that is both compact and intuitive.

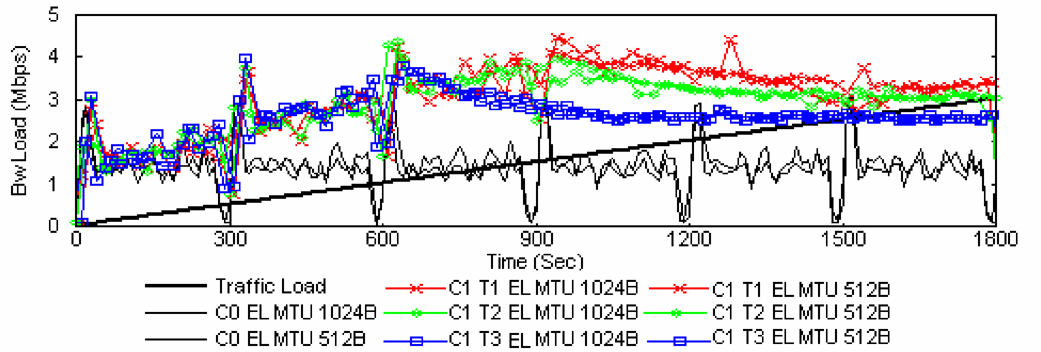
3.3. TRAFFIC GENERATOR

Research has found that the WLAN becomes saturated with a traffic load of approximately 6Mbps [22, 23]. In the experiments described here, the background UDP traffic is tailored such that the total offered uplink (UL) traffic linearly increases to a maximum of 3Mbps over time. The background traffic is relayed through the AP and results in 3Mbps on the downlink (DL). In this way, the total background traffic in the network reaches 6Mbps. In the experiments where there are more than one background traffic source, the offered traffic load is evenly spread across the transmitting background stations. For example, when there is a single MGEN source the offered load was increased from 0 to 3Mbps over a period of 30minutes. Similarly when there are two MGEN sources the offered load per station is increased from 0 to 1.5Mbps over 30minutes. Each test was repeated 3 times using different packet sizes to transmit the same load thus altering the access requirements of the background traffic sources. That is, the smaller the packet size, the more packets that must be sent to achieve the same offered load. Thus, the station has to access the medium more often. A summary of the characteristics of the background traffic used in the various tests can be found in Table 1. The first column indicates the number of contributing sources to the background traffic load. The second and third columns indicate the characteristics of the background traffic and shows the packet size used to achieve the target background load which in turn affects the number of packets per second. Figure 3(a) shows how the offered load per station is increased over time whilst Figure 3(b) shows how the access requirements vary over time to send the same background traffic load.

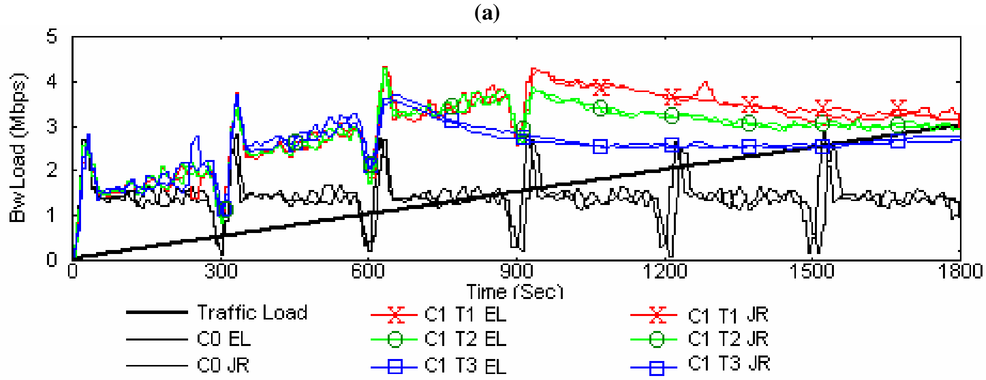
In this paper, we use a naming scheme to identify and compare different test cases. This naming scheme takes the form <Number of Contributing Sources> <Test Name> <Video File> <Hint Track Setting>. Thus, the test ‘C2 T3 JR MTU512B’ refers to the test where there were two sources contributing to the background traffic load, <C2>. Both sources increased their offered load over time to a maximum load of 1.51Mbps using a packet size of 512B, <T3>, resulting in a maximum packet rate of 368 packets per second. The video file that was being streamed during this test was Jurassic Park, ‘JR’, which was streamed using a hint track MTU setting of 512B.

