A Traffic-Aware Intelligent Access Point (TAIAP)

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Abstract — For VoIP, Mouth-to-Ear (M2E) delays over wireless networks can vary significantly due to contention and asymmetries. Real-time applications such as VoIP are very sensitive to delay and Quality of Service (QoS) can be much improved if the M2E delay in a VoIP session is known and can be controlled. We are developing a Traffic-Aware Intelligent Access Point (TAIAP) that will calculate and profile M2E delays in both directions for each active VoIP session in a BSS, to prioritize VoIP sessions that are suffering from lower QoS due to excessive M2E delays. This paper will detail the development of our access point, and we hope to have experimental results by late 2011.

1 INTRODUCTION

The 802.11e protocol prioritizes traffic into four categories with Voice traffic being the highest priority followed by Video, Best Effort and Background. The protocol was developed with time sensitive applications in mind. However, severe congestion leading to unacceptable delays and packet loss can still occur within a prioritized category. The primary goal of our research is to investigate how synchronized time implemented in end terminals and access points can aid in optimizing 802.11e parameters in order to improve QoS for VoIP. This paper details our Intelligent Access Point which is currently in development.

Previous papers [1], [2] have shown the improvements that can be brought about for VoIP QoS by tuning 802.11e EDCA MAC parameters, and [3] where it's based on synchronized time. For our research, we have designed an algorithm to dynamically vary parameters that are informed by the E-Model R-factor, based in turn on per-session each-way delay information. It will implement and maintain, where appropriate and possible, an equalization of R-factors within the Voice access category (AC_VO). This equalization will be carried out by tuning 802.11e EDCA parameters for individual clients.

This project is being carried out in parallel with the development of an extension module for the NS-2 Network Simulator, and the goal is to validate testbed results on a more scalable level via NS-2[3].

The remainder of this paper is structured as follows; Section 2 provides some background information and motivations for our research, section 3 will detail our wireless test-bed in more detail and provide information on the technologies used for our experiments.

2 BACKGROUND

2.1 The ITU-T E-Model

Quality of Service can be defined as a metric that measures the overall user satisfaction with network performance. QoS is application dependent and the QoS requirements may vary widely depending on the application. There are different metrics available to quantify the QoS for VoIP applications. One widely used metric is the ITU recommendation G.107 E-Model.

The E-Model outputs a number rating called the R-factor, which is computed as a function of delay, packet loss, equipment impairment factors, and user quality call expectation. The value is calculated as follows:

$$\mathbf{R = Ro - Is - Id - Ie,eff + A} \qquad \mathbf{[4]}$$

where **Ro** is the basic signal-to-noise ratio (received speech level relative to circuit and acoustic noise). **Is** accounts for the impairments which occur with the voice signal. **Id** is the sum of all impairments due to delay and echo effects. With respect to M2E delay, **Id** (in presence of good echo cancellation) can be approximated by 10 units per 100msec up to 150msec, and by 12-13 units per 100msec thereafter [4]. The **Ie,eff** is the effective equipment impairment factor, taking into account the codec and its tolerance to packet losses. **A** is a bonus factor that models the user expectation of the technology employed and can be used to take into account the fact that the user will tolerate some decrease in transmission quality when using certain technologies.

The major factors affecting QoS in any network are throughput, packet loss, packet delays and jitter. Some of the main problems associated with achieving good QoS, particularly in a wireless environment are that the wireless channel can change with time (due to contention or physical layer issues) and so it is possible that available bandwidth can vary greatly during a voice session.

Our AP will be informed by an R-factor estimator as a measurement of quality for dynamically determining the necessary parameters in order to carry out its functionality. When the network is monitored during experiments, an R-factor will be calculated periodically for each individual session, and these R-factor values will then be used to calculate which sessions require prioritization over other sessions.

2.2 VoIP Optimization based on Time Synchronization

The basis for "intelligence" in our AP is in that it adapts EDCA parameters according to an algorithm that computes their optimal settings based on each way delay calculations. This algorithm will calculate based on real-time delay info, the R-Value for each individual session, and then use this information to tune parameters in order to gradually equalize the overall R-factors for all sessions (Fig. 3/4).

The first goal of our algorithm will be to check whether certain sessions have large M2E delay relative to other sessions, and if so, it will aim to reduce the delay for that session(s), once it does not compromise the QoS of the remaining sessions.

