Resource Efficient Quality-Oriented Wireless Broadcasting of Adaptive Multimedia Content

Gabriel-Miro Muntean, *Member, IEEE*, and Nikki Cranley, *Member, IEEE*

***Abstract*—The performance of multimedia stream delivery is influenced by encoding scheme, streaming solution and network conditions. This paper studies the performance of multimedia streaming when using the Quality-Oriented Adaptive Scheme (QOAS) over an IEEE 802.11b Wireless LAN and compares it to that achieved when using other solutions that do not consider end-user quality in their delivery process such as TFRC, LDA+, and non-adaptive schemes. The performance is assessed in terms of average end-user perceived quality, number of concurrent streaming sessions, loss rate, delay, jitter and total throughput when streaming MPEG-4 encoded content. Simulation results show that the QOAS out-performs these other streaming solutions in all aspects of network delivery. QOAS can support a greater number of concurrent streaming sessions at a higher average quality. In addition, for the same number of clients QOAS achieved a higher average end-user quality, as well as better network delivery streaming performance parameters.**

***Index Terms*—Adaptive multimedia streaming, end-user quality of experience, wireless local area network.**

I. Introduction

FLEXIBILITY of user location, mobility and low deployment costs are key factors that are driving the exponential growth in the usage of wireless technologies for distribution of services at business premises or residential homes [1]. This trend includes the deployment of Wireless LANs (WLAN) that enable users to access various services including those that distribute rich media content anywhere, anytime and from any device. Fig. 1 shows such a WLAN-based solution [2] for efficient distribution of multimedia content over wireless links. The architecture includes a Smart Access Point (SAP) that acts as a server and provides services to multimedia-enabled clients. Content is either received at the SAP via a wired broadband connection or is acquired locally from a source such as a video camera, DVD player, hard disk, etc.

There are many performance issues related to the usage of wireless networks especially in relation to the delivery of high bit rate time-sensitive stream-oriented content using current IEEE 802.11 standards, which were primarily designed for best effort services. Among the most significant are low delivery rates (e.g. theoretically up to 11 Mbps for IEEE 802.11b, but in practice only a throughput of approximately 6 Mbps can be achieved) and high error rates due to delivery media characteristics, contention between stations for access to the medium, back-off mechanisms, collisions, signal attenuation with distance, signal interference, etc. In these conditions it is difficult to provide any Quality of Service (QoS) guarantees.

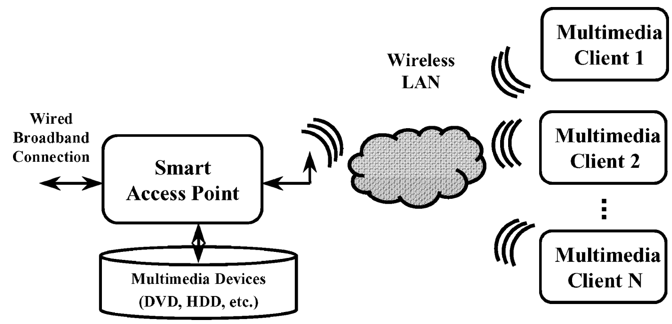

Fig. 1. Wireless architecture for distribution of multimedia content.

There are several significant adaptive solutions that provide a certain level of QoS in variable network delivery conditions such as the TCP-Friendly Rate Control Protocol (TFRC) [3] and the enhanced Loss-Delay Adaptation Algorithm (LDA+) [4]. These solutions offer good network-related results when streaming multimedia over wired networks, but are poorly designed for use over wireless networks and do not include end-user perceived quality in the adaptation process. Recently diverse solutions have been proposed for scalable multimedia transmissions over wireless networks [5] or wireless ad-hoc networks [6]. Among these are adaptive algorithms that operate at the level of layers [5] or objects [7], fine-granular scalability schemes [8] and perception-based approaches [9]. Complementing these approaches, the emerging IEEE 802.11e standard will provide QoS capabilities that can be used to improve end-user Quality of Experience (QoE) by allowing for dynamic prioritization of multimedia services.

This paper presents the Quality-Oriented Adaptation Scheme (QOAS) [10], [11], an adaptive multimedia streaming solution which is based on user QoE that maximizes end-user perceived quality in highly variable and increasingly loaded network delivery conditions. Simulation results for streaming multimedia over an IEEE 802.11b WLAN are compared when using QOAS, TFRC, LDA+ and a non-adaptive (NoAd) approach. Testing involved monitoring network delivery parameters and end-user perceived quality whilst increasing the number of

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concurrent multimedia streaming sessions. Performance is assessed in terms of the average user QoE, number of concurrent streaming sessions, loss rate, delay, jitter and total throughput.

