# PQoS-driven VoIP Service Adaptation in UMTS Networks

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## ABSTRACT

This paper shows the relevance of implementing a dynamic and real-time Voice over IP (VoIP) service adaptation mechanism in order to avoid Perceived Quality of Service (PQoS) degradations. We have studied the most relevant parameters affecting the PQoS experienced by end users in VoIP services over mobile accesses. We will describe the enhancements carried out in the implementation of the simulation model in order to assess VoIP PQoS based on ITU-T E-model and introduce a real-time service adaptation method. This adaptation method involves codec and packetization changes through SIP signalling, jointly to dejittering buffer management, and would be launched upon detection of PQoS degradation, in order to enhance end users' experience.

## Categories and Subject Descriptors

C.2.1 [**Computer Communication Networks**]: Network Architecture and Design – *packet-switching networks, wireless communication.*

## General Terms

Management, Measurement, Performance, Design

## Keywords

VoIP, perceived QoS, E-model, SIP signalling, real-time adaptation

## 1. INTRODUCTION

Voice over IP (VoIP) has become one of the recent key topics for telecommunications industry. One of the main factors to burst VoIP services growth is related to its capability to provide analogue telephony-grade quality at much lower cost in those scenarios where there exist equivalent voice services (i.e. 3G networks). So, it is essential to take into account the Perceived Quality of Service (PQoS) experienced by end users, in order to efficiently design and manage the heterogeneous VoIP network environments previously mentioned.

This paper focuses on the study of the main aspects and parameters which could affect VoIP PQoS, and their relationship with the tunable parameters of the VoIP service, in order to propose an adaptation method which will improve end users' PQoS.

The rest of the paper is organized as follows: Section 2 reviews the VoIP services, focusing on the most accepted encoding scheme for this kind of services and their impact on the protocol overhead. It also reviews related SIP signalling protocol used for the establishment, management and termination of VoIP sessions.

Section 3 establishes the relationship between those network parameters that could affect PQoS and those service parameters which can be modified in order to avoid PQoS degradation. For that purpose, a review of the E-model proposed by the ITU-T in [1] to assess the PQoS is previously introduced.

The proposed study is carried out through different simulations over a single scenario which is described in Section 4. This section also presents some enhancements added to the simulation tool in order to correctly assess the necessary parameters to develop the present study. This section ends with a simple service adaptation method, based on encoding scheme, packetization or dejittering buffer size modification, using SIP signalling protocol, which will lead to a PQoS increase.

Section 5 shows the results obtained for different simulations, while Section 6 illustrates the relevance of implementing a dynamic service adaptation mechanism that avoids the PQoS degradation depending on the network status.


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## 2. VoIP SERVICES

This section describes the main characteristics of the VoIP service provisioning considered for the study. First, we overview the relationship between the codec performance, the voice packetization scheme and the amount of required resources, which is especially relevant for limited capacity systems such as UMTS. Afterwards, we review the main features of the signaling protocols used for session control and adaptation.

### 2.1 AMR Codec

The VoIP service typically reduces the data rate required by voice speech using data compression techniques and encapsulates the voice frames into IP packets.

Among all the available encoding schemes, Adaptive Multi-Rate (AMR) is proposed by the 3GPP as the candidate for VoIP over 3G services [2], since it offers a good quality at low bitrate demands and real-time adaptation capabilities. The AMR codec may work on eight modes at different bitrates, from 4.75 kbps to 12.2 kbps, whose performance has been extensively studied for 3G access networks in [3].

However, the protocol stack introduces a significant overhead in the communication. Following the protocol stack for 3G conversational applications, the voice frame is the payload of a RTP/UDP/IP header, so it results in 40 bytes (12 bytes RTP, 8 bytes UDP and 20 bytes IP) of header overhead. This overhead implies a notable waste of resources, which may be critical for mobile accesses.

The main alternatives to reduce this protocol overhead are header compression and increasing the packetization scheme (number of voice frames per packet). Since this paper focuses on end-to-end real-time service adaptation in a 3G scenario, we will focus on packetization, which shall be increased if the bandwidth limitation occurs within the core network.

Figure 1 illustrates the data rates obtained for different packetization schemes for each AMR mode. The drawback of increasing the packetization factor is that voice frames must be buffered at the sender until all the voice frames sharing an IP packet are generated, and thus introducing additional delays.

