

Seamless Multimedia Communications in Heterogeneous Mobile Access Networks

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Abstract

With the advent of the new wireless technologies allowing for higher data rates, the provision of high quality multimedia services over the upcoming wireless and mobile networks is gaining momentum. These future mobile networks are witnessed to consist of a variety of heterogeneous wireless access technologies connected to a high-speed IP datagram-based core network. The heterogeneity of these networks in terms of network capacity and terminal capabilities among others, is expected to cause an unpredictably capacity-changing network scenario. In these scenarios, the network-layer QoS mechanisms cannot guarantee a stable service because most of the variability is due to the wireless channel itself. Thus, the use of adaptive applications being able to change their settings to adapt to the available network resources is a must. We introduce a new paradigm of user-aware adaptive applications being able to intelligently maintain a good user-perceived QoS even when the network conditions are highly variable. We will show empirical results which demonstrate that the proposed approach is able to outperform traditional multimedia internetworking applications even in extremely variable scenarios such as ad hoc networks.

Keywords: Adaptive applications, Intelligent adaptation, Soft-QoS, User-perceived QoS

1. Introduction

Adaptive applications and services are a key asset for application and service developers to fulfil the new requirements of future heterogeneous networking environments, which are envisioned as consisting of a variety of heterogeneous wireless technologies attached to a high-speed core IP-based network. The dynamic selection of the settings used by multimedia applications during a multimedia session can help at providing a richer user experience and a higher user enjoyment. For instance, a negligible reduction on the coding quality or the frame rate can avoid the perceptual discontent that packet losses produce to the user. We believe that the assessment of the effectiveness of a mobile multimedia session would be hindered if it did not include the quality with which the users perceive the multimedia content. This view is also supported by the definition of Quality of Service (QoS) given by the ITU-T in the recommendation ITU-E.800 [13].

A large body of literature has been devoted to the use of adaptive applications both in wired and wireless networks. Most of it is to do with the dynamic adaptation of the applications' data rates to those supported by the underlying network. For instance, Bolot[1] and Sisalem[2] use adaptive applications to avoid congestion in fixed networks. However, provided that in wireless and mobile networks packet losses are not only due to congestion, new proposals ([3], [4], [5], [6], [7]) came up particularly addressing wireless and mobile networks. However, these papers have usually limited their studies to a reduced number of application settings, without performing a broad analysis of multidimensional adaptations (i.e. those considering the tuning of many settings such as frame rate, video sizes, audio codecs, etc. simultaneously). Moreover, they have almost completely neglected the subjective user-perceived QoS when designing the adaptation of the application-layer data-rates to those instantaneously supported by the network.

We introduce a new category of adaptive applications, which we call user-aware adaptive applications, in which the adaptation logic is driven by the user-perceived QoS. That is, given concrete network conditions, the application will select when needed a new combination of settings (e.g. frame rate, video size, etc.) producing the highest user satisfaction. This approach goes beyond existing works by means of not using low-level QoS metrics such as bandwidth estimations, and packet losses, which do not convey information about the real quality of perception or the user satisfaction regarding the multimedia content. In order to support this approach adaptive multimedia applications need to be challenged with an enhanced adaptation middleware being able to assess the instantaneous user-perceived QoS. Key to this is the ability for the adaptation function to model the subjective components involved in the human perception of multimedia content.

Little work has been reported on the modelling of satisfaction and perception of the user for multimedia content mainly due to the difficulty of modelling the subjective components. Previous work has focused on the evaluation of the individual impact of some of the application settings in the overall quality perception.

Accordingly, Apteker[8] showed a non-linear relation between varying frame rates and the user-perceived QoS. Wijesekera[9] analysed the effect that random packet losses have on the perceived quality, illustrating the high differences in acceptable loss rates for different media.

