IMPROVED QUALITY OF SERVICE UTILISING HIGH PRIORITY TRAFFIC IN HCF IN A DYNAMICALLY CHANGING WIRELESS NETWORKS

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***Abstract* — The demands of the end users has changed in correlation to the advancement of technology, no longer will the end user accept long delays or a poor quality of service. Couple these requirements with the ever increasing demand for greater bandwidth to support such applications as streamed video and VoIP. Adequate Quality of Service (QoS) is now paramount within the dynamically changing nature of wireless networks that experience many issues, specifically in providing good and maintainable levels of QoS for high demand multimedia applications. This paper investigates how a finite bandwidth can be extended to accommodate additional users by utilising a proposed mechanism for the block-frame acknowledgement that exploits the IEEE 802.11 standard. This method takes advantage of the access categories encompassed in the Hybrid Coordination Function to prioritise traffic ensuring adequate quality of service, with no major upheaval**.

***Index Terms*— 802.11, Quality of Service, Hybrid Coordination Function, Wireless Networks.**

I. INTRODUCTION

Quailty of Service (QoS) has been defined as *"The collective effect of service performance which determines the degree of satisfaction of a user of the service"* [1]. QoS is simply a mechanism for provisioning guaranteed bandwidth over a computer network for high demand content. For example, a simple home network may consist of an upstairs PC, and a downstairs living room TV, wirelessly connected to each other. The user may want to watch a digitally stored movie, located on the PC, on the downstairs TV without copying and storing the movie locally. First, the data must be streamed from the PC to the TV. Current streaming technologies (e.g. Windows Media 11 and RealVideo 10, introduced in 2006) permit near-DVD quality streaming at a rate of 500Kbps. This paper focuses on a lower resolution form of video, approx. 320x240 with a frame rate of 20fps (frames per second) [2]. At this resolution and frame rate, video requires around 350Kbps to stream (Windows Media Encoded).

At these speeds, an IEEE 802.11b network (11Mbps) could sustain 31 simulation streams in optimal conditions, providing all streams were running a constant rate of 350Kbps. This does not include any network overhead, or any extrinsic factors affecting the quality of the network connection. The aim of QoS in this situation is to ensure each stream receives as close to 350Kbps as possible, and to insure a seamless change of bandwidth allocation to users when necessary. The goal is to provide a dynamically changing wireless network (that is, a wireless network with devices leaving and entering the network on a ad hoc basis), with the ability to sustain every connected device running streamed multimedia applications with the required bandwidth and network conditions for, satisfactory use. The principle scenario would permit the complexity of multiple devices to simultaneously steam media.

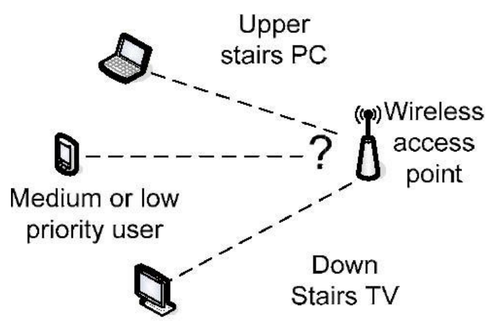


Figure 1, illustration of the home network scenario.

II. QUALITY OF SERVICE (QOS)

A. QoS

Every end user strives for a system that ensures real-time voice and video is delivered without faults or failure, coupled with a guarantee of the required bandwidth. The aim is to achieve a similar QoS that is evident in the telecoms system though this is achieved via a dedicated channel between sender and receiver, which is an expensive luxury in internetworking. Achieving QoS in a wired environment is one thing with the fragmentation of data; these issues are compounded in the wireless environment.

To fully assess how to achieve QoS in a wireless environment, we first need to assess what it is, and what it involves. The aim of QoS is to provide the end user, guaranteed bandwidth and an adequate service. QoS is applicable for high demand applications such as video and audio streaming. These applications require a minimal level of service to function, and seamless transition between levels of service. There are four main aspects of QoS to consider and be aware of:

Reliability: QoS aims to provide reliable and error free data with methods of CRC (error checking). With regard to services such as video or audio streaming, CRC/error checking can be an unnecessary and high cost overhead. Such applications can run adequately with an acceptable degree of data loss. For example, 1 dropped packet during a video stream may only mean one dropped pixel for one frame. During file transfer, error checking is vital – 1 dropped packed during the sending of a file could be the difference between a usable, successfully transferred file, and corrupt, useless data.

