

# ON FAIRNESS AND EFFICIENCY OF ADAPTIVE AUDIO APPLICATION LAYERS FOR MULTIHOP WIRELESS NETWORKS

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## ABSTRACT

*Multimedia applications for networks with wireless links are required to constantly adapt their operations to the changing QoS. In this paper we investigate the effectiveness of a simple adaptive application layer for multimedia streaming over multi-hop ad-hoc networks. Simulations with our hybrid simulation platform are presented that reveal issues involved in end-to-end adaptation over ad-hoc network configurations. Different MAC layers are studied, the standardized 802.11 (DCF) and CSMA, in order to evaluate how these affect the application layer adaptation. Mobility and routing are also applied. We test each of these configurations in 100 randomly generated topologies by gradually introducing adaptive audio connections in place of non-adaptive. We look at results per configuration, per experiment, and per connection, measuring server consumed bandwidth, effective bandwidth, and loss rates. By using the coefficient of variation we report on fairness issues associated with the end-to-end adaptation. Significant improvement is shown when introducing adaptation in all configurations both in loss rates and effective bandwidth. We find that the distribution of server rates is fair in 802.11, especially so, in presence of node mobility. Fairness in effective bandwidth distribution among the connections is significantly increased as adaptivity is introduced. By looking at a connection level we find that when a timeout mechanism exists in the adaptation mechanism, the end-to-end feedback can deliver a consistent view of the RTP loss rates at the server at low mobility and low hop count, even at presence of high competing traffic. The highly oscillatory nature of the loss rates remains for adaptive connections. We argue that a redundant speech captioning scheme is necessary in such conditions. In summary, when connections are adaptive in multi-hop networks perception, efficient bandwidth usage, and fairness is enhanced.*

## 1. INTRODUCTION

A great deal of work is targeted at exploiting adaptive mechanisms at all design layers in order to gain a desired responsiveness in view of unexpected changes in grades of service delivered by the lower design layers. This is especially true in ad-hoc, multihop wireless networks. The application layer plays a significant role in this design space, since it is the layer that interfaces to the user. When the lower layers are expected to present a transient and unstable service such a technique will result in an overall poor user perception. By introducing adaptivity to the application layer and demands placed on the network can be adjusted and user perception may be enhanced by processing the information presented to him with the knowledge of the grade of service supplied by the lower layers. Application models that support adaptivity have been proposed in [8, 9, 10]. An application model suitable for wireless ad-hoc networks can be found in [2]. Architectural considerations are found in [11, 12, 13].

Wireless and embedded computing technologies are expected to become a basic building block in future networks and inter-networks. An increasing number of small size devices will be equipped with wireless communication devices and ad-hoc protocols [7, 15, 16] on Bluetooth, [14] on HomeRF, [17] on sensor networks. Ad hoc multi-hop wireless networks are self-organizing, self-configuring, instantly deployable in response to application needs and independent of a fixed infrastructure existence. As such, they are very attractive to multimedia applications in disaster recovery situations (flood, fire, earthquakes etc), law enforcement (crowd control, border patrol etc), search and rescue in remote areas, sport events, festivals, ad hoc nomadic, collaborative computing, indoor network appliances, and battlefield [18]. Adaptive multimedia for multi-hop networks is discussed in [6].

We investigate adaptive multimedia application layers that continuously within a session, change their demands from the network based on a periodical end-to-end feedback. The change in demand to the lower layers is in response to changing the encoding scheme and/or encoding parameters of the multimedia codec. The decision to choose a layer to use is based on a feedback traversing the end-to-end to path from the client to the server (sender) that contains RTP statistics [19]. Such a system works transparently from any adaptation or QoS provision performed by lower layers.

Initially, we present and validate a new hybrid simulator testbed designed to support experiments where multimedia packets are travelling across parts of both simulated and real networks. As such, it can combine the full complexity offered by real networks with the cost-effective simulated large scale environments as well as produce audible and/or visible results at the receivers, something important for perceptual evaluation. The validation of the platform is based on existing validation work already performed on the GlomoSim [20] platform [21, 22, 23] that forms its basis and on the comparison of a small scale audio streaming experiment performed on both a real multihop testbed and the simulator, presented here.

In defining the environment used in the experiments we introduce a simple client/server adaptation mechanism to a layered multimedia stream, an extension to our QoS Notification Programming Model presented in [2],. Similar work on such mechanisms can be found in [10] where the focus is a traditional wired network and lost packets are mostly caused by congestion. More recent work can be found in [24] and [25] where the network is classified as congested, loaded or unloaded and an additive increase, multiplicative decrease bandwidth estimation is used. We believe our purpose to explore adaptation related issues is served better by not assuming specific strategies.

The characteristics of the streams used are borrowed from an audio stream encoded in 9 layers which differ basically in transmission rate demands and quality, as they are encoded using different codec parameters as well as codec. Namely we borrow characteristics such as rate, inter-send times, quality degradation to loss and others from PCM uncompressed streams encoded at different sampling rates and MPEG layer 3 audio streams for the lower rates [26]. This gives us a wide range of rates to adapt to, from 180Kbps to 8Kbps. We avoided using lower bit rate, speech specific codecs because this range was sufficient to draw our conclusions, and because of significant differences in perceptual quality degradation to loss rate. We consider the work more general by using a combination of robust uncompressed scheme with a widely accepted, audio-perceptual modeled low bit rate codec.

Application layer adaptation experiments may depend highly on random events. For instance, a specific feedback packet being received or not may well change the outcome of an experiment.

Conclusions drawn from a small number of topologies and configuration may not be considered conclusive. This led us to generate a high number topologies (based on constraints) that increase confidence in results. This work is based on 1200 experiments and the conclusions drawn stem from general trends. The large number of experiments directed an aggregation of results in different levels: per configuration, per experiment and per connection.

Special care has been given to exploring fairness. Adaptation has been known to result in unfair situations as reported in [4] for wired networks. By introducing the coefficient of variation of a metric (loss rate, effective bandwidth etc) we find the adaptation improves effective bandwidth long term fairness in multi-hop networks. The work presented here, is to form a basis on which interactions with TCP may be compared to and be better realized.