The paper is structured as follows. The next section provides a brief overview of the IEEE 802.11b standard for WLAN and its use for residential networking. Sections III and IV give details regarding the streaming solutions employed for delivering multimedia during testing: QOAS, TFRC, LDA+ and NoAd. Section V presents simulation setup, scenarios and testing results and includes a brief discussion. The paper ends with conclusions and possible directions for future research.

II. IEEE 802.11 for WLAN

The IEEE 802.11 standard [12] is currently the most popular and widely deployed wireless LAN (WLAN) technology. The IEEE 802.11 operates in the unlicensed Industrial, Scientific, and Medical (ISM) band at 2.4 GHz and supports a mandatory bit rate of 1 Mbps and an optional higher rate of 2 Mbps. In September 1999 the IEEE approved the HR or "high rate" extension to the standard, known as the IEEE 802.11b, which supports data rates up to 11 Mbps. This WLAN standard uses the 802 LLC protocol but provides an independent physical layer (PHY) and medium access control (MAC) sub-layer specification which allows for best-effort wireless communication. There are two modes of operation in WLAN, the distributed coordination function (DCF) and the Point coordination function (PCF). Neither DCF nor PCF provides service differentiation mechanisms that can be used to ensure QoS guarantees such as bounded delays, loss or throughput constraints.

A. DCF

The basic access scheme used in 802.11 WLANs is the distributed coordination function (DCF). STAs can access the medium without the need for a centralized controller using an access mechanism known as carrier sense Multiple Access with collision Avoidance (CSMA/CA). This allows for asynchronous data transfer on a best effort basis where all stations (STAs) must contend with each other to access the medium in order to transmit their data.

CSMA/CA is a "listen-before-talk" access protocol where any STA wishing to transmit must first use the carrier sense mechanism to determine whether the medium is busy or idle. If the medium is busy, the STA defers its transmission until the medium has been idle for a period of time equal to DIFS (or EIFS in the case of an incorrectly received frame). The deferral process uses a collision avoidance mechanism where the STA randomly selects a Backoff Counter (BC) value in units of time slots (TS) (i.e. BC*TS where each TS is 20 us) for the contention window CW that is between CWmin and CWmax, although this is initially set to CWmin. The Backoff counter is decremented when the medium is idle, paused when the medium is sensed as busy, and restarted when the medium is sensed idle again for a period of time that is at least DIFS (or EIFS as appropriate).

When the BC reaches zero, the STA can initiate the transmission of its frame. The backoff time is slotted and a STA is only allowed to transmit at the beginning of a time slot. In DCF all STAs have equal probability to access the medium and share it according to equal data frame rate and not according to equal throughput. When multiple STAs are deferring and go into random backoff, the STA selecting the smallest backoff counter will win the contention. If two or more STAs choose the same BC value, this will lead to a collision whereby the STAs involved will transmit their frames at the same time. In order to resolve collisions between STAs, an exponential backoff scheme is adopted whereby the size of the CW is doubled after each unsuccessful transmission.

Packet priorities are implemented by defining three different length inter-frame spaces (IFS) between the frame transmissions. The IFS intervals are mandatory periods of idle time on the medium. The 802.11 standard defines four different IFS intervals as follows:

- Short Inter Frame Space (SIFS): is used for the highest priority transmissions (i.e. control frames), such as ACK and RTS/CTS frames. In 802.11b, $\mathrm{SIFS} = 10$ us.
- PCF Inter Frame Space (PIFS): is used by the point coordination function (PCF) during contention-free operation. STAs with data to transmit in the contention-free period can transmit after PIFS has elapsed and pre-empt any contention-based traffic. In 802.11b, $\mathrm{PIFS} = 30$ us.
- DCF Inter Frame Space (DIFS): is the minimum idle time for contention-based (i.e. DCF) services. After this interval has expired, any DCF mode frames can be transmitted asynchronously according to the CSMA backoff mechanism. $\mathrm{DIFS} = 50$ us.
- Extended Inter Frame Space (EIFS): is used to recover from a failed transmission attempt. It is derived from the SIFS, DIFS, and the time required to transmit an ACK frame at the basic rate of 1 Mbps.