Delay: delay in network transmission is inevitable, and depending on the distance data must travel to reach the end user, will dramatically vary. Greater distance will usually involve data passing through more devices such as routers. Each device will add its own overhead to the transmission increasing the delay of reaching the end user. With regards to multimedia streaming applications, delay greatly affects the end users experience. As streaming means live data transfer, any stop or delay through a media stream will halt that stream. For example, additional delay in a video stream would pause and skip the video. To avoid this, devices and end applications use buffers to store a 'build-up' of data. The buffer receives the stream, and the application plays from the buffer. If delay occurs on the network, the video can still play from the cached buffer. When the delay subsides, the buffer can replenish. It is these buffers that form a major part to the proposed *Mobiware* additions and enhancements.

Jitter: essentially, the arrival of incoming packets at irregular time intervals. Jitter causes delay. Buffering (as afore mentioned) is a method of 'smoothing' this delay and creating a seamless stream for the end user who is kept unaware of fluctuating network conditions. However, buffering has its own costs, namely additional delay. As buffers hold a cache of the incoming transmission, delay is added at the start of the transmission while the buffer is populated with data. Personally, I believe the initial delay in buffering is more then acceptable if the video or audio stream is later run flawlessly. *Bandwidth*: is the quantity of packets transmitted per second. High demand applications such as audio and video demand high bandwidth. QoS services attempts to manage bandwidth utilisation to provide maintainable QoS.

B. Wireless ATM

The evolution of ATM into wireless networks enables wireless technology to encompass the advantages of the QoS provided by ATM. QoS in a wireless environment, aims to guaranteed bandwidth and service to the end user, fundamentally the QoS is essential for high demand applications such as video and audio streaming. These applications require a minimal level of service to function, and seamless transition between levels of service

WATM claims to provide QoS support that allows multimedia applications to operate transparently during handoff and through heavy QoS requirement fluctuations. Mobiware is a highly programmable middleware platform designed to run between the radio link layer and the application layer [11].

III. TECHNOLOGY

A. Overview

Industry have adopted the Internet Protocol (IP) standard to route data over their corporate network, this enables them to maintain a single infrastructure, which in turn reduces their maintenance costs. The implementation of Wireless Fidelity (WiFi) networking has allowed users to access the corporate network whilst still maintaining mobility. Applications such as Voice over IP (VoIP) has primarily been used on wired networks, implementing this on a WiFi network would give greater flexibility to the user. This type of implementation would enable the corporate user to roam and collect calls from the corporate telecoms network. This scenario also holds true from the home users point of view. The drawback of a wireless implementation is that VoIP demands a good quality of service (QoS) for it to achieve an acceptable level of performance for the end user. It is the type of VoIP application above that necessitates classifications of high priority network traffic providing a good QoS delivery.

B. IEEE 802.11

The IEEE 802.11 wireless networks standard can be configured into two different modes: ad hoc and infrastructure modes. In the ad hoc mode nodes can communicate directly with each other, where as in the infrastructure mode the node communicate via an access point. In early versions of 802.11, it is the Distributed Coordination Function (DCF) that provides the access mechanism for both these communication modes. However the shortcoming of this mechanism is that different applications require varying amounts of bandwidth, packet loss, delay and jitter, but DCF provides all nodes and data flows with the same priority, therefore providing no differentiation of application network requirements. This results in poor QoS for high demand content such as streaming media and VoIP applications. For these applications to be usable it is critical for the 802.11 to support QoS.

Differentiation of application network traffic requirements are achieved via the Hybrid Coordination Function (HCF) this combines both the contention free (CFP) and contention period (CP) based access methods in a single access channel. Basically HCF combined both DCF and Point Coordination Function (PCF) with QoS in enhancements in Enhanced Distributed Coordination Function (EDCF).