Further work has been undertaken by Steinmetz[10] who explored the bounds up to which lip synchronisation does not produce annoyance. Finally, Ghinea[11] evaluated the impact of cognitive styles on the QoS perception. However, these works have not successfully produced a model for the user-perceived QoS. We have used the SLIPPER[12] rule induction algorithm to generate a set of rules which represents the quality that the user perceives. We show that the adaptation mechanism used by the user-aware multimedia application delivers a higher multimedia quality when it uses the rules obtained by SLIPPER.

The remainder of the paper is organized as follows: section 2 explains the process of modelling of the user perception using the rule induction algorithm. Section 3 describes the architecture for user-aware adaptive applications we have developed. Finally, section 4 shows some empirical results derived from the use of our adaptive application approach over an ad hoc network, which is one of the most challenging scenarios regarding the variability of the network conditions.

2. Modelling the user-perceived QoS

There are many combinations of settings which can result in similar data rates, but very different user-perceptual QoS even when the network conditions are optimal. For instance, a user may prefer a bigger video transmitted at 6 fps and encoded using less quantization than a smaller one at 12 fps using a high quantization even when the latter requires more bandwidth.

Taking advantage of the perceptual preferences when adapting multimedia applications requires the characterization of the subjective components which define a user perception of quality as well as its relation with the different parameters which define the behaviour of the real-time multimedia application (e.g. codecs, rates, etc.). Given the difficulty to model the user-perceived QoS analytically, we have used the SLIPPER rule induction algorithm to generate a set of rules which models the user-perceived QoS. These rules have been extracted applying the algorithm to a large number of learning examples which have been evaluated and scored by real users. SLIPPER was mainly selected because it is well-known to outperform other rule extraction algorithms.

The SLIPPER algorithm takes as an input a set of tuples with evaluations that the users have made about a concrete combination of settings in a concrete network scenario, and generates a set of rules which models the QoS perception of the user. Each tuple contains the network bandwidth, the audio and video codecs, the quantization of the video codec, the frame rate used, the video size, the loss-rate and the user's score in the interval $[1,5]$ (as recommended by ITU

recommendation ITU-P.800 [14]). In order to produce all this input tuples, we have set up a scenario using our adaptive video conference application, over a link emulator which emulates different network conditions. A total of 864 different combinations of applications settings (in the ranges shown in table 1) have been tested, and scored by users. We then use the SLIPPER algorithm to extract a set of rules representing the user-perceived QoS. This process is illustrated in figure 1, and described in detail in the remainder of this section.

Parameter	Values	Explanation
BW	{33,...,384}	Bandwidth Limit
LOSS	0..100	Packet loss rate
AUDCOD	PCM, G711-u, G722, GSM	Audio Codec
VIDCOD	MJPEG, H.263	Video Codec
FSIZE	CIF, QCIF, 160x128	Size of video frames
QFVIDEO	5, 10, 15, 30, 60	Quantization of video codec
FPS	{0,...,12}	Frames per second
QoS	1, 2, 3, 4, 5	User-perceived QoS

