Chapter 1

3G M-HEALTH SYSTEM PERFORMANCE

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Abstract

The IPv4 performance of a multi-collaborative wireless telemedicine system operating over Third-Generation (3G) cellular networks is presented. The system is designed to communicate the personnel of an ambulance with medical specialists in a remote hospital through a Universal Mobile Telecommunication System (UMTS) wireless access. Its architecture is based on advanced signalling protocols that allow multimedia multi-collaborative conferences in IPv4/IPv6 3G scenarios. The system offers simultaneous transmission of real-time medical data and videoconference, together with other non real-time services. IPv4 performance results show that the system performs reliably over IPv4-only UMTS accesses (64 Kbps in the uplink). Bandwidth use and jitter buffer design and trade-offs are the main aspects considered. Measurements allow dimensioning system parameters in order to improve transmission efficiency, channel utilization and, finally, the quality of the services offered.

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Introduction

Mobile Health (m-Health) is an emerging area of telemedicine in which the recent developments in mobile networks and telemedicine applications converge. m-Health involves the exploitation of mobile telecommunication and multimedia technologies and their integration into new mobile healthcare delivery systems [1]. Wireless and mobile networks have brought about new possibilities in the field of telemedicine thanks to the wide coverage provided by cellular networks and the possibility of serving moving vehicles.

One of the first wireless telemedical systems that utilized Second-Generation (2G) Global System for Mobile Communications (GSM) networks addressed the Electrocardiogram (ECG) transmission issues [2]. In recent years, several m-Health and wireless telemedical systems based on GSM have been reported [3], allowing the accomplishment of remote diagnosis in mobile environments, as well as communication to geographic zones inaccessible by wired networks. The recent developments in digital mobile telephonic technologies and their impact on mobility issues in different telemedical and telecare applications are clearly reflected in the fast growing commercial domain of mobile telemedical services. A comprehensive review of wireless telemedicine applications and the most recent advances on m-Health systems are presented in [4].

However, 2G-based systems lack the necessary bandwidth to transmit bandwidth-demanding medical data. The Third-Generation (3G) Universal Mobile Telecommunications System (UMTS) overcomes the limitations of first and second mobile network generations supporting a large variety of services with different Quality of Service (QoS) requirements. However, this fact makes network design and management much more complex. New applications require networks to be able to handle services with variable traffic conditions keeping the efficiency in the network resources utilization. The UMTS air interface is able to cope with variable and asymmetric bit rates, up to 2 Mbps and 384 kbps in indoor and outdoor environments, respectively, with different QoS requirements such as multimedia services with bandwidth on demand [5]. In this kind of scenario, the emergence of 3G mobile wireless networks will permit to extend the use of m-Health applications thanks to their higher transmission rates and flexibility over previous mobile technologies [6].

UMTS introduces the IP Multimedia core network Subsystem (IMS) [7], an IPv6 network domain designed to provide appropriate support for real-time multimedia services, independence from the access technologies and flexibility via the separation of access, transport and control. The fundamental reason for using IPv6 is the exhaustion of IPv4 addresses. Support for IPv4 is optional, but since network components require backward compatibility, it is clear that a dual stack configuration (IPv4 and IPv6) must be provided. The IMS uses the Session Initiation Protocol (SIP) as signalling and session control protocol [8]. SIP allows operators to integrate real-time multimedia services over multiple access technologies such as General Packet Radio Service (GPRS), UMTS or, ultimately, other wireless or even fixed network technologies (interworking multimedia domains).

This chapter presents the IPv4 performance of a 3G m-Health system (Fig. 1) [9] designed for different critical and emergency medical scenarios. With this system, medical specialists in a hospital take part in a multipoint conference with the medical personnel of an ambulance, receiving compressed and coded biomedical information from the patient, making it possible for them to assist in the diagnosis prior to his reception. The m-Health system includes intelligent modules such as information compression and coding, and QoS control (data prioritization, congestion control and jitter control) [10] to significantly improve the transmission efficiency of joint real-time and non real-time data over wireless channels in a more appropriate way than previous systems [11]. IPv4 performance results validate the QoS mechanisms used and allow to dimension jitter buffers in order to improve the quality of real-time services.

m-Health system overview

The wireless m-Health system has been built using standard off-the-shelf hardware, instead of developing propriety hardware as in [12], uses free software and commercially available 3G wireless UMTS cellular data services. In the first stages of its design, user requirements and functional specifications were established in collaboration with medical specialists, in order to create a portable and modular system that could be easily integrated in any environment, using any underlying network technology capable of supporting IP multimedia services.