C. Enhanced Distributed Coordination Function (EDCF)

The EDCF manages the wireless medium access in the CP while the HCF is responsible for the CFP and the CP [4]. The 802.11e specification [5] allows packets to gain priority by utilising traffic classes (TC) initially eight TC's (see Table 1) were defined but these were reduce down to four access categories, each with its own queue, these are voice, video, best effort and background, e.g. email and ftp, [6].

Using EDCF, nodes try to send data after detecting the medium is idle using the listen-before–transmission method, after waiting a period of time defined by the corresponding traffic category called the Arbitration Interframe Space (AIFS). AIFS is at least the length of DIFS but differentiates for each AC [8]. It is the AIF that aids the management of the TC as a higher-priority traffic category will have a brief AIFS while lower-priority traffic category will have much longer periods of AIF. Collisions are avoided within a TC by the node counting down a random number of time slots, known as a contention window (CW), before attempting to transmit data. If another node starts transmitting before the countdown has ended, the node waits for the next idle period, when it continues the countdown. The timings for EDCF are illustrated in Figure 2.

Priority	Access Category (AC)	Designation
1	0	Best Effort
2	0	Best Effort
0	0	Best Effort
3	1	Video Probe
4	2	Video
5	2	Video
6	3	Voice
7	3	Voice

TABLE 1 PRIORITY TO ACCESS CATEGORY MAPPING [7]

Therefore QoS is achieved by prioritising the application that demands greater bandwidth to be useable while the applications with lower priority such as FTP or email, has to wait longer for access to the wireless medium. Consequently higher priority traffic will be transmitted before lower priority traffic. Fundamentally EDCF has enabled multi-backoff instances delivered by MAC Service Data Units (MSDU's) on one node as illustrated in figures 3. The shortcoming of EDCF is that potentially medium or lower access class traffic may never be granted bandwidth.

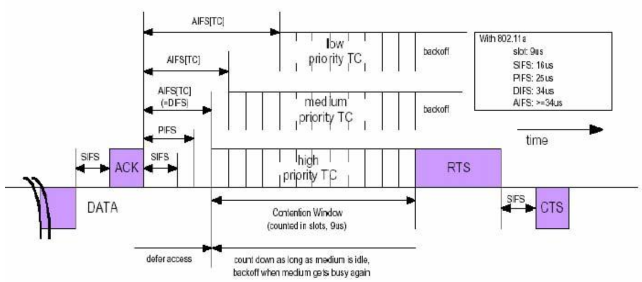


Figure 2 Enhanced Distributed Coordination Function (EDCF) time relationship [3]

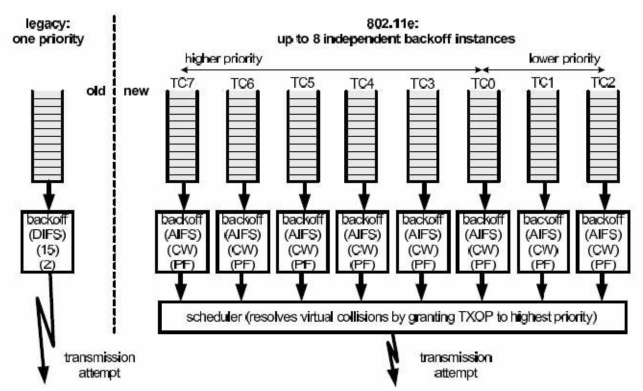


Figure 3 IEEE802.11e MAC structure [3]

D. Hybrid Coordination Function

The Hybrid Coordination Function (HCF) extends the polling mechanism of PCF. A hybrid controller polls stations during a contention-free period. The polling grants a station a specific start time and a maximum transmit duration controlling the channel access [9].

One node employing HCF is responsible for the management of the wireless medium access. It is able to apportion Transmission Opportunity (TXOPs) to itself after a gap of PIFS. This leads to priority over other EDCF nodes that cannot access the wireless medium until the completion of AIFS_DIFS_PIFS.

