ADAPTIVE MULTIMEDIA APPLICATIONS TO IMPROVE USER-PERCEIVED QOS IN MULTIHOP WIRELESS AD-HOC NETWORKS

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Adaptive applications are a key concept to take into account when dealing with multimedia internetworking in wireless and mobile environments, in which network conditions may change abruptly and not always due to congestion –as it happens in fixed networks- but also to interference, mobility and so on. This paper presents an adaptation architecture which allows applications to minimize the impact of adverse and changing network conditions, by using self-adjustment techniques at high layers, obtaining information from the network. The objective is maintaining the QoS perceived by the user at an acceptable level. Results are presented by means of some tests in different test-case scenarios.

1 Introduction

There are many real-time multimedia applications which are able to distribute audio, video and many other streams. However, few of these applications have mechanisms for taking advantage of QoS and network status information like packet losses, delay variation or available bandwidth. These applications are also not able to adapt automatically to changing network conditions to offer a good user-perceived QoS. Thus, adverse network conditions –which are common in ad-hoc and mobile environments– like packet losses, abrupt bandwidth changes and substantial delay variations, cause excessive degradation in the quality which is perceived by the user. Some studies like [1] show that user-perceived audio quality starts becoming extremely bad when packet losses goes over 20% even when packet retransmission techniques are used to replace lost packets. This demonstrates the need for adaptive applications and the appropriate signaling mechanisms.

The idea of adaptability was already known for fixed networks [2]. However, these results are not directly applicable to wireless and mobile scenarios: in fixed networks packet losses are usually due to network congestion, while in a wireless environment there are many other factors such as access to the radio link, interferences, fading, etc. In that work, changes of data rate were enough to improve the audio or video quality. There are works focused on wireless networks ([3, 4]) which offer some interesting ideas like using speech recognition or text files as an alternative. However, they handle the adaptation only by changing the sampling rate and packet size. In addition, the bandwidth value needs to be provided by the user, and the source and destination are required to be synchronized via NTP to calculate the variation delay. These requirements may not be always met in present or future mobile and wireless networks.

Typical QoS provisioning solutions have been always based on the idea of trying to reserve or assure certain network guarantees so that packets coming from delay or bandwidth sensitive applications receive a better treatment in the network. This approach has been demonstrated to work very well in fixed networks. However, in wireless networks – especially those based on multihop ad-hoc nodes – it is not always possible to offer any guarantee, due to continuously changing conditions and unpredictable radio link consumption. We will demonstrate that in such kind of networks a good QoS level can be achieved if the application tries to adapt itself by reacting to what it senses from the network rather than relying on non-guaranteed bandwidth reservations.

Our approach differs from other solutions in that many other settings like codec, codec-specific parameters (e.g. Quality factor for MJPEG), frame sizes, selection of components, etc. are taken into account. In fact, none of the previous solutions considers video transmission, where these parameters take most of the importance. In addition, we use a novel logic to deal with signaling packet losses due to bad network conditions. In this paper we implement a sub-set of BRENTA [5] functionalities, specifically the application layer QoS signaling architecture.

The reminder of the paper is organized as follows: Section 2 describes the architecture for adaptive applications and our implementation and Section 3 shows some empirical results derived from the use of our adaptive approach in extreme network conditions: a vertical handover scenario and an ad-hoc network

2 Architecture for application adaptation

Quality of Service (QoS) is defined in ITU-T recommendation E.800 [6] as *the collective effect of service performance, which determines the degree of satisfaction of a user of a service*. It is characterized by a combination of service performance factors such as operability, accessibility, retainability and integrity. Placing some additional features in the application layer would allow presenting a better QoS to the user in environments in which traditional solutions would perform badly. The main items in this architecture are shown in Figure 1.

