Analysis and Measurement of a Wireless Telemedicine System

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***Abstract*— The analysis and measurement of a multi-collaborative wireless telemedicine system operating over Third-Generation (3G) mobile networks is presented. It is designed to communicate the personnel of an ambulance with medical specialists in a remote hospital through a Universal Mobile Telecommunication System (UMTS) mobile access. The system architecture is based on advanced signalling protocols that allow multimedia multi-collaborative conferences in IPv4/IPv6 3G scenarios. The system offers real-time transmission of medical data and videoconference, together with other non real-time services. It has been optimized specifically to operate over 3G mobile networks using the most appropriate codecs. Measurements show a reliable performance over IPv4 UMTS accesses (64 Kbps in the uplink) and also allow to dimension certain system parameters in order to improve the quality of the services offered.**

Keywords-3G, IPv6, m-Health, Real-Time, Telemedicine

I. INTRODUCTION

Mobile Health (m-Health) is an emerging area of telemedicine in which the recent development 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 their capacity to serve 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 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 is presented in [4].

However, 2G-based systems lack the necessary resources to transmit bandwidth-demanding real-time medical data. The Third-Generation (3G) Universal Mobile Telecommunications System (UMTS) overcomes this and other limitations in order to support a large variety of services with different Quality of Service (QoS) requirements, thus extending the use of flexible telemedicine applications with higher transmission rates. UMTS introduces the IP Multimedia Subsystem (IMS) [5], an IPv6 network domain designed to provide appropriate support for real-time multimedia services. IMS uses the Session Initiation Protocol (SIP) as signalling and session control protocol. SIP allows operators to integrate real-time multimedia services over multiple access technologies (interworking multimedia domains).

This paper presents an analysis and measurement of a wireless telemedicine system designed for different emergency scenarios, as shown in Fig. 1. Medical specialists in the hospital take part in a multipoint conference with the personnel of an ambulance, receiving compressed and coded biomedical information from a patient, making it possible for them to assist in the diagnosis prior to his reception.

II. WIRELESS TELEMEDICINE SYSTEM ARCHITECTURE

The wireless telemedicine system (Fig. 2) has been built using standard off-the-shelf hardware, instead of developing propriety hardware as in [6], 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.

The details of the wireless telemedicine system architecture are shown in Fig. 3. The system comprises of the signaling and session control, medical user services and application control subsystems that will be described later 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 [7]-[8]. Information compression, coding and QoS control (data

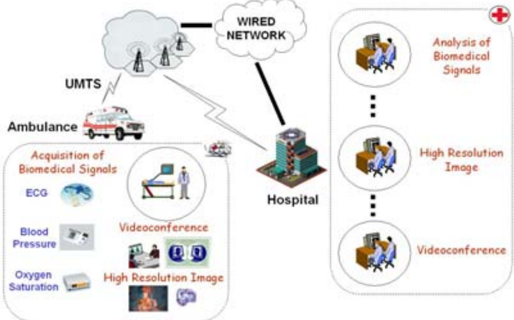

Figure 1. Typical medical emergency scenario

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Figure 2. Wireless telemedicine system

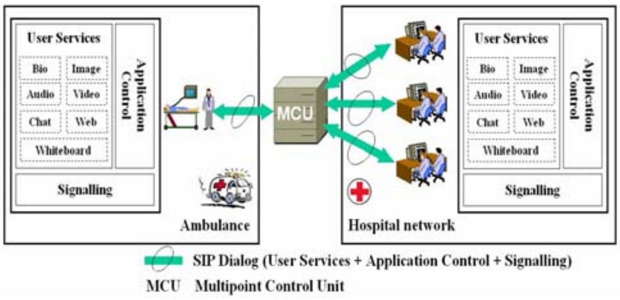

Figure 3. Block diagram of the wireless telemedicine system architecture and subsystems

prioritization, congestion control and dejittering) modules improve transmission efficiency of joint real-time and non real-time data over wireless channels in a more appropriate way than previous systems [9]. In addition, and also unlike [9], this system follows a multi-collaborative design, integrates new real-time multimedia features intended for 3G wireless networks, supports IPv4/IPv6 interworking 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: IPv4 ambulance connecting to an IPv6 hospital (first stages of IPv6 deployment in 3G networks), IPv6 communication through IPv4 islands and, finally, native all-IPv6 communication between them.