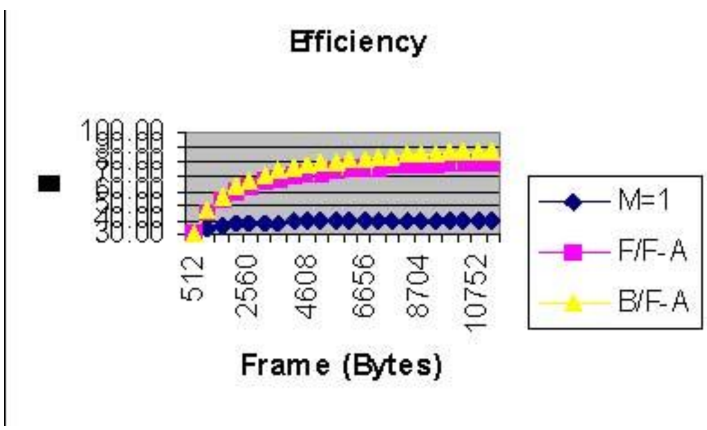
HCF has two TXOPs:

- EDCF-TXOP for contention – wireless medium acces via EDCF methods;

• polled-TXOP. EDCF-TXOPs can be obtained only in the CP while polled-TXOPs can be apportioned by the HCF [4]. This enables the HCF to take control by sending a QoS CF-poll frame after a PIFS-idle medium (eliminating the backoff). The TXOPs in the CFP are allocated by HCF with CF-poll frames including the starting point and the maximum duration. A CF-end frame of the HCF or the point of time announced in the beacon frame leads to the end of CFP.

E. High Priority Traffic

A mechanism utilising HCF for the block-frame acknowledgement that is compatible with the IEEE 802.11 standard, has been proposed [5] to support quality of service [1], with no major upheaval. This has been analysed, and is valid for all type of traffics low, medium and high priority classes. Figure 4 illustrates EDCF part of the HCF, the MAC protocol. The value of the DIFS depends on the priority classes.
fragment has a sequence number and a flag indicating the end of the transmission. We suggest that in this case when a fragment is corrupted a negative acknowledgment is generated after a frame space called IIFS (Intelligent Interframe Spacing), which is less than the SIFS space of the acknowledgment. The sender receives the corrupted negative acknowledgment as noise, it waits until the channel is clear, then after SIFS it transmits the next fragment, unaware of the previous unsuccessful transmission. Figure 6 demonstrates the efficiency as a percentage for Non-fragmented (M=1), Fragment-by-Fragment ACK (F/F-A), and Block Frame ACK (B/F-A) for noiseless channel (p=.5, q=0), as a function of the frame length.



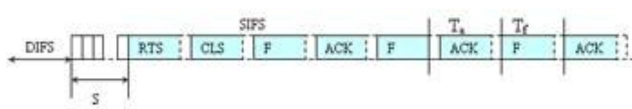


Figure 4: EDCF part of the HCF, the MAC protocol. The value of the DIFS depends on the priority classes [13].

F. 3 Fragment-by-Fragment Acknowledgment

In the IEEE 802.11 standard, for reliability each fragment is acknowledged. If a fragment is corrupted a negative acknowledge is send back. In the case where, the negative acknowledge is lost, the sender times out and retransmit the fragment. We assume that the length of the time out period is equal to the acknowledgment time [13]. Figure 5 illustrates the efficiency as a percentage for Non-fragmented (M=1), Fragment-by-Fragment ACK (F/F-A), and Block Frame ACK (B/F-A) for a noiseless channel (p=q=0), as a function of the frame length [13].

Figure 6: The efficiency (%) for Non-fragmented (M=1), Fragment-by-Fragment ACK (F/F-A), and Block Frame ACK (B/F-A) for noiseless channel (p=.5, q=0), as a function of the frame length.

H. Mobiware Mobiware, is based on the latest distributed system technology and claims to be a highly programmable middleware platform designed to run between the radio link layer and application layer of future next-generation wireless systems, such as base stations and WATM switches. Built on distributed systems
and Java technology, it uses adaptive algorithms to transport scaleable transmission flows.