Our experiments conclude on the promising effects on perception, efficient bandwidth usage, and fairness of adaptive applications in multihop instantly deployable wireless environments.

Previous results from [3] have suggested a highly oscillatory shape of the RTP loss rates versus time, based on real multihop experiments. We validate this oscillatory behavior with our larger scale simulation experiments. This also motivated us to draw conclusions regarding the effectiveness of our aggressive speech layering technique by use of redundant captioning presented in [2, 3]. Experiments in both our wireless multi-hop testbed and our newly introduced hybrid simulation platform show that the communication quality is substantially upgraded even when adverse network conditions persist for a long time.

The rest of the paper is organized as follows: in the next section we present the specifics of the environment used for the experiments and provide an additional validation for our phase 1 hybrid simulator used here. Section 3 describes in detail the results obtained in all aggregation levels and in relation to our speech captioning scheme. In section 4 we present our conclusions and in section 5 we discuss future work.

## **2. ENVIRONMENT**

### **2.1 HYBRID SIMULATION CONCEPT**

The concept of using a hybrid simulation platform for evaluating speech in large wireless networks has been introduced during the WAMIS project at [1]. The hybrid simulation platform introduced and used here is a new platform built from scratch. It is, however, based on the concept introduced in [1, 27, 28] and has been extended to support adaptive application layer techniques over the new GlomoSim platform. It enables us to run large scale adaptive experiments using the large set of options for the lower layers supported by GlomoSim.

Detailed simulation models of complex heterogeneous systems can be extremely resource intensive to develop. An alternative is to use a hybrid model i.e., a partially implemented design, where some components exist as simulation or analytic models and others as operational subsystems realized in hardware or in software [27, 28]. Use of hybrid models allows an analyst to ascertain the impact of design changes in one subsystem without developing detailed simulation models of the entire system; instead certain system modules can be directly replaced by simulation models of alternative designs, considerably reducing the modeling overheads. The hybrid simulation platform described is designed specifically to support adaptive multimedia experiments.

On the input side of our hybrid simulator we have the ability to feed into a simulated subsystem, packet traces produced in an external system. Since the interface to the simulator is the trace file format, the external system can be just about anything that can produce those traces. The traces may come directly from a codec or a streaming application. Alternatively the traces may be captured after travelling a real network subsystem or a different simulation platform subsystem. On the output side we may feed the traces on a next real subsystem. In the case of application layer adaptation to differently encoded streams the process may be significantly simplified by pre-capturing the traces and choosing which to use at run-time based on the adaptation mechanism. Under development, is an application that by reading the output traces and plays them out in order to perform end user perception evaluation.

## **2.2 ADAPTIVE APPLICATION LAYER**

Our multimedia adaptive application design consists of a server who encodes and packetizes audio data and a client that receives them, buffers them to alleviate the jitter and plays them out. The application

is based on the RTP standard recommendations and also uses the RTP statistics to adapt to changes in the network.

### **2.2.1 ADAPTIVE SOURCE ENCODING**

The server is able to encode the data in 9 different layers. These layers differ in bandwidth requirements, inter-send times and quality of the encoded audio as they are encoded at different sampling rates, using different compression ratios and possibly using different encoders.

In previous work [3, 2] based on real wireless testbed experiments we have shown the payload size to be an application parameter that can affect lower layer response. From the same experience we note that accordance of payload size with codec application layer buffers may significantly simplify processing for some codecs that process data in blocks (for example ADPCM, CELP). The payload size also affects perception of audio, especially when the audio application runs on top of non specific lower layer configurations that normally apply their own error control mechanism and may not deliver partially damaged packets. In this work, in order to keep complexity low, we chose a useful payload size of 480 bytes to carry the audio in all layers.

The 9 layers used are 180, 128, 88, 64,32, 24, 20, 16, 8 Kbps. As mentioned previously, this gives us a wide range of rates to adapt to and allow for generalization. We use a combination of robust uncompressed scheme with a widely accepted, audio-perceptual modeled low bit rate codec (Mpeg). They also allow us to use static values for the loss rate threshold of our adaptation mechanism as discussed in the next paragraph (and as opposed to a speech specific codec, for example, where quality degrades differently and in complex relations to FEC schemes etc).

In order to set the loss rate thresholds for our experiments we performed perceptual evaluation of the codec perceptual degradation to loss rates. Measurement of sound perception quality, in general, is a difficult problem, issues of which are discussed in [29] (relevant software work in [30]). This is due to the inability of formalized methods to reflect the highly subjective perception of human ear. MPEG compresses audio by removing the redundancy of the audio signal using statistical correlation and by reducing the irrelevancy of the signal by considering psycho-acoustical phenomena like spectral and

temporal masking. As such, it is not based on any specific assumptions for the audio stream –as the CELP codec for example assumes speech content. Based on the widely used metric of SNR (signal to noise ratio), for source coded PCM and MPEG-1 audio the perceptual degradation can be assumed linear to an evenly distributed bit loss rate in the time domain, as is the underlying assumption in [31]. There, this provides a base for their quality loss measurements for a processor-optimized MPEG codec. We carried out our own experiments that allowed us to listen performance degradation of the different layers used in our experiments due to packet loss. These experiments helped us decide on the maximum tolerable loss rate for our adaptation mechanism. As expected (and discussed in [32]), perception is significantly impaired at a 20% loss rate for the audio codecs we use. The need to simplify our adaptation technique with a single static maximum tolerable loss rate influenced this last decision. In fact, a 20% loss rate in packets caused significantly lower quality in the lower bandwidth streams independently of codec used. This is due to the quantization caused by the 480 payload size which, in turn, caused an uneven bit loss distribution in the time domain as the number of packets expected per second decreased. This indicates that adaptation techniques developed for specific types of networks can gain in perception by introducing soft and dynamic rather than hard and static thresholds.