Depending on the scenario and the desired outcome, there are varying combinations of EDCA parameters that can be tuned. In a scenario where there are multiple VoIP sessions running on an 802.11 network (such as in Fig. 1), and the goal is to equalize M2E delays for all sessions, as mentioned earlier, the algorithm uses the R-factor to measure the QoS for each session. Where R-factor is below 70 for example, a user would experience a "medium" level of QoS, and some users would be dissatisfied according to the E-Model. QoS for individual sessions would then be calculated in order to determine how 802.11e EDCA parameters will be tuned. We are building this into our access point as the basis of our delay equalization algorithm.

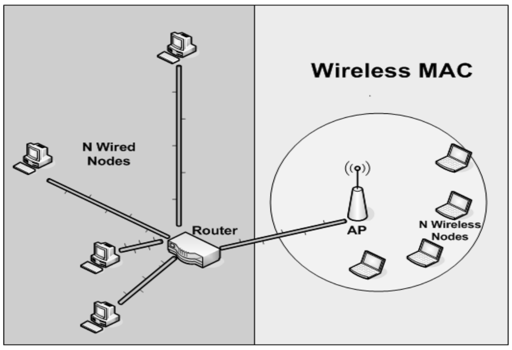

Fig. 1: Example scenario: Multiple VoIP sessions in a network with varying one way delays

As outlined by ITU-T recommendation G.114[5], the acceptable limit of M2E delay for VoIP applications is approx 150ms. Although wireless MAC delays for multiple VoIP sessions through a single access point will be roughly equivalent, the overall M2E delay will typically vary greatly. For example, some calls take place over a LAN, with typical network delays < 10ms, and others over long distance with network delays > 100ms (Fig. 2).

Fig. 2: Different Baseline Delays

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 ITU-T G114 recommendation of 150msec.

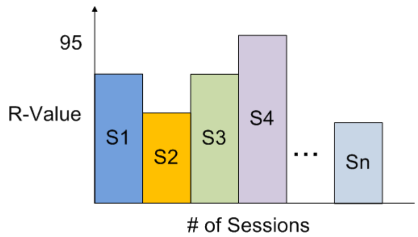

Fig. 3: One way delays can vary for VoIP sessions, which will in turn effect the R-Value

All sessions within the 802.11e EDCA voice category will have roughly similar delays at the wireless MAC layer.

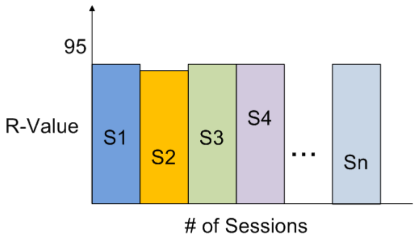

Fig. 4: With Knowledge of each way delays, we can achieve acceptable QoS by equalizing the R-factor among Voice Sessions by dynamically tuning EDCA parameters

The challenge of finding the correct parameters for optimal tuning depends on many factors that have already been addressed in the literature [1],[2]. For example obtaining a lower AIFSN value than competing STAs provides some priority to a node in an 802.11e BSS, this will be a possible element in our algorithm.

2.3 Dynamic Parameter Tuning

Where a VoIP session is deemed to have a low QoS, tuning EDCA parameters will be used to try and improve the delay values. Much research has been carried out on finding the optimal parameters to alter. In [6], the author tunes the AIFSN and CW parameters to optimize overall performance of heterogeneous traffic sources where the delay constraints of real-time stations are met and the throughput of data stations is maximized. In [7] we prioritized a node by assigning a lower AIFSN parameter than the remaining nodes and we did find some improvement in delay. This will most likely be our first parameter to tune when we begin testing.

3 WIRELESS TEST-BED & ACCESS POINT

3.1 Overview

Our experiments involve running multiple VoIP sessions over a wireless network, each connecting to a wired client via an asterisk PBX server. Once the sessions have been initiated, we must calculate M2E delays in each direction on the AP for each session. This information can be computed by analyzing RTCP Sender Reports (SR) and Receiver Reports (RR) at the Access Point (Fig. 5). These RTCP reports contain an NTP timestamp (TS), which allows accurate computation of delays when all nodes are synchronized with NTP.