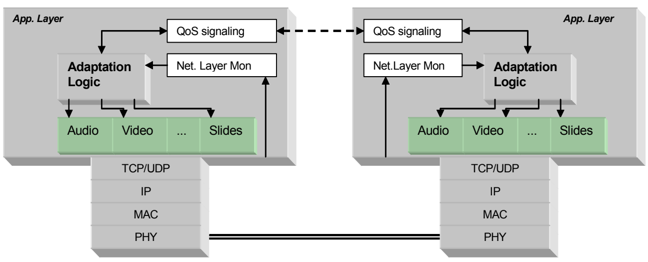

Figure 1. Framework for adaptive applications

The QoS signaling mechanism is the protocol in charge of sending and receiving reports describing the network conditions from the other end. When such a report is received it is passed to the Adaptation Logic as an additional input. Additionally, the Adaptation Logic is in charge of deciding which set of parameters is best suited to the current network conditions.

2.1 Adaptation Logic

The main problem at the application layer with respect to mobile and wireless networks is that of adapting data transmitted to or received from the application to the characteristics of the different networks, including throughput and delay concerns.

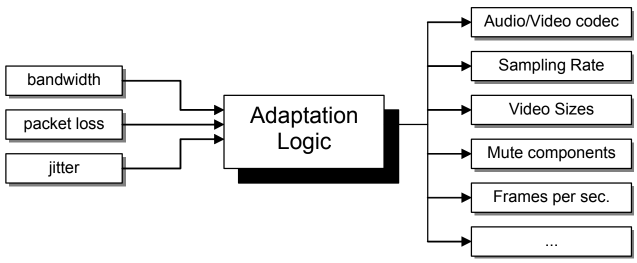

Figure 2. Operation of the adaptation logic

Therefore, as Figure 2 shows, the Adaptation Logic can be seen like a somewhat complex function, where output has to take into consideration the subjective aspects of user preferences. So it seems reasonable – and in fact is being prepared for future work – to apply artificial intelligence techniques like fuzzy clustering to a user-created set of rules to generate a new set of rules in terms of the standard inputs.

Most of the bad effects perceived by the user are due to packet losses, and therefore the most important input for the Adaptation Logic will be the end-to-end percentage of packet losses per reporting period. End-to-end delay problems may also make the user feel a bad quality but this problem can usually be avoided with a proper buffer management without needing to reduce the used bandwidth.

The detailed description of the Adaptation Algorithm which is applied to every QoS report received is presented in Figure 3. As shown, a downgrade in the quality will only be performed when the end-to-end packet losses percentage exceed a 5% or 3 consecutive QoS Reports are lost – possibly by a network congestion or too much interference. Additionally, the quality is upgraded to the estimated bandwidth value whenever 4 consecutive QoS reports indicating 0% packet loss arrive. These parameters have been set according to our own experience. However, they could be tuned according to the testbed trials and they could also be dynamically calculated according to the concrete network scenario. Finally, intelligent techniques like supervised learning could also be applied to calculate these values. The difficulty of applying such techniques derives mainly from the existence of subjective components in the user-perceived QoS which may vary from person to person and which are very difficult to model mathematically.

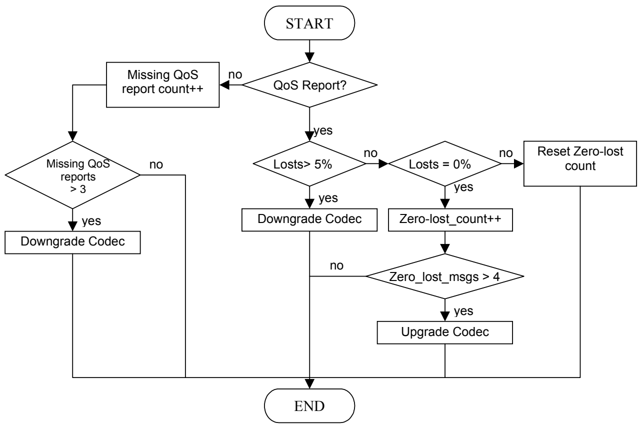

Figure 3. Schematic diagram of the adaptation logic

2.2 QoS Signaling Mechanism

The QoS signaling is another key point of the adaptation architecture as it is the only feedback that the source has from the other end. It is basically an end-to-end transport mechanism for signaling data; no special protocol is needed. In fact, it may be enough with a TCP/UDP socket in both sides, and even a standard protocol like Session Initiation Protocol (SIP).