B. PCF

PCF attempts to support time-sensitive traffic flows using a contention free service. The Point Coordinator (PC) periodically sends a beacon frame to broadcast network identification and management parameters specific to the wireless network. PCF splits the time into a contention free period (CFP) and a contention period (CP). Only STAs polled by the PC may transmit during the CFP. The CFP ends after the time announced by the beacon frame or by a CF-End Frame. Although PCF can offer some priority access, it cannot differentiate between traffic sources with time-sensitive data. Furthermore, the start time and duration of the CFP varies since the PC must contend with other STA to gain control of the medium.

III. Quality-Oriented Adaptation Scheme (QOAS)

Several adaptive schemes, including TFRC and LDA+ have been proposed that demonstrate good network-related results; however their adjustment policies are not directly related to the viewers' perceived quality. In contrast, QOAS bases its adaptation process on estimates of the end-user perceived quality made at the receiver. Perceived quality is estimated in-service using the no-reference Moving Picture Quality Metric (MPQM) proposed in [13] that describes the joint impact of MPEG rate and data loss on video quality.

QOAS is a distributed solution that consists of server-side and client-side components indicated in Fig. 2. QOAS makes use of a client-located Quality of Delivery Grading Scheme (QoDGS)

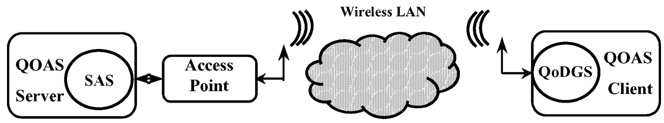

Fig. 2. QOAS components.

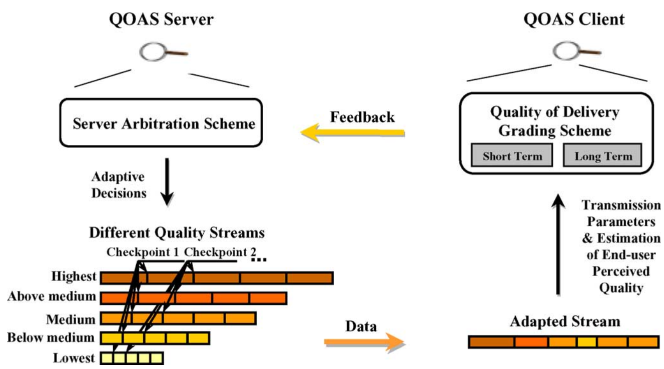

Fig. 3. QOAS adaptation principle, illustrated for pre-recorded streaming.

and of a Server Arbitration Scheme (SAS) that co-operate in order to implement the feedback-controlled adaptation mechanism. The principle behind QOAS is schematically illustrated in Fig. 3 for pre-recorded multimedia streaming, and is briefly described in the next section.

A. Principle of Quality-Oriented Adaptive Scheme

Multimedia data is received at the client where the QoDGS continuously monitors both some network-related parameters such as loss rate, delay and jitter and the estimated end-user perceived quality. According to their values and variations of these values, QoDGS grades the quality of delivery (QoD) in terms of application-level quality scores ($\mathrm{QoD}_{\mathrm{Scores}}$) that are sent to the server as feedback. These scores are analysed by the SAS that may suggest making adaptation decisions in order to maximize the end-user perceived quality in existing delivery conditions. These decisions affect an internal state defined for the QOAS server component that was associated with the streamed multimedia clip's quality as shown in Fig. 3. The figure presents the five-state quality model used during testing with the following states: excellent, good, average, poor and bad. Any QOAS server state modification affects the multimedia data transmission rate. For example, when increased traffic in the network affects the client-reported quality of delivery, SAS switches to a lower quality state. This results in a reduction in the bit rate of data sent, thus helping to recover from congestion in the network and improve the end-user perceived quality of the multimedia streaming session. This is performed as research has shown [14] that viewers prefer a controlled reduction in multimedia quality to the effect of random losses on the streamed multimedia data. In improved delivery conditions, the QOAS server component gradually increases the transmitted stream quality and therefore the transmission rate. In the absence of loss this causes an increase in end-user perceived quality.