Communication between the remote ambulance 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 [10], which requires the existence of a Multipoint Control Unit (MCU). System users and the MCU exchange information associated with the different medical user services and its presentation (application control), and also communication and service quality management data (signalling and QoS control).

A. *Signalling and session control subsystem*

The developed signalling allows the exchange of the characteristics associated to the different information flows between the system elements and is based on standard protocols that favor 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. In order to do that, each element in the system has a SIP UA, slightly modified in the MCU to allow the use of multiple simultaneous dialogs.

Multipoint conference establishment, management and termination are performed by exchanging SIP messages between the different users. When a user connects, he creates a SIP dialog with the MCU, joining the conference. During the conference, SIP messages are exchanged between users and the MCU, varying conference characteristics and therefore allowing its management. In a similar process to that of conference joining, when a user wants to leave it, this fact must be communicated to the MCU with the necessary SIP messages. SIP messages also serve as the mean of transport of SDP messages with the description of medical user services.

In addition to session control functions (establishment, management and termination of the multipoint conference), the SIP protocol is also useful for user mobility purposes inside the IMS environment.

B. *Wireless medical user services subsystem*

The medical user services included in the wireless telemedicine system are associated with information shared in a multi-collaborative environment. Specifically, the system has services to share audio, ambient video, medical data information, high-resolution still images and graphical and textual information, as well as a web service that allows remote access to clinical information databases. In addition to these services, there is a service designed to exchange control information (application control), which is discussed later. Each kind of information is associated with a medical user service and uses a transport protocol and a codec according to its characteristics (see Table I). Hence, real-time services (audio, video and medical data information) use the Real-Time Transport Protocol (RTP), whereas the rest of the services use the Transmission Control Protocol (TCP). Furthermore, the exchanged information can be very sensitive and requires a secure communication. The wireless telemedicine system uses an IP Security protocol (IPSec) implementation supporting public key certificates in tunnel mode. This protocol ensures private communication of selected services.

The ECG signal is stored both in transmission and reception following the SCP-ECG standard. It is well known

TABLE I. CODEC OPERATION MODES FOR 3G REAL-TIME WIRELESS MEDICAL USER SERVICES

Medical user service	Codec	Codec Rate
Audio	AMR[a]	4.75, 5.15, 5.9, 6.7, 7.4, 7.95, 10.2, 12.2 (Kbps)
Medical data	WT[b]	5, 10 (Kbps)
Video	H.263	5, 10 (Frames per second, FPS)

[a] Adaptive Multi-Rate
[b] Wavelet Transform

that for an efficient transmission, an ECG compression technique has to be used. In our case, a real-time ECG compression technique based on the Wavelet Transform is used [11]. This is a lossy compression technique, therefore there is a trade-off between transmission rate and received ECG signal quality. Thus, there exists a minimum transmission rate to be used to transmit a useful ECG for clinical purposes, which was selected in collaboration with cardiologists after different evaluation tests. The minimum transmission rate used in our implementation (625 bits per second and per ECG lead) leads to a clinically acceptable received ECG signal. Regarding blood pressure, oxygen saturation and heart rate, these signals have low bandwidth requirements and, therefore, are not compressed.

The videoconference module captures, sends and plays audio and video information obtained by a web camera and a microphone. In order to reduce the bandwidth, these data are compressed and coded. The video signal is compressed following the H.263 standard, whereas the audio signal uses the Adaptive Multi-Rate (AMR) codec, recommended for UMTS by the 3GPP [12]. This module provides the basic functionality for starting, pausing and stopping video signals acquisition and representation, as well as volume control for the microphone (capture) and the speakers (reproduction). Due to the fact that each participant in the conference receives a unique video signal, the system allows the user to select the particular video signal among all the users connected.