Table 1: Attributes and values to learn the user-perceived QoS

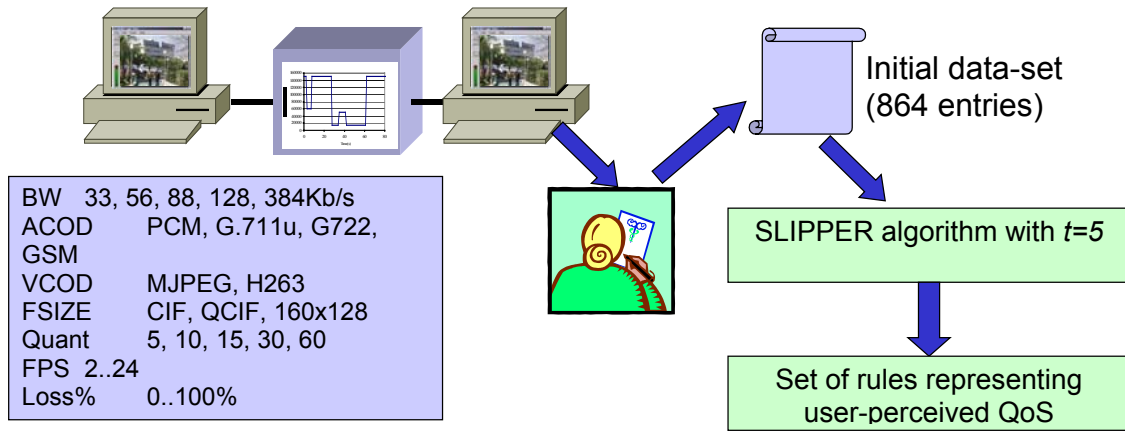


Figure 1: Illustration of the overall learning process

Our goal is to obtain a model of the users' evaluation of QoS, let it be denoted with ξ , representing how they perceive the quality of the multimedia service. SLIPPER produces an approximation ξ' for that user perception. It can be defined, as shown in equation (1), as a function producing a MOS-like score taking as inputs the specific network conditions and the particular settings that the multimedia application is using.

$$\xi': BW \times AudCod \times VidCod \times Size \times Qual \times FPS \times Loss \rightarrow QoS \quad (1)$$

The SLIPPER algorithm makes use of a configuration parameter, noted as t which corresponds to successive refinements of the rule set. The higher the value, the more accurate the model and the bigger its complexity (in number of resulting rules). We applied SLIPPER to the aforementioned data set, using values of $t \in [1, 15]$. In figure 2 we have represented the evolution of the classification error during learning and during testing and the complexity of the rule set in number of rules. It is clear that there is a trade-off between classification error and complexity.

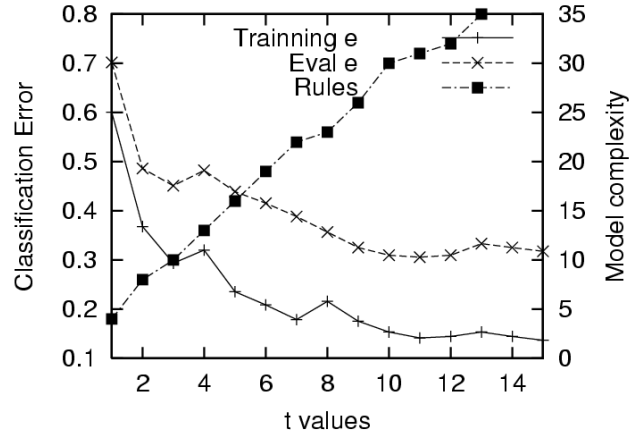


Figure 2: Trade-off between classification error and complexity

From our point of view, the model with $t=5$, represents an example of a good trade-off consisting of a total of 12 rules. This configuration produces following rule set, in which the resulting QoS score is obtained as the sum of the output of all the rules whose left-hand antecedent is met:

