Real-Time Transmission of 2D Echocardiograms over WiMAX Networks

Eva Cavero, Álvaro Alesanco, José García

Aragón Institute of Engineering Research, University of Zaragoza, Zaragoza, Spain

Abstract

This study presents a methodology to transmit 2D echocardiogram video over WiMAX networks in real-time. Before transmission the echocardiogram has to be compressed. In order to compress the echocardiogram the 3D SPIHT algorithm was used. To guarantee the clinical quality of the compressed echocardiogram a clinical evaluation of the compressed echocardiogram was carried out. The evaluation yielded the minimum transmission rate necessary for an accurate diagnosis, 200 kbps. However, the channel produces errors and distorts the video, so in order to reduce these errors a reliable application protocol was proposed using a hybrid solution in which retransmission was combined with Forward Error Correction (FEC) techniques. The transmission was simulated in WiMAX networks. The proposed protocol achieved 100 % of time with guaranteed clinical quality (200 kbps) for bit error rate up to 0.254, being this value lower than for the retransmissions and FEC working separately.

1. Introduction

Mobile healthcare (m-health) is an emerging concept that has seen the evolution of e-health systems from traditional desktop telemedicine platforms to wireless and mobile configurations. With emergency wireless technologies, such as WiMAX, patients can access healthcare services not only from hospitals, but also from rural healthcare centers, ambulances, ships, trains, airplanes and homes. However, wireless channels are band limited, time varying and error prone. This is a problem especially for medical video streaming applications. Echocardiogram video streaming is a very demanding application in terms of bandwidth and delay and hence requires compression for transmission while at the same time maintaining high image quality on reception in order to avoid loss of diagnostic information.

Thanks to the development of efficient video compression techniques the development of medical video streaming applications has been made possible. In [1] the improved performance of 3D Set Partitioning In Hierarchical Trees (SPIHT) in terms of PSNR as compared to H.264/AVC and Xvid codecs for echocardiogram compression was shown.

However, it is very important to note that lossy compression modifies the original video and decreases its quality. In medical images, a minimum quality is required to be able to make an adequate diagnosis. For example, in [2] a testbed that unifies and reflects the clinical evaluation procedure was developed. This test gives a precise evaluation of the real degradation in compressed echocardiograms and provides a minimum recommended transmission rate required for each echocardiographic mode to achieve adequate clinical quality.

In order to provide an accurate diagnosis, it is not only necessary to have a compression method that guarantees clinical quality, but it is also essential to be able to guarantee the integrity of the video during the transmission process. It is well known that wireless channels are error prone. Thus, digital transmission may be affected by erroneous bits that distort the reconstruction of the video on reception. Hence, reliable data transfer is required. The two main techniques for reliable transmission are retransmissions and Forward Error Correction (FEC). For telemedicine applications, error resilient implementations for robust diagnosis performance have been addressed for example in [3].

This paper presents a proposal for echocardiogram transmission in real time over WiMAX channels. A codification technique is proposed and a reliable application protocol is designed using retransmissions and FEC techniques. The minimal transmission rate to guarantee a suitable diagnosis for the 2D mode of echocardiogram video has been established and a study of echocardiogram transmission over WiMAX has been performed. The proposed protocol is compared with retransmissions and FEC techniques working independently.

2. Materials and methods

2.1. Codification: 3D SPIHT

The 3D SPIHT algorithm is proposed for the compression of clinical video because of the advantages described in [1]. As in [1], the frame resolution (number of frames that are coded together) used for the 3D SPIHT compression was 16 frames. In order to define a minimal transmission rate that guarantees an adequate diagnosis for the echocordiogram, an evaluation was performed following the procedure described in [2]. An evaluation of all the modes was not carried out, because would have been very costly in terms of time. Also, the 2D mode is the most common mode and it has the highest requirements of bandwidth (see the recommended transmission rates in [2]). It is expected that similar or inferior transmission rates would be obtained for the rest of the modes.

To obtain a minimum recommended transmission rate, an evaluation was carried out by three cardiologists who assessed nine 2D mode echocardiograms. The echocardiograms were recorded with three different ultrasound devices (SonoSite, Philips Envisor, and Philips IE33), three videos for each device. Each video had a duration of at least 10 minutes, 25 frames per second, and a resolution of 720x576 pixels. Table 1 shows the Clinical Distortion Index (CDI) values (mean $\pm$ standard deviation of scores obtained by the cardiologists) for the nine echocardiograms and three transmission rates (150, 200, and 250 kbps). As was established in [2], values above 0.25 for the CDI scores were not permitted (values above 0.25 are shaded in Table 1). The recommended transmission rate is the lowest with all the CDI values lower than 0.25, that is 200 kbps. Thus a block of 16 000 bytes (16 frames) is generated every 0.64 seconds. Because of the block length, the blocks are fragmented into packets (of X bytes) for their transmission, having a total of N packets (16 000/X). If all the fragments of the blocks (Npackets) reach the receiver, an adequate diagnosis is possible. Otherwise, the echocardiogram is visualized with inferior quality and adequate diagnosis is not guaranteed.