The high-resolution still image module obtains high quality images with a Charge Copled Device (CCD) colour camera connected to the computer through an image acquisition card. This module includes options to preview the captured images and modify their main characteristics in real time: brightness, contrast, hue, etc. Captured images can be stored and transmitted in different formats, with various qualities and compression levels. These images are sent automatically to the electronic whiteboard module of the remote users, allowing to select and mark fixed areas in a multi-collaborative fashion to facilitate a diagnostic clinical procedure.

C. 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 [13]. The QoS metrics selected are packet loss ratio, delay variation (jitter) and octect-based IP packet throughput (bandwidth). In addition, the wireless telemedicine system has application dejitter 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 transmitted information 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, that 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.

According to the previous 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 due to congestion conditions, channel errors or handoffs, whereas packet drops occur in end application queues due to excessive jitter values. 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 [14], problems derived from channel errors and handoffs turn into transmission congestion problems. In addition, congestion conditions can also appear in the rest of the network. Thus, 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 to control 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 [9], 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. This is carried out to protect the quality of the high priority service. Finally, this service can be limited if worse conditions appear, but always keeping a minimum allowable quality. If congestion conditions disappear for a fixed time, the system raises codec rates in the reverse service priority order.

Regarding jitter caused by the variable nature of wireless links and the joint transmission of all services, each real-time service has an application dejitter buffer associated with it that tries to mitigate its effects. These buffers have been properly dimensioned to minimize jitter, delay and packets dropped.

D. Application control subsystem

The MCU forwards the information generated by each medical service according to the presentation spaces defined using the control service. Each medical service has a presentation space associated with it that defines the way in which the information has to be transferred and its destination.

The MCU simply forwards the information it receives, but has a special treatment for the audio, video, medical data and control services. Regarding the Audio service, the MCU decodes the signal of each user, mixes it with the decoded signal of the other conference participants and codes the result in order to transfer a unique audio signal to each user. On the other hand, the MCU only forwards one video signal to each conference participant. The particular video signal forwarded to each user is selected by using the control service. Finally, the medical data service is similar to the video service: medical data are only generated in the remote location, whereas the other conference participants can only receive them.

III. WIRELESS TELEMEDICINE SYSTEM MEASUREMENT

In order to measure the wireless telemedicine system performance and improve the quality of offered services by dimensioning dejitter buffers, several tests have been carried out using the system over 128/64 Kbps IPv4 only UMTS accesses in urban areas (good coverage level and speeds up to 50 Km/h, as well as static situations). The measurement scenario used is shown in Fig. 4. As the uplink is more restrictive, the results we are presenting here correspond to this 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 in 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. Finally, it is important to note that all IP datagrams transmitted have been received during the tests, which means that only datagram drop situations have occurred in our tests.

A. Average bandwidth results

Table II presents the results about average IP-level bandwidth used by real-time medical 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, 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

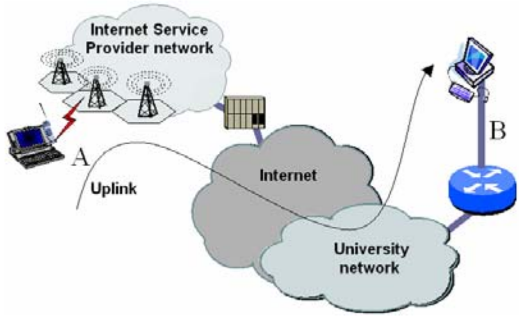

Figure 4. Measurement scenario

TABLE II. AVERAGE IP-LEVEL BANDWIDTH USED BY REAL-TIME USER SERVICES

Medical user service	Operation Mode		IP Bandwidth (Kbps)
Audio	Samples/Packet	Codec Rate (Kbps)	
	1	4.75	21.2
	1	12.2	28.8
	3	4.75	10.5
	3	**12.2**	**18.1**
Medical data	Bit Rate (Kbps)		
	5		5,3
	10		**10,3**
Video	FPS		
	5		16
	10		**24**

of audio samples per packet that can be used because more audio samples per packet yield more audio delay. For example, a more efficient transmission mode including four audio samples per packet every 80 ms causes four times the delay including only one audio sample per packet every 20 ms. 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. Regarding the video service, it is worth noting that the bandwidth shown in Table II 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 efficiency.

