802.11e EDCA Parameter Optimization Based on Synchronized Time

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Abstract—

IEEE 802.11 based wireless Local Area Networks (WLANs) are fast becoming the standard in homes, businesses, and public hotspots. As well as supporting conventional Internet applications such as email, file transfer and web access, WLANs are increasingly required to support Quality of Service (QoS)-centric applications such as Voice over IP (VoIP) and Video over IP, which are delay sensitive and require a certain level of throughput. This creates an urgent need for supporting QoS in 802.11-based WLANs. Whilst 802.11e goes some way towards meeting this need, severe congestion leading to unacceptable delays and packet loss can still occur. In this paper, we outline our plans to investigate how synchronized time implemented in end terminals can aid in optimizing 802.11e parameters to improve QoS for VoIP. Testing will be done using a combination of simulation using the NS-2 network simulator, with a practical test bed to back up our research.

Keywords –

802.11e, M2E Delay, EDCA, Synchronized Time, E-Model, NS-2

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I INTRODUCTION & BACKGROUND

The IEEE 802.11 standard defines two access methods: 1) Distributed Coordination Function (DCF), also known as the basic access method, which is a carrier sense multiple access protocol with collision avoidance (CSMA/CA), and 2) Point Coordination Function (PCF), which is a polling-based access method and uses a point coordinator to arbitrate access among stations. In DCF, all the data traffic is transmitted on a first come first serve, or best-effort basis. There are no priorities, and all the stations in the basic service set (BSS) compete for the wireless medium with the same priority. DCF uses an exponential back-off process, which doubles the size of the Contention Window (CW) after each transmission failure. Back-off intervals are chosen randomly from the range [0–CW]. DCF does not provide any means for differentiating traffic classes.

PCF, on the other hand, provides QoS to a certain extent. It was designed to support time-bounded traffic, and defines periods between consecutive transmissions of two Delivery Traffic Indication Message (DTIM) beacon frames: Contention Free Period (CFP) and Contention Period (CP). Beacon frames are sent periodically by the access point (AP), and carry synchronization and network / Basic Service Set (BSS) information. In particular, DTIM beacon frames are used to indicate the start of a CFP. However, PCF has not been implemented in many actual devices and will not be considered here.

To address QoS concerns of time sensitive data, 802.11e was developed as an approved amendment to the IEEE 802.11 standard. It defines a set of enhancements for wireless LAN applications through modifications to the MAC layer. The standard is considered of critical importance for delay-sensitive applications, such as Voice & Streaming Multi-Media over Wireless IP. The amendment has been incorporated into the published IEEE 802.11-2007 standard. In particular, the IEEE 802.11e standard defines a new coordination function called the Hybrid Coordination Function (HCF). HCF includes both a contention-based channel access method, called the Enhanced Distributed Channel Access (EDCA) mechanism, for contention based data transmission, and a controlled channel access, referred to as the HCF Controlled Channel Access (HCCA) mechanism, for contention free data transmission. The HCCA mechanism has not been implemented very much in industry so our research is based on the EDCA mechanism.

In 802.11e EDCA, there are four Access Categories (AC's), and there are different parameters that control how and when a node accesses the medium e.g., the Contention Window (CW), the inter-frame space and the transmission opportunity (TXOP) are different depending on the priority level, in order to "prioritize" data over transmissions from higher Access Categories (AC's).

Much research [1] [3] [4] has shown how 802.11e can differentiate between different traffic classes and greatly improve QoS. Nonetheless, where there is significant contention within a traffic class, increased levels of contention delay and jitter can occur resulting in unacceptable Mouth-to-Ear (M2E) or End-to-End (E2E) delays for some sessions. In [5], the use of synchronized time to facilitate precise delay measurements was shown to significantly improve VoIP QoS. In this paper we investigate the potential of synchronized time to bring about further performance improvements of 802.11e through parameter optimization of EDCA.

The remainder of this paper is structured as follows. Section 2 will look at the operation of 802.11e EDCA in more detail, and the parameters which are used to optimise QoS. Section 3 will summarise the ITU-T E-Model, and how we plan to quantify any QoS improvements. Section 4 will review the work of [2] [6] [8] which demonstrated how Synchronized time can improve QoS through improved receiver play-out algorithms. In section 5 we outline our plans to use synchronized time and the resulting precise M2E delay information to improve QoS by dynamically optimising 802.11e parameters within the VoIP traffic class. Section 6 will summarize the current status of our work, and we conclude with section 7.