An important perceptual degradation when audio data are streamed over a network occurs from the non-uniform distribution of errored packets. In fact, it has been shown that significant quality is lost due the bursty nature of errors when non-adaptive MPEG is streamed over the internet [33]. In that same paper a technique to interleave audio packets at the server is presented specifically to deal with this type of quality degradation.

### **2.2.2 ADAPTATION MECHANISM AND QOS NOTIFICATION**

The server packetizes (in RTP payload) and sends the audio according to parameters of the currently selected layer. In our implementation each packet contains a number (from 1 to 9) which indicates all the necessary parameters to the client who will perform the playback. The client uses this information to play out the packets using the specified parameters and calculate the RTP statistics as well. The loss rate is

measured over an interval of one second and sent back to the server. The server uses the loss rate information to adapt its rate as in the procedure shown in figure 1.

The QoS feedback packet itself, has to traverse the network back to the server, and the probability of it actually making it there on time is inversely proportional to its importance (we assume a QoS feedback packet to be important when it carries information about a congested network). In experiments performed in a real wireless testbed we concluded that a UDP transport is much more appropriate to carry the feedback than a transport using retransmissions like TCP. TCP retransmissions result in stale QoS information especially on a congested network. With UDP, a mechanism can be employed at the server so that when a few QoS packets are lost the server assumes a congested network. In our implementation we choose to downgrade the service whenever we detect  $TOTimesThreshold$  lost QoS packets every  $TOMaxTicks$  (see Table 1 and Figure 1). A soft timeout mechanism is proved to be necessary because at high congestion feedback packets have a low probability of reaching the server. In a more complex adaptation mechanism, intuitively, an improvement in efficiency can be expected by dynamically computing these thresholds based on current bandwidth consumption and past experience. As mentioned, a more complex adaptation mechanism is not fitted for this work.

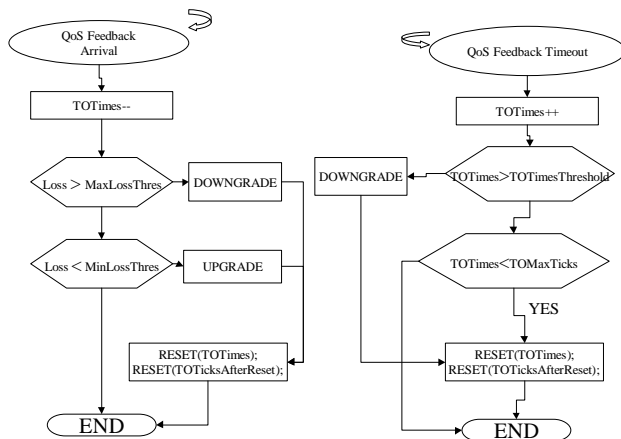


Figure 1: Adaptation mechanism

<i>MaxLossThreshold</i>	20%
<i>MinLossThreshold</i>	5%
<i>TOTimesThreshold</i>	5
<i>TOMaxTicks</i>	7

Table 1: Adaptation mechanism parameter values

<i>MAC</i>	<i>Network</i>	<i>Application</i>	<i>Mobility</i>
802.11	STATIC	NO-ADAPT	NO
CSMA	BELLMAN FORD	MIXED (ADAPT5)	RANDOM WAYPOINT 2.9km/hour
		ADAPTIVE (ADAPT10)	

Table 2: Configurations tested

## 2.3 MULTIHOP NETWORKS



We extensively experimented with non-adaptive audio and mixed traffic of adaptive and non-adaptive in order to produce clearer results. The different options tested are shown in Table 2.

A fair number of nodes is used in these experiments. Specifically 25 nodes were randomly placed in a 60 by 60 unit area, each equipped with a 2Mbps radio of 30 units range. A large number of experiments with randomly placed nodes was needed to increase confidence, applicability and generalization of the results. This instructed the 30 unit range provides reasonable probability of the generated topology being disjoint and connections will interfere with each other and share bottlenecks. 100 topologies were produced. The same 100 topologies were used in each experiment. All of them were inspected using our topology visualization tool –based on gnuplot. 10 different servers and 10 different clients were defined, either producing adaptive or non-adaptive audio. The only traffic in the network is these 10 audio connections. The resulting routes are mostly 1, 2, or 3 hop averaging at 1.8 hops.

A large interval (100 seconds) in the beginning of the simulation allows the free exchange of bellman ford tables among the nodes. After that, the audio traffic is generated. In case of static topologies the routing table exchange is stopped. In the experiments with mobility the bellman-ford exchange of routing tables is allowed to go on. The mobility model is the random waypoint mobility model [34]. It is a model widely accepted and used. The nodes randomly choose a destination point and move 1 distance unit towards it by randomly choosing an axis on which to move. In our experiments they take 5 seconds to complete the one move. Then they wait for 10 seconds and choose another destination. This gives an average speed of 720 units/hour. In all, this scenario resembles a real “low-mobility” situation. By accepting one unit to be 4 meters then we get a “real” open transmission range of 120 meters, where people (nodes) walk (move) at 2.9km/hour within an area of 240 by 240 meters.

Initial experiments showed that the traffic cannot be carried sufficiently by the network at full rate (180Kbps each). In this way the effectiveness of adaptivity may be clearly shown while specific adaptivity issues are easily addressed. The configuration options across layers are tabulated in **Table 3**.

<i>Simulation Time</i>	<i>400 sec</i>
<i>Propagation Function</i>	<i>Free Space [36]</i>
<i>Radio Bandwidth</i>	<i>2Mbps</i>

<i>Radio Type</i>	<i>No Capture</i>
<i>MAC Queuing</i>	<i>FIFO</i>

Table 3

## 2.4 SPEECH CAPTIONING EXTENSIONS

As mentioned before, partial results motivated us to apply loss rates distribution results to our speech captioning technique presented in [2]. Synchronization, adaptation and perception issues of its application have been addressed in [3]. We use captioning as an ultimate compression technique. A text transcription is associated with the speech stream using a speech recognition engine and is piggybacked to the audio packets in a redundant fashion. When the locally calculated RTP statistics suggest switching to this bottom layer, the speech can be reproduced using the transcription with a text-to-speech synthesizer. In this way, we are able to sustain meaningful communication even at times when interference allows none but a single packet to be correctly received every 2 seconds. For more details see [2],[3].

