Enhanced 3G-Based m-Health System

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Abstract — **The architecture of a multi-collaborative mobile telemedicine system operating over Third-Generation (3G) mobile networks is presented. The system architecture is based on advanced signalling protocols that allow multimedia multi-collaborative conferences in IPv4/IPv6 3G scenarios. It is designed to communicate the personnel of an ambulance with medical specialists in a remote hospital through a 3G mobile access, providing appropriate support for real-time transmission of medical data and videoconference, together with other non real-time services. The system has been optimized specifically to operate over 3G mobile networks using the most appropriate codecs and Quality of Service (QoS) mechanisms to improve the services offered.**

Keywords — **m-Health, 3G, Telemedicine, IPv6, QoS**

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. The wide coverage provided by cellular networks and their capacity to serve moving vehicles, have brought about new possibilities in telemedicine.

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 [1]. In recent years, several m-Health and wireless telemedical systems based on GSM have been reported [2], allowing remote diagnosis in mobile and hardly accessible environments. 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 latest advances on m-Health systems is presented in [3].

However, 2G-based systems lack the necessary resources to transmit bandwidth-demanding real-time (RT) 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. UMTS also introduces the IP Multimedia Subsystem (IMS) [4], an IPv6 network domain designed to provide support for RT multimedia services. The IMS uses the Session Initiation Protocol (SIP) as signalling and session control protocol [5]. SIP allows operators to integrate RT multimedia services over multiple access technologies.

This paper presents the architecture of a 3G-based m-Health system designed for different emergency scenarios (Fig. 1). With the aid of this system, several medical specialists in a hospital take part in a multipoint conference with the personnel of an ambulance, receiving compressed and coded biomedical information from a patient, assisting them in a diagnosis prior to his reception.

II. 3G M-HEALTH SYSTEM ARCHITECTURE

The 3G m-Health system (Fig. 2) has been built using standard off-the-shelf hardware, instead of developing propriety hardware as in [6], uses free software and commercial 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.

The details of the 3G system architecture are shown in Fig. 3. The system comprises of the signalling and session control, medical user services and application control subsystems, which 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]. Information compression, coding and

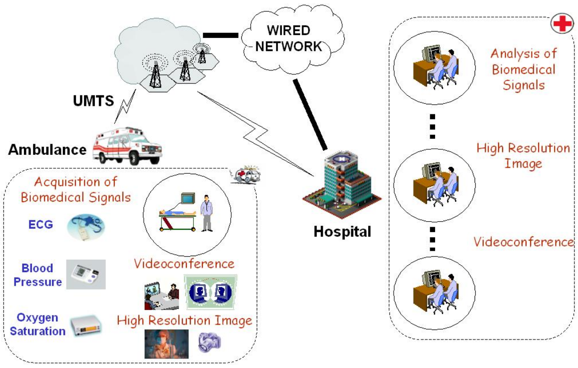

Fig. 1. Typical emergency scenario.

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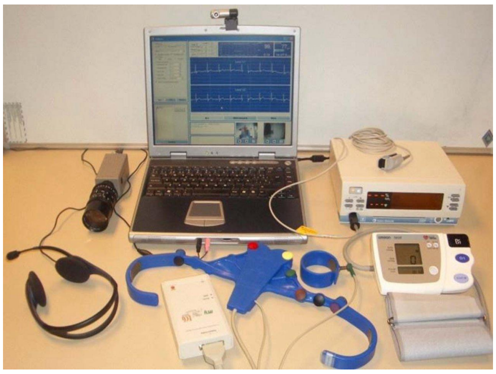

Fig. 2. 3G m-Health system.

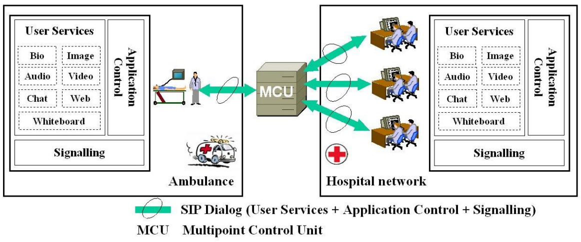

Fig. 3. System architecture.

QoS control modules improve transmission efficiency of joint RT and non real-time (NRT) data over wireless channels in a more appropriate way than previous systems [8]. In addition, and also unlike [8], this system follows a multi-collaborative design, integrates new RT 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). An 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 different types of multimedia traffic. The conference model selected is the *tightly coupled* [9], that requires a Multipoint Control Unit (MCU). The MCU maintains a dialog with each participant in the conference and is responsible for ensuring that the media streams that constitute the conference are available to the appropriate users.

