Performance Analysis of Multiplexed Medical Data Transmission for Mobile Emergency Care Over the UMTS Channel

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***Abstract*—In this paper, a third-generation universal mobile telecommunications system (UMTS) solution for the delivery of biomedical information from an ambulance to a hospital is presented. The joint transmission of voice, real-time video, electrocardiogram signals, and medical scans in a realistic cellular multiuser simulation environment is considered, taking into account the advantages and particularities of UMTS technology for such transmission. The accomplishment of quality of service constraints for different services is investigated and quantitative results are provided in order to demonstrate the feasibility of using UMTS technology for emergency care services on high-speed moving ambulance vehicles.**

***Index Terms*—Mobile telemedicine, multimedia traffic, quality of service (QoS), universal mobile telecommunications system (UMTS).**

I. Introduction

TELEMEDICINE takes advantage of telecommunications services to offer expert-based medical care to any place where health care is needed and at any time. Thanks to the technological advances of recent years, today we can affirm that telemedicine is a real possibility. When the first telemedicine services were provided, the means to transmit the medical information between remote zones were based on fixed networks like plain old telephone service or integrated services digital network [1], [2]. 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 capability of providing service to moving vehicles. Then, via satellite-based solutions [3]–[5] and those based on second-generation (2G) mobile technology global system for mobile communications (GSM) [6]–[8] appeared, allowing the accomplishment of remote diagnosis in mobile environments as well as communication to geographic zones inaccessible by fixed networks.

Today, telemedicine is involved in a context where the emergence of new multimedia mobile services joint to the strong growth of the Internet will lead to a dramatic change in the concept of telecommunication networks and services. In fact, the new multimedia bearer services will be the result of the confluence of telecommunications, computers, and audiovisual industries. The result of this confluence is a new concept of society, the so-called Information Society, where the user will have access to any information, in any place and any time. This claim should be satisfied with the third-generation (3G) mobile systems, in which the universal mobile telecommunications system (UMTS) is included as the European proposal chosen by the European Telecommunications Standards Institute. Even though 3G telecommunications systems are not yet in widespread operation throughout the world, they will bring new possibilities but also challenges to operators, service providers, and vendors.

GSM is the technology of today's 2G voice mobile phones. It can be used to carry circuit-switched data at a maximum bit rate of 9.6 kb/s, requiring an end-to-end connection for each call and for the whole duration of that call. These restrictions make it infeasible to support a multimedia application like multiplexed medical data transmission over this network. The enhanced GSM schemes (2.5G cellular systems) use multislot operations to give users access to multiple channels, so they can attain higher data rates. The General Packet Radio Services (GPRS) system works over voice GSM wireless networks and is principally aimed at bringing convenient wireless access to web type services providing data rates up to 171.2 kb/s under ideal conditions. However, as GPRS traffic is in contention with voice traffic for access to the network, the major constraint with the 2.5G packet technologies is cell capacity, because packet-switched data share the same limited physical channels (time slots in time-division multiple access (TDMA) frames) with GSM circuit-switched voice connections. Therefore, during times of high traffic load, users will compete for available timeslots with voice connections, which usually have higher priority. This means that the practical data rate for the current packet technologies can be very low (compared to the theoretical maximum rate) due to network congestion, not being able to make an efficient use of the multislot function. Thus, GPRS is suitable for delay tolerant data applications, such as consumer mobile web access, where significant volumes of data are sent irregularly.

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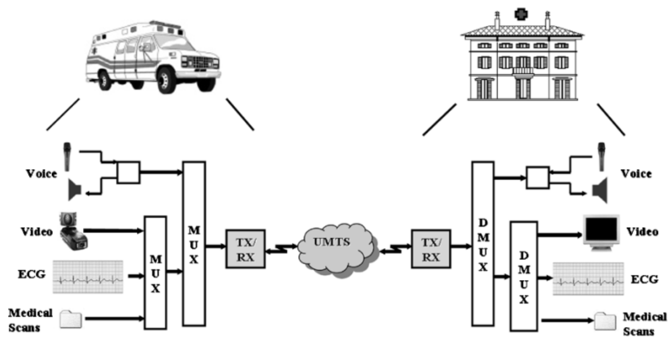


Fig. 1. Structure of the mobile emergency care over the UMTS system.

UMTS overcomes these limitations supporting a large variety of services with different QoS requirements. However, this fact makes the network design and management much more complex when compared to GSM, and even to 2.5G systems such as GPRS. New applications require networks to be able to handle services with variable traffic conditions keeping the efficiency in the network resources utilization. UMTS air interface has to be able to cope with variable and asymmetric bit rates, up to 2 Mb/s and 384 kb/s in indoor and outdoor environments, respectively, with different quality of service (QoS) requirements (mainly bit and frame error probabilities and delay) such as multimedia services with bandwidth on demand [9]. Effective access protocol is also essential for the UMTS air interface to handle bursty real-time and nonreal-time data [10].

In this kind of scenario, where the presented study is carried out, the emergence of 3G mobile wireless networks will allow us to extend the use of telemedicine applications thanks to the provided higher transmission rates and flexibility in front of previous technologies such as GSM or GPRS [6], [11]. Although previous studies exist on communications between an ambulance and a hospital [12], the work presented here is related to the accomplishment of this communication by means of UMTS. Today, UMTS technology can support the joint transmission of not only voice signals, but also visual information, such as high-quality still images and video, and biological signals, such as electrocardiogram (ECG) and blood pressure (Fig. 1).