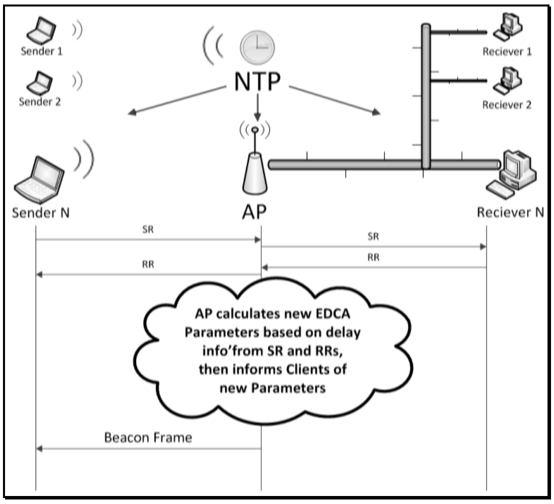

Fig 5: Sender Reports (SR) and Receiver Reports (RR) are used to calculate delays between clients and AP.

A script file running on the AP will use these delay values provided by Tcpdump on the AP, to generate a full picture of delays for all active sessions. This information will then be plugged into the E-Model to generate R-Values for each session. A simple algorithm will prioritize certain VoIP sessions based on their R-Values. This prioritization will be implemented by the issuance of new 802.11e EDCA parameters to all clients via broadcast beacon frames or by sending unicast beacon frames to relevant clients.

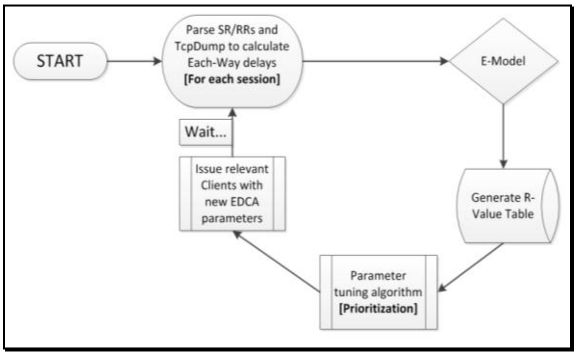

Fig. 6: Life cycle of role of IAP in BSS

Fig. 6 shows an iteration of the script on the access point. This program will control the functions of the AP by calling various functions.

When an experiment is started, a c program will begin on the AP. This will call a function which will run a tcpdump command to capture RTCP SR and RR packets and write filtered data from them to a database. This function will simultaneously write the time of arrival at the AP, of each packet, to the database in order to allow us to calculate the M2E delay in each direction.

A second function will then be called periodically that will read the relevant data from the database in order to calculate an E-Model R-value for each session in each direction.

Once our R-Values are calculated, they will be written sequentially back to a table in the database. A third function will use this data to determine whether or not to prioritize certain sessions, and how to tune 802.11e EDCA parameters for sessions in the experiment as discussed in section 2.3.

3.2 Each-way delay calculations

In order for our TAIAP to have a full view of the network, it must calculate all delay values between each end of a VoIP session, as well as the delay between a client and the AP (see Fig. 7). We will use information from RTCP sender reports (SR) and receiver reports (RR) headers [8] to calculate delays between all clients and the AP.

In order to get a full picture of delay information, the AP must calculate the following delays;

- One Way Delay (wired-wireless / wireless-wired (D1/D2))
- Wireless MAC Delay (wireless client-AP / AP-wireless client (D3/D4))
- Non-Wireless MAC Delay (AP-wired client / wired client-AP (D5/D6))

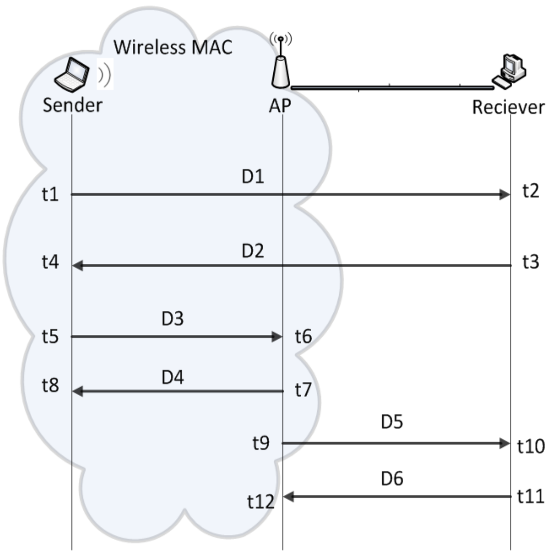

Fig. 7: Delays to be calculated to provide full picture of network.