A. Signalling and session control subsystem

The developed signalling allows the exchange of the characteristics associated to the different information flows between 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 services are described. In order to do that, each element in the system has a SIP User Agent (UA), slightly modified in the MCU to allow the use of multiple simultaneous dialogs.

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.

Multipoint conference establishment, management and termination is 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.

As the MCU constitutes a potential point of failure, the SIP protocol also allow system users to establish direct point-to-point sessions between them.

B. Wireless medical user services subsystem

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 information (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. 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 (Table I). Hence, RT services (audio, video and medical data) use the Real-Time Transport Protocol (RTP), whereas the rest of the services use the Transmission Control Protocol (TCP).

Table I also shows codec operation modes for each RT service. Every codec operation mode consumes a portion of the total channel bandwidth, that translates into different values of bandwidth at the IP level. The maximum average values of this bandwidth have been measured and are also displayed in Table I. On one hand, the total average IP bandwidth consumed by RT services is 52.4 Kbps, which gives enough room for NRT services in a 64 Kbps UMTS access. On the other hand, NRT services use TCP built-in mechanisms to adapt to the available network resources, so they can be considered as Available Bit Rate (ABR) services. Thus, the IP-level bandwidth consumed by them varies depending on channel utilization.

C. QoS control subsystem

The QoS in this system is mainly determined by the fluctuant characteristics of the UMTS link, 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

TABLE I: CHARACTERISTICS OF MEDICAL USER SERVICES

	Timing requirements	Bandwidth requirements	Transport protocol	Codec	Codec operation modes	Maximum average IP bandwidth (Kbps)
Medical data	RT	Low	RTP	WT[a]	5 10 (Kbps)	10.3
Audio	RT	Medium	RTP	AMR[b]	4.75 5.15 5.9 6.7 7.4 7.95 10.2 12.2 (Kbps)	18.1
Video	RT	High	RTP	H.263	5 10 (Frames per second, fps)	24
Chat	NRT	Low	TCP	-	-	ABR[c]
Electronic whiteboard	NRT	Low	TCP	-	-	ABR
Still image	NRT	Medium	TCP	-	-	ABR

a. Wavelet Transform b. Adaptive Multi-Rate c. Available bit rate

Y.1540 [10]. The QoS metrics selected are packet loss ratio, delay variation (jitter) and octect-based IP packet throughput (bandwidth).

The first measure adopted to optimize the use of shared resources among all medical services has been to prioritize them. The most priority services are the medical data, audio and ambient video (in this order), that will take up most of the channel resources. The characteristics of these services are monitored at transmission and reception and are taken into account to increase or decrease codec rates by sending SIP/SDP messages. NRT services, on the other hand, will be treated best-effort, adapting to the spare network resources using TCP built-in mechanisms.

One of the main causes of poor QoS is packet loss. As the commercial 3G wireless cellular service used in this system operates in the Acknowledged Data Transfer mode at the radio link layer, packet losses are considered to be produced only by network congestion. Part of the QoS control process is based on congestion control, whose signalling can be implicit, activated by packet drops in transmission queues, or explicit, initiated in reception. The congestion control algorithm selected is that presented in [8], but considering RT service priorities.

Regarding jitter, it can be caused by the variable nature of wireless links or by the joint transmission of all services. Each RT service has an application dejitter buffer associated with it that has been properly dimensioned to minimize jitter, delay and packets dropped. They are First In-First Out (FIFO) buffers, but considering that 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.

D. *Application control subsystem*

The MCU forwards the information generated by each medical service according to its associated presentation space defined using the application control service (Fig. 4). A presentation space defines the way in which the information has to be transferred and its destination. The MCU forwards the information it receives, but first mixes all the audio signals in order to transfer a unique signal to each user. It also forwards only one video signal to each conference participant. Medical data are only generated in the ambulance for the rest of the users.

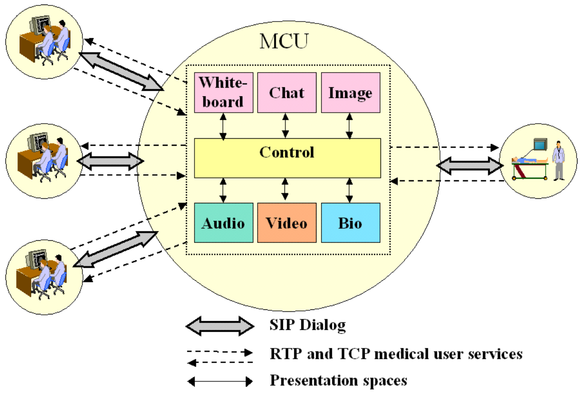

Fig. 4. Presentation spaces.