### 2.2 SIP Signalling Protocol

Session Initiation Protocol (SIP) [5] has become the main signalling protocol in VoIP services. It is used for session establishment and management and is based on request-response messages. SIP is an end-to-end oriented protocol which means that all the logic and the session information is stored in end users' devices. However, at least one proxy/register is usually needed in order to correctly route SIP messages.

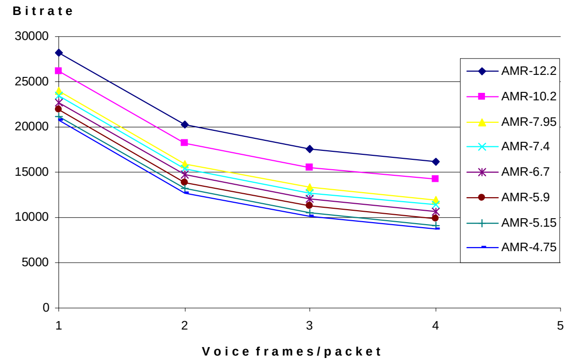


**Figure 1. Bitrate vs. packetization**

The session information contained in SIP headers is just the necessary to exchange the messages, that is why it is commonly used together with the Session Description Protocol (SDP) [6]. SDP is used for real-time session descriptions, which allow the announcement and negotiation of multimedia session capacities.

Although these protocols are commonly used in the initial part of a session establishment in order to agree on the session characteristics, they can be also useful for session re-negotiations. In our approach, the required information for adaptation mechanism will be included into the SDP information. The exchanges of this SDP information will be made through the SIP re-INVITE method.

Some authors [7] propose carrying out inband adaptation procedures, this is, signalling is transmitted in the RTP flow. But, this type of inband signalling is still experimental and under study. Therefore, we have chosen SIP and SDP protocols because they are commonly used signalling protocols and because they offer IMS compliance, which could be useful for Next Generation Networks (NGN).

## 3. PQoS OF 3G VoIP SERVICES

We consider the PQoS as the main driver for end-to-end service adaptation. Therefore, we must first take into account how to dynamically assess the quality experienced by end users. The considered E-model allows relating the end-to-end performance metrics to the service levels. After providing a brief review of the ITU-T E-model, we focus on the study of the end-to-end performance metrics and the main sources of impairments that may degrade the PQoS.

### 3.1 E-model

The predominant method for assessing the PQoS of VoIP services is the E-model, defined by the ITU-T in [1]. This model gives the value of a scalar quality rating value, R, which provides an evaluation of the communication impairment, taking into account most of the possible sources of impairment.

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$$R = Ro - Is - Id - Ie\text{-}eff + A \quad (1)$$

Ro is the basic signal-to-noise ratio. Is represents a combination of all impairments which occur simultaneously with the voice signal. Id is the factor due to all delay impairments. Ie-eff represents impairments caused by low bit rate encoding schemes and packet losses. Finally, A is defined as an advantage factor for compensation in certain service scenarios.

The relationship between R and the Mean Opinion Score ($MOS_{CQE}$) is determined by (2):

$$\mathrm{MOS_{CQE}} = \begin{cases} 1 & \mathrm{R} < 0 \\ 1 + 0.035R + R(R-60)(100-R)7 \cdot 10^{-6} & 0 < \mathrm{R} < 100 \\ 4.5 & \mathrm{R} > 100 \end{cases} \quad (2)$$

When the R score is computed with default values proposed in [1] it takes a 93.2 value and corresponds to an estimated MOS score of $MOS_{CQE}$=4.41, which means a high end users' satisfaction.

Among all the impairment factors in (1), we can point out the Id and Ie-eff factors as the main sources of impairment due to the provision of VoIP over UMTS Packet Switching services. As previously mentioned, Id is defined as the impairment due to delay effects. If perfect echo cancellation is considered, Id is determined by the total one-way delay at application layer (Ta) as in (3):

$$Id = \begin{cases} 0 & \mathrm{Ta} \le 100\mathrm{ms} \\ 25\left\{\left(1+X^{6}\right)^{\frac{1}{6}} - 3\left(1+\left[\frac{X}{3}\right]^{6}\right)^{\frac{1}{6}} + 2\right\},\ X = \dfrac{\log\left(\frac{Ta}{100}\right)}{\log 2} & \mathrm{Ta} > 100\mathrm{ms} \end{cases} \quad (3)$$