## 2.5 ADDITIONAL VALIDATION OF HYBRID SIMULATOR AUDIO BASE FUNCTIONALITY

Validation of the first phase of the hybrid simulator is based on validating GlomoSim and validating the software built on top of GlomoSim. For the first see publications [20, 21, 22, 36]. Validation of the application components has been performed by performing small scale experiments in both real and simulated networks. RTP loss rates in both have same shape and magnitude.

## 3. EXPERIMENTS

### 3.1 OVERALL AVERAGED RESULTS

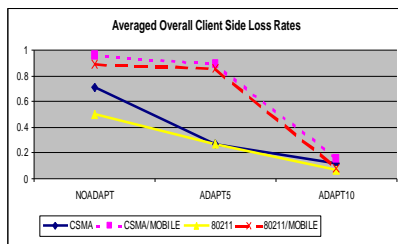


Figure 2: Averaged per connection and per experiment client loss rates vs adaptivity

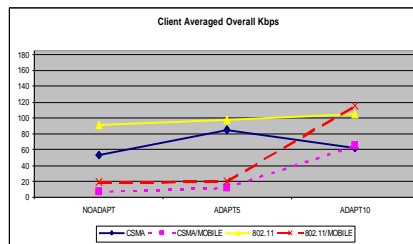


Figure 3: Averaged per connection and per experiment effective bandwidth vs adaptivity

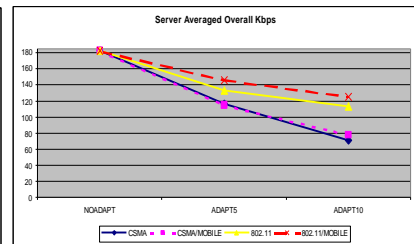


Figure 4: Averaged per connection and per experiment server consumed bandwidth vs adaptivity

The points in figures 2-4 show the metrics when none of the connections is adaptive, half are adaptive and when all servers are adaptive (experiments named accordingly NOADAPT, ADAPT5, ADAPT10). The server consumed bandwidth, shown in Figure 4, does not depend only on the actual RTP loss rates received but also, on how effectively the QoS architecture through the QoS end-to-end feedback mechanism managed to convey this information at the server side. In high loss rate situations, for example, the server will keep pumping at his current layer until he receives a QoS feedback report or until it times out waiting for it, as per our scheme. For this reason, we see that in face of mobility, even though more packets are expected to be lost (and are lost as seen in Figure 2) the servers pump more packets because they fail realize the QoS change through the end-to-end mechanism. This effect also appears later.

Figure 6 shows, that by reducing the rate at the server side we increase the effective bandwidth (the bandwidth from packets that actually reached the client). This is especially true when mobility is introduced. Even at the low speed mobility, an average drop in a server rate of 60 Kbps (33%) -performed in a timely fashion as directed by the adaptation mechanism- results in a 100 Kbps more (500%) effective bandwidth (802.11, adapt all, mobility). For CSMA, a 100Kbps drop (55%) results in a 47Kbps (265%) increase in effective bandwidth.

There is one case in the graph that does not follow the general trend. In CSMA having half (5) of the connections adapt results in more effective bandwidth than having all of them adapt (the losses are significantly decreased as per trend though). This is because the adaptation process is based on a RTP type loss which is sampled on a short past interval. Since we upgrade layers (bps) less reluctantly than we downgrade then such cases may exist In this implementation we have a threshold of MINLOSSRATE 5% below which we upgrade and a MAXLOSSRATE of 20% above which we downgrade.

In Figure 2 by observing the overall averaged loss rates we see that when we replace non-adaptive connections with adaptive ones the loss rates drop. This is also an indication of better quality provided that we promptly adapt to the changes and the buffering mechanism works well. We go on

looking at those experiments closer, to increase detail of understanding and applicability of the graphs presented in this section.

### **3.2 PER EXPERIMENT RESULTS: FAIRNESS AND ADAPTIVITY**

An important issue in a network with adaptive streams is fairness. In this work as discussed in the opening paragraphs we will concern ourselves with fairness among the audio connections. In a distributed environment where many audio connections are sharing parts of the network, it is desirable to equally distribute the network resources among them to avoid the case where one connection gets perfect QoS while another sharing the same bottlenecks suffers. As mentioned fairness problems are common in adaptive connections [4].

#### **3.2.1 SERVER RATES**

In this paragraph we look at fairness in server side rates as those are directed by the adaptation mechanism. The adaptation mechanism orders downgrades and upgrades in layers resulting in an average rate for the lifetime of the session. It is desirable and efficient to stream at similar rates among connections that face similar network conditions. For our 400 experiments, where all of the connections adapt, we define and graph in Figure 5 the coefficient of variation in average (among the 10 audio connections) server side rates. The coefficient of variation is a metric of the dispersion from the mean that is comparable among series with different magnitude of mean averages. The COV (or CV) expresses the standard deviation as a percentage of the mean. It is defined as the standard deviation divided by the mean and has been used extensively as a fairness metric.

In our case, the graph is not independent of the rates of the different layers. It depends on the gaps between the layers. It can be made independent by assigning equal spaced numbers to each layer. This would be desirable when comparing adaptation mechanisms. In our case the graph of the COVs of layer numbers looks very similar to the COVs of the server rates (in all experiments the adaptation mechanism directs to downgrade similarly).