QoS provision lies with an efficient sharing and managing of the radio resources, in this case power and interference, in addition to time allocation. An efficient implementation of radio resource management (RRM) strategies such as power control, adaptive transmission rate, time scheduling, and call admission control, allows us to increase the number of users, while new call blocking and handover call dropping probabilities are minimized. In this context, we propose an adaptive rate scheme for multimedia medical data multiplexing in order to reduce the required resources from the telemedicine application adapting them to the time variant channel conditions of the system. In order to evaluate the capability of UMTS for supporting QoS and, particularly, the proposed adaptive rate multiplexing scheme, an application scenario lying in a telemedicine service in the environment of an ambulance or mobile care unit has been implemented. This allows us to evaluate the system performance both in a qualitative and a quantitative way. In this paper, the joint transmission of voice, ECG, video, and medical scans has been investigated by using a model of UMTS network programmed in C++. For the accomplishment of the proposed objectives, the characteristics of real time, reliability, interferences, and requirements of bandwidth pointed in [13] have been considered and both a physical level and a system level simulator have been developed for the UMTS terrestrial radio access (UTRA) frequency-division duplex (FDD) system.

The remaining paper is organized as follows. In Section II, a brief overview of the UMTS system functionalities, services, and service capabilities is given. In Section III, the traffic features and quality constraints of the service components included in the multimedia application are described. According to these features, the selected multiplexing scheme is presented. We discuss performance results in Section IV and, finally, conclusions are provided in Section V.

II. UMTS System Overview and Applications

The main technological feature of the UMTS system is the use of the wide-band code-division multiple access (WCDMA) scheme in the air interface [14], [15]. In code-division multiple access (CDMA), user data information is spread over the shared bandwidth using a distinct spreading code that allows almost perfect isolation of each user's transmission resources from those of all other users. One of the key requirements of the UMTS air interface is the support for multiplexing different services with different QoS requirements on a single connection. Table I shows the different traffic classes defined in the UMTS standard, according to their QoS requirements [16]. The main advantage of the WCDMA air interface to provide these differentiated services is the option of variable transmission data rates through different spreading factors (ratio between shared bandwidth and user data rate), multicode transmission, and coding schemes. However, this flexibility requires a more complicated management in front of GSM/GPRS, given that the real available capacity is limited by the overall interference in the system.

The acceptance of a new user connection must be conditioned by the fact that signal-to-interference ratio values can be achieved by each existing connection once a new one is activated. Once a new connection is accepted, achieving each user transmission rate, delay requirement, and error rate is closely connected with power allocation. In fact, power is the common shared resource for users. In the downlink, the total transmitted power of a radio frequency carrier is shared between the users transmitting for the base station (BS), whereas, in the uplink there is a maximum tolerable interference level at the BS receiver that is shared between the transmitting MSs in the cell, each contributing to the interference (Fig. 2). So, a good interference handling performed by radio resource allocation schemes plays an important role to guarantee the performance and to increase the system capacity.

Power contribution of a particular user to the total interference depends on the required data rate and bit and packet error rate. The latter are related to the signal-to-interference ratio. Users moving at high speeds have worse channel conditions and, thus, require higher bit energy to interference (E_b/N_o) values

TABLE I
UMTS QoS CLASSES

Traffic class	Conversational class	Streaming class	Interactive class	Background
Fundamental characteristics	Preserve time relation (variation) between information entities of the stream. Conversational pattern (stringent and low delay)	Preserve time relation (variation) between information entities of the stream.	Request response pattern. Preserve payload content.	Destination is not expecting the data within a certain time Preserve payload content
Example of the application	Voice	Streaming video	Web browsing	Background download of emails

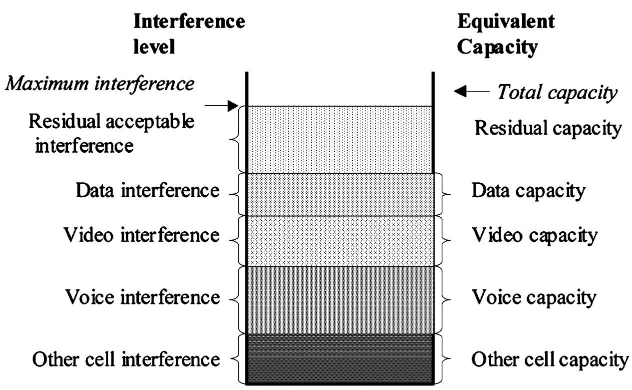


Fig. 2. Capacity distribution in a WCDMA system.

for the same QoS constraints, where the term N_o includes the thermal noise in addition to interference level. As shown in Fig. 2, the maximum interference level takes into account both intercell and intracell interference.