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if matchConfidence {
  [QFVIDEO >= 60, VIDCOD = MJPEG, FSIZE = QCIF, LOSS <= 10, FPS >= 6] -> 2.8792
  [AUDCOD = GSM, BW >= 80, QFVIDEO >= 30, FSIZE = QCIF, FPS <= 6] -> 1.4357
  [AUDCOD = GSM, BW >= 128, LOSS = 0, QFVIDEO >= 30, FPS >= 3, VIDCOD = MJPEG] -> 1.7013
  [] -> -2.4188
} > 0 then 5 else if matchConfidence {
  [BW >= 384, QFVIDEO >= 40, FSIZE <= 2] -> 2.7121
  [QFVIDEO >= 30, VIDCOD = MJPEG, LOSS <= 3, AUDCOD = G722, BW >= 80] -> 1.1756
  [FSIZE = CIF, QFVIDEO >= 30, LOSS <= 3, AUDCOD = G722, BW >= 80] -> 1.4437
  [] -> -1.5044
} > 0 then 4 else if matchConfidence {
  [LOSS >= 30] -> 2.1188
  [QFVIDEO <= 5] -> 1.4142
  [LOSS >= 16, FPS <= 3] -> 1.5438
  [] -> -1.0984207275826066
} > 0 then 1 else if matchConfidence {
  [LOSS >= 16] -> 1.9109
  [QFVIDEO <= 10, FSIZE = QCIF] -> 1.5861
  [FSIZE = 160X128, QFVIDEO <= 40, VIDCOD = H.263] -> 1.2546
  [] -> -0.3953
} > 0 then 2 else 3

```

The rule set extracted by SLIPPER allows us to assess how different combinations of settings impact on the user-perception of QoS. In order to get the evaluation of the current settings the application just needs to sum the value of all the consequents of all the rules whose antecedent is matched. For instance, the rules imply that the higher the frame rate the better the quality, but the user prefers changing from a higher frame rate to a lower one (never below 3 fps), provided that the video size is increased. Of course, these are just examples of QoS preferences which can be extracted from this rule-set. However, the important point, is that these rules can be used by any multimedia application to assess the user's satisfaction for a particular combination of network and application parameters. In fact, these rules are used by the user-aware adaptation logic presented in the next section.

3. User-aware adaptive applications

Traditional multimedia applications have very limited adaptation capabilities and usually they cannot adapt at all. In order to support user-aware adaptive applications we add some components to the architecture of a traditional real-time multimedia application. These new components are in charge of the signalling of QoS information, and the provision of the

intelligence to keep the user-perceived QoS at an acceptable level. So, the overall architecture, as shown in figure 3, consists of a set of multimedia components (e.g. audio, video, etc.), a QoS signalling mechanism and an adaptation logic which is in charge of taking the adaptation decisions. We describe these components in the following subsections.