The above delay values can only be calculated with information contained in RTCP receiver and sender reports (RR/SRs), and the program assumes a synchronized network.

A sender sends a packet at time t1 and it is received at the receiver at time t2, D1 and D2 are calculated using these values which are contained in the SR NTP timestamp and RR DLSR (delay since last SR) respectively. D3-D6 require a logging of times that packets pass through the AP (t6, t7, t9, t10), these values are not contained in RTCP headers and therefore must be logged to our database separately using tcpdump. The delay values D3-D6 are calculated using logged AP packet send/arrival times along with SR/RR information. This process is executed for every active voice session on the AP and allows the TAIAP to determine what sessions, if any, should receive a higher priority.

3.3 Implementation Progress

Our test-bed (Fig. 8) consists of a Linksys wireless router running dd-wrt mega firmware, with 32MB of flash memory. Tcpdump is a powerful command line packet analyzer [7] that is installed on our router and will be called by our program to filter packets passing through the AP. When called, tcpdump filters RTCP packet data and writes it to a SQLite database located on the AP, this allows for efficient read/write operations as all data is stored locally. The X-Lite softphone application is used for making all VoIP calls, as it fully implements RTP and RTCP, therefore all information required from Sender/Receiver Reports can be obtained from packets sent by X-Lite.

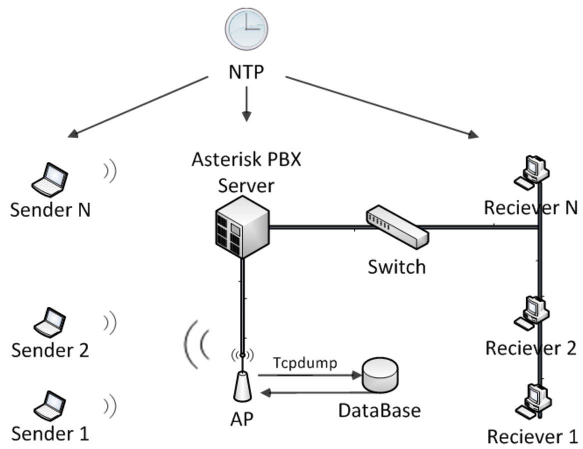

Fig. 8: Test-bed Setup

All processing on the TAIAP is handled by a c program as it is lightweight and doesn't consume much of the routers resources. As memory is the main limiting factor in development on the router, we decided to cross compile all code on a separate linux machine into a single executable that we then transfer across, we also want to make this code as lightweight as possible so that it will be easily upgradable for future development. This program as detailed in section 3.1 will run continuously on AP for duration of experiments.

NTP will be installed at both ends of each VoIP call, and on the AP, to allow us to accurately calculate delay values. This is centric to the experiment as well as to the overall development of our research goals.

The AP is connected to an Asterisk PBX server which handles the setup of all VoIP calls between wired and wireless clients. This is a very useful facility when setting up multiple VoIP sessions.

In order to issue new EDCA parameters to the wireless clients, we must edit the periodic beacon frames that are broadcast to all nodes by the AP. The primary challenge here is the fact that we need to issue different parameters to different wireless clients.

4 CONCLUSION

The development of a TAIAP would greatly enhance the QoS of multimedia, particularly VoIP, in an office environment, where network administrators could implement this as part of a dedicated wireless voice communications framework. Currently we are developing our program for the AP that will handle all the functions of the TAIAP.

Future work consists of completing development of this Access Point (TAIAP) and obtaining experimental results that show the improvements that can be brought about by dynamically tuning 802.11e EDCA parameters for sessions based on delay values calculated with the help of synchronized time within the wireless MAC layer. We are also finishing work on our NS-2 extension module

We hope to have experimental results by late 2011.

Acknowledgments

This research is funded by Údarás na Gaeltachta and The Higher Education Authority (HEA), and is carried out at NUI Galway city campus and at Acadamh na hOllscoilaiochta Gaeilge, Carna[10]. This research is also carried out within the Performance Engineering Laboratory (PEL) [11] research group located at a number of Irish universities including NUI Galway, UCD and DCU.

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