Provision of QoS in the UMTS air interface is related to functionalities of the radio interface protocol architecture, which is shown in Fig. 3. The radio resource control [17] configures the characteristics of the lower layer protocol entities in the air interface, including parameters for the physical, transport and logical channels, whereas, it offers services to higher layers for signaling mobility management, call control, and session management. The packet data convergence protocol (PDCP) [18] contains compression methods, which are needed to get better spectral efficiency for services requiring Internet protocol packets to be transmitted over the radio interface. The radio link control (RLC) protocol [19] provides segmentation and reliable transmission services for both user and control data. This reliability is provided by means of forward-error correction (FEC) and automatic repeat request (ARQ) techniques. To reduce bit-error rate, the first uses coding data adding overhead, whereas, the second uses retransmissions of erroneous packets increasing delay. Depending on the specific requirements of the data to be sent, it is necessary to choose one of these techniques or a combination of them. In fact, the UMTS system considers three different modes in RLC configured by the RRC: transparent mode (TM), unacknowledged mode (UM), and acknowledged mode (AM). TM and UM are based on FEC and are suitable for hard delay constraint services where the presence of some packets with errors is not critical. AM, based on joint FEC and ARQ, is suitable for nondelay constrained applications which require to be error-free.

Another important feature of the RLC mode is the use of early discard, which allows us to drop in the transmitter packets that have exceeded the maximum tolerable delay, reducing the delay of the following packets. The time a packet is allowed to stay in the RLC buffer is controlled by a timer (discard timer) whose value is signalled by upper layers. We have considered a packet-dependent value according to

$$T_{\text{Disc},i} = T_{\text{margin}} + T_{\text{TX},i} = T_{\text{margin}} + \frac{L_i}{R_{\text{target}}} \qquad (1)$$

where L_i is the length of packet $i, T_{\text{TX},i}$ the time to transmit packet i at R_{target}, which is the mean rate required by the service, and T_{margin} an additional provided margin.

The main functions of the medium access control (MAC) [20] layer for transmitting multimedia traffic include logical and transport channel (TrCh) mapping, selection of transport format, priority handling, and dynamic scheduling. A set of logical channel types is defined for the different kinds of data transfer services offered by MAC. They can be dedicated, shared, and common channels. A logical channel is defined by the type of transferred information, dedicated traffic channel (DTCH) is a point-to-point channel dedicated to one mobile station (MS) for the transfer of user information, whereas, dedicated control channel (DCCH) transmits control information between the MS and the network. Several logical channels, belonging to different services (e.g., video, audio, etc.) can be jointly transmitted using different TrChs. Each logical channel can be assigned to a different TrCh, or some logical channels can be multiplexed over the same TrCh. The set of TrChs is multiplexed over the coded composite TrCh [21], [22]. Each of the multiplexed TrChs may have variable data rate on a transmission time interval [(TTI) whose allowed values in UMTS are 10, 20, 40, and 80 ms] by TTI basis, including zero rate. Each combination of rates on the individual channels results in a certain data format to be transmitted, defining the total number of bits per 10-ms radio interface frame and their assignment to the individual channels [transport format combination (TFC)]. The identity of the transport format is transmitted on a physical control channel (DPCCH), whereas, the multiplexed local channel data forms a physical data channel (DPDCH).