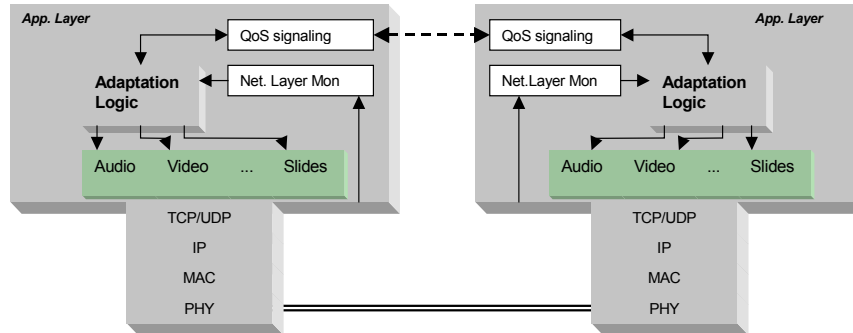


Figure 3: Architecture of the adaptive application

3.1. Components of the multimedia application

Real-time multimedia applications usually consist of a number of multimedia components dealing with different types of data such as audio, video, white boards, etc. In our case, we will focus our work on the videoconference service consisting of audio and video components. The implemented extensions allow the adaptation logic to choose in real time different audio (GSM, G.722, G.711 and PCM) and video (MPEG, H.261, H.263 and MJPEG) codecs, and dynamically change the video frame rate (0..25 fps), the video size and the quantization factor of the video codec.

3.2. QoS signalling mechanism

The QoS signalling is a key element of the adaptation architecture as it is the only feedback that the source has from the other end. The received QoS information is used by the adaptation logic to make the decision on which re-configuration of the application settings to perform. As we target heterogeneous networking scenarios, we use an end-to-end feedback mechanism. The QoS signalling module computes the loss-rate and mean delay experienced by the data packets in the network. This information is carried in a special signalling packet, called "QoS Report", which is sent back to the source. A sequence number is used to deal with delayed QoS Reports. The rate at which QoS Reports are sent has to represent a good trade-off between highly dynamic adaptation and a reduced traffic load. At the moment in our implementation it is fixed at one QoS report per second. However the use of adaptive rates is being investigated.

To overcome from losses of signalling messages one can consider among other alternatives: (i) prioritising QoS reports or (ii) allowing the applications to detect such losses. As the former approach requires a tight coupling with the network layer, we have found more appropriate for heterogeneous networks to make the receivers send periodic reports towards the sources. In this way, whenever network problems come up, the adaptation logic at the sender can detect missing reports. This information can be used by the adaptation logic to implement some heuristics based on that information (e.g. downgrading the bandwidth consumption when a certain number of QoS Reports is lost).