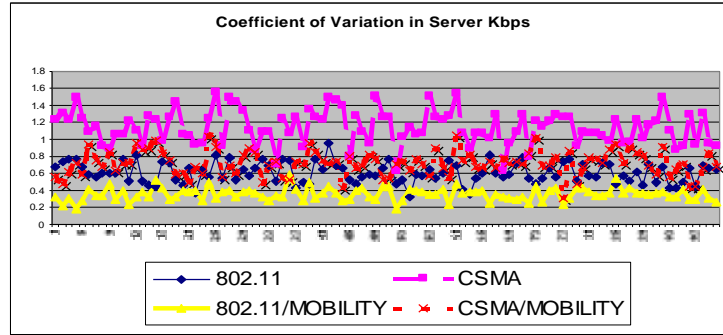


Figure 5: Coefficient of variation of server rates of the 10 connections (y axis) for all 100 generated topologies (x axis) when all connections are adaptive

We notice that adaptive applications appear more unfair in apportioning server rates when are tested in static topologies (notice the similarity in absolute rates in Figure 4). When mobility is introduced fairness in server rates is improved dramatically. The COV is reduced by 100% in both CSMA and 802.11 cases. Mobility introduces randomness in the loss rates and the feedback effect and so fairness is increased. In general 802.11 is fairer in apportioning server rates than CSMA. From this set of experiments we may conclude that fairness concerns in end-to-end adaptation are not directly applicable to ad-hoc networks. We go on looking into fairness issues closer, by comparing COVs in client kbps and loss rates.

### 3.2.2 EFFECTIVE BANDWIDTH

In Figures 6 through 9 fairness in client effective bandwidth is increased (COV is decreased) when we adapt. This is true for all cases of 802.11 or CSMA MAC and mobility or not. When half of the audio connections adapt fairness is not considerably improved but remains comparable with the case of NOADAPT. These results are based on 1200 experiments performed and the COVs have a clear trend in all 100 experiments of the same type.

A notable graph is shown in Figure 6. In 802.11 with no mobility adapting in half the connections results in improved fairness (same as adapting in all). This is the only exception, since in other cases we see an improved fairness only when we adapt to all connections. This is not an “out of trend” result though. Consider that the adaptation mechanism works well when the feedback works well. When the feedback travels in a congested –not released from adaptive connections- network, the result is a less

responsive and less network-aware servers. The case of 802.11 is the one that constantly throughout the experiments delivered increased effective throughput. This happened naturally in the non-mobile experiments as compared to the mobile. The case of 802.11 with no mobility is the case where the adaptation mechanism works best. This is why when half the audio connections adapt we were able to notice the fairness effects noticed usually when all the connections were adaptive. See also Figure 3 (client rates) where 802.11, non mobile is by far better in the ADAPT5 case of interest.

These graphs also show that CSMA is in general more unfair and less efficient than 802.11 (dominant COV in CSMA is 3 whereas in 802.11, 2 in non-adaptive cases). However making the connections adaptive has an equal proportionally positive effect on fairness in both MACs.

COVs of loss rates graphs are not in keeping with the client kbps COVs. Loss rate COVs across clients become higher with adaptive connections. This cannot lead to any valid conclusion though. By looking at the absolute per connection loss rates it becomes clear that when connections are not adaptive the network is unable to deliver the packets altogether. For example in experiment with topology #0 on 802.11 (no adaptation, no mobility) the 10 non-adaptive connections get a loss rate per connection tuple of (69, 98, 100, 100, 100, 100, 100, 100, 100, 100). When these connections are turned adaptive in an otherwise all similar experiment the loss rates tuple becomes (17, 51, 0, 42, 1, 0, 39, 20, 8, 8). The last one has obviously a higher COV. Measuring fairness directly in loss rates using COVs thus is misleading.

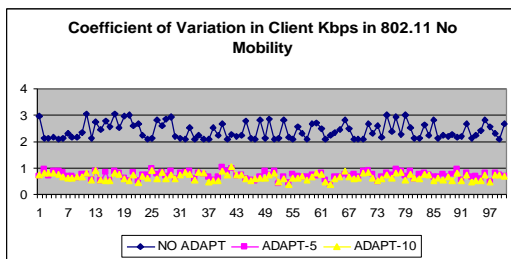


Figure 6: Coefficient of variation of client rates of the 10 connections (y axis) for all 100 generated topologies (x axis) with 802.11 MAC and no mobility

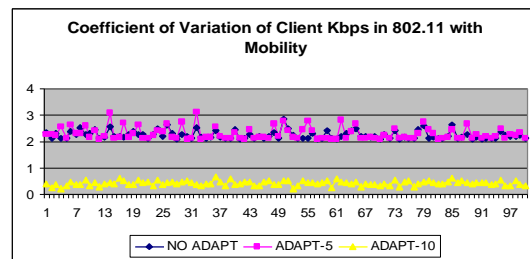


Figure 7: Coefficient of variation of client rates of the 10 connections (y axis) for all 100 generated topologies (x axis) with 802.11 MAC and mobility

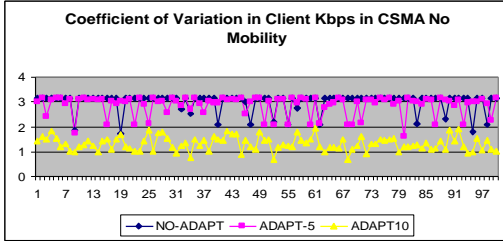


Figure 8: Coefficient of variation of client rates of the 10 connections (y axis) for all 100 generated topologies (x axis) with CSMA MAC and no mobility

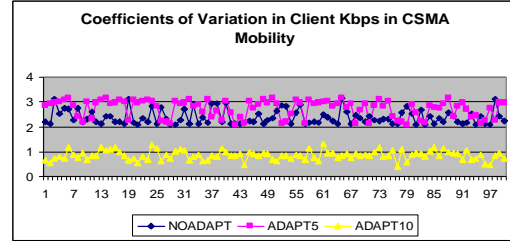


Figure 9: Coefficient of variation of client rates of the 10 connections (y axis) for all 100 generated topologies (x axis) with CSMA MAC and mobility

### 3.3 PER CONNECTION RESULTS: RTP LOSS RATES TIME DISTRIBUTIONS

In this section we look at individual connections rather than aggregated metrics. In particular, we notice the RTP loss rates versus time for two typical connections. In Figure 10 we can see the randomly generated topology for experiment 10. Its aggregated results can be noticed in all the graphs presented so far. Here we choose connection 1-21, a typical 2-hop connection that adapts (in the adaptivity experiments) and contests with the other connections for its bandwidth, and connection 0-20 which does not adapt (in the experiments that half the connections adapt).