B. Quality of Delivery Grading Scheme (QoDGS)

The QoDGS maps transmission related parameters such as loss, delay and jitter and variations of these parameters as well as estimations of the viewers perceived quality on application level scores that describe the quality of multimedia streaming session as perceived by the end-users. The quality of the streaming process is monitored and analysed over time in both short-term and long-term. Short-term monitoring is important for learning quickly about transient effects, such as sudden traffic changes, and for quickly reacting to them. The long-term variations are monitored in order to track slow changes in the overall delivery environment, such as new users in the system. These short-term and long-term periods are set to be an order and two orders of magnitude (respectively) greater than the feedback-reporting interval in the experiments described here.

There are three stages in the QoDGS grading mechanism. The first stage records the instantaneous values of the transmission related parameters and saves them in different length sliding windows where both the short-term and long-term variations are assessed. At the same time, session-specific lower and higher limits are maintained for each parameter, allowing for corresponding partial scores to be computed. In the second stage, the relative importance of all the monitored parameters in this delivery infrastructure is considered by weighting their contributions. The partial weighted scores are used to compute the quality of delivery grades in the short-term ($\mathrm{QoD}_{\mathrm{ST}}$) and long-term ($\mathrm{QoD}_{\mathrm{LT}}$). In addition, this second stage considers estimates for short-term and long-term end-user perceived quality. In the third stage, $\mathrm{QoD}_{\mathrm{ST}}$ and $\mathrm{QoD}_{\mathrm{LT}}$ are weighted to account for their relative importance and the overall client score ($\mathrm{QoD}_{\mathrm{Score}}$) is computed.

Extensive tests were performed in order to make sure that the design of QoDGS to optimize this grading process and ensures that the best results are obtained in terms of adaptiveness, responsiveness to traffic variations, stability, link utilization, and end-user perceived quality in local broadband IP-networks.

C. Server Arbitration Scheme (SAS)

SAS makes adaptation decisions based on the values of a number of recent feedback reports, in order to minimize the effect of noise in the $\mathrm{QoD}_{\mathrm{Scores}}$. This arbitration process is ***asymmetric***, requiring fewer feedback reports to trigger a decrease in quality than for a quality increase. This ensures a fast reaction during bad delivery conditions, helping to eliminate their cause and allowing the network conditions to improve and stabilize before any quality upgrade. These adaptation decisions are taken to maintain system stability by minimizing the number of quality variations. The late arrival of a number of feedback messages is considered as an indication of network congestion, and triggers quality degradations. This permits the streaming scheme to work even if feedback is not available.

IV. Other Multimedia Streaming Solutions

Among the proposed adaptive solutions for streaming multimedia are TFRC and LDA+. Both these adaptation algorithms have been widely used and researched. Due to the simplicity of

TABLE I
MAC SETTINGS USED DURING SIMULATIONS

MAC Settings			
Bit Rate	11 Mbps	SIFS	10 us
CW Min	21	Preamble Length	144
CW Max	1023	Short Retry Limit	7
Slot Time	20 us	Long Retry Limit	4

non-adaptive streaming schemes, they are also widely deployed. Such a non-adaptive solution is also considered in this paper.

A. TCP-Friendly Rate Control Protocol

The TCP-Friendly Rate Control protocol (TFRC) [3] uses a TCP model [15] to compute the transmission rate based on the loss rate that is measured as the inverse of the weighted average loss interval. This enables the scheme to adapt only to longer sustained periods of loss. TFRC provides additional delay-based congestion avoidance by adjusting inter-packet sending delay. As a result of these improvements, the scheme's sending rate is more stable, while still providing high responsiveness to changing traffic conditions.

B. Loss-Delay-Based Adaptation Algorithm

The Loss-Delay-based Adaptation Algorithm (LDA+) [4] adapts its transmission rate based on network conditions and on the bandwidth share already utilized. During periods of loss, the sending rate is decreased by a factor of the square root of (1-loss rate), being limited to the rate suggested by the TCP throughput model [15]. During periods of no loss, the sending rate is additively increased with a figure that is the minimum between three values. The first value is computed as the inverse of the bandwidth share utilized by the current flow. The second value limits the rate increase to be not greater than the bottleneck link bandwidth. The third value is determined such that the rate should not increase faster than that of a TCP connection sharing the same link.

C. Non-Adaptive Solution

The Non Adaptive (NoAd) solution streams multimedia content at the encoding rate regardless of the delivery conditions. Consequently the transmission to be performed is with variable bit rate or constant bit rate. This paper considers a constant bit rate approach. NoAd does not employ any feedback nor does it use any algorithm to modify either the content or the transmission rate.