3.3. User-aware adaptation logic

The adaptation logic can be seen as a function which uses the QoS Reports and additional local information to decide which settings need to be configured in each of the different multimedia

components. The adaptation logic solves the problem of adapting multimedia flows to the characteristics of the different networks or terminals.

Traditional adaptation logics not considering the user-perceived QoS, are usually driven by fixed thresholds like the loss rate to reduce the data-rates, the number of consecutive 0% QoS reports before trying to increase the data rates, etc. In our case, instead of having fixed thresholds based on network-layer parameters, the user-aware adaptation logic will use QoS perceptual thresholds. Therefore, application reconfigurations will only occur whenever the user-aware adaptation logic detects that the user-perceived QoS is not good enough.

If that happens, the application logic selects from all the feasible configurations of the multimedia components the ones which produce the best user-perceived QoS with those network conditions. Finding such an optimal set of settings requires the adaptation logic to be able to assess the perceptual QoS using the rules shown in the previous subsection.

4. Empirical results

In order to evaluate the effectiveness of our proposal, we have set up a real wireless ad hoc testbed, on which we have compared the performance of real-time videoconferencing both with traditional applications and with user-aware adaptive applications. The testbed has been deployed in the basement of the CS Faculty at the Univ. of Murcia (see Figure 4).

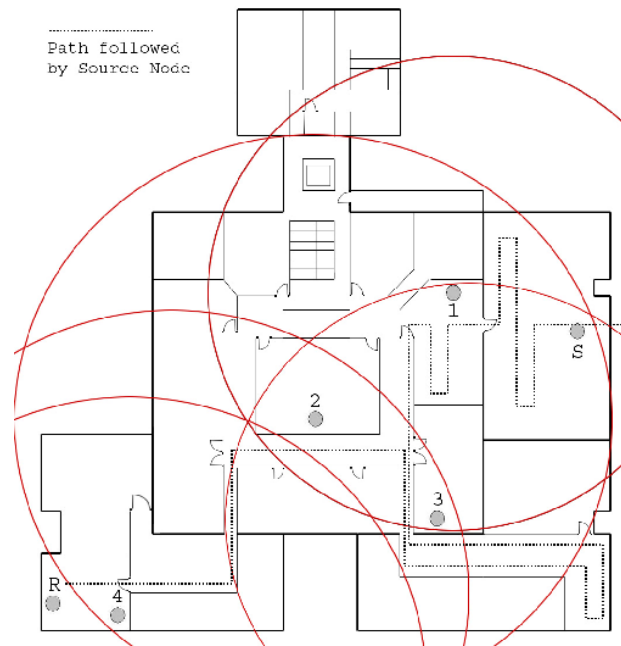


Figure 4: Architecture of the adaptive application

We use the Multicast MANET Ad hoc Routing Protocol (MMARP[6]). The source (S) follows (at walking speed) the path which is shown in the next figure while it runs a videoconference session with node (R). The trials have been performed using the extended adaptive videoconferencing application described in the previous subsection.

The results which we present are extracted from the RTP traces which are generated by the videoconferencing application. We have used the same route, at the same speed in the same network conditions for the adaptive and non-adaptive trials. The path followed by the node R has been specifically selected so that link breaks and MMARP route changes take place during the videoconferencing session. Furthermore, there were changes in signal strength due to the variation of the distance to MMARP nodes and the number of intermediate walls to traverse. This made the available bandwidth vary during the session.