In our implementation this quality will be measured in terms of percentage of packets lost and mean delay experienced by the data packets in the network. This information is carried in a special signaling packet called “QoS Report”. A QoS report message presents the structure shown in Figure 4.

Type	% Lost	Delay	User Preferences	Estimated BW

Figure 4. Format of a QoS report message

Some experiments performed in [7] demonstrate that a UDP transport is much more appropriate to carry the feedback than a transport using TCP. TCP retransmissions result in stale QoS information especially on a congested network. An additional issue is that the 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. That is, a feedback packet is most important when it carries information about a congested network and it is not important when it is just saying that all is going well. There are at least two approaches for solving that: prioritization of QoS reports, and periodical sending of reports. The first one is related to the idea of giving a higher priority in every router within the intermediate network to these signaling messages. In this way, when intermediate routers have data to send, the will firstly forward signaling messages and then the rest of the data packets. Thus signaling messages are given a higher probability to reach the destination. This mechanism works very well but its main drawbacks are related to its difficult implementation – especially in ad-hoc networks. We have used the second approach which is based on the idea of the clients sending periodic reports towards the sources. This way, whenever network problems come up, the sender can detect missing reports because it is expecting them at periodical intervals. The sender uses the heuristic of downgrading the quality when a certain number of QoS Reports are lost.

2.3 Bandwidth Signaling Mechanism

Apart from the discussed end-to-end QoS signaling, applications can benefit from a local feedback on actual bandwidth capabilities of the network in use. They may not only model the characteristics of its outgoing data flow, but also send information to their peers, referring to the incoming bandwidth limitations.

This signaling mechanism is complementary to the first one in the sense that the locally sensed bandwidth of the network cannot always be assumed to be a good guess on the real bandwidth available. Thus, data streams are not to be shaped only based on bandwidth, but in all the rest of network parameters.

2.4 Adaptation Capabilities

The ISABEL [8] application has been extended to dynamically (and in real-time) adapt its behavior to the available resources. It handles different RTP [9] flows for each service, but our main focus will be in audio and video. The most important adaptation capabilities implemented are as follows:

- **Codecs.** The application may handle contents encoded in several standards such as H.263/MJPEG for video, or GSM/G.722/G.711 for audio, choosing among them at user request or based on information lower from layers.
- **Sampling rate.** Transmitting at a lower frame (or sampling) rate means saving bandwidth, and a fair quality is often achieved at less than 24 fps.
- **Component size.** In scarce bandwidth environments the user will prefer seeing smaller videos than bad quality ones in which most of the frames are lost.
- **Component use.** In very constrained bandwidth scenarios the user may prefer using some components instead of using all of them with a poor quality.

- **Buffering.** Intelligent and dynamically adaptable buffers help offering a better quality in adverse network conditions, lessening the effects of jitter and delay.

3 Results

Experiments have been conducted over two different scenarios, so that the adaptive capabilities of the application have been demonstrated in contrast with the behavior found in usual ones. First scenario deals with a simulated vertical handover, second one with ad-hoc networks. Tests have been done using the audio part of ISABEL, as a sample of a most sensitive service, but could be extrapolated to a case with different flows. Adaptation has been made possible changing the audio codec.

3.1 Handover Scenario

3.1.1 Testbed infrastructure

The testbed (Fig. 5) consists on a number of standard desktop computers that provide all the routing functionalities and mobility agents, and a Laptop, which is used as the mobile node (MN). Network links are mostly Ethernet (10 Mbps), and an ISDN PPP link (64 Kbps), which serves the purpose of limiting the effective bandwidth in one of the branches of the testbed.