II 802.11E EDCA

The widely implemented 802.11 DCF mechanism provides nodes with an opportunity to access the medium, but in fact tends to favour successful transmissions, leading to possible channel domination by a single sender. Even if Quality of Service (QoS) mechanisms are added at higher network layers, the MAC layer must provide sufficient services to support QoS. 802.11e EDCA classifies the traffic flows into 4 Access Categories (ACs). They are Voice (VO), Video (VI), Best Effort (BE) & Background (BK). Each AC is associated with a single MAC transmission queue, where each AC has its own adjustable MAC parameters and behaves independently of others. The basic MAC parameters of each access category are labelled as:

Arbitration Inter-frame Space (AIFS),
Minimum Contention Window (CWmin)
Maximum Contention Window (CWmax)
Transmission Opportunity (TXOP)

Every AC behaves as a single DCF contending entity and each entity has its own contention parameters, (CWmax[AC], CWmin[AC], AIFS[AC] and TXOP Limit[AC]). The smaller the values of the parameters, the shorter the channel access delay for the corresponding AC, and so the higher priority to access the medium [3]. The AIFS is an IFS interval with arbitrary length as follows: *AIFS[AC] = SIFS + AIFSN[AC] × slot time*, where AIFSN[AC] is called the arbitration IFS number and determined by the AC and the physical settings, and the slot time is the duration of a time slot, (SIFS: Short Inter-frame Space).

The TXOP is defined in IEEE 802.11e as the interval of time when a particular QSTA has the right to initiate transmissions [9]. There are two modes defined, the *initiation of the TXOP* and the *multiple frame transmission within an EDCA TXOP*. An initiation of the TXOP occurs when the EDCA rules permit access to the medium. The multiple frame transmission within the TXOP occurs when an EDCA Function (EDCAF) retains the right to access the medium after already completing a frame exchange sequence [10]. The TXOP limit duration values are contained in the EDCA Parameter Set in Beacon frames. During a TXOP, a STA is allowed to transmit multiple MAC protocol data units (MPDUs) from the same AC with a SIFS time gap between an ACK and the subsequent frame transmission. A TXOP limit value of 0 indicates that a single MPDU may be transmitted for each TXOP.

Control information for QoS management is contained in the Beacon frame, which is generated by the AP at the beginning of each beacon interval (usually 100ms). The EDCA Parameter Set of the beacon frame is used to send the EDCA QoS parameters "Parameter Record Fields" to the QSTA [9].

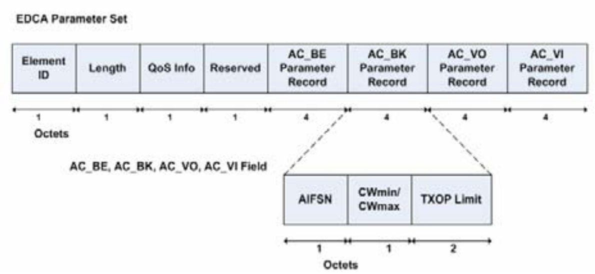

Fig. 1: *EDCA Parameter Set*

III Assessment of Call Quality

The ITU-T E-Model Recommendation defines five categories of end-to end speech transmission quality that act as a guide in establishing different speech transmission quality levels in telecommunications networks. The five categories are defined in terms of "user satisfaction", which have ratings given by the transmission planning tool of Recommendation G.107. The ratings take the combined effects of various transmission impairments into account. The E-Model is independent of any specific technology that may be used in different types of network scenarios [12].