Ie-eff is defined as the impairment factor due to the combined effect of the encoding scheme and packet losses. In (4), Ie is the equipment impairment factor, and reflects the degradation due to codification. Thus, it is a codec-dependant parameter. When there are packet losses, the consequent degradation is determined by the packet loss ratio (Ppl), the codec-dependant Packet-loss Robustness Factor (Bpl), which reflects the resiliency of a codec to losses, and by the Burst Ratio (BurstR), which is determined by the bursts intensity of experienced losses. The Ie and Bpl values for AMR-12.2 mode (compatible with GSM EFR) can be set to 5 and 10 respectively [8]. But it should be taken into account that if another AMR mode is selected, the correspondent Ie and Bpl values should be introduced into (4).

$$Ie\text{-}eff = Ie + (95 - Ie) \cdot \frac{Ppl}{\frac{Ppl}{BurstR} + Bpl} \quad (4)$$

## 3.2 VoIP and PQoS Parameters Relationship

For this study, we consider a scenario where a mobile terminal sends voice traffic using the UMTS Background class, selecting the Acknowledged Mode (AM) among the available Radio Link Control (RLC) modes. The other endpoint is considered to be directly connected to the GGSN through a wire connection.

RLC AM provides typical ARQ error recovery method to assure the correct delivery of Service Data Units (SDUs). Although AM recovery mechanisms are faster than TCP's (and therefore provides a better performance with TCP applications), it may not be so advantageous for UDP real-time applications such as VoIP since recovered frames show much higher delays. Furthermore, it could be possible that those recovered packets do not arrive in time to the end user and would be lost anyway. A method for solving this problem could be an increase of the dejittering buffer size, but it also affects the end-to-end delay. So it is necessary to reach to an optimal trade-off between packet losses and end-to-end delay.

With this configuration, the end-to-end delay can be calculated as (5): [4]

$$\mathrm{Ta} = \mathrm{d_{cod}} + \mathrm{d_{decod}} + \mathrm{d_{pack}} + \mathrm{d_{UTRAN}} + \mathrm{d_{CN}} + \mathrm{d_{jit}} \quad (5)$$

$d_{cod}$ and $d_{decod}$ are the codification and decodification delays. $d_{pack}$ is the delay introduced by the packetization scheme. $d_{UTRAN}$ is defined as the delay introduced in the UMTS Terrestrial Radio Access Network (UTRAN), which includes the transmission delay and the additional delay due to RLC functions, such as the delay introduced by the recovery mechanism of the RLC AM. $d_{CN}$ is the delay introduced by the transport of packets through the UMTS Core Network, which may include transmission and queuing delays. $d_{jit}$ is the delay introduced by the dejittering buffer, which depends on the buffer size.

The total Ppl and BurstR values are determined by the individual contributions of each packet loss ratio due to the different sources of losses.

$$\mathrm{Ppl} = f\ \{\ \rho_{UTRAN},\ \rho_{CN},\ \rho_{jit}\} \quad (6)$$

$\rho_{UTRAN}$ is the packet loss ratio caused by packet lost in the UTRAN due to Discard Timer expiration. $\rho_{CN}$ is the packet loss ratio through the core network, mainly due to congestion effects. $\rho_{jit}$ is the packet loss ratio experienced in the dejittering buffer, due to voice frames arriving later than its play out time.

In the general case, while $\rho_{netw}$ depends only on core network dimensioning, both $\rho_{UTRAN}$ and $\rho_{jit}$ are dependant on the values configured for the SDU Discard Timer and the dejittering buffer size, as well as on the end-to-end

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delays (Ta), which at the same time is determined by the statistical characteristics of the SDU losses in the UTRAN.

In order to simplify the conditions or the more general case study, some assumptions are made. The combined characteristics of the UTRAN loss pattern and the SDU Discard Timer are supposed to be such that UMTS recovers most of the packets lost in the radio link. This means that this part of the network only introduces additional delays and no packet losses. Therefore, packet losses are due to congestion conditions in the core network ($\rho_{CN}$) and due to packets arriving to the dejittering buffer after their corresponding playout time ($\rho_{jit}$).