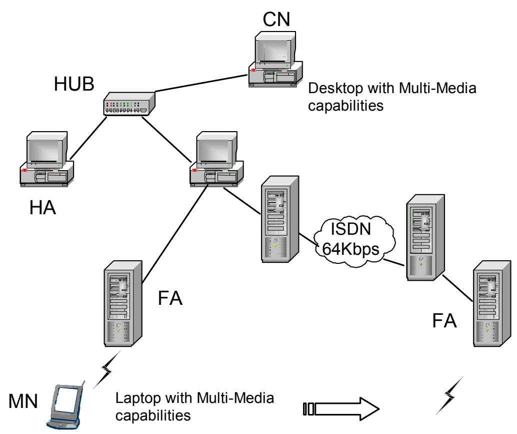

Figure 5. Testbed infrastructure.

The MN is forced to perform a handover between the two lowermost hosts, which act as access points for the two separate networks. This resembles a typical scenario of switching from a broadband network to one with low resources (vertical handover).

All the systems run Linux Kernel 2.4.18, and have dual IPv4/IPv6 stack. Hierarchical Mobile IP (HMIP [10]) has been used in these tests, but results should be quite close when using any other mobility protocol.

Two codecs have been used, one for high bandwidth situations, and the other one for low bandwidth, as shown in Table 1.

Table 1. Used codecs.

Codec	S. Rate	S. Depth	Bit rate (mono)
PCM	8000 Hz	16 bit	~128 Kbps
GSM	8000 Hz	16 bit	~14 Kbps

Switching between one and another codec is done in the basis of a report indicating the actual bandwidth of the network after each handoff.

Data shown in the following tests has been acquired from the RTCP statistics collected at the MN side.

3.1.2 Operation without adaptation

The first set of tests covers the functioning of a simple application with no adaptive capabilities. Figure 6 shows the effect that a handover to the ISDN bandwidth limited network from the broader Ethernet has in the transmission:

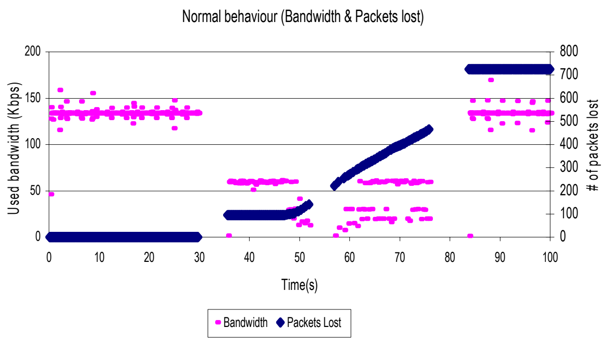

Figure 6. Bandwidth without adaptation

The first handover occurs between seconds 30 – 35, so all the packets are lost in this period. [That latency in the handover is only ascribed to the particular settings in this HMIP installation and does not affect the result of the tests, which do not deal explicitly with mobility procedures]. The many packets in excess after changing to the 64 Kbps ISDN branch are lost (after an initial transitory, while the UDP buffers are being emptied). Packet loss is critical in real-time applications, and the effect is disturbing the audio quality experienced by the end user.

The transitory can be explained through the graphic of Jitter (Fig. 7):

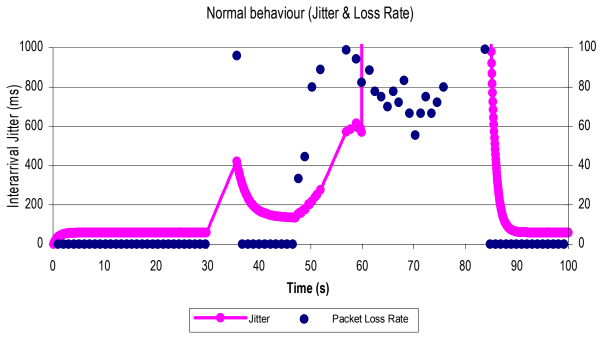

Figure 7. Jitter without adaptation

It is noticeable that the jitter is highly variable during transit in the ISDN network (while being 60 ms before it), being this also related to an elevated packet loss rate; this fact worsens the audio experience.