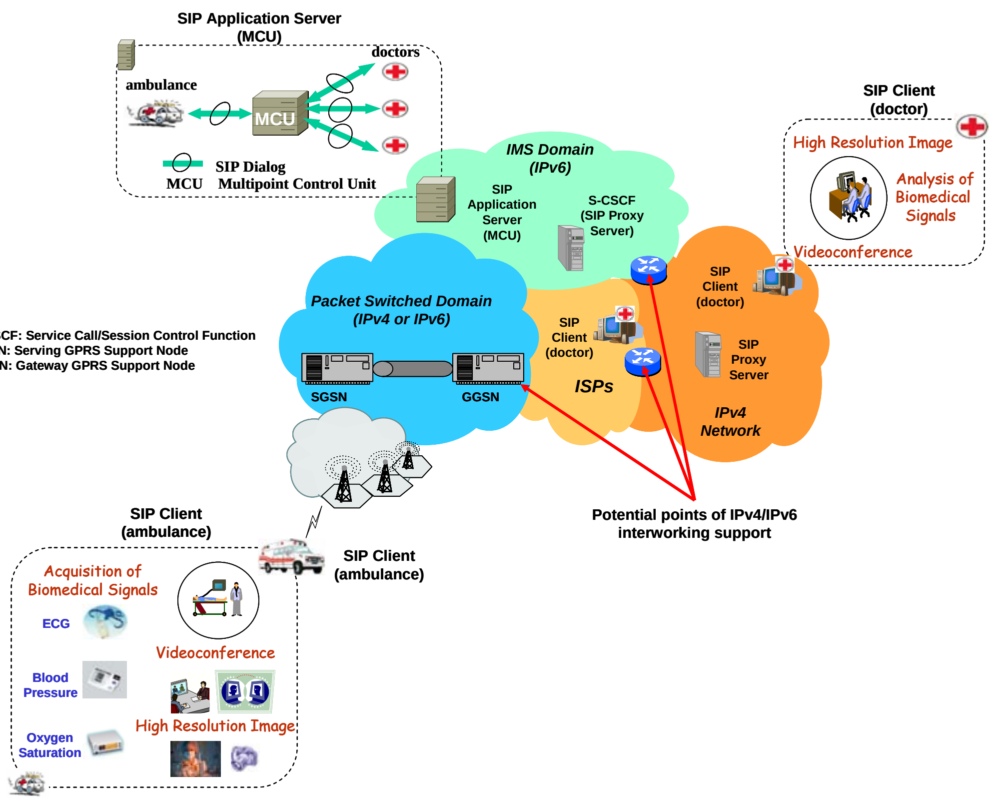


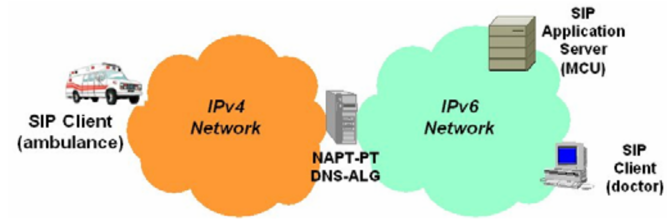
Figure 1: Wireless m-Health system

The details of the 3G m-Health system architecture [9] are shown in Fig. 1. The system comprises of the signalling and session control, medical user services and application control subsystems, together with the QoS control internal subsystem. Several intelligent modules allow the acquisition, treatment, representation and simultaneous media transmission, rather than only one media at a time [13]-[14]. Information compression, coding and QoS control (data prioritization, congestion control and jitter control) modules [10] improve the transmission efficiency of joint real-time and non real-time data over wireless channels in a more appropriate way than previous systems [11]. In addition, and also unlike [11], this system follows a multi-collaborative design, integrates new real-time multimedia features intended for 3G wireless networks, supports IPv4/IPv6 interworking [15] and uses SIP as the service control protocol, including messages defined specifically for the IMS by the 3rd Generation Partnership Project (3GPP). The IPv4/IPv6 SIP dual stack is the basis to integrate the ambulance and the hospital in any possible 3G scenario [16] (Fig. 2): IPv4 ambulance connecting to an IPv6 hospital, IPv6 communication through IPv4 islands and, finally, native all-IPv6 communication.

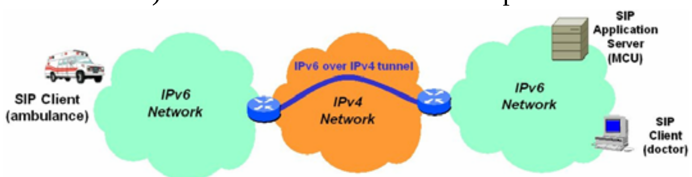
In the first stages of IPv6 deployment in 3G networks, there will be GPRS networks that will only be able to provide IPv4 connections. As it is shown in Fig. 2a, if an IPv4 ambulance needs to communicate with an IPv6 hospital, a transition mechanism is required. One of the possibilities is the use of a Network Address Protocol Translator-Port Translator (NAPT-PT) working together with a dual stack Domain Name System (DNS) server with a DNS-Application Level Gateway (DNS-ALG). The main problem with NAPT-PT is that it modifies packets, therefore security cannot be guaranteed if an IP Security (IPsec) association is used. But this is exactly the same problem that exists today with the use of IPv4 private addressing for peer-to-peer applications. On the other hand, the NAPT-PT mechanism provides an interoperability framework for more than one user terminal and any kind of application.

The next step of IPv6 deployment will be a common scenario when operators start providing IPv6 communications and customers start connecting to IPv6 services. Some IPv6 islands will still need to be connected through an IPv4 network (Fig. 2b) as it is done today in fixed networks. In this case, the most appropriate transition mechanism is tunneling. Tunneling means adding an IPv4 header instead of translating it, which could cause an important bandwidth waste in the Radio Access Network (RAN). However, this tunneling will occur in the fixed world.