Once degradation sources are identified, and taking into account that our research work focuses on end-to-end real-time service adaptation, it was necessary to identify which voice application parameters were modifiable:

- The coding rate of the AMR mode.
- The packetization scheme.
- The dejittering buffer size of the destination end user.

Summarizing, it can be established the following relationship between the main source of PQoS impairments and the service modifiable parameters:

- The end-to-end delay is mainly affected by the coding rate of the AMR mode, the packetization scheme and the dejittering buffer size.
- The packet losses depend mainly on the congestion experienced within the core network and on the dejittering buffer.

# 4. SIMULATION ENVIRONMENT

In order to study the most relevant parameters affecting the end users' PQoS impairments, a VoIP network scenario is implemented using OPNET©. Figure 2 depicts network topology which is made up of a mobile end user terminal which establishes a VoIP session with a fixed VoIP workstation through an UMTS network.

The VoIP session is established and managed by SIP. Although this is an end-to-end signalling protocol, it is necessary to include a SIP proxy server in the network in order to correctly route SIP messages.

SIP implementation included in OPNET© is not IMS fully compliant, but there are some available contributions (see [9] for example) which allow a complete simulation of IMS environments.

Regarding the voice application configuration, it is considered that both end users support the eight AMR modes, selecting the highest AMR mode (12.2 kbps) to start

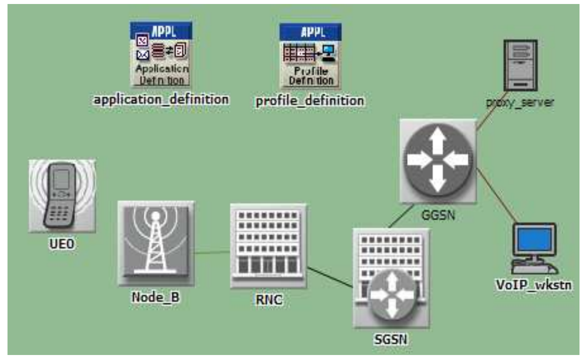


**Figure 2. Simulation environment**

with. The voice traffic is marked as Best Effort and the lowest packetization is used, this is one voice frame per packet. On the reception side, the optimum dejittering buffer size for VoIP services [10] is initially configured to 60ms. The mobile user is placed in an outdoor to indoor and pedestrian environment, sending voice information through the background class, which has an uplink and downlink bitrate of 64 kbps and 384 kbps, respectively. The RLC Acknowledged Mode (AM) is used.

## 4.1 VoIP Simulation Enhancement

In order to obtain accurate simulation results, we have introduced some modifications in OPNET©'s model processes and libraries, briefly summarized in this section.

### *4.1.1 Voice Service*

Since this paper focuses on AMR encoding schemes, a new Voice application attribute is created in order to take into account the eight AMR modes. These AMR modes should be previously defined in the Voice Encoder Schemes attribute, setting parameters such as: the coding rate, the frame size (20 ms for AMR modes), the DSP processing ratio, etc. Figure 3 shows a table containing the structure of the new attribute.

In addition to the name of the AMR mode, it is necessary to indicate the type of the encoding scheme. This capability allows end users to configure different codec modes to send and receive information, which was not previously supported in OPNET©.

### *4.1.2 E-model*

There are some aspects that should be taken into account to obtain an accurate assessment of end users' PQoS.

As described in subsection 3.1, PQoS is estimated using the E-model, whose computation is based on the end-to-end delay. This delay depends on the codification and decodification delays, so first of all, it is necessary to obtain the encoding scheme being used for transmission and reception.

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| | Encoder | Type |
|---|---|---|
| 0 | AMR_12.2 | sendrecv |
| 1 | AMR_10.2 | sendrecv |
| 2 | AMR_7.95 | sendrecv |
| 3 | AMR_7.4 | sendrecv |
| 4 | AMR_6.7 | sendrecv |
| 5 | AMR_5.9 | sendrecv |
| 6 | AMR_5.15 | sendrecv |
| 7 | AMR_4.75 | sendrecv |

**Figure 3. AMR modes implemented**

Due to the introduction of all AMR modes in both directions of the communication, it is necessary to carry out some modifications in OPNET© in that respect. Now, the encoding scheme selection is made through the SDP protocol, so the obtaining of the incoming and outgoing codecs should be done after the session establishment. Moreover, as the MOS score is computed in OPNET©, to obtain the PQoS of any of the end users, it is necessary to make all the computation with the outgoing encoder scheme of the other party, this is, the incoming encoder scheme of the current party.