As it can be checked, 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 II, achieving a reasonable trade-off between bandwidth, transmission efficiency, delay and loss ratio. During normal operation, codec modes can vary in response to congestion conditions with the aid of the congestion control algorithm mentioned in subsection II.C.

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 of the points. Therefore, the network does not modify traffic characteristics regarding bandwidth and packet loss.

B. Jitter results

48 tests have been carried out every hour during 2 days, with all real-time medical services operating at the same time and at the highest codec rate. These tests are useful for observing the influences between the 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 140 ms of jitter due to the time it takes the 64 Kbps UMTS uplink channel to transmit big-sized medical data packets (around 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. It is important to note that we are not trying to model the network and understand all the causes of its behaviour, but designing the application to adapt to its variations and make a better use of the resources it offers.

Regarding the effects of audio and video over the medical data service, none of them have a significant influence 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 dejitter buffer.

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

C. Dejitter buffer dimensioning

Packet captures in point B are very useful to obtain the instantaneous application dejitter 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 subsection III.B, with all real-time medical services operating at the same time and at the highest codec rate, several buffer occupancy

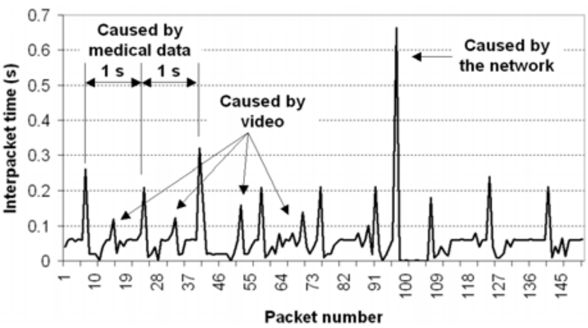

Figure 5. Audio interpacket time

calculations have been carried out. In the calculations, buffer size and threshold can be varied in order to obtain useful results to dimension dejitter buffers properly. First of all, the buffer threshold must be able to support the jitter effects caused by the real-time services presented in the previous subsection. Subsequently, the total buffer size must be able to support the 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.

1) 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. Due to the small jitter effects present on the medical data service, the buffer threshold can be selected to be minimum. 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, and as it has been stated before, the ambient video service is not critical, so the buffer threshold selected has been the minimum, i.e., 1 packet.

2) Buffer size

As it has been noted before, the total buffer size must be able to support the 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. 6, 7 and 8 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

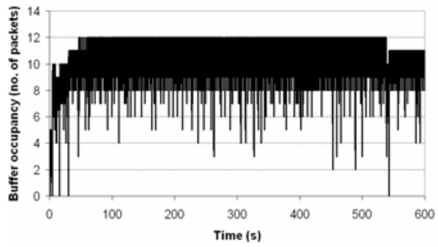

Figure 6. Audio service buffer occupancy (buffer threshold = 4 and infinite buffer)

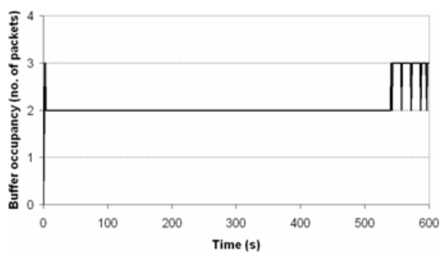

Figure 7. Medical data service buffer occupancy (buffer threshold = 3 and infinite buffer)

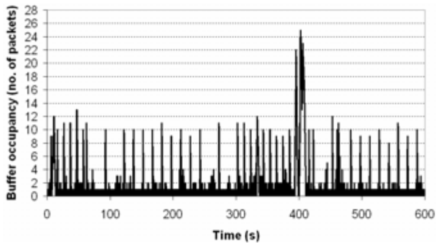

Figure 8. Video buffer service occupancy (buffer threshold = 1 and infinite buffer)

TABLE III. 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

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 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 conditions.