The E-model is a useful tool for assessing the combined effect of all parameters and hence differentiating between categories of speech transmission quality. The primary output of the E-model is a transmission rating factor R. *Table 1* gives the definitions of the categories of speech transmission quality in terms of ranges of Transmission Rating Factor R provided by Recommendation G.107. Also provided are descriptions of "User satisfaction" for each category

R Value	Speech Transmission Quality Category	User Satisfaction
$90 \le R<100$	Best	very satisfied
$80 \le R < 90$	High	satisfied
$70 \le R < 80$	Medium	some dissatisfied
$60 \le R < 70$	Low	many dissatisfied
$50 \le R < 60$	Poor	most dissatisfied

Table 1: *Categories of Speech Transmission Quality*

The calculation of the transmission factor R is defined in [12]. The calculation takes loss, delay, echo, codec type and noise caused by the signals properties and network characteristics into consideration to produce a single R-rating. We will use this system to quantify any QoS improvements for voice sessions within the voice access category, arising from our dynamic tuning of EDCA parameters.

IV Potential of Synchronized Time to Optimize QoS

There are a number of methods of measuring delay in a network, including Round Trip Times (RTT), distributed synchronized time and a variable delay estimation mechanism within routers, the latter requiring a specific protocol format [6]. RTT is an inaccurate mechanism for measuring uni-directional delay, because delays (and sometimes paths) can be quite different in either direction. However, synchronized time is now becoming more widely available facilitating precise delay measurements in each direction. This is due to the more widespread deployment of the Network Time Protocol (NTP) and the availability of accurate time sources like GPS receivers.

To cope with network jitter, much research on receiver-based adaptive buffering and on VoIP has been carried out, such as [2]. Such mechanisms compensate for network jitter by dynamically delaying play-out to facilitate the arrival of delayed packets at the expense of adding to the overall end to end delay. They do so without any knowledge of actual delays. Such an approach is shown in *Fig. 2*:

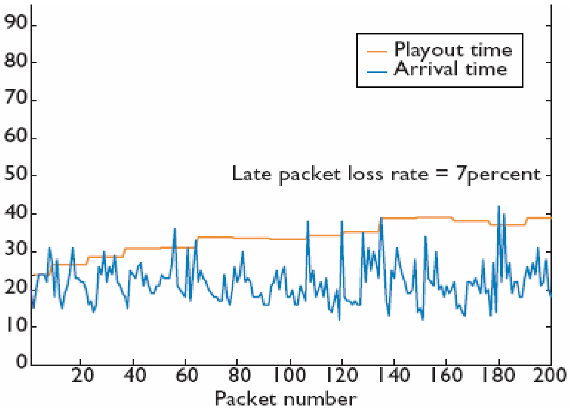

Fig. 2: *Adaptive buffer performance. The y-axis scale illustrates the inter-packet arrival-time variance (jitter) resulting principally from variable queuing delays. The dark line indicates the play-out time of each packet relative to its arrival time.*

Note that packets that arrive after the play-out are dropped, which causes distortion unless some packet loss strategy is used. The advantages of using Synchronized time were shown in [8] where an informed fixed play-out delay was shown to significantly improve voice quality. According to the International Telecommunications Union's (ITU-T) G.114 recommendation, one-way delays should not exceed 150 milliseconds [11]. Therefore, if actual delays are precisely known and well within the G.114 limit, there is room to "manoeuvre" by increasing the play out time to avoid losing any late packets, as shown in *Fig. 3*. The improved QoS is based on the fact (borne out by the ITU-T E-Model G.107 [12]) that users are more tolerant of increased delay (once it is within the G.114 limit) than of increased late loss [12].

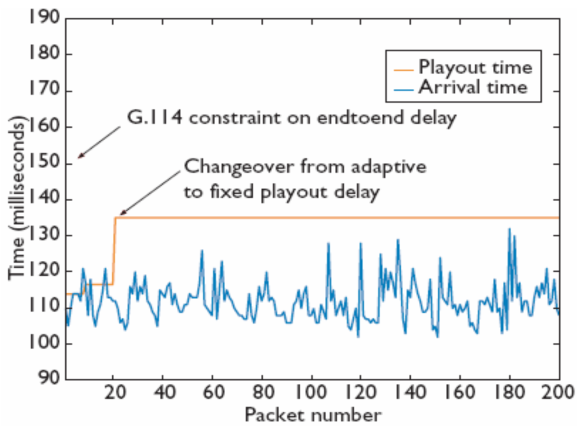

Fig. 3: *With time synchronization, switching from an adaptive to a fixed play-out delay minimises late-arriving packet loss and silence period distortion. In this case, it results in an end-to-end delay of 135 milliseconds.*

For more details on this see [8].