In all experiments, the server and the client had a similar view of the loss rates throughout our experiments. The feedback mechanism worked well even at the presence of low mobility and high loads. Intuitively, however, increased hop counts must be explored. In general, the network is heavily loaded, the feedback packets have high loss rates. This indicates that a properly fine-tuned time-out mechanism is necessary. As mentioned before in these experiments we have incorporated into our simple adaptation mechanism two variables for soft timeout support (see section 2).

#### 3.3.1 STUDY OF AN ADAPTIVE CONNECTION

In figures 11 through 22 we see the loss rates distribution in time for the connection 1-21 as the network around it (and itself) becomes adaptive. The improvement in RTP loss rates distribution in time is shown clearly for the case of 802.11 with and without mobility (connection 1-21 that adapts in the Adapt-5 experiment series). In fact, the shape of the distribution of the RTP loss rates in time is notably similar in all cases, whether the network has adaptive connections or not, and the change is seen in the magnitude of the loss rate.

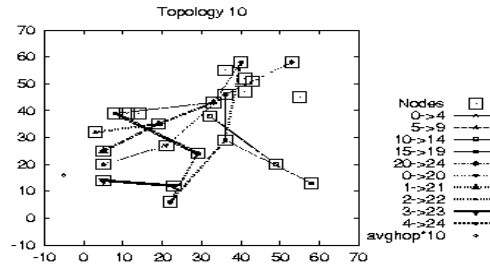


Figure 10 : Randomly generated topology #10.

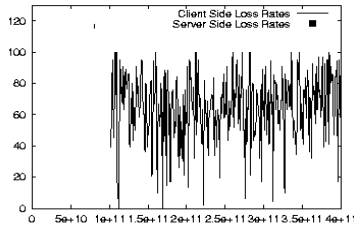


Figure 11: No Adapt No Mobility 802.11 Connection 1-21 (loss rates vs time)

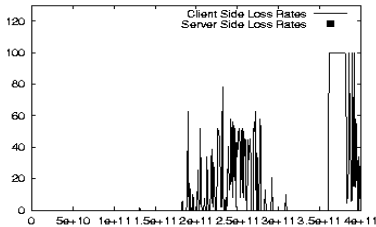


Figure 14: No Adapt, Mobility, 802.11 Conn 1-21 (loss rates vs time)

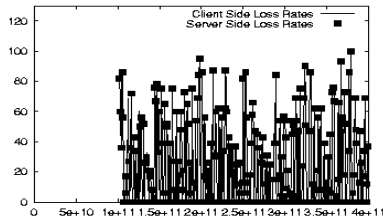


Figure 12: Adapt5 No Mobility 802.11 Connection 1-21 (loss rates vs time)

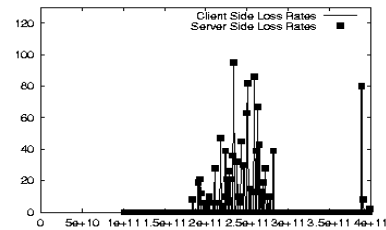


Figure 15: Adapt5, Mobility, 802.11 Conn 1-21 (loss rates vs time)

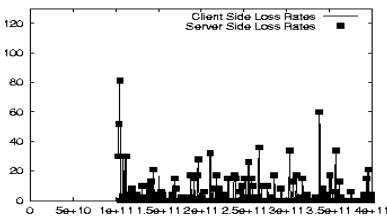


Figure 13: Adapt10 No Mobility 802.11 conn 1-21 (loss rates vs time)

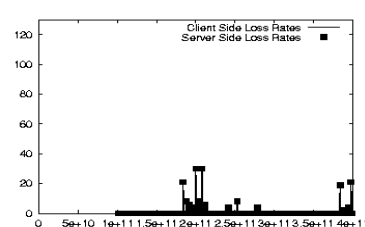


Figure 16: Adapt10, Mobility 802.11 Conn 1-21 (loss rates vs time)

In Figures 17 to 23 the same results are presented for a CSMA MAC in order to evaluate the effect of the different MAC layer. The general conclusions reached for the 802.11 experiments apply here as well. The CSMA MAC layer resulted for all the experiments in graphs that exhibited similar trends with 802.11 but with a less obvious convergence to the conclusions. This is due to CSMA's more deterministic parameters that result in behavior that favors or shuts down completely connections for some period of time depending on periodic transmissions, as those typically seen in audio connections. This deterministic behavior is diminished when one more degree of randomness is introduced by mobility



(see figures 20, 21, 22). In that case, the CSMA graphs closely match the trends of the 802.11 graphs (see aggregated results in paragraphs above). This difference in behavior is illustrated in Figure 35 and 36 where the sequence numbers of the packets are graphed against their arrival time.

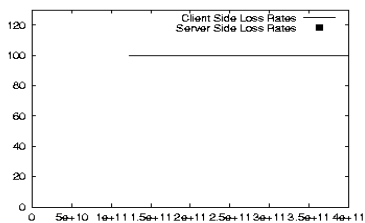


Figure 17: NoAdapt No Mobility CSMA Conn 1-21 (loss rates vs time)

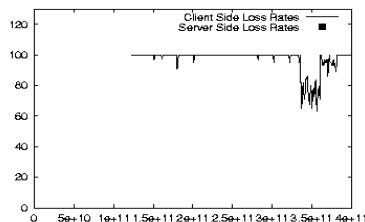


Figure 20: No Adapt, Mobility, CSMA Conn 1-21 (loss rates vs time)

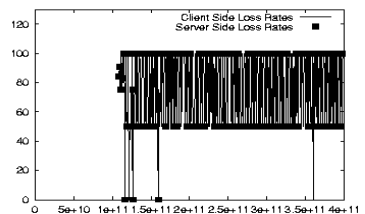


Figure 18: Adapt5 No Mobility CSMA Conn 1-21 (loss rates vs time)