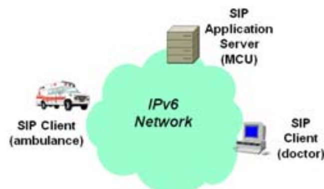
Finally, the expected final scenario in IPv6 deployment (Fig. 2c) does not require any transition mechanism provided that the m-Health system is IPv6 capable and ready to be integrated in an all-IPv6 3G world.



a) IPv4 ambulance and IPv6 hospital



b) IPv4 islands between IPv6 networks



c) Native IPv6 communication

Figure 2: IPv4/IPv6 migration process

Communication between the remote medical personnel and medical specialists is established by means of multipoint multi-collaborative sessions through several network environments capable of supporting the different types of multimedia traffic. The conference model selected is the tightly coupled conference model [17], which requires the existence of a Multipoint Control Unit (MCU) (Fig. 1). System users and the MCU exchange information associated with the different medical user services and their presentation (application control), and also communication and service quality management data (signalling and QoS control). The developed signalling allows exchanging the characteristics associated to the different information flows between the system elements and is based on standard protocols that favour interoperability. Signalling tasks, performed by the SIP protocol, begin with the establishment of a SIP dialog with the MCU in which, by means of Session Description Protocol (SDP) messages, the different medical services are described.

The medical user services in the m-Health system are associated with information shared in a multi-collaborative environment. Specifically, the system has services to share audio, ambient video, medical data (ECG, blood pressure, heart rate and oxygen saturation), high-resolution medical still images, chat, electronic whiteboard and a web service to access clinical information databases. Each kind of information is associated with a medical user service and uses a transport protocol and a codec according to its characteristics (Table 1). Hence, real-time services (audio, video and medical data) use the Real-Time Transport Protocol (RTP) [18], whereas the rest of the services use the Transmission Control Protocol (TCP). In addition, audio and video services use the codecs recommended by the 3GPP [19], and the medical data service uses the Wavelet Transform (WT) [20].

Table 1: Characteristics of medical user services

Medical user service	Timing needs	Bandwidth needs	Transport protocol	Codec	Codec operation modes	Maximum average IPv4 bandwidth (Kbps)
Audio	RT[a]	Medium	RTP	AMR[c]	4.75-5.15-5.9-6.7-7.4-7.95-10.2-12.2 (Kbps)	28.80
Medical data	RT	Low	RTP	WT	5-10 (Kbps)	10.30
Video	RT	High	RTP	H.263	5-10 (Fps[d])	18.24
Chat	NRT[b]	Low	TCP	-	-	ABR[e]
Electronic whiteboard	NRT	Low	TCP	-	-	ABR
Still image	NRT	Medium	TCP	-	-	ABR

a. Real-Time
b. Non Real-Time
c. Adaptive Multi-Rate
d. Frames per second
e. Available Bit Rate

QoS control subsystem

The QoS in this system is mainly determined by the characteristics of the UMTS link. Mobile links are very fluctuant, therefore a QoS control process is required in order to obtain a good network performance. This process uses IP packet transfer performance metrics recommended by the International Telecommunication Union (ITU) in its Recommendation Y.1540 [21]. The QoS metrics selected are packet loss rate, delay variation (jitter) and octet-based IP packet throughput (bandwidth). In addition, the wireless telemedicine system has application jitter buffers to mitigate channel effects.

The QoS control process is especially important in end points because it is there where the QoS-related decisions are taken. When an end point detects that a particular communication does not operate properly, it needs to modify the characteristics of its multimedia session in order to improve QoS and thus, it renegotiates the corresponding session by sending SIP/SDP messages. Hence, the system end points and the MCU can modify certain upper-level protocol parameters (codecs used, transmission rates, compression ratios, etc.) in order to adapt the information transmitted to network performance. This process is possible thanks to a transport library that provides a uniform interface to send the information generated by medical user services and different QoS metrics measurement tools developed for several types of links. This transport library also offers different queuing policies.

Due to the variable and scarce wireless channel resources shared between all medical user services, it is necessary to prioritize them to provide an adequate treatment to real-time and non real-time ones. Real-time services are very sensitive to channel conditions (mainly bandwidth, delay, jitter and packet loss rate), whereas non real-time ones can adapt well to varying environments thanks to the built-in flow control and reliability of TCP. For that reason, the most priority services are the medical data, audio and video services, which will take up most of the channel resources. Non real-time services will be treated best-effort, adapting to the spare network resources using TCP built-in mechanisms (Table 1). According to the discussion, real-time services are the most priority, but, among them, a clinically acceptable ECG signal is more important than a clear audio conversation that, in turn, is more important than the ambient video signal. Thus, the medical data service has high priority, the audio service medium priority and the ambient video service low priority. The characteristics of these services are monitored at transmission and reception and are taken into account to increase or decrease codec rates.

Two of the main causes of poor QoS are packet losses and packet drops. Packet losses are produced inside the network, whereas packet drops occur in application queues. Regarding packet losses, they can be caused by congestion conditions or channel errors. As the commercial UMTS 3G wireless cellular service used in this system operates in the UMTS Acknowledged Data Transfer mode at the radio link layer [22], packet losses are considered to be produced only by network congestion. Thus, part of the QoS control process is based on congestion control. Congestion control signalling can be implicit, activated by packet drops in transmission queues, or explicit, initiated in reception. Implicit signalling allows controlling the congestion in the UMTS link, whereas explicit signalling is used when congestion is detected in the rest of the links in the communication path. The congestion control algorithm selected is that presented in [11], but applied to the three real-time medical user services of this system. Using codec rate adaptation according to service priorities, the codec rate of the low priority service is decreased first, and then the codec rate of the medium priority service is varied. Finally, the high priority service is limited.