V. TESTING SETUP AND RESULTS

A. Simulation Setup, Multimedia Clips and Models

Simulations use Network Simulator version 2.27 (NS-2) [16] and NOAH (No Ad-Hoc) wireless routing agent that supports only direct communication between base stations and mobile nodes. Table I presents the MAC settings used. Fig. 4 shows the simulation topology based on an IEEE 802.11b WLAN. It involves a SAP streaming multimedia content to a number of N clients (deployed at nodes Ci, $i = 1$,N). SAP includes N senders deployed at nodes Si, $i = 1$,N. Si-B1 ($\mathrm{bandwidth} = 100\,\mathrm{Mbps}$, $\mathrm{propagation\ delay} = 5\,\mathrm{msec}$) and B1-B2 (200 Mbps, 5 msec)

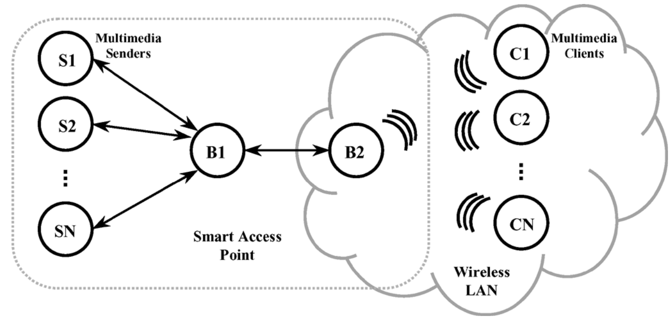

Fig. 4. Simulation setup.

TABLE II
PEAK/MEAN BIT RATE RATIOS FOR ALL MPEG-4 ENCODED QUALITY VERSIONS OF THE CLIPS USED DURING SIMULATIONS

	MPEG-4 - Average Rate (Kbps)				
Clip	**64**	**128**	**256**	**384**	**512**
DH	3.92	3.85	4.46	4.56	4.46
RE	6.86	4.50	4.32	4.31	4.31
DW	4.18	3.91	3.90	3.90	3.90
JP	4.63	3.26	3.20	3.19	3.19
FM	4.75	3.79	3.78	3.78	3.78

links are over-provisioned so that the only packet drops and significant delays are due to WLAN delivery. Buffering at intermediate nodes uses drop-tail queues. Clients' buffer size was such set that it causes no loss.

Five five-minute long multimedia clips with different degrees of motion content were considered: DH—high, RE—average/high, JP—average, DW—average/low and FM—low. They were encoded at five different rates using the MPEG-4 scheme. The MPEG-4 clips have their average bit rates evenly distributed between 64 and 512 kbps respectively. Among these different bit rate versions, the content is selected during adaptive streaming. All the clips have a frame rate of 25 frames/sec, IBBP frame pattern and 9 frames/GOP. Peak/mean bit rate ratios for all multimedia sequences used during testing are presented in Table II.

Testing was performed using NS-2-built QOAS, LDA+, TFRC and NoAd models that follow the descriptions made in Sections II and III. In order to increase feedback accuracy QOAS employs very high inter-feedback intervals (100 msec) and makes use of small feedback report packets (40 B). This balances the need for the most up-to-date information with the requirement of low overhead. TFRC implementation had a 5 sec update interval as suggested in [3] for delays greater than 100 msec, as in our setup. LDA+ implementation used an RTCP feedback interval of 5 sec as suggested in [4].

B. Scenarios

QOAS, LDA+, TFRC and NoAd approaches were used in turn for streaming MPEG-4-encoded clips to a number of N clients. In successive tests N was increased from 1 to 15 in a step-wise fashion increasingly loading the delivery network. This number of clients ensures that the average user QoE when using the best streaming solution–QOAS–is still above