Finally, audio tests present certain 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 the size. The packet drop ratio has been averaged for all the tests and the results are shown in Table III as a function of buffer size. 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 buffer size. If, for example, a packet drop ratio of less than 1% is desired, 7 packets would be choice.

IV. CONCLUSIONS

This paper has presented a feasible wireless telemedicine system targeted specifically for critical and emergency medical scenarios. 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 networks and advanced signalling protocols (SIP/SDP) that allow the integration of real-time multimedia services over multiple access channels that support IPv4 and IPv6 interworking depending on current commercial UMTS releases. The system has the following features: 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. The system has been optimized specifically to operate over 3G mobile networks using the most appropriate codecs. Evaluation results show a reliable performance over IPv4 UMTS accesses (64 Kbps in the uplink) and also allow to dimension dejitter buffers in order to improve the quality of real-time services.

REFERENCES

[1] R.S.H. Istepanian, and J.C. Lacal, “Emerging Mobile Communication Technologies for Health: Some Imperative notes on m-Health”, Proc. of the 25th Silver Anniversary International Conference of the IEEE Engineering in Medicine and Biology Society, Vol. 2, pp. 1414-1416, 2003.

[2] R.S.H. Istepanian, E. Kyriacou, S. Pavlopoulos, and D. Koutsouris, “Wavelet Compression Methodologies for Efficient Medical Data Transmission in Wireless Telemedicine Systems”, Journal of Telemedicine and Telecare, Vol. 7, No. 1, pp. 14-16, 2001.

[3] R.S.H. Istepanian, B. Woodward, and C.I. Richards, “Advances in telemedicine using mobile communications”, Proc. IEEE Engineering Medicine and Biology Society, Vol. 4, pp. 3556–3558, 2001.

[4] R.S.H. Istepanian, S. Laxminarayan, and C.S. Pattichis (Eds), “M-Health: Emerging Mobile Health Systems”, New York, Springer, 2006.

[5] 3GPP TS 23.228 V6.8.0., “IP Multimedia Subsystem (IMS); Stage 2”, Release 6, 2005.

[6] J. Cullen, W. Gaasch, D. Gagliano, J. Goins, and R. Gunawardane, “Wireless mobile telemedicine: En-route transmission with dynamic quality-of-service management”, National Library of Medicine Symposium on Telemedicine and Telecommunications: Options for the New Century, 2001.

[7] S. Pavlopoulos, E. Kyriacou, A. Berler, S. Dembeyiotis, and D. Koutsouris, “A novel emergency telemedicine system based on wireless communication technology – AMBULANCE”, IEEE Trans. Inform. Technol. Biomed, Vol. 2, pp. 261-267, 1998.

[8] E. Kyriacou et al, “Multi-purpose HealthCare telemedicine systems with mobile communication link support”, BioMedical Engineering OnLine, Vol. 2, No. 7, 2003.

[9] Y. Chu, and A. Ganz, “A Mobile Teletrauma System Using 3G Networks”, IEEE Trans. Information Technology in Biomedicine, Vol. 8, No. 4, pp. 456-462, 2004.

[10] J. Rosenberg, “A Framework for Conferencing with the Session Initiation Protocol”, Internet draft, 2004. Work in progress.

[11] A. Alesanco, S. Olmos, R.S.H. Istepanian, and J. García, “A Novel Real-Time Multilead ECG Compression and De-Noising Method Based on the Wavelet Transform”, Proc. IEEE Computers Cardiology, Los Alamitos, CA, IEEE Comput. Soc. Press, pp. 593-596, 2003.

[12] 3GPP TS 26.235 V6.3.0, “Packet switched conversational multimedia applications; Default codecs”, Release 6, 2005.

[13] ITU-T Rec. Y.1540, “IP Packet Transfer and Availability Performance Parameters”, Dec. 2002.

[14] 3GPP TS 25.301 v6.2.0, “Radio interface protocol architecture”, Release 6, Mar. 2005.