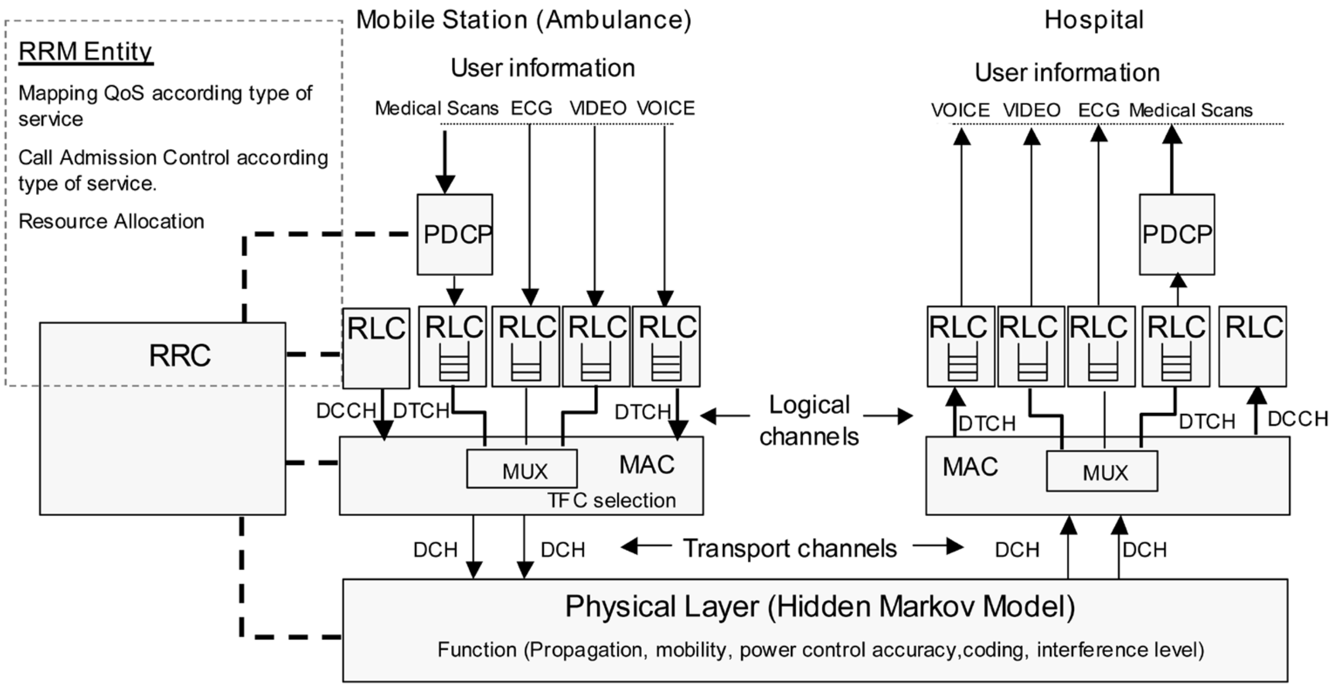


Fig. 3. UTRA-FDD radio interface protocol architecture

For successful transmission of user information, it is necessary to select suitable channels to match the QoS requirements. The biomedical multimedia application can be considered as a set of traffic components of different nature and different QoS requirements, including loss-sensitive, delay-sensitive traffic, and best-effort traffic sources, all requiring differentiated treatments by the network. It is possible to penalize a low priority service to better utilize radio resources. In fact, it could be also possible to consider that a user is enabled to assign itself a different degree of importance (static priority) to each component. The penalization lies in a temporary reduction of the bit rate assigned to that service. If propagation channel conditions become worse, it is also possible to temporary suspend the transmission of a service.

III. Service Multiplexing for the Telemedical Application

The purpose of this work is to emulate a realistic environment for a real-time telemedical mobile system such as the communication between an ambulance and a hospital. Thus, the goal is to achieve the correct transmission of all the important medical signals in the uplink (from the vehicle to the hospital). The UMTS system can provide the resources to accomplish this, but an intelligent resource management is required to get the QoS requirements guaranteeing system efficiency. In this section, traffic sources are described first since they determine the decision of the service multiplexing scheme.

A. Traffic Sources

Four different traffic sources have been considered [voice, video, ECG signal, and medical data, via file transfer protocol (FTP)]. The models that have been selected to generate the transmitted information for each service are described next. Their QoS requirements are shown in Table II.

TABLE II
QoS Requirements

Services	Data Rate	Maximum Delay	Packet Loss
Audio	4-25 kbps	150 – 400 ms	3 %
Video	32-384 kbps	150 – 400 ms	1%
ECG	1-20 kbps	~ 1 s	Zero
FTP	N.A.	N.A.	Zero

To model voice traffic, a classical two state ON–OFF Markov model has been used. In this model, packets are only generated during talk spurts (ON state) with fixed interarrival time (adaptive multirate (AMR) speech codec at 12.2 kb/s). The times spent in ON and OFF states are exponentially distributed with means T_{ON} (0.4 s) and T_{OFF} (0.6 s), respectively. Voice traffic transmission implies long silent periods (OFF state), which can be used to distribute the liberated resources among the least priority services. A streaming video traffic model, based on measurements of H.263 encoded video, has been implemented. It is based on a simple linear model described in [23]. By letting the model parameters be state dependent, and by letting a Markov chain control the states, the model has the flexibility to capture nonstationary dynamics present in the real streaming video traces. The target bit rate is set to 32 kb/s. The model assumes a constant video frame rate (7.5 frames per second), with variable frame size. In FTP applications, a session is modeled by a sequence of file (or medical scans) transfers, separated by reading time. The model proposed in [24] has been considered in this work. The two main parameters of an FTP session are the size of the file to be transferred and the reading time (time interval between end of download of the previous file and the user request for the next file). File sizes are randomly generated