TABLE III
PERFORMANCE RESULTS WHEN STREAMING MPEG-4 CLIPS OVER IEEE 802.11B WLAN USING QOAS, NOAD, LDA+ AND TFRC

Clients	QOAS (MPEG-4)					NoAd (MPEG-4)					LDA+ (MPEG-4)					TFRC (MPEG-4)				
	Quality (1-5)	Loss Rate (%)	Delay (ms)	Jitter (ms)	Throughput (Mbps)	Quality (1-5)	Loss Rate (%)	Delay (ms)	Jitter (ms)	Throughput (Mbps)	Quality (1-5)	Loss Rate (%)	Delay (ms)	Jitter (ms)	Throughput (Mbps)	Quality (1-5)	Loss Rate (%)	Delay (ms)	Jitter (ms)	Throughput (Mbps)
1	4.50	0.02	12.64	0.51	0.48	3.99	3.11	26.4	1.14	0.50	4.34	0.00	12.15	0.25	0.34	4.46	0.27	16.10	1.86	0.49
2	4.47	0.08	14.11	1.91	0.93	2.11	16.11	59.0	2.58	0.86	4.39	0.02	13.54	1.50	0.76	4.32	1.03	21.57	5.37	0.92
3	4.42	0.17	16.31	4.23	1.26	1.44	22.03	72.0	3.55	1.20	4.37	0.01	14.16	2.75	1.08	4.26	1.39	26.53	8.17	1.35
4	4.42	0.10	18.20	5.38	1.65	1.00	33.94	99.5	2.94	1.35	4.28	0.04	15.72	3.44	1.23	4.13	2.13	36.85	11.92	1.62
5	4.38	0.19	22.52	7.05	1.95	1.00	38.58	106.4	3.02	1.57	4.24	0.16	17.86	4.53	1.45	4.09	2.36	42.62	13.03	1.89
6	4.34	0.27	25.31	8.28	2.19	1.00	42.04	113.8	2.90	1.78	4.23	0.23	20.51	5.63	1.74	4.04	2.68	49.47	14.01	2.21
7	4.29	0.35	28.30	10.18	2.34	1.00	44.36	117.8	2.73	1.99	4.21	0.06	20.73	6.33	1.88	3.96	3.32	54.26	14.41	2.42
8	4.25	0.48	31.76	11.48	2.50	1.00	47.42	121.1	2.73	2.15	4.17	0.07	25.77	7.21	2.02	3.90	3.76	59.80	14.36	2.61
9	4.22	0.48	33.32	11.83	2.66	1.00	47.76	124.2	2.73	2.41	4.09	0.29	28.55	7.56	2.04	3.88	3.93	63.74	14.72	2.76
10	4.20	0.56	37.32	12.76	2.90	1.00	49.38	127.0	2.97	2.59	4.03	0.32	30.27	7.75	2.08	3.84	4.19	66.95	14.59	2.85
11	4.15	0.74	41.58	14.11	3.00	1.00	>50.00	>127.0	>2.97	>2.59	4.04	0.52	41.15	9.49	2.35	3.80	4.58	71.32	14.95	2.96
12	4.10	1.07	45.63	14.30	3.10	1.00	>50.00	>127.0	>2.97	>2.59	3.99	0.73	46.05	9.72	2.43	3.76	4.93	74.80	14.49	3.09
13	4.07	1.05	53.03	17.42	3.17	1.00	>50.00	>127.0	>2.97	>2.59	3.96	0.70	43.68	10.12	2.51	3.71	5.43	77.38	14.45	3.21
14	4.05	0.93	48.21	15.59	3.23	1.00	>50.00	>127.0	>2.97	>2.59	3.95	0.71	47.36	10.36	2.67	3.71	5.37	82.11	14.50	3.35
15	4.00	1.51	51.75	16.56	3.43	1.00	>50.00	>127.0	>2.97	>2.59	3.85	1.47	62.18	10.40	2.66	3.68	5.60	84.39	14.29	3.42

the "good" perceptual level on the ITU-T R. P.910 five-point subjective quality scale [17]. This level of end-user perceptual quality is considered in this paper as the minimum level of interest. However the solution can support even a higher number of simultaneous streaming sessions, but their quality is expected to be lower.

The clients randomly select both the movie clip and the starting point from within the chosen clip. This ensures that all movie types are used during testing. It also provides independence of the simulation results from the natural bit rate variation in time within each of the streamed movies. Consecutive streaming sessions were started at intervals of 25 sec and the transitory periods are not considered when processing the results reported in this paper. During the stable periods of 600 sec, the loss rate, one-way delay, delay jitter, end user QoE and total throughput were measured and analysed. The end-user perceived quality was estimated using the non-reference Moving Picture Quality Metric [13] and expressed on the ITU-T R. P.910 five-point quality scale [17].

C. Results

Table III presents the complete set of testing results when QOAS, LDA+, TFRC and NoAd streaming solutions were used in turn for delivering multimedia clips over the IEEE 802.11b wireless LAN. The tests were performed with increasing number of simultaneous clients and for each case the end-user perceived quality, loss rate, delay and jitter as well as the total throughput are monitored and averages are computed and indicated in the table.