Regarding jitter, it can be caused by the variable nature of wireless links or by the joint transmission of all services, therefore each real-time service has an application jitter buffer associated with it that tries to mitigate its effects (Fig. 3). These buffers have been properly dimensioned to minimize jitter, delay and packet drops. They are First In-First Out (FIFO) queues that are filled with network packets as they arrive. The application periodically empties one packet from the buffer at the required rate, but with the following considerations:

- If the buffer becomes empty, it does not serve packets to the application until a predefined buffer occupancy threshold is reached.
- From that point on, the buffer serves packets until it becomes empty. If, on the other hand, the buffer fills completely and a new packet arrives, the first packet stored in the buffer (the oldest one) is dropped to make room for the new one.

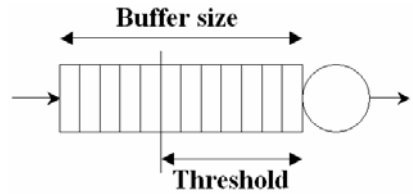


Figure 3: Application jitter buffer model

IPv4 performance results

In order to measure the wireless telemedicine system performance and to improve the quality of the services offered by dimensioning jitter buffers, several tests have been carried out using the system over 64/128 Kbps (Uplink/Downlink bandwidth at IP level) IPv4-only UMTS accesses in urban areas (good coverage level and low speed, as well as static vehicles).

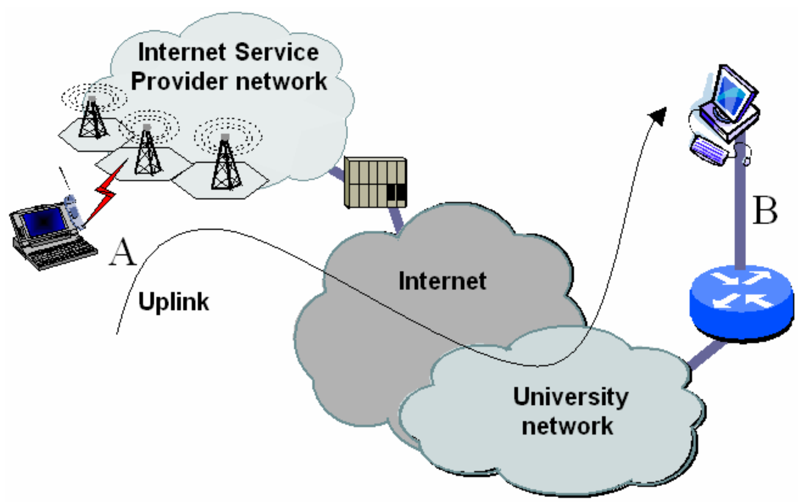


Figure 4: Measurement scenario

The measurement scenario used is shown in Fig. 4. As the uplink is more restrictive, the results presented here correspond to that connection sense. Packets have been captured in points A and B. Measurements in point A have been used to obtain the characteristics of the traffic injected into the uplink (IP-level bandwidth and jitter), whereas measurements in point B allow to obtain network behaviour (packet loss rate and jitter). Several tests have been carried out during several weeks, staggered along the whole day, all days of the week. The duration of these tests has been selected according to the average service time of an ambulance, which has been considered to be 10 minutes in a medium-sized Spanish city like Zaragoza.

Average bandwidth results

Table 2 presents the results about the average IP-level bandwidth used by real-time medical user services in point A. 48 tests have been carried out every 30 minutes during one day, using isolated user services and varying codec operation modes. As it can be observed, the audio service adapts well to the theoretical expected average IP bandwidth (see Table 3 and Eq. 1). In addition, considering more audio samples per network packet reduces bandwidth use, since transmission efficiency (information carried by each packet to total packet size ratio) is increased. However, there is a limit in the number of audio samples per packet that can be used because more audio samples per packet yield more audio delay (audio samples are generated every 20 ms, see Eq. 2). Moreover, if an audio packet is lost, all the audio samples carried by it are lost and, therefore, a reduced number of audio samples per packet is more suitable to error-prone environments. That is the reason why the maximum number of samples per packet has been limited to 3 in the final system, although Table 2 also shows the results for 4 and 5 samples per packet. Regarding the video service, it is worth noting that the bandwidth shown in Table 2 can vary substantially with the movement of the video scene captured. Finally, the medical data service adapts well to the codec rate specified because medical data frame sizes are long enough to obtain good transmission efficiencies.