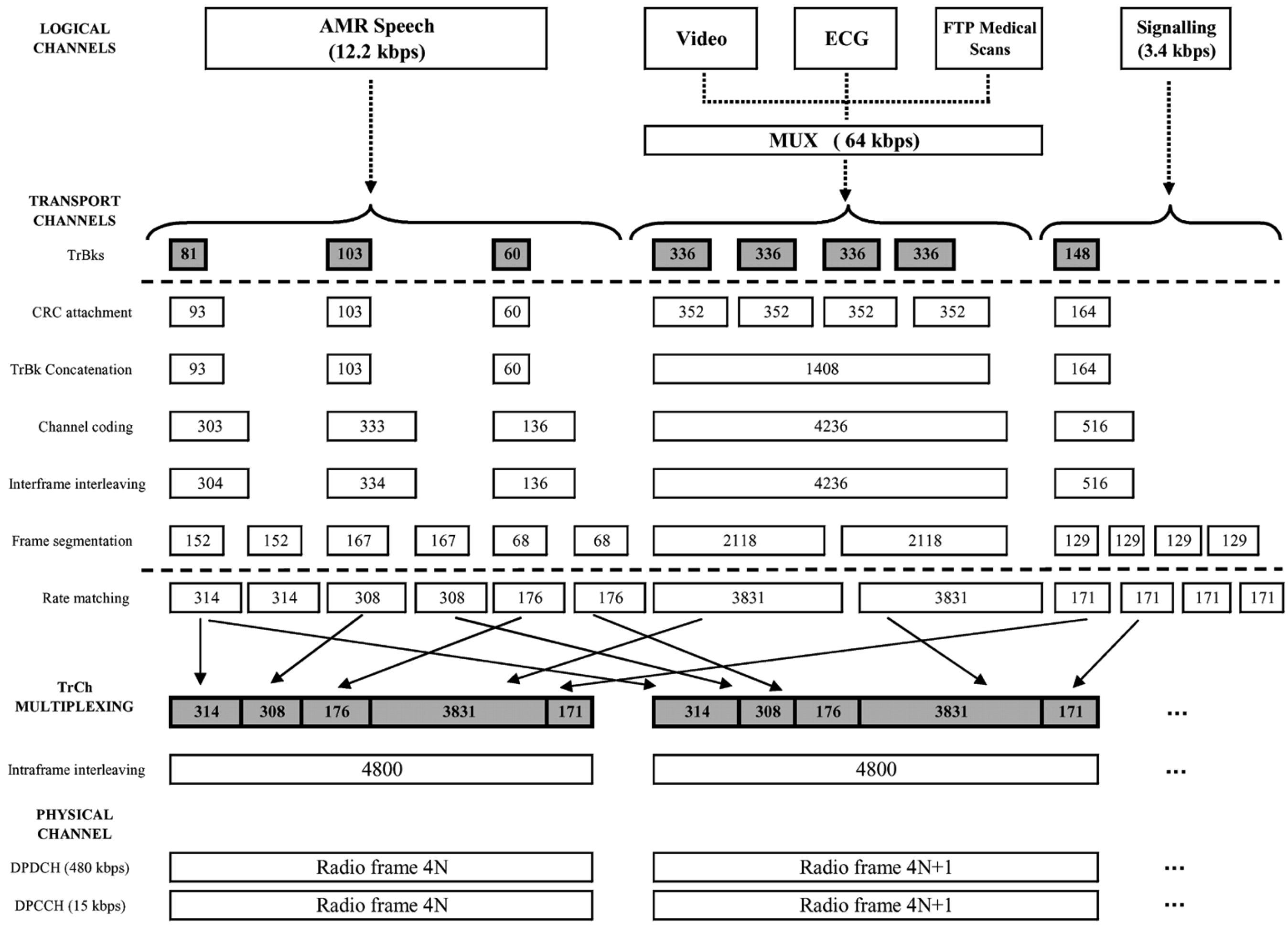


Fig. 4. Channel coding, mapping of logical channel onto TrChs and multiplexing scheme.

using a truncated lognormal distribution (Mean: 2 MB. Deviation: 0.722 MB. Maximum: 5 MB) and the reading times are exponentially distributed (Mean: 180 s).

The ECG signal to be transmitted results from the monitoring of the patient in the ambulance. The analog signal is digitalized: sampled at 250 Hz and quantized with 12 bits per sample. Three leads have been considered enough to provide the necessary clinical information although up to 12 leads can be transmitted. Some real-time time techniques can be developed for ECG data compression [25], allowing us to get compression rates up to 8 : 1. This compression reduces the transmission rate allowing to assign the saved bandwidth to other multiplexed services.

B. Service Multiplexing

The multiplexing scheme shown in Fig. 4 has been decided taking into account the different service features regarding both QoS parameters and traffic nature. Five different services (logical channels, DTCHs) are transmitted over three available channels (TrChs) multiplexed in one physical data channel (DPDCH + DPCCH). A 3.4-kb/s signalling bearer is required to exchange control information of higher levels. The 12.2-kb/s AMR speech service is split into three TrChs [26]. Video, ECG, and medical scans via FTP have been mapped onto a single TrCh (DCH) using different transport formats (64, 48, 32, and 16 kb/s). In this study, voice is considered as a priority service, so first it is decided if, according to power constraints, voice could be transmitted. Once voice is allocated, video, ECG, and FTP traffic demands are taken into account.

Different RLC modes are selected for both services. The TM mode is considered for voice traffic, since it does not have hard restrictions about errors (3% packet error rate) but it is delay sensitive. AM is selected for the multiplexing of video, ECG, and FTP applications. Early discard will not be applied to the ECG and FTP transmissions, as they require data integrity. However, video traffic is error tolerant, so video packets will be discarded early in the transmitter when their delay is higher than acceptable, reducing the delay of the following packets. Erroneous video RLC segments will be retransmitted making use of the available bandwidth providing that the delay is acceptable. In this way, packet loss for video transmission is reduced, keeping delay constraints.

When different services are transmitted over the same TrCh, the MAC entity has to multiplex upper layer packet data units (PDUs) from the different RLC entities into the transport block sets delivered to the physical layer. In [27], a dequeueing strategy is proposed: RLC PDUs are stamped with a maximum waiting time (WT) that depends on the QoS requirements of the service. RLC PDUs with a lower WT are first dequeued. For video packets, this time has been considered equal to the discarding time. For ECG packets, the maximum WT is set to

TABLE III
MULTIPLEXING SERVICES COMBINATIONS

	Services (priority order)					CCTrCh	
	Signaling	**AMR Speech**			**Video + ECG + FTP**		
	TrCH0	TrCH1	TrCH2	TrCH3	TrCH4	Rs (kbps)	Es/No (dB)
TFC 0	ON		Audio ON		64 kbps	480	-5.980
TFC 1	ON		Audio ON		48 kbps	480	-6.497
TFC 2	ON		Audio ON		32 kbps	240	-5.101
TFC 3	ON		Audio ON		16 kbps	240	-6.905
TFC 4	ON		Audio ON		-	120	-6.913
TFC 5	ON		Audio OFF		64 kbps	480	-6.574
TFC 6	ON		Audio OFF		48 kbps	480	-7.728
TFC 7	ON		Audio OFF		32 kbps	240	-6.117
TFC 8	ON		Audio OFF		16 kbps	120	-5.546
TFC 9	ON		Audio OFF		-	30	-5.252
TFC 10	ON		-		64 kbps	480	-6.769
TFC 11	ON		-		48 kbps	240	-4.946
TFC 12	ON		-		32 kbps	240	-6.474
TFC 13	ON		-		16 kbps	120	-6.193
TFC 14	ON		-		-	15	-5.406
TFC 15	-		-		-	0	-