Another aspect to take into account in the PQoS assessment in this simulation tool is the way the end-to-end delay and the packet losses are computed. The end-to-end delay is computed as in (7):

$$Ta = d_{netw} + d_{cod} + d_{decod} + d_{compr} + d_{decompr} \qquad (7)$$

Where, $d_{netw}$ is the network delay introduced by the transport of packets, $d_{cod}$ and $d_{decod}$ are the codification and decodification delays and $d_{compr}$ and $d_{decompr}$ are the compression and decompression delays, respectively.

However, neither the packetization delay nor the dejittering buffer delay are considered in (7). So, in order to accurately relate the PQoS scores as the E-model determines in (5) to the sources of impairments, those delays should be computed and introduced in the end-to-end delay.

Regarding to the packet losses calculation, if a packet arrives out of sequence, it could be possible to insert it in the dejittering buffer, so it would not be a loss. OPNET© takes into account both fixed and adaptative play outs. In the fixed mode, the dejittering buffer size remains the same all the simulation long, but in the adaptative mode, the buffer size is computed when each packet arrives. In this situation, the problem appears when a packet out of sequence is received. As it was implemented, if the dejittering buffer size were big enough, it would be possible to insert the packet in the buffer, although it arrived too late. This happened because OPNET checked its conditions according to the packets' sequence number and it did not consider the delay experienced by the packets. In order to correctly determine the packet losses due to the dejittering buffer size, we have based the previous algorithm on time variables and not on the packets' sequence numbers.

### *4.1.3 SDP Implementation*

SIP signaling protocol is commonly used with SDP, as mentioned in subsection 2.2. This protocol allows the inclusion of streaming media parameters description for the purpose of multimedia session initiation and management. SDP provides a negotiation between end users to allow them agree on a media type and format.

With the inclusion of the new encoding scheme attribute, end users could decide to start the VoIP session with different AMR modes for sending and receiving. Then, the encoding scheme negotiation must be included in the multimedia session initiation via SDP.

For the SDP implementation in OPNET© it is necessary to define a new type of structure with all the session description parameters needed. For the purpose of our work and considering initial negotiation only, the following fields are necessary: the media description and the initial AMR modes that end users want to start with. We assume that both users support all the AMR modes. If this were not the case, the list of supported AMR modes should be also included, in order to avoid the selection of not supported encoding schemes. Additionally, once the session is established, the adaptation actions shall include another field with the packetization scheme to be adopted.

When initiating a VoIP session, end users' voice application processes obtain their configuration attributes and set them into the SDP structure (such as the list of AMR modes). Once voice application processes have filled all the necessary SDP fields for initial negotiation, it is included into the SIP call information, which forms the SIP messages' body.

As said before, SDP can be also used for session management purposes, so it can be introduced in any SIP message, not only in initial negotiation ones. In this way and for adaptation actions, we implement a new SIP method called: re-INVITE, described in the following subsection.

### *4.1.4 SIP re-INVITE Method Implementation*

As it is mentioned in [5], it is possible to use re-INVITE SIP messages in order to re-negotiate some multimedia session aspects. Although SIP UPDATE method [11] could be also used for this purpose, according to the definition of the IETF this kind of messages does not allow modifying some other characteristics of the established dialog. Then, considering future extensions, we decided to use re-INVITE.

Despite the re-INVITE method implementation is quite similar to the INVITE implementation, there are some differences. First of all, the exchanged SIP methods that

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follow a previously re-INVITE sent message. Once one of the end users has sent a re-INVITE, the other party sends an OK method to accept the modification. If this party does not accept the change, the terminal sends an error response code. Anyway, both messages receive an ACK as a response.

Another difference in the implementation of re-INVITE methods is the way the OPNET©'s SIP processes interact with the voice application ones. As described in the previous subsection, when an end user wants to establish a VoIP session with another user, first the voice application process set the SDP information with the configuration parameters. Then, there is a multimedia session negotiation via SDP and SIP protocols. In the re-INVITE cases, these messages bring the new SDP information, so when this kind of method is received, the correspondent voice application processes should adopt the new configuration. Figure 4 summarizes SIP-SDP procedures when initiating or adapting a session.