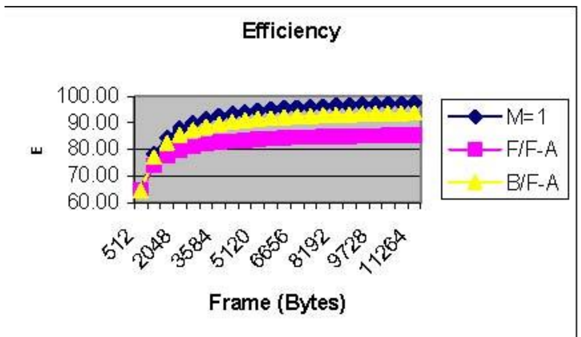


Figure 5: The efficiency (%) for Non-fragmented (M=1), Fragment-by-Fragment ACK (F/F-A), and Block Frame ACK (B/F-A) for a noiseless channel (p=q=0), as a function of the frame length [13].

G. Block-Frame Acknowledgment

In block-frame acknowledgement an acknowledgement is sent when all the fragments generated from the same frame have been received. From the IEEE 802.11 standard, each A very interesting aspect of *Mobiware* is its application specific ‘flow adaptation policy’. This policy in its basic form characterises each transmission stream (flow) of data and recognises its acceptable minimal level of QoS. Using this information, *Mobiware* is capable of scaling each stream to match available bandwidth while attempting to ensure each transmission stream at least maintains these acceptable QoS levels. A further claim of *Mobiware* is provision of QoS support that allows multimedia applications to operate transparently during handoff and through heavy QoS requirement fluctuations. Figure 6 shows the architecture makeup of *Mobiware* [11].

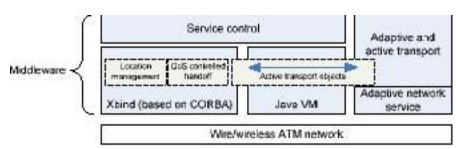


Figure 6. Mobiware Architecture, [11]

Mobiware has set out to improve on multi-rate multimedia connections. These are difficult to achieve with current widely deployed technology, namely the Internet. And, with the rapid continual expansion of the Internet, and the increasing demand for this type of usage, solutions must be found. Multi-rate connections also require management to ensure a seamless change in transmission quality to the end user. *Mobiware* achieved this through end-to-end QoS control using two methods: resource binding between devices and *Mobiware's* adaptive algorithms as illustrated in figure 1, [11]:

QoS controlled handoff: providing signalling of handoff events, used to represent flows, aggregation of these flows to/from devices and re-routing negotiation. **Adaptive network service**: provisions QoS guarantees based on available resources. **Adaptive and active transport:** supports multilayer transmission flows via *Mobiware's* API.

IV. INTELLIGENT ADAPTIVE BUFFER CONTROL (IABC)

The proposed intelligent Adaptive Buffer Control (iABC) will integrate low-level network protocols and end, high-level user applications. It is envisaged that this will maintain QoS within a saturated network, while permitting short-term use for additional applications. Incorporating Mobiware's 'flow adaptive policy' with application intelligence could achieve this. Mobiware guarantees the QoS though a secured bandwidth, the drawback being that this is a finite resource, iABC proposes to dynamically alter the size of the receiving applications buffer once additional demands arise. iABC provides an interesting proposition to provide all parties with their respective requests, while maintaining a high and maintainable QoS for all.

Figure 7, illustrates n High priority users (H1, H2, …, Hn) sharing the available bandwidth, Br, with the required QoS. For instance, each high priority user requires 350Kbytes/sec bandwidth for video streaming. During the window of opportunities, window of accepted requests (WAR), we assume that m low or medium priority users (L1, L2, …, Lm) have requested bandwidths for QoS of the same magnitude of the high priority request for a limited duration only. For instance, each low or medium priority user requires a video clip to get football news, and then releases the bandwidth. We denote by Ba the remaining bandwidth which BT - nBr, which the total bandwidth takes away the requested bandwidth by all the high priority users. In the figure below we denote by Tp, Tb, Te and Ts the durations of the WAR, the buffering time, the buffering emptying time and the viewing or streaming time respectively. Finally we refer to the additional buffer size for each user as bf.