4. RESULTS

The tests were conducted over a testing period of 30minutes. Even though the video clips used had a duration of 5minutes, they were streamed in a constant loop for the duration of each test. Each video clip was hinted with an MTU packet size of 512B and 1024B. Figure 4(a) show the variations over time for the BW_{LOAD} measured at the AP whilst streaming the clip ‘EL’. The tests indicated by the thin black line <C0 EL MTU 1024B> and <C0 EL MTU 512B> show the variations in the BW_{LOAD} when there is no background traffic present. The repeating pattern every 300seconds represents each loop of the video stream. In addition, it can be seen that there is a difference between the measured BW_{LOAD} using the different hint track settings. We have found that by using a hint track MTU setting of 512B increases the BW_{LOAD} by approximately



(a)



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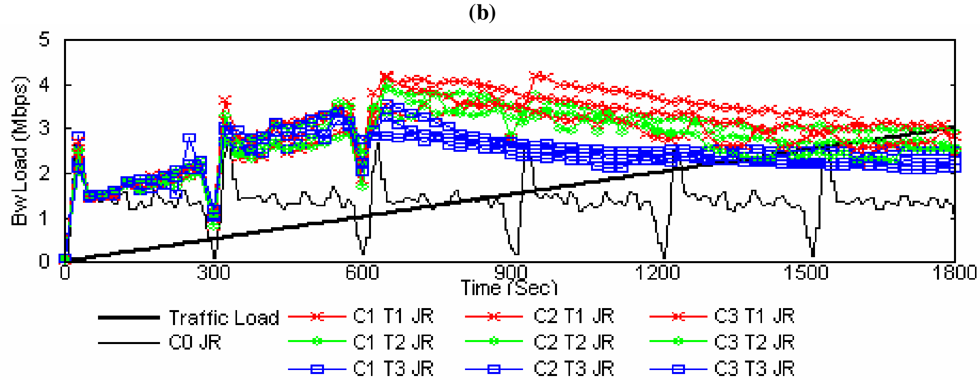

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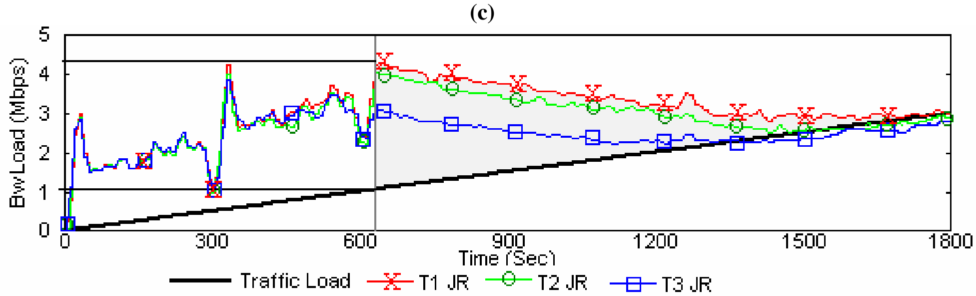


(d)

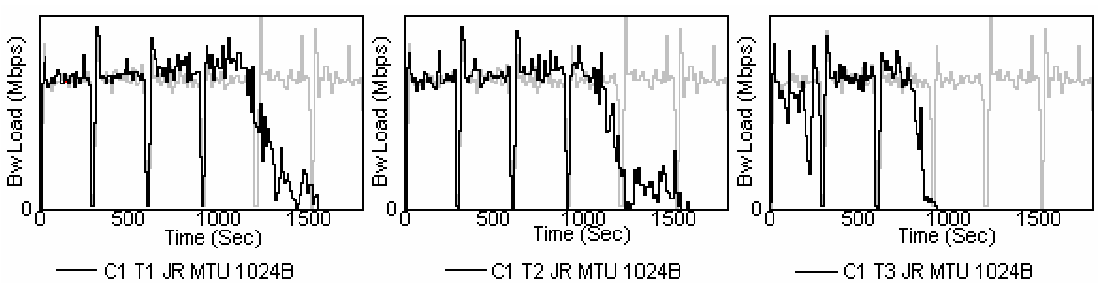
Figure 4. Variations in BW_{LOAD} over time and increased background traffic load (a) for video clip 'EL' with hint track 512B and 1024B (b) averaged behaviour for video clips 'JR' and 'EL' and hint track settings (c) averaged behaviour for clip 'JR' for one to three sources (d) averaged results overall tests

20% due to the additional packet header overhead that needs to be sent and the increased number of ACKs that need to be sent to acknowledge each packet [24]. This difference in BW_{LOAD} can be approximated using the throughput analysis described in [25] for a given load transmitted using different packet sizes.