## 4.2 VoIP Real-time Adaptation

As explained in subsection 3.2, there is a close relationship between the network situation and the PQoS experienced by end users.

In order to maintain the MOS score at suitable levels during all the VoIP session, we introduce a real-time service adaptation. Since this paper focuses on service adaptation, only encoding scheme, packetization scheme and dejittering buffer size modifications are considered.

When an end user detects that its MOS score has decreased under a pre-established threshold (for this study, a 3.5 MOS score is considered and the MOS calculation is carried out when each packet is received), it tries to determine which the source of PQoS degradation is. As mentioned in subsection 3.2, there are two main sources: delays in the UMTS network or packet losses in the core network. This subsection also sets the relationship between these sources and the service tunable parameters.

This way, if the end user detects that the main source of impairment is the delay in UMTS network, it will send a SIP re-INVITE message including updated SDP information. This SDP structure indicates that the other party should change: its outgoing encoder scheme, using one with lower coding rate, or its packetization scheme, increasing it. In these cases the end-to-end delay experienced low values. However, if the main source of impairment is the packet loss, the SDP information will contain a higher dejittering buffer size value. Figure 5 depicts this adaptation mechanism.

For this purpose it is necessary that end users' voice application processes store the established multimedia session information. This way, it is possible to determine

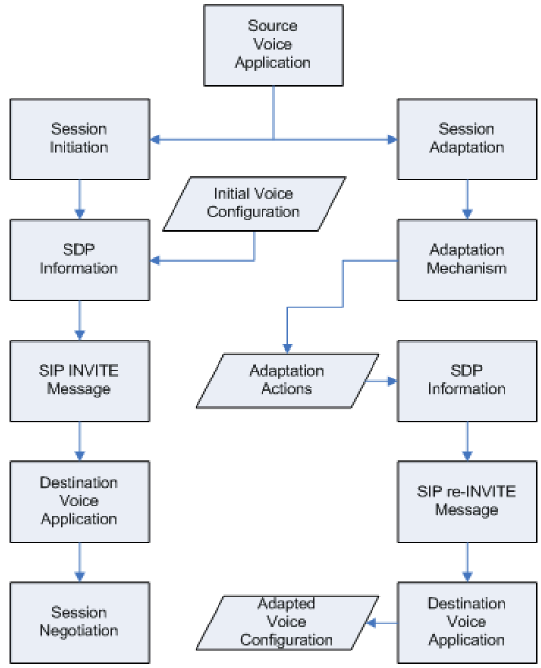


**Figure 4. SIP-SDP procedures**

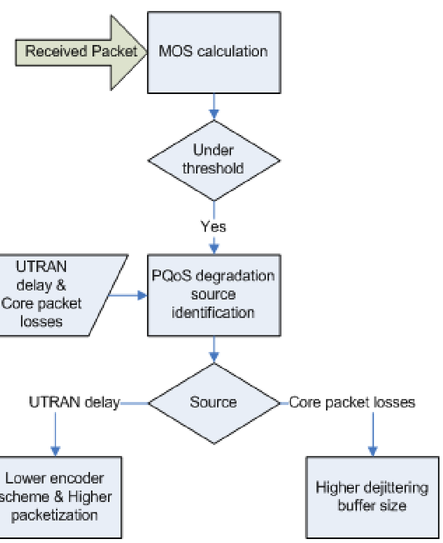


**Figure 5. Adaptation mechanism**

which adaptation actions need to be set into the SDP information.

# 5. RESULTS

In order to simulate PQoS impairments, we have set up a specific configuration in the simulation environment. For example, congestion conditions are obtained setting uplink

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background traffic in a core network link at the middle of the simulation time.

## 5.1 No Adaptation

Figure 6 depicts the evolution of the MOS score curve when no adaptation mechanism is introduced. As it can be seen, the MOS score finally reaches its lowest value. It also shows how the curve would be without sources of PQoS degradation.

## 5.2 With Adaptation

The results present in this subsection are obtained with the service adaptation mechanism launched when congestion situations result in MOS score going below the specific admissible threshold.

Figure 7 depicts the packet end-to-end delay when congestion conditions are deliberately introduced and AMR encoding reduction is used, changing from the initial AMR 12.20 to the AMR 7.40 mode.