Fig. 5 plots the average viewers perceived quality as a function of the increase in the number of simultaneously served clients. The graph clearly shows how QOAS outperforms all the other streaming approaches, including LDA+ that has the best results among the other approaches. For example when streaming multimedia to 1 client, the perceived quality is 4.50 when using QOAS, 4.34 when using LDA+, 4.46 when TFRC is employed for streaming and 3.99 when non-adaptively delivering the multimedia content. These values are recorded on the 1-5 ITU-T five-point scale. As the number of simultaneous clients increases, the scores begin to diverge more significantly. For example, when the multimedia content is streamed simultaneously to 10 clients over the IEEE 802.11b WLAN. When using QOAS the end-user perceived quality reaches 4.20 in comparison to only 4.03 when using LDA+, 3.84 when TFRC-based streaming and the lowest value of 1.00 when NoAd approach was used. The most significant results are obtained in highly loaded delivery conditions. For 15 clients when using QOAS the end-user perceived quality is still at the "good" quality level, whereas for all the other approaches it drops much below this level: 3.85—LDA+, 3.68—TFRC and 1.00—NoAd.

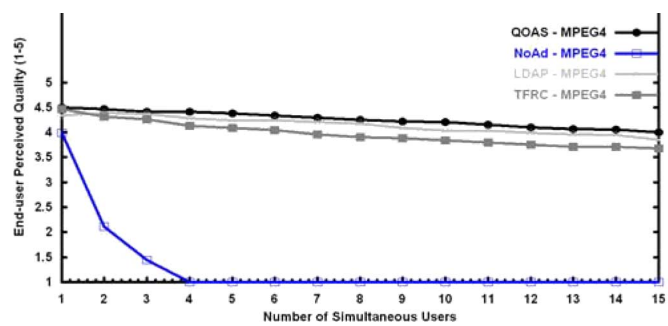

Fig. 5. Comparative quality.

Fig. 6 shows how the average loss rate increases with the increase in the number of concurrent sessions when different streaming approaches are employed for multimedia delivery. It can be clearly seen that QOAS and LDA+ outperform the other

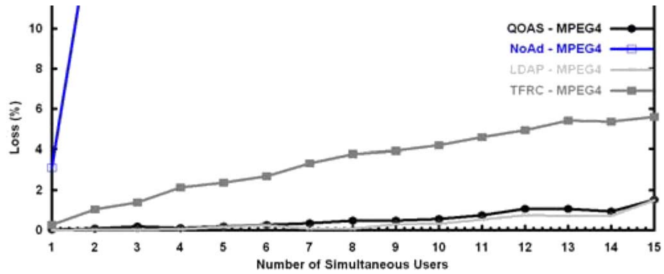

Fig. 6. Comparative loss rate.

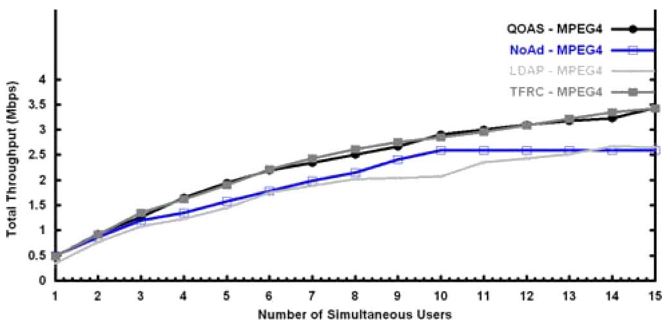

Fig. 7. Comparative total throughput rate.

solutions when streaming MPEG-4-encoded content. They successfully maintain an average loss rate below or in the region of 1% much lower than those recorded when TFRC and NoAd approaches are used. For example when streaming multimedia content to 10 clients the average loss rates are 0.56% when using QOAS, 0.32% for LDA+, 4.19% when using TFRC and 49.38% for the NoAd approach.

Fig. 7 presents the total throughput recorded when using different streaming approaches for delivering multimedia to an increasing number of simultaneous viewers. The figure shows how QOAS and TFRC achieve much better total throughput than the other approaches regardless of the number of multimedia clients. For example when streaming clips to 10 simultaneous clients using QOAS and TFRC approaches the total throughput is 2.90 Mbps and 2.85 Mbps respectively in comparison with that measured when using LDA+ and NoAd which is 2.08 Mbps and 2.59 Mbps respectively. Both QOAS and TFRC achieve similar high total throughput when delivering multimedia content to 15 clients. In this situation the total throughput is 3.43 Mbps and 3.42 Mbps respectively in comparison with only 2.66 Mbps when using LDA+.