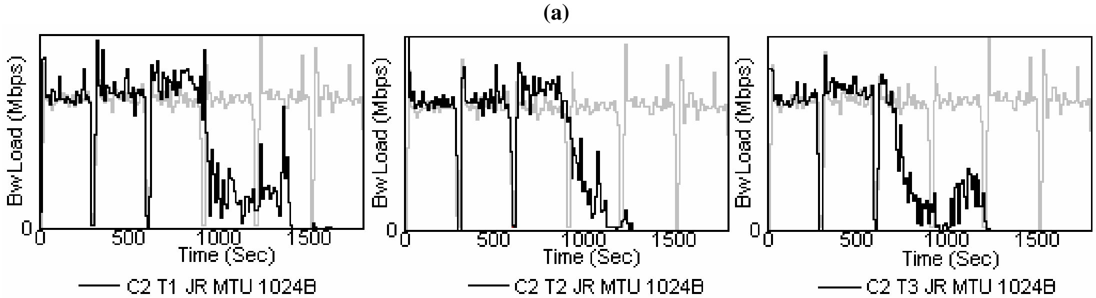
The thick black line shows the background traffic load. The lines marked with an 'x' in tests <C1 T1 EL MTU 1024B> and <C1 T1 EL MTU 512B> show the overall BW_{LOAD} at the AP which includes both the video streaming traffic and the background traffic load generated with a packet size of 1400B. It is expected that as the background traffic load increases, the overall BW_{LOAD} will deviate from the repeated patterns observed when there is no background traffic. It can be seen that at approximately 900seconds, the background traffic load is at approximately 1.5Mbps. At this time, the BW_{LOAD} reaches a maximum value of 4.4Mbps and then begins to decrease over time. This is due to the fact that the AP has become saturated with the offered load. With this total offered traffic load, the client video stream has become starved of data and the video server terminates the video streaming session. Once the video streaming session has been closed, there are still a considerable number of packets in the AP buffer that must be flushed from the queue which results in the gradual decrease in the BW_{LOAD} over time. In particular, it can be seen that the packet size of the background traffic load plays an important role in determining the background traffic load that can be supported before the video stream is severely corrupted and ultimately terminated. It can be seen that the lines marked with a square in <C1 T3 EL MTU 1024B> and <C1 T3 EL MTU 512B> reach a maximum BW_{LOAD} value of 4Mbps at approximately 600seconds when the background traffic load uses a packet size of 512B.

It can be seen that it is the packet rate of the background traffic and the traffic load have the biggest impact on the point of saturation of the AP and the collapse of the video stream. The experiments were repeated using the clip 'JR' and the very same trends were observed when using the different video file. Since we have found that there is a small difference between the measured BW_{LOAD} for streaming video with an MTU or 512B and 1024B, the results for the different video clips in these tests are averaged in Figure 4(b). This graph demonstrates that the video clip being streamed does not alter the behaviour of the system. Figure 4(c) shows the variations in the BW_{LOAD} for one to three sources contributing to the background traffic load using three different packet sizes of 1400B, 1024B, and 512B in tests T1, T2, and T3 respectively averaged over the video clip 'JR' streamed with a hint track MTU setting of 1024B and 512B. It can be clearly seen that the packet size of the offered traffic load plays a more important role in the performance of the WLAN than the number of sources contributing to the traffic load.