The results presented in figure 5 show that the use of adaptive applications is able to reduce the overall packet losses both for audio and video to approximately one third. As it was expected, the differences are higher in the periods in which there is less bandwidth available.

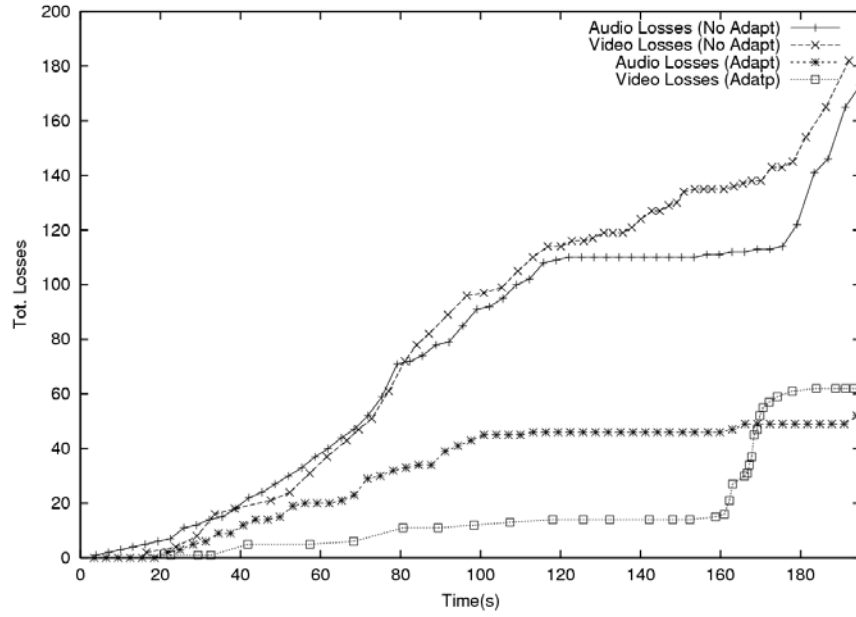


Figure 5: Total losses during the trial

This is also noticed in the variation of the delays depicted in figure 6. During the same critical periods, the non-adaptive approach is not able to control the growing of the end-to-end delay, whereas the adaptive one is able to quickly restore the original steady-state.

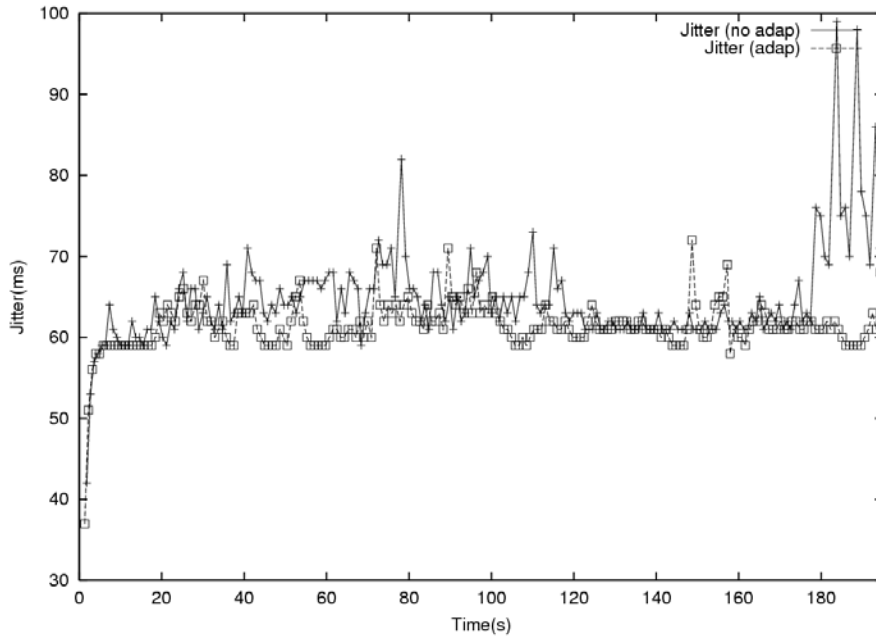


Figure 6: Jitter measurements over the trials

The overall packet losses is a good reference to identify the points of the trial in which the network conditions are most critical. This is identified by an increase in the slope of the total packet loss curve. However, what really affects the user perception of QoS is the instantaneous loss-rate, which is what causes most of the user's annoyance.

As long as the instantaneous loss-rate is the responsible for most of the user's annoyance, in figure 7, we compare the statistical histogram for the distribution of the audio loss-rate for both approaches.

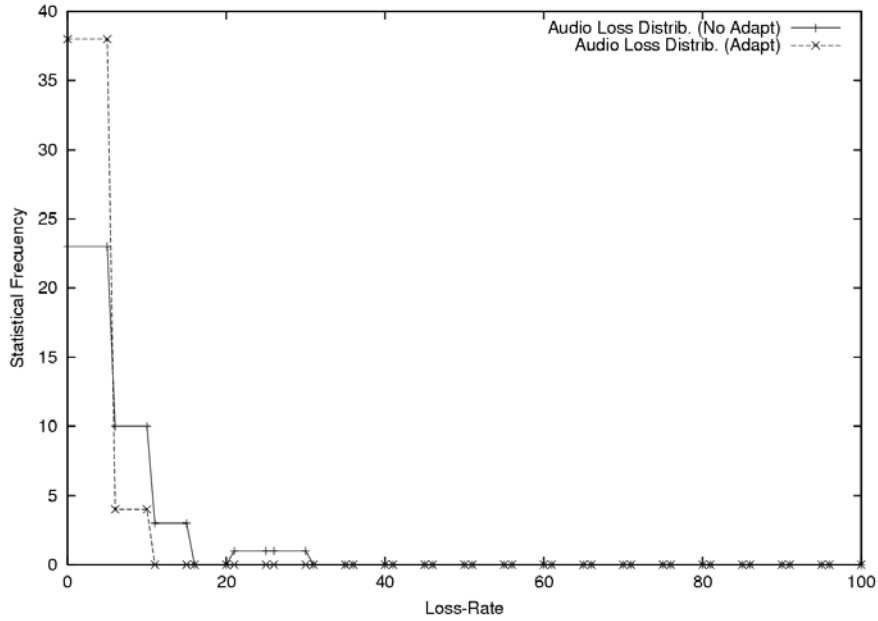


Figure 7: Loss-rate histogram for the audio session

The same statistical analysis is performed for the video flow in figure 8. For the audio flow, the adaptive application approach is able to keep the loss-rate below 10% all the time. In fact, it keeps the loss-rate below 5% during the 91% of the time. For the video flow, the loss-rate is kept under the 5% the 64% of the time, and its has been under the 10% the 78% of the time.

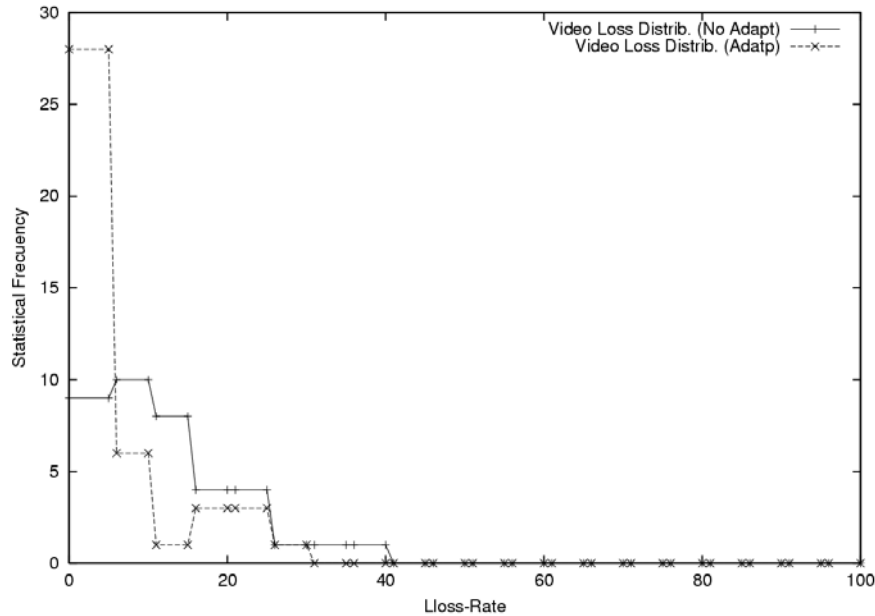


Figure 8: Loss-rate histogram for the video session

Although the previous two figures demonstrate that the user-perceived QoS is improved when using our novel machine learning driven adaptation, we have conducted an additional evaluation