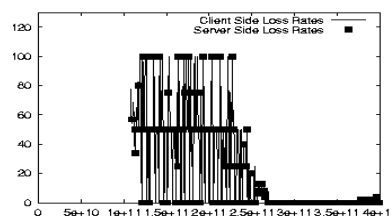


Figure 21: Adapt5, Mobility, CSMA Conn 1-21 (loss rates vs time)

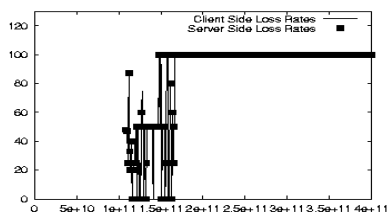


Figure 19: Adapt10, No Mobility, CSMA Conn 1-21 (loss rates vs time)

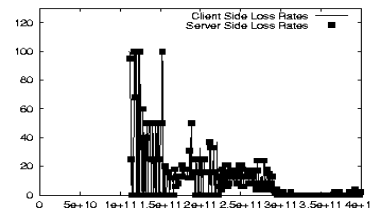


Figure 22: Adapt10, Mobility, CSMA Conn 1-21 (loss rates vs time)

### 3.3.2 STUDY OF A NON-ADAPTIVE CONNECTION

A different 2-hop connection from the same experiment is being looked at closely in this paragraph. The connection does not adapt in the experiment ADAPT5. The purpose of this paragraph is to see how the loss rate distribution in time is affected for a non-adaptive connection when other connections that share the links adapt, and how this is compared to the case where the connection itself adapts.

In 802.11 with mobility experiments we see clearer the improvement between the three experiments (NOADAPT, ADAPT5- non adaptive connection, ADAPT10- adaptive connection). Note a shut down period from 300sec to 380sec approximately. This dis-connectivity is due to routing (bellman

ford is used in the mobile experiments). During that time no feedback arrives at the server. The server adaptation mechanism times-out and downgrades the rate repeatedly. The audio connection is resumed after the routing tables successfully track the node positions, in the reduced rate. Reducing the rate in this case bears the significant advantage of allowing more bandwidth for other connections (and the uninhibited inband exchange of routing tables) but does not have any quality effect at the client. In other words, downgrading in this case does not serve the “dual purpose of adaptation”.

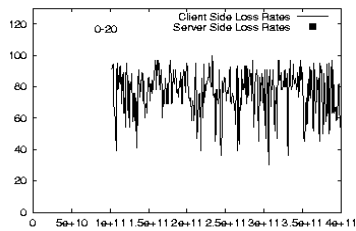


Figure 23: No Adapt No Mobility 802.11 Connection 0-20 (loss rates vs time)

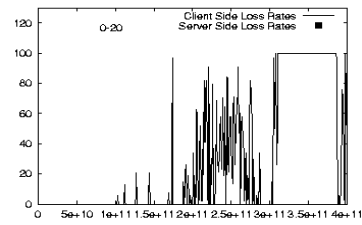


Figure 26: No Adapt Mobility 802.11 Connection 0-20 (loss rates vs time)

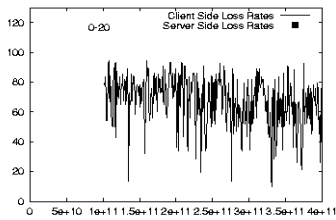


Figure 24: Adapt5 No Mobility 802.11 Connection 0-20 (non-adaptive) (loss rates vs time)

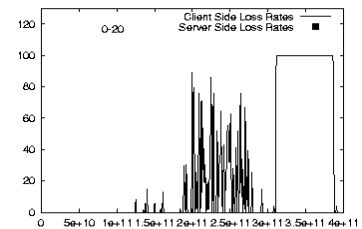


Figure 27: Adapt5 Mobility 802.11 Connection 0-20(non adaptive) (loss vs time)

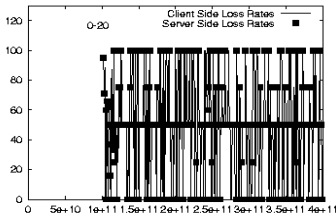


Figure 25: Adapt10 No Mobility 802.11 Connection 0-20 (loss rates vs time)

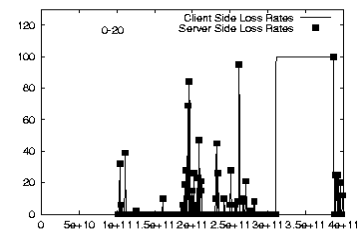


Figure 28: Adapt10 Mobility 802.11 Connection 0-20 (loss rates vs time)

In Figure 29 we see the CSMA, no mobility results for the same connection. When the other connections adapt, we note a dramatic improvement in loss rates. From 100% loss through out the experiment the server takes advantage of the bandwidth that becomes available from other connections rate adaptation. See also Figure 18 for the exact experiment on a competing connection that adapts.

In the next Figure 31 we see how the loss rates are changed in this case when the connection becomes adaptive itself as do all network connections. Now the connection itself reduces the rate and allows other connections to ‘capture’ the channel. This results in higher loss rates for the adaptive connection. The ‘capture’ effect has been studied for CSMA and CSMA/CD environments in [35]. Here we also include Figure 35 and Figure 36 which illustrate the different behavior of the two MAC protocols used in these experiments.