To avail of synchronized clocks, the Network Time Protocol was implemented at each end. Furthermore a mechanism was needed to relate the RTP time stamps to absolute time. RTCP sender reports (SR), which are nominally used to lip-synch audio/video sessions from the same end device by relating RTP timestamps to common device NTP time, were used for this. RTCP packets also currently allow senders to periodically determine round-trip-delay (RTD) time. In a synchronized time environment, RTCP-SR packets will let a sender determine the delay for incoming packets and thus, with the knowledge also of RTD, delays for both legs of the round trip are known in real-time [6]. We propose to use this mechanism as detailed in section VI.

It is important to note that all work by Melvin and Murphy was carried out on wired networks, where synchronization between participants in a VoIP session was << 10msec. Whether synchronized time can be implemented on wireless networks to this precision is a separate research question being undertaken at NUI, Galway. For our research, we assume that synchronized time can be implemented to a satisfactory level on access points and wireless endpoints.

V POTENTIAL OF SYNCHRONIZED TIME TO OPTIMIZE 802.11E PARAMETERS

As outlined previously, the acceptable limit of M2E delay for VoIP applications is approx 150ms. As shown in *Fig. 4*, higher delays can be tolerated with some user dissatisfaction.

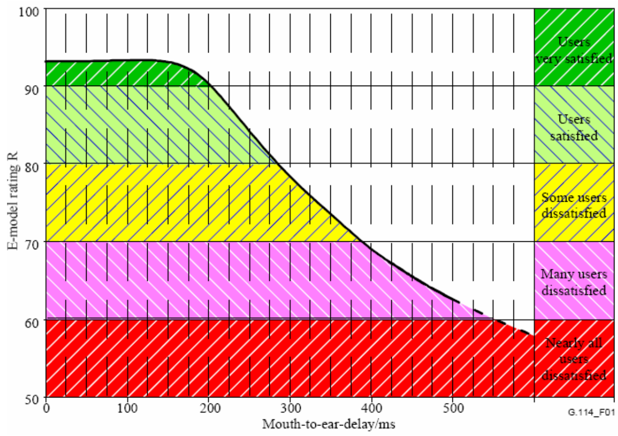

Fig. 4: *G.114 – Determination of the effects of absolute delay by the E-model [11]*

The M2E delay for simultaneous VoIP calls through a single access point will typically vary greatly. For example, some calls maybe over a LAN (with typical network delay < 10ms), and others long distance (with network delay > 100ms). With such latter network delays the total M2E delay (Sender including MAC contention, Network and Receiver) of packets could be close to, or may exceed the International Telecommunications Union's (ITU-T) recommendation (G114) of 150msec.

We propose that if the precise M2E delay information for each VoIP session is known in both directions, EDCA parameters can be configured differently between VoIP sessions to optimise the channel delays, so that the sessions with the higher M2E network delays receive higher priority treatment at MAC level relative to other VoIP sessions with lower delays. As indicated in section II, much research has already been done to investigate the impact of 802.11e parameter tuning. We thus aim to build on this.

VI WORK TO DATE

Research in [13] and [14] finds upper bounds for the number of voice calls that can run concurrently in an 802.11e infrastructure network, while maintaining an acceptable level of QoS. This number can vary depending on factors such as network transmission rate, voice codec, extent of background traffic present and wireless access mechanism used. Once an upper bound for the number of simultaneous voice calls has been reached, *Fig. 5*, we can optimize QoS across multiple VoIP sessions by prioritizing packets for voice sessions that have higher baseline delays as in *Fig. 7*, thus equalizing the delays for all sessions. For more details on VoIP over 802.11e see [13] [14] and [15].

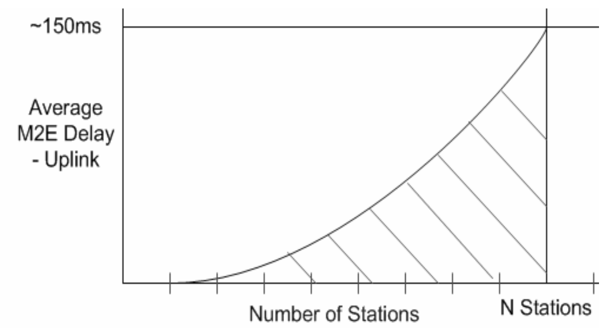

Fig. 5: *Average uplink delay; MAC delay + Other delays = M2E delay*

When N stations are running voice sessions simultaneously, they all have the same contention delays at the MAC layer. Packets that have larger (non-MAC) delays will be prioritized at the MAC layer, in order to equalise the overall (M2E) uplink delay.