Table 2: Average IP-level bandwidth used by real-time user services

Medical user service	**Operation mode**		**Average IPv4 bandwidth (Kbps)**
Audio	Samples/Packet	Codec rate (Kbps)	
	1	4.75	21.20
	1	12.2	28.80
	2	4.75	13.21
	2	12.2	20.81
	3	4.75	10.50
	3	**12.2**	**18.10**
	4	4.75	9.22
	4	12.2	16.82
	5	4.75	8.41
	5	12.2	16.03
Medical data	Bit rate (Kbps)		
	5		5.30
	10		**10.30**
Video	Frames per second		
	5		8.05
	10		**18.24**

Table 3: Audio sample size

Audio codec rate (Kbps)	**Sample size (Bytes)**
4.75	13
12.2	32

$$IPv4\ Bandwidth\ (Kbps) = \frac{8 \times (20 + 8 + 12 + Samples\ per\ Packet \times Sample\ Size)}{Samples\ per\ Packet \times 20} \qquad \text{Eq. (1)}$$

$$Fixed\ delay\ (ms) = Samples\ per\ Packet \times 20 \qquad \text{Eq. (2)}$$

As it can be checked in Table 2, the total bandwidth consumed by all real-time medical user services fits in a 64 Kbps UMTS channel, even when the most bandwidth-consuming codec rates and the lowest transmission efficiencies are used. Thus, according to the previous discussions, the initial codec operation modes selected in this wireless telemedicine system have been those highlighted in Table 2, achieving a reasonable trade-off between bandwidth, transmission efficiency, delay and loss rate. During normal operation, codec modes can vary in response to congestion conditions with the aid of the congestion control algorithm mentioned in subsection "QoS control subsystem". The average IP-level bandwidth obtained in point B is very similar to that obtained in A. In addition, no packet losses have been observed in any point. Therefore, the network does not modify traffic characteristics regarding bandwidth and packet loss.

Jitter results

48 tests have been carried out every hour during 2 days, with all the real-time medical user services operating at the same time and at the codec rates highlighted in Table 2 (the highest possible codec rates). These tests are useful for observing the influences between traffics generated by each real-time service.

As all possible jitter effects can be observed in point B, Fig. 5 presents a zoom over 9 seconds of audio interpacket time taken in a test in point B. Audio packets are generated every 60 ms, so this is the theoretical time that should appear in Fig. 5. Medical data packets are generated every second (approximately every 17 audio packets), therefore their effects over the audio service appear uniformly spaced. It can be observed that they cause more than 200 ms of jitter due to the time it takes the 64 Kbps UMTS uplink channel to transmit big-sized medical data packets (about 1300 bytes at IP level). Regarding video, packets are smaller and not uniformly spaced because they depend on image movement. Thus, the effects of video over audio packets are smaller. In addition, other jitter effects are caused by the network. Finally, all the effects can overlap at the same time.

Regarding the effects of audio and video over the medical data service, none of them have a significant influence (Fig. 6) due to the fact that medical data packets are very spaced between them (1 second, ideally), and a jitter effect of less than one second is not noticed in reception with the aid of a minimal jitter buffer. This result can be checked in Fig. 6, which presents medical data interpacket time in point B for a particular test.

The last real-time service, the ambient video, also suffers jitter effects caused by the rest of real-time services (Fig. 7). However, the most serious effects are produced by the video codec used and big reception buffers are recommended. In addition, this service has the lowest priority and video motion softness is not critical, therefore a big jitter buffer is enough to support all the possible jitter effects.