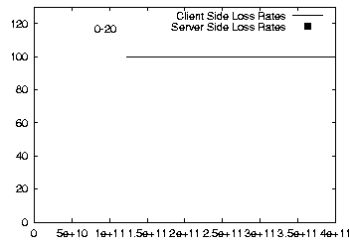


Figure 29: NoAdapt, No Mobility, CSMA Connection 0-20 (non-adaptive) (loss rates vs time)

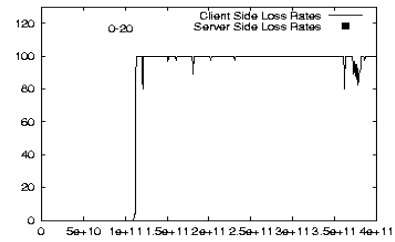


Figure 32: No Adapt Mobility CSMA Connection 0-20 (loss rates vs time)

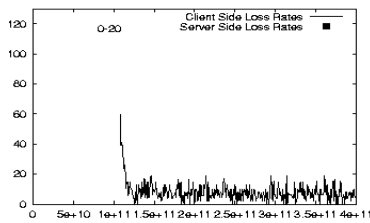


Figure 30: Adapt5 No Mobility CSMA Connection 0-20 (non-adaptive) (loss rates vs time)

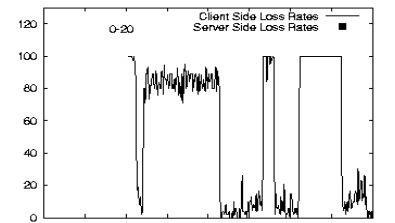


Figure 33: Adapt5 Mobility CSMA Connection 0-20 (non-adaptive) (loss rates vs time)

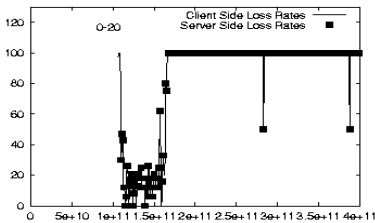


Figure 31: Adapt10 No Mobility CSMA Connection 0-20 (loss rates vs time)

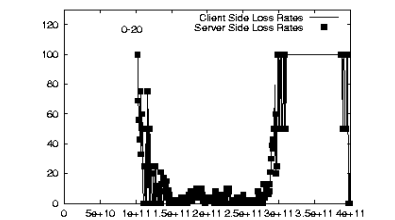


Figure 34: Adapt10 Mobility CSMA Connection 0-20 (loss rates vs time)

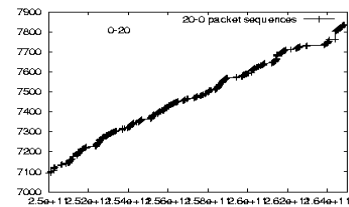


Figure 35: Sequence Number of packets vs arrival during 250 to 265 sec in the 802.11, mobility, adapt5 experiment for connection 0-20

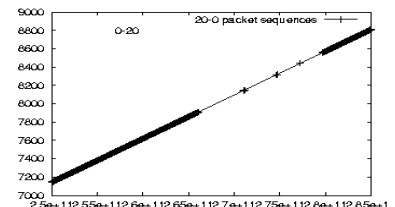


Figure 36: Sequence Number of packets vs arrival during 250 to 285 sec in the 802.11, mobility, adapt5 experiment for connection 0-20

### **3.4 SWITCHING TO TEXT GENERATED SPEECH**

The necessity of our speech captioning scheme presented in [2, 3] is indicated by these results. The oscillatory nature of loss rates there proved the necessity of the scheme in order to keep a speech connection alive. Here, in much larger scale networks we find that even at best case of averaged loss rates –for example as those seen in 802.11, no mobility- the loss rate distribution in time is still highly oscillatory and frequently above a 20% even for averaged loss rates of 5 to 10%. As shown in [2], our scheme is necessary and effective in such conditions.

## **4. CONCLUSIONS**

In this work we adopted a simple application layer adaptive architecture for multimedia applications. The adaptation mechanism -validated and used in previous work on real multi-hop testbed- was fine tuned according to audio perceptual results. By using our hybrid simulation platform, we run extensive experiments in order to explore the behavior and effect of adaptive audio and speech connections in ad-hoc multi-hop wireless networks. 1200 experiments formed the basis for the conclusions in this paper. 100 randomly generated topologies with similar characteristics were used to clearly denote the trends. The traffic included mixes of non-adaptive and adaptive audio connections running over CSMA and 802.11. No mobility and a low mobility scenarios were used. Audio connections borrowed their characteristics from known widely used codecs. They adapted to 9 layers of different codecs and a rate range from robust PCM of 180Kbps to low bit rate MPEG audio at 8Kbps. We looked at loss rates, client and server rates and used their COV's to explore fairness issues. The results were aggregated in 3 different aggregation levels.

At the configuration level we clearly noted the effect of adaptation on the networks under study. When all the audio connections adapt there is significant improvement when compared to non-adaptive traffic both in loss rates and effective bandwidth.

At the experiment level, apart from looking at the absolute averaged loss rates, client effective bandwidth and server rates, we looked at fairness issues by looking at the COV's of these in each experiment. Based on the similarity constraints of the experiments we compared those and found:

- In multihop mobile networks over 802.11 our simple adaptation mechanism fairly distributed the rate at which servers should transmit. This became particularly evident when mobility was introduced. CSMA exhibited similar behavior but double COV's than 802.11
- As connections become adaptive, the distribution of effective bandwidth was consistently fairer especially for 802.11 networks.

At individual connection level, we noticed closely scenarios with competing adaptive and non-adaptive connections and the end-to-end feedback effectiveness. In general, when a timeout mechanism exists to drop layers in the adaptation mechanism, the feedback can deliver a consistent view of the RTP loss rates at the server at low mobility and low hop count. The highly oscillatory nature of the loss rates in all cases studied indicated the necessity and effectiveness of an additional, local to the client layer switching –such as the one we introduced in [2] [3]. It has been shown to enhance speech perception and be necessary to sustain meaningful communications in such network conditions.

In general, we explored issues related to end-to-end adaptation for multimedia applications over multihop wireless networks, concluded on the very promising effects on perception, efficient bandwidth usage, and fairness, and formed a basis for comparison and understanding of adaptive connections in multihop networks.

## **5. FUTURE WORK**

There is a considerable number of parameters that has been left out in this study and require examination. Similar connections were used and explored on how being replaced with adaptive ones would change the dynamics of the network. External interference was left out and is considered future work. By introducing external interference and TCP, jitter statistics are affected and buffering issues become of key significance to multimedia applications. Initial experiments that carry TCP traffic have been performed and present complicated scenarios to be studied. The development of our hybrid simulator platform continues in directions that will allow us to have perceptual feedback from the simulated network and on-line reaction to events occurring in different components either simulated or real.

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