Our methodology will be to employ a mixture of simulation and real testbed experiments. With NS-2, we are currently simulating multiple VoIP sessions in the NS-2 network simulator over 802.11e, both with and without background traffic present in the network. Baseline delays (non-MAC delay) can be configured as shown in Fig 7. An NS-2 extension module is currently being designed that will dynamically alter EDCA parameters between VoIP sessions based on precise one way delay information.

The design of our real test bed is shown in *Fig. 6*. It consists of:

- 2 Access Points
- Multiple VoIP clients
- Network Emulator
- NTP or equivalent time synchronization

A network emulator such as NISTNet will be used to introduce different delays for different sessions. NTP/RTCP will be used to determine the delays of incoming packets. Using this information along with Round Trip Delay (RTD) from RTCP, it will enable the outgoing delays to be determined also.

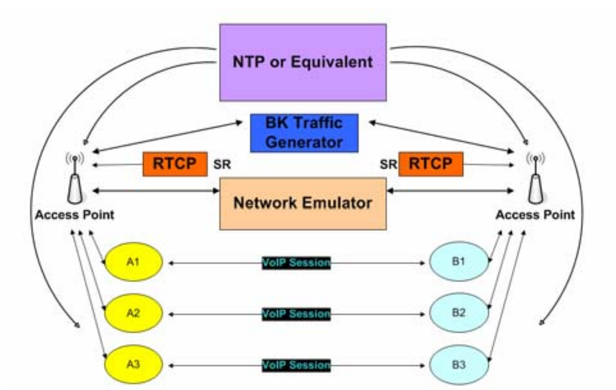

Fig. 6: *Test bed*

Fig. 6 & 7 illustrate how the emulator will be used to emulate calls with different network delay characteristics. In particular *Fig.7* shows 3 VoIP sessions running concurrently, with typical delays for geographical distances such as US – Ireland, East Europe – Ireland and an internal LAN.

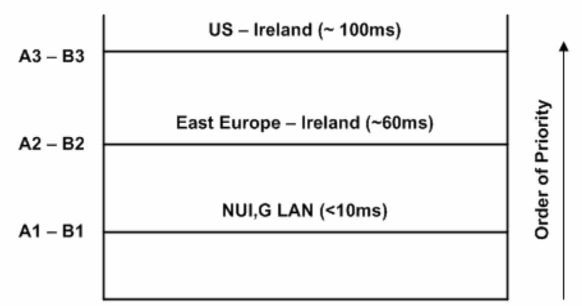

Fig. 7: *Different Baseline Delays*

At the client end, a station with a packet that belongs to a voice session which has a high relative overall M2E delay as mentioned above, can be assigned priority over other stations by tuning the EDCA parameters for that particular station so that its MAC contention delay is reduced.

The precise contribution of the AP in above scenarios will depend on its degree of 'intelligence' about the concurrent VoIP sessions and we envisage a number of possibilities. In the absence of any AP information, co-ordinated management between the different clients will be difficult, and some pre-installed intelligence will be required at each client to implement this in an effective manner. Optimisation will thus only be on the uplink. On the other hand, if the AP is more informed and involved, it can act as the coordinating intelligence and dynamically tune uplink parameters for each station based on precise knowledge of each session. Individual tuning of downlink sessions is then possible but would require use of virtual queues or equivalent within the AP.

VII Conclusion

802.11 based Wireless LANs are becoming the standard in homes, businesses and hotspots. As well as supporting conventional internet applications, there is an increasing need to support QoS centric applications. While 802.11e goes a long way towards meeting this need, severe congestion leading to unacceptable delays and packet loss can still occur. This paper describes our work in progress in which we plan to use synchronized time to provide QoS improvements for VoIP through 802.11e parameter optimization. We aim to have results by Summer 2009.

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