Figure 8 depicts the MOS score obtained: when an AMR mode reduction adaptation takes place (from AMR 12.20 to AMR 7.40); and when we use two different packetization adaptations: one changing from one to two voice frames per packet and another, from one to three voice frames per packet.

As Figure 8 shows, it is not worth applying a packetization adaptation mechanism, since the increase of the number of voice frames per packet may result in an increase of the voice frames lost due to the dejittering buffer size (if the packet can not be inserted in the dejittering buffer, there are more voice frames lost, resulting in a higher MOS score impairment).

But if the dejittering buffer size increases (i.e. 100 ms size), there would more packets to be inserted into the dejittering buffer size, so the total amount of voice frames received would be higher. Figure 9 depicts this situation in which the dejittering buffer size is 100 ms. It shows that there is no such a big difference between the encoding scheme adaptation and the packetization scheme adaptation. Furthermore, during the simulation there are some moments in which each of the different adaptations provides the highest MOS score.

Figure 10 shows a detailed area of Figure 9. As previously mentioned, each adaptation mechanism proposed provides the highest MOS score in different periods along the simulation. So, different adaption possibilities seem to be valid in this kind of scenarios, depending on the specific network status. Therefore such heterogeneous nature of these VoIP network environments demands the analysis of end users' PQoS in their design and management.

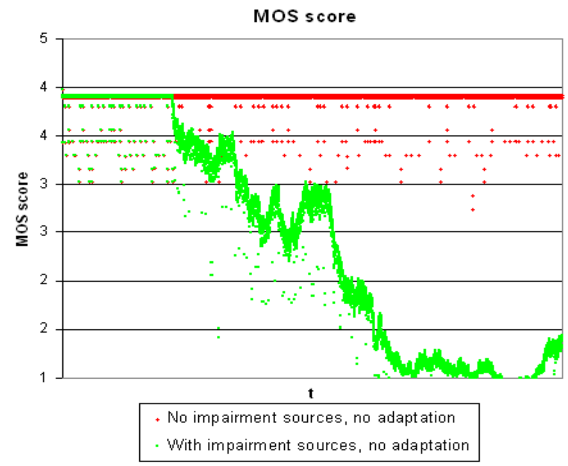


**Figure 6. MOS score with no adaptation**

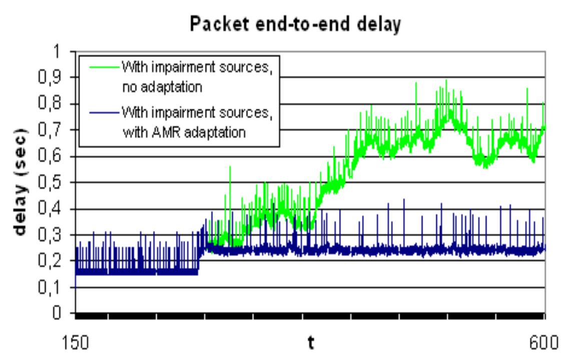


**Figure 7. Packet end-to-end delay**

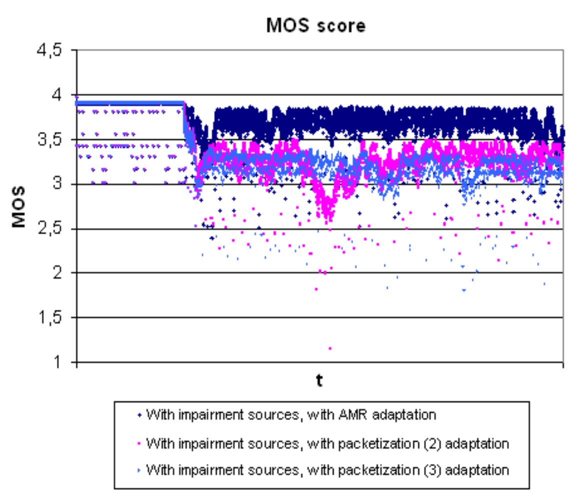


**Figure 8. MOS score with different adaptations**

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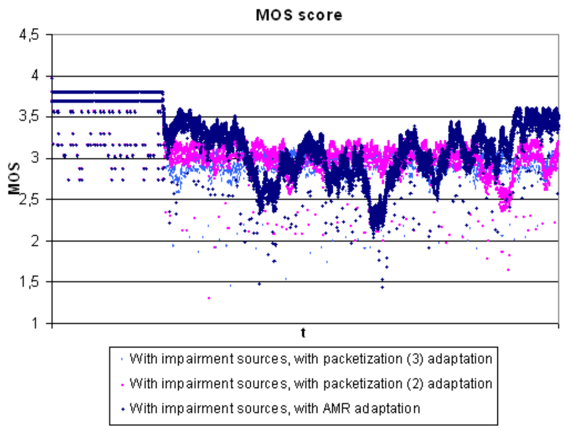