We analyse three schemes. During the WAR period low priority use the remaining bandwidth **Ba** to broadcast their requests and requirements for the streaming period. The requests could be denied, if not enough remaining bandwidth is available, or the requirements cannot be met. Each high priority user adjust dynamically its buffer with the number of requests and the remaining bandwidth **Ba/n**, while maintaining a connection to the server at the **Br** rate for QoS.

$B\ nB$

ar

$$*T = T_b * B \Rightarrow T_p = *T_b \quad (1)$$

r

$n^{p}\ B$

a

The additional buffer size for each High priority user would be

B

a

$$Bf_H = *T_p = B *T_b \quad (2)$$

r

n

and for each low priority user would be

$$Bf_L = \frac{B_T}{} *T_b \quad (3) \; m$$

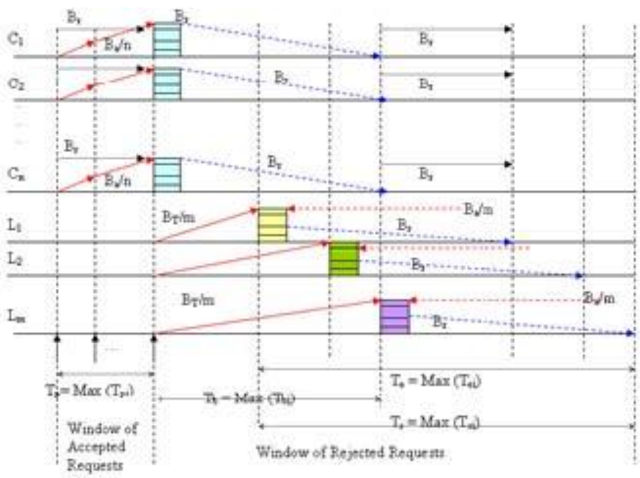


Figure 7, High priority users (H1, H2, …, Hn) sharing the available bandwidth, Br, with the required QoS. [10]

In these three schemes if the total bandwidth is more than enough for all high and low priority users (BT > (m+n) Br), the viewing time starts immediately (Tv = 0). Otherwise, the bandwidth will be shared equally among them. However, if the total bandwidth is enough for high priority users only (BT <= nBr) all requests from low priority users will be denied and the viewing time is infinity (Tv = ∞). In all the other cases, (nBr < BT < (m+n) Br), requests will be accepted with durations as computed in each case with the expense of extra buffering as follows. There are no preferences between low priority users. It is possible to implement preferences within low priority users as demonstrated in section M.

A. Fully Loaded Buffer before Streaming (FLB)

In this scenario the lower priority users start streaming, when their buffers are full. The streaming period is equal to the buffer emptying time (Ts = Te). Each low priority user fills it buffer at the rate of the total shared bandwidth (BT/m).

¾ 1569014582

mB

r

$$\frac{B_T}{}*T_b = B*T \Rightarrow T_b = *T$$

$rs\ s$

$m\ B_T$

(4) Combining equations 1 and 4 and after some manipulations, we deduce the duration of the viewing delay as

mB

r

$$T = T_p + T_b = *T \quad (5)$$

vs

$B_T - nB$

r The buffer sizes at for high and low priority users can simply be derived from using equation 4 into 2 and 3 respectively. The results are shown in the figure 2,3 and 4.

B. Partially Loaded Buffer before Streaming (PLB)

In this scenario the lower priority users start streaming, when their buffers are filled up to a threshold value. During the streaming, each lower priority user relies on the remaining bandwidth to keep filling it buffer at a lower rate. The streaming period is equal to the buffer emptying time (Ts = Te).

$mB - B$

$a\ ra$

$$T_b * \frac{B_T}{} + T * {}^{b} = T * B \Rightarrow T_b = *Ts \quad (6)$$

$s\ sr$

$mm\ B_T$

Using equations 1 and 6 we can deduce the viewing time as

mB

$$T = T_p + T_b = ({}^{r} -1)*T \quad (7)$$

vs

$B_T - nB$

r

to fulfil their QoS requirements while the remaining share the left over bandwidth.

X = (BT -nBr)/Br If X is greater than 1 then n = n + $\lfloor X \rfloor$ // Low priority user is promoted to High

priority

Ba = BT – //Recalculate the new

n Br

remaining

bandwidth

m = m – ⌊x⌋ // the remaining share the new

available

bandwidth

else

All requests are denied.