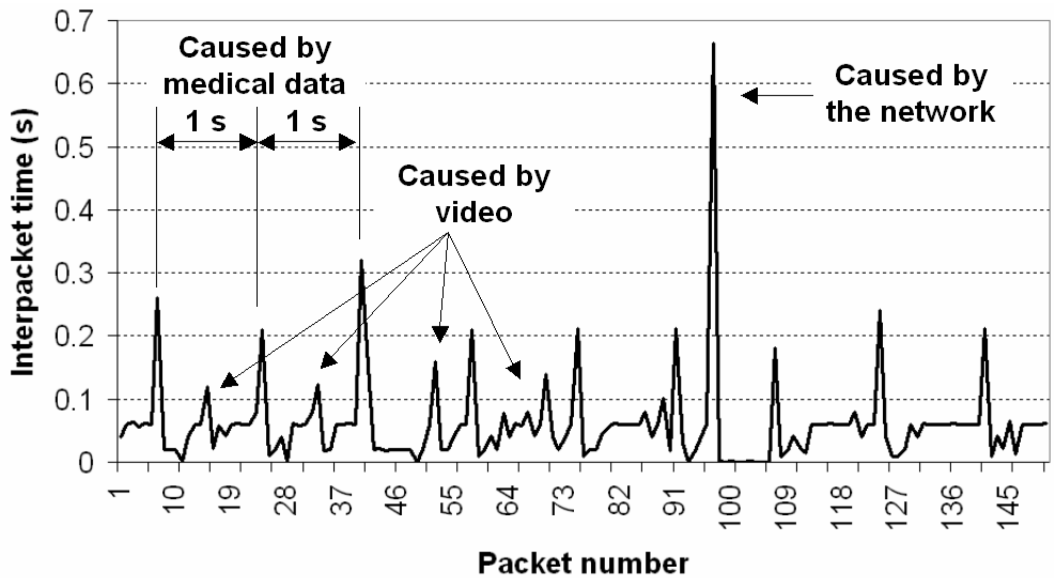


Figure 5: Audio interpacket time

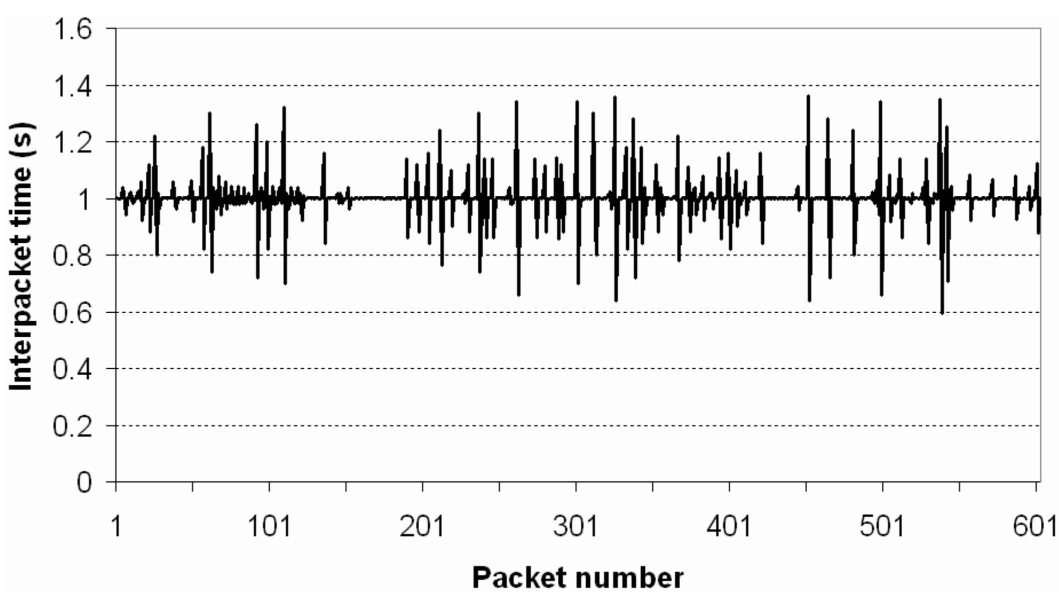


Figure 6: Medical data interpacket time

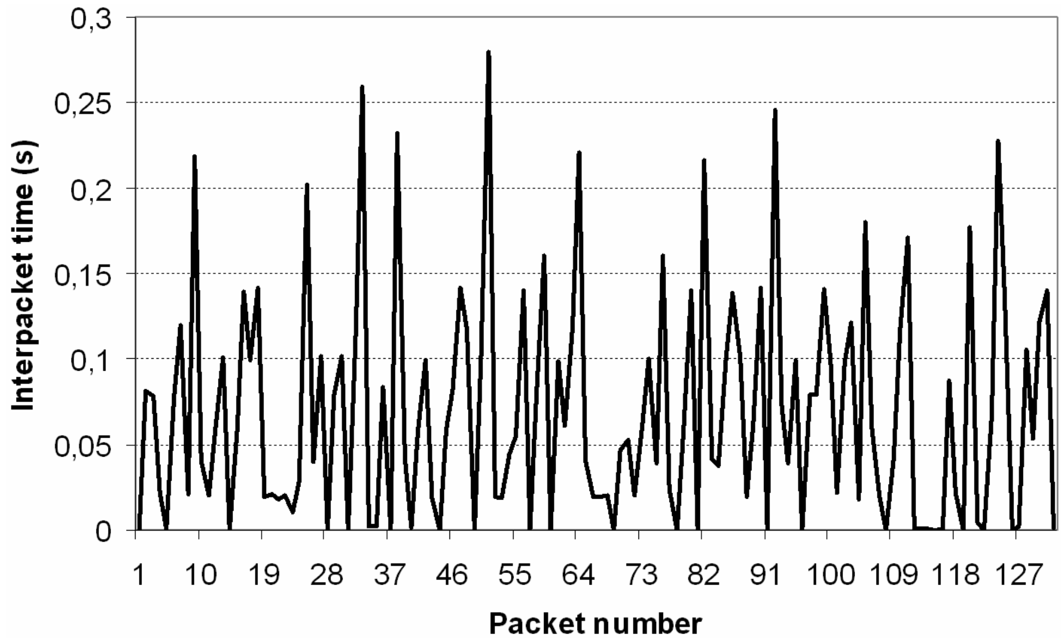


Figure 7: Video interpacket time

Jitter buffer dimensioning

Packet captures in point B are very useful to obtain the instantaneous application jitter buffer occupancy for all the real-time services. Using packet timestamps and theoretical packet buffer empty times, the instantaneous buffer occupancy can be calculated. Using the same tests of the previous subsection, with all the real-time medical user services operating at the same time and at the codec rates highlighted in Table 2, several buffer occupancy calculations have been carried out. In the calculations, buffer size and threshold values can be varied in order to obtain useful results to dimension the jitter buffers properly. First of all, the buffer threshold must be able to support jitter effects caused by the real-time services. Subsequently, the total buffer size must be able to support jitter effects caused by the network. High values of the threshold cause fewer situations in which the buffer becomes empty, allowing a continuous reproduction, but, on the other hand, introduce a bigger fixed delay. A value too low reduces the fixed delay, but at the expense of causing frequent interruptions in the reproduction. Regarding the buffer size, bigger buffers allow less packet drops than smaller ones, but also entail a bigger delay on enqueued packets.

Buffer threshold

The most important jitter effects caused by other services that the audio service suffers are produced by the medical data service. Their value can vary, but rarely exceeds 200 ms. Considering that audio packets are generated every 60 ms, a threshold of 4 packets (240 ms of audio stored in the buffer) is enough in order to support them. Due to the small jitter effects present on the medical data service, the buffer threshold can be selected to be minimal. A value of 2 packets would be enough, but to ensure a more robust behaviour, and considering that fixed delay is not relevant, 3 packets has been used as the threshold for this service.