TABLE IV
TRANSPORT FORMAT PARAMETERS

		Rate	**Channel Coding**	**TTI**	**QoS Requirements**		**Eb/No**
					BER	**BLER**	
Signalling	TrCH0	3,4 kbps	1/3 Convolutional	40 ms	-	1.0E-02	4.05
AMR Speech	TrCH1	12,2 kbps	1/3 Convolutional	20	5.0E-04	-	4.49
	TrCH2		1/3 Convolutional	20	1.0E-03	-	3.36
	TrCH3		1/2 Convolutional		5.0E-03	-	3.31
Data (Video + ECG + Medical Scans)	TrCH4	64 kbps	1/3 Turbocoding	20	-	1.0E-02	2.25
		48 kbps					2.25
		32 kbps					2.77
		16 kbps					3.45

300 ms. Since FTP transfer does not have delay restrictions, its packets do not have a specific WT and they are enqueued always behind video and ECG packets, so they are transmitted in the remaining bandwidth after video and ECG transmission.

Table III summarizes every possible TFC, that is to say, which TrChs are active and which transmission rates they have associated. The main TrCh parameters are included in Table IV. QoS constraints and required E_b/N_o of these services in a fast fading vehicular environment (120 Km/h) are also included.

This selection is adaptively done TTI to TTI according to the traffic demands and the maximum allowed transmission power (depending on the mobile equipment and the restrictions imposed by the network). Information is transmitted at the highest available speed. Transmission rate is only reduced if there are not enough packets in the buffer. In this table, TFC15 is the null combination, which means that, due to power restrictions, it is not possible to transmit anything. Audio ON represents the ON state in the traffic voice model. According to 3GPP specifications, in the OFF state, a special indication in the first frame of a silent period is sent, which is modeled as Audio OFF. Silent periods are represented as no-voice transmission.

IV. PERFORMANCE EVALUATION

In order to assess the performance of the multiplexing scheme and the different proposed strategies, a system level simulator for the UTRA (UMTS Terrestrial Access Radio) FDD system, programmed in C++ [27], [28] has been developed. The system level simulator allows us to evaluate the performance of different RRM strategies, including several traffic sources, propagation conditions, mobility models (vehicular speed from 50 to 120 km/h) [29], [30], and results from a physical layer simulator [31], [32]. The UMTS protocol stack (PDCP, RLC, and MAC) and the multiplexing of logical and TrChs are implemented. Off-line results from the physical level simulator (bit-error distribution according to E_b/N_o) are included through a Hidden Markov chain. This physical level simulator considers different aspects concerning the physical layer (Doppler shift, channel estimation, synchronization, power control, etc.) including channel coding and interleaving for all TFCs. Handover procedures are assumed to be ideal in the simulations. Anyway, UMTS allows for the use of soft handover (the mobile equipment transmits and receives

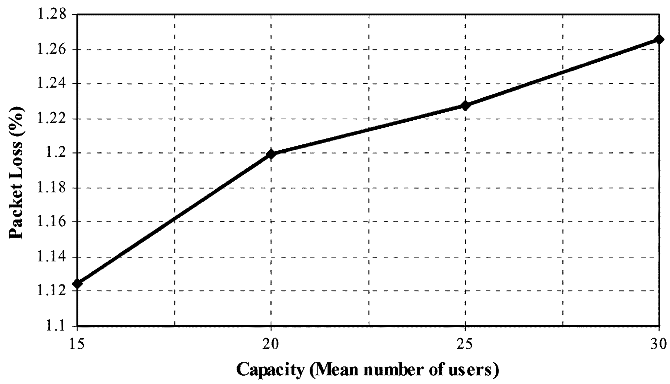


Fig. 5. Packet loss for voice transmission.

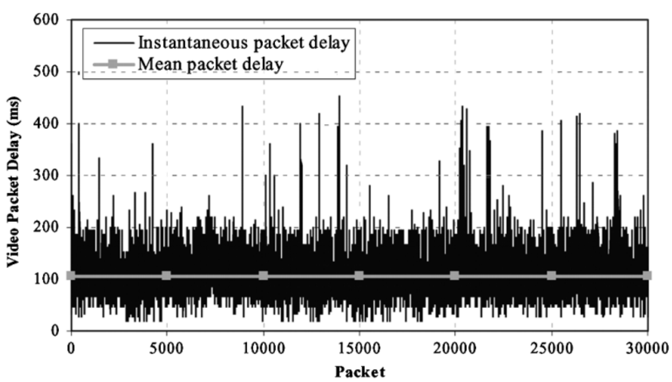
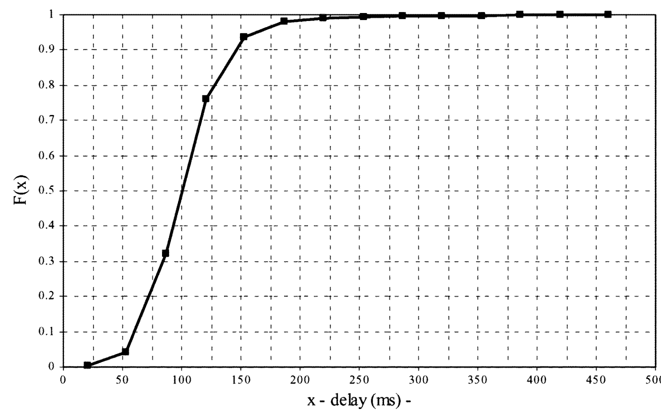
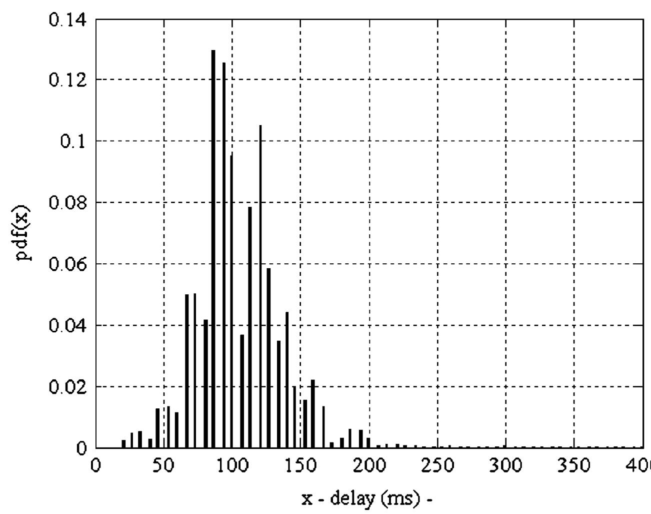