A second handover takes place between seconds 76-84, switching again to the high bandwidth link and returning to usual values (Fig. 6 and 7).

3.1.3 Operation with adaptation

The adaptive behavior of the application is triggered when it senses, by any means, that actual bandwidth has changed. In the simplest case, which is the one tested, the reaction consists only in the selection of the lowest bandwidth codec available (GSM). Figure 8 shows the expected behavior.

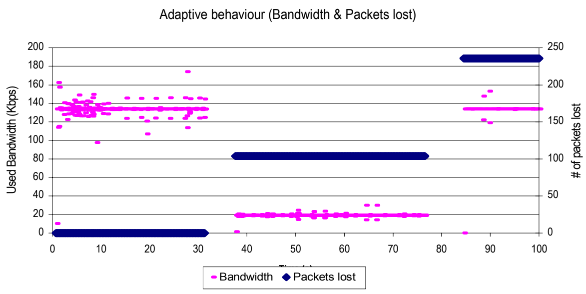

Figure 8. Bandwidth with adaptation

After the handover ends (around second 36 in the graph), the application in informed of the new bandwidth situation. Before the overflow of the UDP buffers, the sender side is told to adapt, and almost immediately the new codec is selected: bandwidth consumption is reduced to less than 20 Kbps, enabling all the packets to go though the ISDN with no packet loss (flat part in the middle of the graph). This way, audio quality is considerably better than in the first case, without adaptation. Jitter is shown in Figure 9.

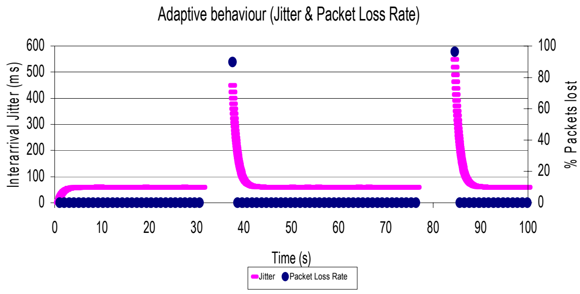

Figure 9. Jitter with adaptation.

Situation is utterly different from that described in the previous section: jitter is constant and limited to 60 ms, except on the point where the handovers take place: all packet traffic is processed in its proper time.

After the second handover (seconds 76-84) situation is reversed, and again the PCM codec is used. Audio quality is ensured by the high bandwidth Ethernet.

3.2 Multihop Ad-hoc Network

To demonstrate how adaptive multimedia applications can improve the user-perceived QoS in the case of ad-hoc networks, we have deployed a network testbed in which the source and destination of multimedia data are connected by a 1-hop ad-hoc network. As Figure 10 shows, an additional host is introduced in the same radio range in order to generate interferences – both by taking part of the bandwidth and also accessing to the shared media. The idea is not to simulate a real environment but to test the application reaction under different interference load. Thus, after interference periods of 100s we introduce periods of 50s without interference so that the different tests can start from the same initial conditions.

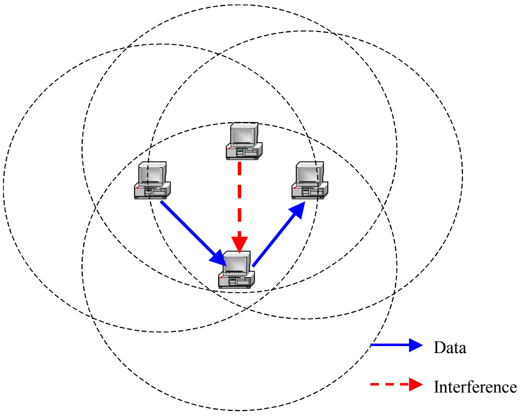

Figure 10. Multihop ad-hoc network

A traditional application in such kind of environment starts losing packets as the interference periods are having a higher load (Fig. 11). When adaptive behavior is used, the overall number of packets is strongly reduced in a 66% (from 6981 packets to 2387). The packet losses which are produced in the adaptation part are due to the losses not exceeding the loss threshold (5%) that the application requires to trigger the adaptation process. As we show in the next figures, these small losses are not enough for damaging the user perception of the audio because the loss percentage in most of the cases is maintained below the 20% which many studies [1] have identified as the point from which the audio quality starts getting noticeably poor.