E. Results

Results at 5000Bkps a normal IEEE 208.11a with access point for different users request a short period of the bandwidth at the MPEG rate of 350Kbps, with 10 high priority users. The size of the buffer for the PLB is lower in all cases. For instance for 6 requests during the WAR period, the buffers of the high priority and low priority users will increase to 42KBytes*60 = 2520Kbytes and 6000Kbytes for a streaming time of 60sec.

The viewing time defined as the time taken to start viewing the video clip is lower for both PLB and CLB (figure 8). For instance for 6 requests, on average for a 1 minute of video streaming, the viewing time will start about .4*60sec = 24Sec.

2.5

C. Continuously Loaded Buffer while Streaming (CLB)

In this scenario the lower priority users start streaming, while buffering. During the streaming, each lower priority user relies on the remaining bandwidth to keep filling it buffer at a lower rate. The streaming period is equal to the filling and

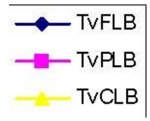
View/Streaming Times

2

1.5 1

0.5

0



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emptying times of the buffer (Ts = Tb + Te).

Num ber of Requests

b

a

$T_b * \frac{B_T}{} - B * T_b + T * - T B = 0$

$re\ re$

mm

$\underline{B_T} b$

a

$T_b * - B * T_b + (T - T_b) * - B (T - T_b) = 0$ (8)

$rs\ rs$

mm

$\Rightarrow T_b = ((m + b\ n - B_T$

r

$)$ $) * T$

s

nB

r

Using equations 1 and 8 we can deduce that the viewing time (Tv = Tp) is similar to equation 7.

D. Preferences between low priority users using PLB scheme

In this case we assume that low priority users have preference, and when enough bandwidth remain, some of them will use it

Figure 8, viewing time defined as the time taken to start viewing the video clip [12]

In figure 9, we vary the number of High priority users. For a total bandwidth of 2500Kbits/sec with 5 low priority requests, we notice that as the number of high priority increases, the is insufficient bandwidth left to assign to low priority, in which case the viewing time tends to infinity. For 3 users, and 5 requests, the viewing time for PLB and CLB is about 20% if the streaming time giving 12sec for a video clips of 1-minute duration.

REFERENCES

Viewing/Streaming Times

4.0
3.5
3.0
2.5
2.0
1.5
1.0
0.5
0.0

1

2345678910

Number of High Priority Users

Figure 9, Viewing time with a variable number of High priority users [12].

V. CONCLUSION

We will consider an intelligent sender that will use the noise from the negative acknowledgments as indication to a wrong transmission. The receiver will have to send negative acknowledgment to overcome duplications. The sender has a time out mechanism to retransmit non-acknowledged fragments. We believe that this will improve equation 2 further. We will also look at the impact of multi-hops on the efficiency of the proposed scheme.

We propose to combine both methods the IIFS and iABC methods to achieve a good QoS, as theoretically each posed a viable solution individually. This will enable iABC to be viable within the boarder wireless medium environment and not only with WATM. The classification elements that HCF provides in respective of determining the TC, will enable the identification for the required traffic priority which will aid the decisions made by the iABC for initiating the buffering mechanisms. Fundamentally this will work in a similar way to the adaptive flow priorities in

Mobiware.

Utilising an intelligent sender will remove the issues of the client only prioritising traffic and controlling the priority allocation could also be verified. The current TC's will enable the iABC to determine the bandwidth allocation and to continue utilising the buffer control mechanism for the relevant traffic.

Despite all solutions, QoS methods, additional bandwidth etc, we are trying to make a resource that is finite appear infinite. Ultimately, bandwidth can only support a certain number of devices at a certain level of quality. When that limit is reached, additional requests will simply have to wait for freed resources. Networks, like every other aspects of IT are continually and rapidly evolving. The Internet is starting to outgrow its roots, and unless new technologies like those covered in this paper can be refined and implemented on a mass and commercially viable basis, the Internet will eventually cease to function in any usable state, and will certainly be unable to cope with the future of high demand real-time content.

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