III. RESULTS

The 3G m-Health system has been tested in order to measure its performance and to improve QoS by dimensioning dejitter buffers. Several tests have been carried out in urban scenarios (good coverage level and low speed, as well as static vehicles) using the system over 64/128 Kbps (Uplink/Downlink) IPv4 UMTS accesses. As the uplink is more restrictive, the results presented correspond to this connection sense.

Several tests have been carried out during various 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, that has been considered to be 10 minutes in a medium-sized Spanish city like Zaragoza.

A. *Average bandwidth results*

As Table I shows, the total bandwidth consumed by all RT and NRT medical user services fits in a 64 Kbps UMTS channel, even when the most bandwidth-consuming codec rates are used. During normal operation, codec operation modes can vary in response to congestion conditions with the aid of congestion control mechanisms. In addition, no packet losses have been observed in any test, therefore the network does not modify traffic characteristics regarding packet loss.

B. *Jitter results*

48 tests have been carried out every hour during 2 days, with all RT medical services operating at the same time

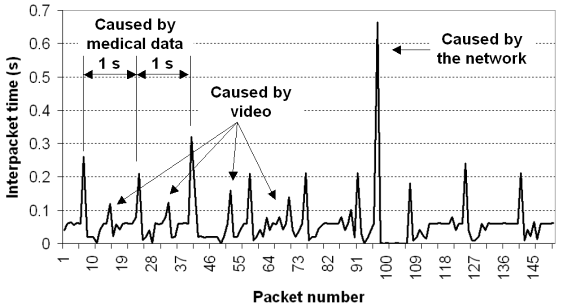

Fig. 5. Audio service interpacket time.

and at the highest codec rate. These tests are useful for observing the influences between traffics generated by each RT service.

Fig. 5 presents a zoom over 9 seconds of audio interpacket time taken in a test at the receiver. Medical data packets are generated every 1 second, therefore jitter effects over the audio service appear uniformly spaced. Regarding video, packets are smaller and not uniformly spaced, so the effects of video over audio packets are smaller. In addition, other jitter effects are caused by the network.

The audio and video services do not have a significant influence over the medical data service due to the fact that medical data packets are very spaced between them (1 second, ideally), and a small jitter effect is not noticed in reception with the aid of a minimal dejitter buffer. The last RT service, the ambient video, also suffers jitter effects caused by the rest of RT 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, so these jitter results are not presented here. In short, a big dejitter buffer is enough to support all possible jitter effects.

The buffer threshold value must be able to support jitter effects caused by RT services, whereas the total buffer size value must be able to support jitter effects caused by the network. High values of the threshold cause less situations in which the buffer becomes empty (playback interruptions), but, on the other hand, introduce a bigger fixed delay. In the same manner, bigger buffers allow less packet drops, but also entail bigger delays to enqueued packets.

Using packet timestamps and theoretical packet buffer empty times from the tests several buffer occupancy calculations varying buffer size and threshold have been carried out in order to dimension dejjiter buffers. For example, Fig. 6 shows the different values of the average buffer packet drop ratio as a function of the total buffer size for the audio service. Not only packet drop ratio, but also the maximum delay that a particular buffer size causes are the relevant parameters in order to select buffer size. The dejitter buffer parameters selected for each RT service are presented in Table II.

IV. CONCLUSIONS

This paper has presented the architecture of a feasible 3G-based m-Health system targeted specifically for emergency medical scenarios that can also be used in the home telecare area. The system architecture is based on 3G networks and advanced signalling protocols (SIP/SDP) that allow the integration of RT multimedia services over multiple access channels that support IPv4/IPv6 interworking depending on current commercial UMTS releases. The system has the following features: simultaneous transmission of RT clinical data, videoconference and other NRT medical services. 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 to improve the quality of RT services.

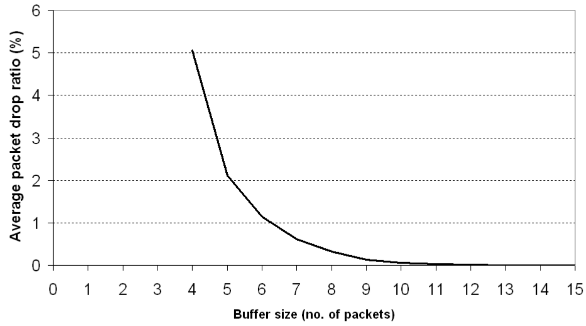

Fig. 6. Average packet drop ratio for the audio service (threshold = 4).

TABLE II: DEJITTER BUFFER PARAMETERS

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

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