test with real users giving scores of their QoS perception during these trials as well as an overall session score. As we did in the generation of the learning examples, we have tried to follow the MOS ITU-recommendation as much as possible. Eight different users have given their QoS evaluations with scores in between 1 and 5. The higher the score the better the quality (i.e., 1=poor, 2=bad, 3=fair, 4=good, 5=excellent). Figure 9 shows the mean evaluation from these 8 users in different points of the path. The figure also depicts the mean of all the overall QoS perception values. These points are identified by the elapsed time since the starting. These specific moments in time in which we asked for the evaluation were specifically selected in points in which the network conditions were most critical.

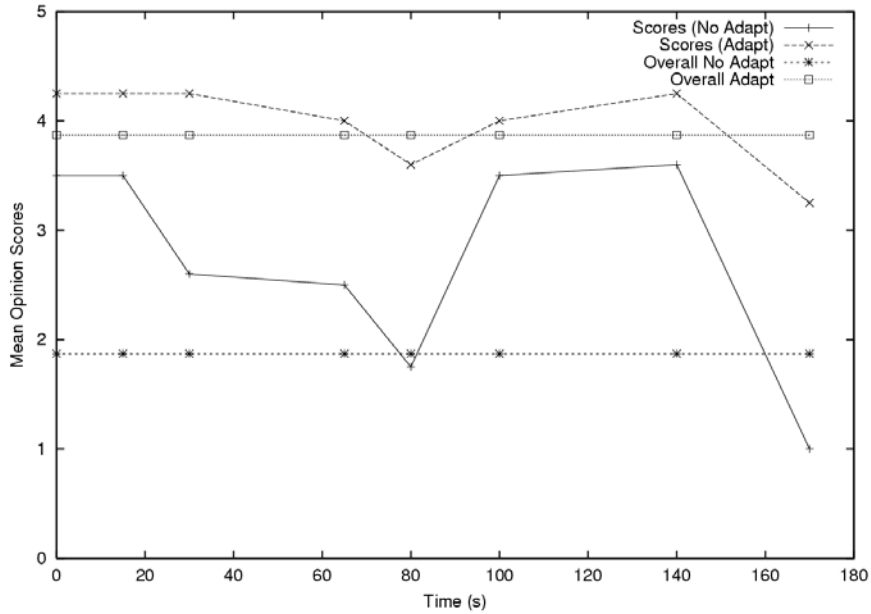


Figure 9: MOS scores given by real users

As it is depicted in the figure, the user-aware adaptation logic clearly outperforms the traditional multimedia adaptation because the proposed approach is able to maintain a good QoS level even when the network resources are extremely scarce and variable. As can be observed in the previous figure, the mean overall perception for our adaptive approach is around 3.87 which means that most of the users scored the overall QoS as good. However, the non-adaptive approach has scored in average 1.87, which means that most of the users scored the overall QoS as bad.

5. Conclusions and future work

We have proposed a novel machine learning driven adaptation approach for real-time adaptive multimedia applications. This approach, which we call user-aware adaptive applications, differs from other analogous systems as it takes into account the instantaneous user-perceived QoS when taking the decisions about when and how to adapt the multimedia applications to the underlying networking environment. Our proposed user-aware adaptation logic has demonstrated to be effective compared to traditional multimedia applications when used in real networks. These results are clearly supported by the evaluations which the users have done on the overall performance in the wireless ad hoc testbed.

As short-term future work, we are extending the proposed adaptation logic with reinforcement learning modules. The goal is not only to generate a rule-based model of

the user, but to be able to automatically enhance it over time according to the user's profile and scores.

6. Acknowledgements

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8. Vitae

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