Fig. 6. Instantaneous video packet delay (ms).



(a)



(b)

Fig. 7. (a) Distribution function and (b) histogram for video packet delay (20 users).

simultaneously to and from two or more BSs), providing a macrodiversity gain that avoids the increase of packet losses leading to a seamless handover connection.

Video, ECG, FTP, and voice transmissions have been simulated according to the multiplexing scheme described in Section III. Simulations have been carried out with different load conditions. The mean number of video users in the same cell ranges from 15 to 30. The considered QoS parameters [16] for voice and video transmission are a maximum delay of 400 ms and a packet loss of 3% and 1%, respectively. ECG transmission is considered to have a packet delay lower than 300 ms without packet loss. For video transmission, early discard tries to match the delay constraints removing packets with higher delay from the RLC queue. In order to limit this delay to 400 ms on average, the selected T_{margin} (1) is fixed to 267 ms, considering a mean interarrival time of 133 ms. For ECG and FTP services, which require data integrity, packet loss is not allowed, so early discard is not used.

The useful parameters to determine the performance of the system are service dependent. Fig. 5 shows the percentage of packet loss for voice transmission. The QoS requirement (packet loss lower than 3%) is covered in all the simulated load conditions. Performance slightly decreases when system load increases thanks to an appropriate network planning and the efficiency of the applied RRM techniques.

For video transmission, both discarded packets and packet delay are interesting QoS parameters. In Fig. 6, variation of video packet delay is shown. These results have been obtained considering only a mean number of active users equals 20. In addition, Fig. 7 shows the distribution function and the histogram of video packet delay for the same selected capacity.

Variations in the instantaneous delay are due to the inherent variability of the WCDMA air interface capacity. In addition to stochastic variations of the available capacity, when the ambulance suffers from bad conditions (adverse propagation channel, high load), the transmission data rate must be reduced, leading to an increase in packet delay. This increase in video packet delay is regulated through the early discard. Fig. 6 shows that maximum delay is kept around 400 ms because packets with a higher delay are discarded in the transmitter. On the other hand, the discard probability obviously increases as load grows, as shown in Fig. 8. Anyway, the percentage of packet loss is lower than 1% for all load conditions, so AM with the use of early discard can provide a reliable real-time transmission for the video service.

Assuming that ECG transmission is error free, the only performance criterion is packet delay. Figs. 9 and 10 show instantaneous ECG packet delay and its distribution function and the histogram, respectively. The low transmission rate required by this service allows us to guarantee the delay constraints keeping data integrity.

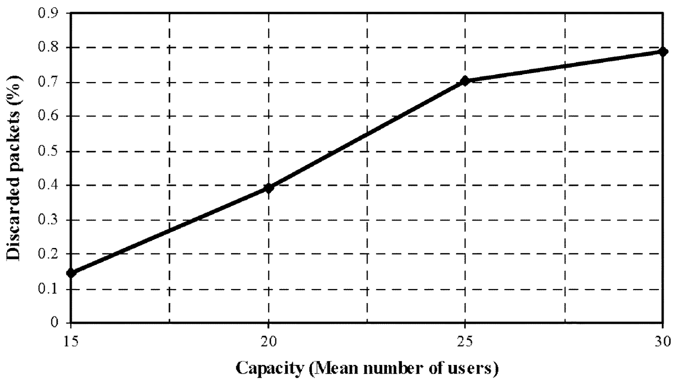


Fig. 8. Percentage of dropped video packets.

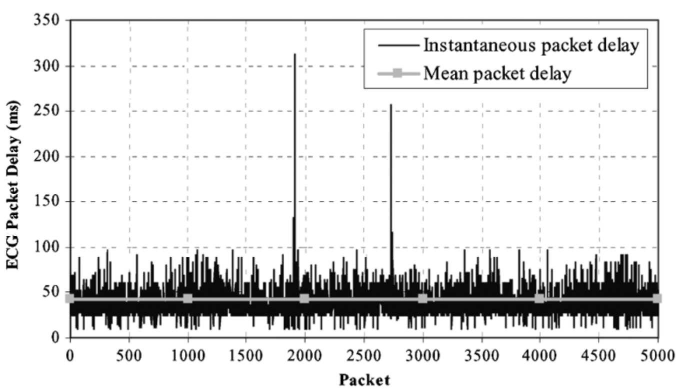


Fig. 9. Instantaneous ECG packet delay (ms).