Another significant aspect of the comparison between these streaming solutions is the number of simultaneous multimedia sessions supported for a given target end-user quality level. For example Fig. 8 presents a comparison between the number of simultaneous streaming sessions supported when using QOAS, NoAd, LDA+ and TFRC approaches respectively in order to maintain a "good" perceptual quality level. The figure shows that by using QOAS fifteen times more simultaneous clients can be served with MPEG-4-encoded multimedia content than by using NoAd. This result is somehow expected, as NoAd does not perform any content or transmission rate adaptation to loaded delivery conditions. However the significance of the QOAS performance is highlighted by the fact that the number of customers

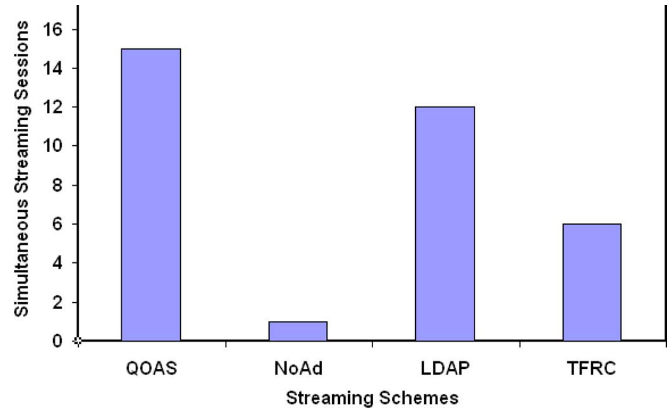

Fig. 8. Comparative number of simultaneous streaming sessions at "good" quality level.

served when using QOAS is 150% higher than those served when TFRC is used and 25% greater than the number of streams supported at "good" QoE level when LDA+ is used. These results are obtained although both approaches QOAS is compared to are adaptive and have shown significant results when delivering multimedia content.

Looking at other performance parameter values presented in details in Table III, it can be seen that when using QOAS, the delay remains at very low levels in spite of the high increase in the number of simultaneous multimedia sessions. In contrast, streaming using the other approaches such as TFRC and NoAd incurs a significant increase in the delay that may eventually affect end-user QoE. The slight increase in jitter when using QOAS with the high increase in overall traffic over the IEEE 802.11b WLAN in comparison with when the other streaming solutions are employed can be coped with using client buffering and is not expected to affect the viewers.

These results show significant performance gains when using QOAS for streaming multimedia over IEEE 802.11b WLAN mainly in terms of viewers perceived quality, loss and total throughput in comparison to when other schemes are employed. Of particular importance, is the achieved increase in WLAN's overall delivery capacity and efficiency when using QOAS, which is a highly desirable attribute as a higher number of simultaneous viewers that experience the same "good" perceived quality can be served from an existing infrastructure.

VI. Conclusions and Further Work

This paper compares our Quality Oriented Adaptation Scheme (QOAS) with three other solutions: TFRC, LDA+, and a non-adaptive (NoAd) approach, when streaming MPEG-4-encoded content over an IEEE 802.11b WLAN. This comparison is performed in terms of average end-user perceived quality, number of concurrent streaming sessions supported, loss rate, delay, jitter and total throughput.

Simulation results show that for the same average end-user quality, QOAS can accommodate a significantly higher number of simultaneous clients while also having higher total throughput. For example a 25% increase in the number of simultaneous clients that can be served at "good" quality level was achieved when using QOAS in comparison when LDA+

was used and much higher gains when the other approaches were employed. The significant results achieved by QOAS enable network operators, service providers, and the residential users to use their infrastructure more efficiently. Commercial companies using QOAS would see an eventual increase in their revenues since they can serve a higher number of customers at the same "good" quality level.

It is also significant to mention that for the same number of simultaneous multimedia clients, the average values of the performance metrics such as end-user perceived quality, loss, delay and total throughput were always consistently better for QOAS than for all the other solutions studied. This result shows that by using QOAS an increased level of QoE can be provided to each viewer.

Work in progress investigates QOAS performance when streaming multimedia over WLAN in the presence of other traffic. Further work will include comparisons of QOAS with other adaptive schemes when streaming over IEEE 802.11e WLAN. It is also planned to perform subjective tests on a prototype system to verify all these simulation results.

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