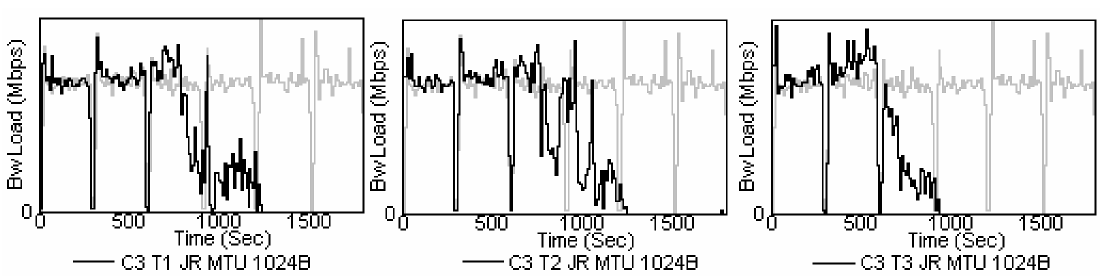
To demonstrate this more clearly and summarise this behaviour, we have averaged the results for each packet size as can be seen in Figure 4(d). We can see that just after 600seconds when the traffic load reaches about the 1Mbps, the AP has reached maximum throughput of 4.4Mbps, 4Mbps and 3Mbps for the tests T1, T2 and T3 respectively. At this point, the video client is slowly starved until it is finally terminated. However, many packets relating to the video stream remain in the buffer which need to be flushed from the system and can be seen as the greyed area in Figure 4(d). The smaller the packet size of the background traffic load, the quicker the system can recover. During this flushing period, the average BW_{LOAD} was 3.35Mbps, 3.05Mbps and 2.4Mbps for the tests T1, T2 and T3. Whilst the BW_{ACCESS} was 2.6Mbps, 2.8Mbps and 3.5Mbps for the tests T1, T2 and T3. It is expected that as the packet rate is increased, the transmitting station spends more time gaining access to the medium and will also use more bandwidth to send the ACKs for each load packet. The relationship between the BW_{ACCESS} and BW_{LOAD} has been investigated in [24]. In contrast, by averaging over the number of contributing sources, the



(a)



(b)



(c)

Figure 5. Variations in BW_{LOAD} over time (a) with one background source, C1 (b) with two background sources, C2 (c) with three background sources, C3

Table 2: Failure Time of Client Streaming Session and Background Traffic Load

Number of Contributing Sources	Test Name	Background Packet Size (B)	Video MTU 1024B		Video MTU 512B		Mean	
			Failure Time (Sec)	Offered UL Load (Mbps)	Failure Time (Sec)	Offered UL Load (Mbps)	Failure Time (Sec)	Offered UL Load (Mbps)
C1	T1	1400	1310	2.20	1350	2.26	1329	2.22
	T2	1024	1230	2.06	1190	2.00	1210	2.03
	T3	512	860	1.45	930	1.57	895	1.50
C2	T1	1400	1400	2.35	1190	2.00	1295	2.17
	T2	1024	1120	1.88	1440	2.42	1280	2.15
	T3	512	930	1.57	890	1.50	910	1.53
C3	T1	1400	1240	2.08	930	1.57	1200	2.02
	T2	1024	1250	2.10	940	1.58	1095	1.83
	T3	512	940	1.58	940	1.58	940	1.58
Mean Values:			1142.2	1.92	1088.9	1.83	1128.2	1.9

average BW_{LOAD} was 3.2Mbps, 2.9Mbps and 2.7Mbps for the tests C1, C2 and C3. Whilst the BW_{ACCESS} was 3.2Mbps, 2.9Mbps and 2.7Mbps for the tests C1, C2 and C3.

Of particular interest is the effect of the traffic load on the video streaming session at the client. Figure 5 shows the measured BW_{LOAD} at the client for each test case. The grey line shows the variations in BW_{LOAD} over time when there is no background traffic and clearly shows the repeated loops in the video streaming session. This is used as a reference to compare the observed BW_{LOAD} with the known BW_{LOAD} at the client. It can be seen that as the background traffic load increases, the BW_{LOAD} received at the video client is reduced. When the background traffic reaches a certain level, the video client is slowly being starved of data until the streaming session fails. Failure is determined as being the time at which the received bit rate at the client falls below the bit rate required to send only the I-frames for a sustained period of 20sec from which the adaptive streaming session cannot recover causing the session to be terminated by the server. This bit rate is approximately 180kbps since on average there are 2.5 I-frames per second in the encoded video sequence and each I-frame is on average 8990B. In Figure 5(a) it can be seen that the video client is relatively unaffected by the background traffic load until the very end of the fourth loop of the video stream at approximately time 1190sec. At this point, the background traffic has reached such a level that it begins to degrade the received stream at the client. When the client detects lost packets and increased packet delays, the RTCP-RR feedback from the client cause the server to adapt the quality of the transmitted stream until only the I-frames are being sent. It can be seen that even though the background traffic load increases at a constant rate, the packet rate of the background traffic load affects the failure time of the client. For example, when there is a single background traffic source, <C1>, that is generated with a packet size of 1400B <T1>, the video client fails at time 1310sec which corresponds to a traffic load of 2.2Mbps. However, for the same number of background traffic sources but using a packet size of 512B <T3>, the video client fails at 860sec and corresponds to a background traffic load of 1.45Mbps. The same behaviour can be seen in Figure 5(b) for two background traffic sources and Figure 5(c) for three background traffic sources.