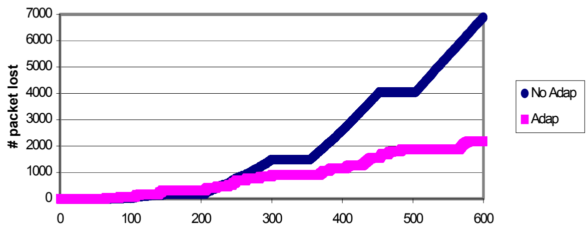

Figure 11. Packet losses with and without adaptation

If we observe the variation of the instantaneous packet loss percentage over time – which is shown in Figure 12– we can see that, as expected, the adaptive application tends to use the lower-bandwidth codecs in highly loaded periods (350-450 sec and 500-600 sec). This way, even in such heavy loaded situations, most of the packet loss percentages are successfully kept at 0%.

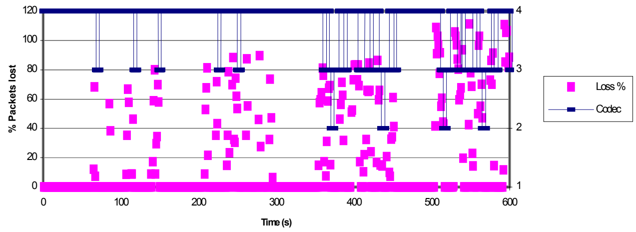

Figure 12. Losses distribution and codec changes

These results are also supported by the figures of the packet loss distribution during time. Its mean value has been calculated for both approaches and we have found that using adaptive applications, the mean value is 14% versus the 34% in the case of no adaptation. However, this value is not really significant if it is not taken in conjunction with the distribution. As we present in Figure 13, in the period of higher load (500 to 600 sec), when no adaptation is used, although the mean value is 34%, most of the values are really in the range 75%-100%. However, in our case although the mean value is 14%, most of the values are in the range 0%-14% being 0% the most repeated value. This demonstrates that except for very punctual peaks, the adaptive approach offer a good quality audio (under 20% losses) in situations in which traditional applications are unable to offer less than 75% packet losses.

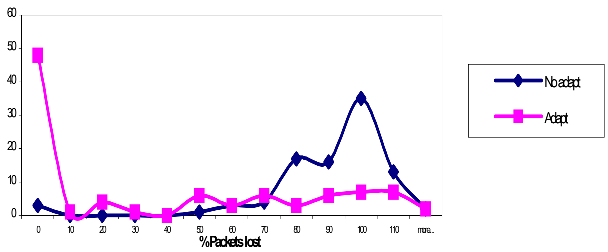

Figure 13. Packet loss distribution with high interference

4 Conclusions and future work

We have defined and implemented an adaptive multimedia application framework for mobile and wireless applications. Its performance has been shown in wireless routing scenarios. Through a sample testbed and some experiments, this paper has demonstrated the benefits that such a technology can provide to enhance the experience of an end user in a typical changing conditions environment, ensuring an adequate service level even in dramatic bandwidth changes. It has been shown how applications can help in offering a good QoS even in situations in which the network is not able to guarantee anything.

As a future work, improvements in reaction are related to the use of intelligent techniques for the selection of the application parameters which better suit the current network conditions. In addition, total integration with lower layer QoS control facilitates the final objective to reach a quite fair level of service quality.

5 Acknowledgements

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