In order to guarantee a continuous display of both video streaming and ECG data, a dejitter buffer should be applied in the receiver at the hospital. This buffer will introduce an additional delay to those provided in the previous figures. Although no interstream synchronization has been explicitly taken into account, all ECG, video, and voice are transmitted in real time with their QoS delay constraints, guaranteeing that the delay differences between these different streams are almost imperceptible for a medical application.

Finally, FTP transmission of medical scans does not have packet delay constraints. Its only requirement is data integrity. An interesting parameter is the effective transmission rate, shown in Fig. 11, which is determined by the available bandwidth in the TrCh after transmission of video and ECG services. Results show that the UMTS air interface provides the necessary flexibility to multiplex all the service components of the telemedical application through a single connection, a particular treatment according to the specific characteristics of each service is also guaranteed.

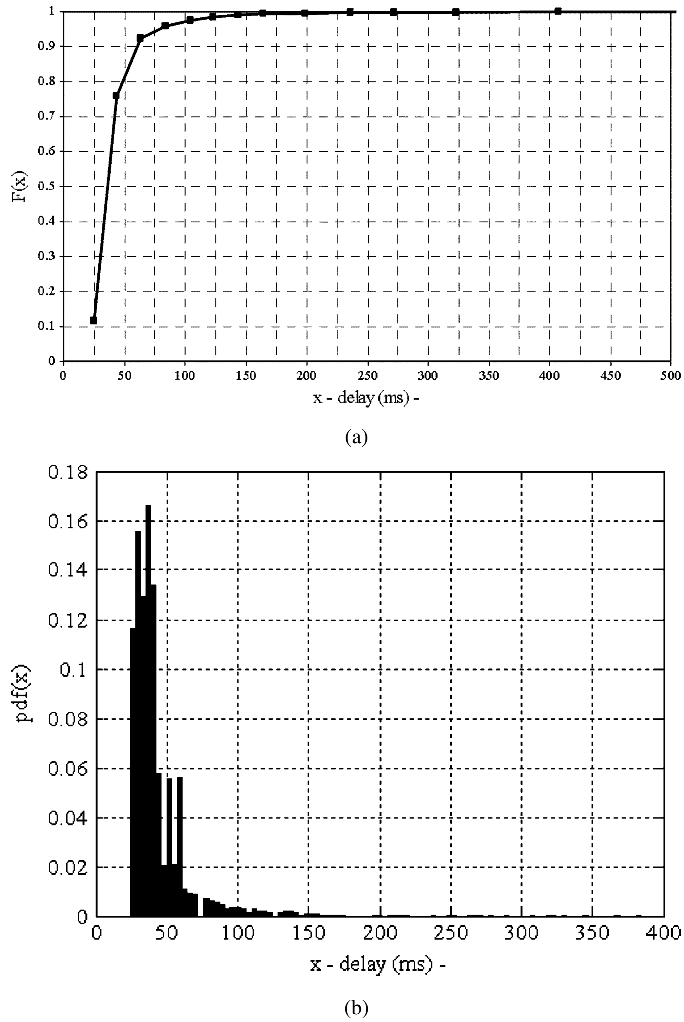


Fig. 10. (a) Distribution function and (b) histogram for ECG packet delay (20 users).

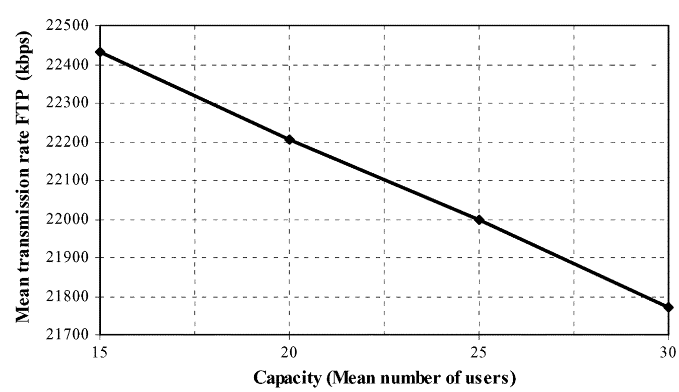


Fig. 11. Mean file transfer rate.

V. Conclusion

A multiplexing scheme for providing a multimedia connection between an ambulance and a hospital based on UMTS technology has been proposed. Transmission of voice, real-time video and ECG signals, and files with interesting clinical information as still images have been taken into account. Performance of the proposed scheme has been evaluated by means of a realistic UMTS simulator under different load conditions and in a high mobility situation. It has been shown that a careful system design mainly related to RRM must be carried out in order to have an efficient share of the available capacity. Results show the feasibility of UMTS to provide different QoS requirements for a multimedia application, consisting of four data services.

Thus, UMTS belongs to a new generation of eHealth systems and provides a reliable technology for supporting telemedical applications consisting of heterogeneous health information sources. Although these kinds of applications and services involve a considerable change in clinical practice for health professionals, it allows us to provide immediate and ubiquitous care in emergency situations and can reduce significantly the mortality since the best possible treatment can be given to the patients.

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