Finally, the ambient video service is not critical, so the buffer threshold selected for it has been the minimum, i.e., 1 packet.

Buffer size

As it has been noted before, the total buffer size must be able to support jitter effects caused by the network. The first step is to consider an infinite buffer with all the 10-minute tests carried out to calculate the maximum buffer size that would have been needed in order not to drop any packet. Figs. 8, 9 and 10 show the instantaneous buffer occupancy obtained for a particular test of the audio, medical data and video services, respectively. In these cases, a buffer size of 12 packets for the audio service, 3 packets for the medical data service and 25 packets for the audio service would have been enough.

The same results have been obtained in all medical data tests (3 packets), therefore a value of 4 packets as jitter buffer size is a good choice in order to ensure a robust behaviour and no packet drops even if worse conditions appear. Regarding the video service, the maximum buffer occupancy obtained in all the tests has been 25 packets. A more conservative value of 30 packets has been selected to support even worse working conditions.

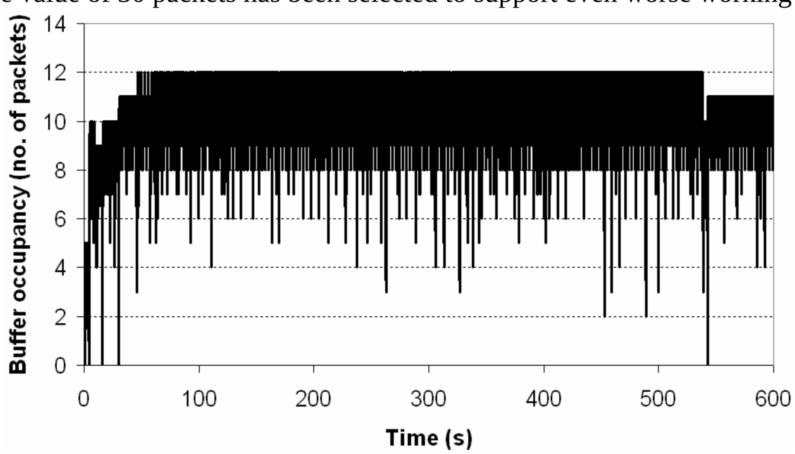


Figure 8: Audio service buffer occupancy (threshold = 4 and infinite buffer)

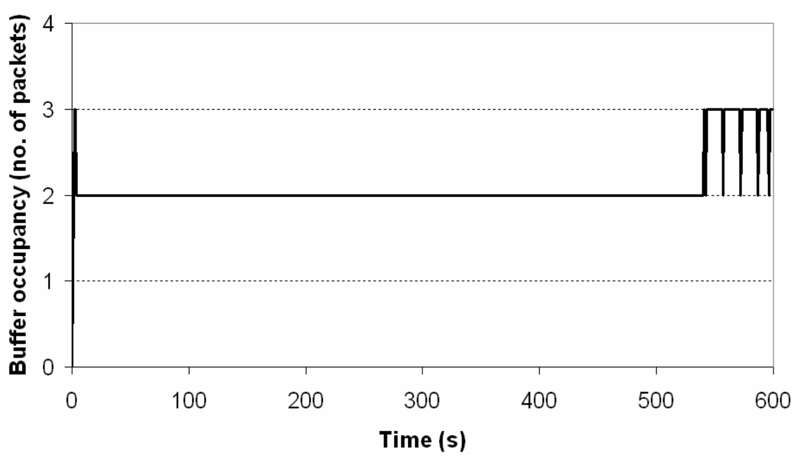


Figure 9: Medical data service buffer occupancy (threshold = 3 and infinite buffer)

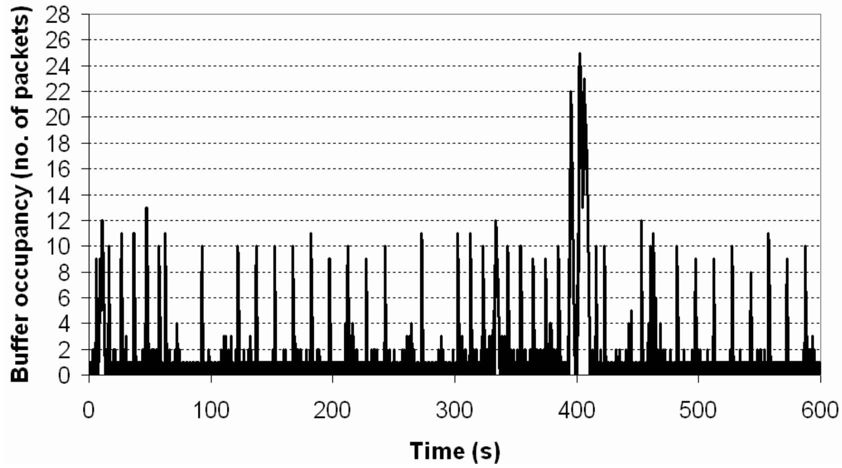


Figure 10: Video buffer service occupancy (threshold = 1 and infinite buffer)

Finally, audio tests present a small variability in the maximum buffer size. In order to obtain a suitable buffer size, finite buffer sizes have been considered, producing different packet drop ratios depending on their value. The packet drop ratio has been averaged for all the tests and the results are shown in Fig. 11 as a function of buffer size. Numeric results are also presented in Table 4. Not only packet drop ratio, but also the fixed and maximum delays that a particular buffer size causes are the relevant parameters in order to select its value. If, for example, a packet drop ratio of less than 1% is desired, 7 packets would be choice. Taking into account all the previous discussions, Table 5 presents the jitter buffer parameters finally selected for a proper operation of the m-Health system.