These results are summarised in Table 2. It can be seen that as the packet size of the background traffic source is reduced, the failure time of the video stream is reduced and the load at the time of failure is reduced. In addition, as the number of sources contributing to the load is increased, the failure time is reduced. It can be seen that the maximum background traffic load of 4.4Mbps can be supported, that is 2.2Mbps on the UL and 2.2Mbps on the DL, although, this is reduced with the number of contributing sources and packet rates of the background traffic. Figure 6 shows the mean values over the various video clips tested and hint track settings. It can be seen that the packet size of the background traffic plays an important role in determining the failure time of the video streaming session. This is expected for a number of reasons. Firstly, the smaller the packet size, the greater the number of packets that are in the queue at the AP awaiting to be transmitted onto the sink station increasing the likelihood of the packet being dropped at the AP. With a greater number of packets, in the queue, the video packets are more likely to be delayed longer since they must wait for the AP to gain access to the medium for each of the packets in the queue ahead of it. It also increases the delay of the video packets and therefore increases the likelihood of a packet arriving past its playout time and being effectively lost. Secondly, with a greater number of packets each station must gain access to the medium more often to transmit its offered load. This increases the level of contention between the stations, increasing the likelihood of a station having to back off. Thirdly, with more packets being transmitted, there is an increased number of ACKs and an increased likelihood of collisions resulting in retransmissions, both increasing the overall load of the system. In addition, by increasing the number of sources contributing to the background traffic load, this increases the contention levels and likelihood of collisions between the stations as mentioned above in the second and third points.

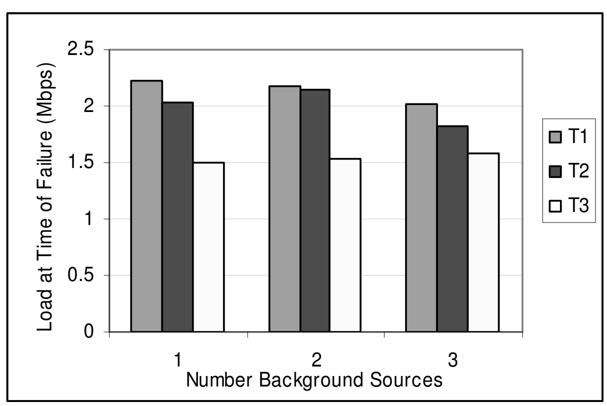


Figure 6. Mean failure times with packet size and number of contributing sources

5. CONCLUSIONS

In this paper we have evaluated the performance of a typical video streaming application in a WLAN environment. The primary goal of this work was to monitor the resource utilisation of the video streaming application under loaded conditions in the WLAN test bed. The performance of the system is measured using a WLAN probe. The probe is used to monitor WLAN resource utilisation in terms of its MAC bandwidth components. In particular, we monitor the load bandwidth (BW_{LOAD}) component that is associated with the transport of data packets and is related to the throughput of the station.

Through experimentation, we have found that the packet size and packet rate of the traffic in the network have a large impact on the video streaming session. In the experiments a video streaming session was established between the video client and server and the traffic load was increased steadily over time. The background traffic load was varied in terms of the packet size and the number of contributing sources to the load. As the load is increased, the throughput reaches a maximum and the AP becomes saturated. We found that after 600seconds with a traffic load of approximately 1Mbps, the AP reached maximum throughput of 4.4Mbps, 4Mbps and 3Mbps for a background packet size of 1400B, 1024B and 512B respectively. At this point, the video client is slowly starved of bandwidth until the streaming session can no longer be supported and the streaming session is finally terminated. However, there are many packets in the buffer relating to this video session which must be flushed from the network. The smaller the packet size of the background traffic load, the quicker these packets can be flushed from the buffers and the network can recover. In contrast, we observed that the number of sources contributing to the load did not make a significant impact on the behaviour of the network. In particular we looked at the effect on the video client whilst it is slowly starved of data. We found that the packet size and packet rate of the background traffic seriously affected the video client. We found that a traffic load of 2.1Mbps, 2Mbps and 1.5Mbps with packet sizes of 1400B, 1024B and 512B respectively was sufficient to starve the video client and result in it being terminated.

Currently work is in progress that investigates the effects of loaded network conditions on video streaming applications. In particular, we are looking at how the bursty nature of video traffic is queued in the AP causing sawtooth delay patterns under different background load conditions and traffic characteristics. Future work is planned to apply knowledge of the behaviour of multimedia streaming applications to enable radio resource management and the provision of statistical QoS guarantees in IEEE 802.11e.

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