**Figure 9. MOS score with higher dejittering buffer size**

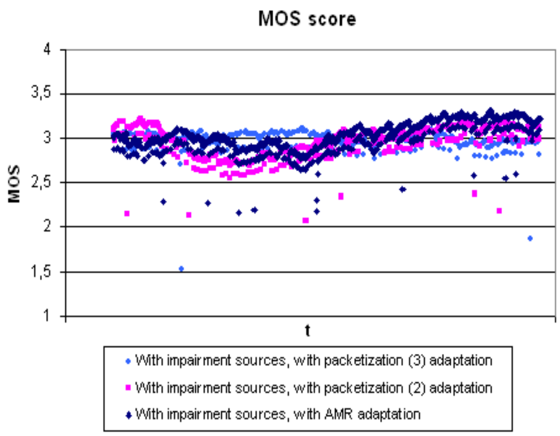


**Figure 10. Detailed area of Figure 9**

## 6. CONCLUSIONS

This paper provides a study of the relationship between the VoIP tunable parameters (encoding scheme, packetization scheme and dejittering buffer size) and the main sources of PQoS degradation (end-to-end delay and packet losses).

We have carried out different simulations over a generic VoIP over UMTS scenario using OPNET© simulation tool. In order to achieve more accurate results, we have introduced some enhancements, such as the implementation of the E-model and the support of the eight AMR modes.

Introducing some network congestion conditions in the simulation scenario, we have obtained MOS score degradations due to different sources of impairment. In order to avoid such PQoS decrease, we have introduced a real-time service adaptation mechanism based on VoIP tunable parameters and carried out using SIP-SDP signalling. Due to the heterogeneous nature of VoIP network environments, the obtained results have not shown the existence of a single optimal adaptation method. Each of the adaptation methods proposed is able to provide the highest MOS score in different moments of the simulation.

So, it becomes necessary to implement a dynamic and real-time service adaptation mechanism into VoIP network scenarios, able to suit the VoIP service to the network conditions. Thus, it would be possible to provide the highest QoS to end users even if the network status changes from one source of PQoS degradation to another.

## 7. ACKNOWLEDGEMENTS

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## 8. REFERENCES

[1] ITU-T Recommendation G.107. 2007. “The E-model, a computational model for use in transmission planning”.

[2] 3GPP TS 26.235. 2008. “Packet switched conversational multimedia applications; Default codecs”.

[3] 3GPP TR 26.975. 2007. “Performance characterization of the Adaptative Multi-Rate (AMR) speech codec”

[4] Poppe, F., De Vleeschauwer, D. and Petit, G. H. 2001. “Choosing the UMTS Air Interface Parameters, the Voice Packet Size and the Dejittering Delay for Voice-over-IP Call between a UMTS and a PSTN Party”. IEEE INFOCOM 2001.

[5] Rosenberg, J., Schulzrinne, H., et al. 2002. “SIP Session Initiation Protocol”, IETF RFC 3261.

[6] Handley, M., Jacobson, V. and Perkins, C. 2006. “SDP: Session Description Protocol”, IETF RFC 4566.

[7] Johansson, I. 2006. “VoIP Shim for RTP Payload Formats”. IETF draft.

[8] ITU-T Recommendation G.113. 2001. “Transmission impairments due to speech processing. Provisional planning values for the equipment impairment factor Ie and packet-loss robustness factor Bpl”

[9] Vázquez, E. Simulation of SIP in the 3GPP IP Multimedia Subsystem (IMS). http://www.dit.upm.es/asignaturas/opnet/

[10] Barbaresi, A., Mantovani, A. 2007. “Performance Evaluation of Quality of VoIP Service Over UMTS-UTRAN R99”. IEEE ICC ’07.

[11] Rosenberg, J. 2002. “The Session Initiation Protocol (SIP) UPDATE Method”, IETF RFC 3311.

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