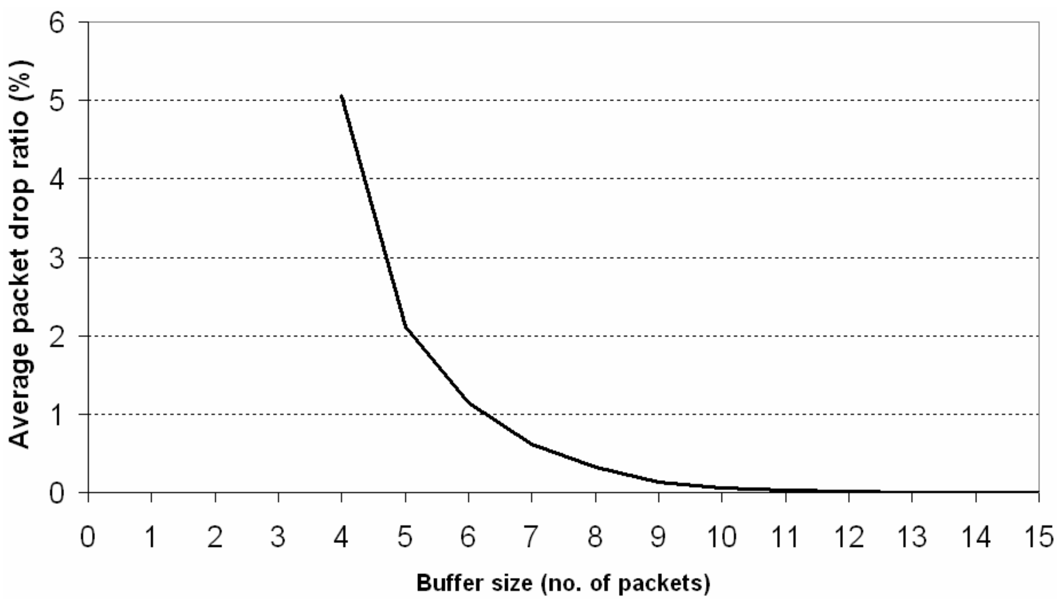


Figure 11: Average packet drop ratio for the audio service (threshold = 4)

Table 4: Average audio packet drop ratio vs. delay (buffer threshold = 4)

Buffer size (packets)	**Average packet drop ratio (%)**	**Typical deviation**	**Fixed delay (ms)**	**Maximum delay (ms)**
4	5.059	0.441	240	240
5	2.113	0.237	240	300
6	1.137	0.198	240	360
7	**0.619**	**0.153**	**240**	**420**
8	0.321	0.098	240	480
9	0.137	0.063	240	540
10	0.060	0.039	240	600
11	0.028	0.019	240	660
12	0.010	0.010	240	720
13	0.004	0.006	240	780
14	0.001	0.003	240	840
15	0.000	0.000	240	900

Table 5: Jitter buffer parameters

Medical user service	**Buffer threshold (packets)**	**Buffer size (packets)**	**Fixed delay (ms)**	**Maximum delay (ms)**
Audio	4	7	240	420
Medical data	3	4	3000	4000
Video	1	30	-	-

Conclusions

This chapter has presented the IPv4 performance of a wireless telemedicine system targeted specifically for critical and emergency medical scenarios. It offers simultaneous transmission of real-time clinical data (including ECG signals, blood pressure and blood oxygen saturation), videoconference, high-resolution still image transmission and other facilities such as multi-collaborative whiteboard, chat and web access to remote databases. Due to the nature of the services transmitted and since it is IP-based, home telecare and chronic patient telemonitoring are other application areas in which this wireless telemedicine system can be used.

The system architecture is based on 3G wireless networks and advanced signalling protocols (SIP/SDP) intended for setting up multimedia communication sessions between one or multiple clients, that also allow the integration of real-time multimedia services over multiple access channels supporting IPv4 and/or IPv6 protocols (depending on current commercial UMTS releases).

Real-time multimedia data transmission has been optimized specifically to operate over 3G wireless networks using the most appropriate codecs. IPv4 performance results show that the system performs reliably over IPv4-only UMTS accesses (64 Kbps in the uplink). The total bandwidth used fits in a 64 Kbps UMTS channel even when the most bandwidth-consuming codec rates and the lowest transmission efficiencies are used. Regarding jitter, IPv4 measurements allow dimensioning jitter buffers that improve playback quality of real-time services. The parameters selected for each jitter buffer and the design trade-offs and occupancy calculations are also presented.

The migration process towards next generation Internet and Fourth-Generation (4G) networks will require the use of the IPv6 protocol. As it has been stated before, this wireless telemedicine system is ready for IPv6, but it has been dimensioned for IPv4-only accesses so far. Therefore, the study of all system parameters and how could the IPv6 protocol affect system performance in any possible IPv4/IPv6 transition scenario will be the next step. The final objective of this work is to build a useful m-Health system ready for the current and envisioned technologies in the years to come. Finally, and taking into account current technologies, the integration of the wireless telemedicine system into a real IMS environment will also be considered in the future.

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