Table 1: CDI values for the 2D mode

Bit rate	150 kbps	200 kbps	250 kbps
SonoSite	0.25± 0.07	0.09± 0.06	0.09± 0.06
	0.80± 0.00	0.06± 0.06	0.02± 0.03
	0.75± 0.07	0.06± 0.10	0.06± 0.06
Envisor	0.20± 0.14	0.11± 0.15	0.07± 0.08
	0.20± 0.00	0.20± 0.15	0.16± 0.09
	0.25± 0.07	0.16± 0.19	0.07± 0.08
IE33	0.15± 0.07	0.12± 0.21	0.08± 0.07
	0.25± 0.07	0.23± 0.11	0.08± 0.14
	0.20± 0.00	0.20± 0.04	0.08± 0.09

2.2. Transmission protocol and monitoring process

Since the integrity of the clinical video in the monitoring process is essential, a protocol that retransmits or correct losses is needed. Since it is well known that Transmission Control Protocol (TCP) does not work properly in wireless networks with a high bit error rate, a User Datagram Protocol (UDP) based protocol is used. However, UDP does not introduce reliability. For this reason, reliability is added to the designed protocol creating an application layer. In order to decrease header overheads, reduce packet loss and increase security over noisy wireless links, RObust Header Compression (ROHC) has been used [4]. This standard compresses Internet Protocol (IP) and UDP headers to just 3 bytes.

A reliable protocol is implemented in the application layer that uses both retransmissions and FEC code techniques. With retransmission techniques, only missing packets are retransmitted. If the channel delay is long or if several retransmissions of a same packet are required because of channel errors, the resulting delay would be intolerable for real-time applications. With FEC, redundant bits are added in the transmitted data and the decoder uses these added bits to correct the errors. The amount of redundancy embedded can be more than is necessary to correct the errors, using more bits than with the retransmission mechanism. However, if the amount of redundancy is less than is necessary, no error can be corrected. But this technique does not need a feedback channel and reduces the time needed to recover missing packets. In this way, retransmissions adapt the bits used to the amount needed and the FEC code provides extra protection. The FEC code used is Reed$-$Solomon (RS), which is a systematic and block code. The RS code is applied to the N packets that correspond to a block, so as not to introduce more coding delay and to avoid the effects of burst errors. The RS code generates K packets, N packets with the coded video and $(N-K)$ packets with the correction packets. The K packets are transmitted into the network, but Y packets arrive at the receiver. If N or more packets arrive correctly, the RS code is able to recover the lost packets and the video is visualized with guaranteed quality. Otherwise, the RS code is not able to recover the lost packets and the video is visualized with inferior quality, thus an adequate diagnosis is not guaranteed.

In order to efficiently implement retransmissions, a monitoring buffer has to be used in reception that enables the continuous flow of data when a packet is retransmitted. Thus there are two buffers: a reception buffer which contains all the packets that get the receiver and a monitoring buffer which contains the decoded frames that are visualized. The monitoring process starts passed the buffer time

(T_B) after the first packet is received. T_B is selected for the user.

2.3. Wireless channel model

WiMAX networks are a flexible choice for providing telemedicine services in both fixed and mobile environments. Suitable scenarios for these networks include small clinics communicated with heath-care centres and prehospital treatment or follow up in a mobile scenario (e.g. in an ambulance).

The wireless channel model used is the same as that described in [5]. A two-state Markov model has been used to simulate the lost bit patterns in wireless fading channels. Two channel states are considered, described as good and bad in terms of bit errors. The model specification is completed by the bit error rate (BER) and the average burst length (ABL) that depend on the fading margin (*F*) and the speed of the transmitter.

The used channel parameters are carrier frequency of 3500 MHz, maximum bandwidth of 500 kbps, mean delay of 47 ms and standard deviation delay of 20 ms, as same as in [3]. The simulated mobile speed values were from 5 to 60 km/h. The fading margin took values ranging from -5 to -30 dB. A total of 42 different channel conditions were simulated. The BER values obtained are between 0.00068 and 0.266, presenting a wide variety of values in order to evaluate the transmission from normal to extreme channel conditions, being the transmission errors introduced in both communication directions.

The network simulator used for this study was OPNET. The transmissions of 30 min of echocardiogram at 200 kbps, repeated four times, have been simulated. Thus, a total of 2 hours have been simulated for each point shown in the results figures.

2.4. Quality parameters

In order to measure the quality, the following parameters have been analyzed.

- *Bandwidth (BW):* This quantifies the amount of bits per second (in bps) used in the communication.
- *Percentage of time with guaranteed clinical quality:* The percentage of time with guaranteed clinical quality is the percentage of the time that the video is visualized with the minimal bandwidth given by the clinical evaluation.
- *Delay:* This is the time from when the video is visualized in the transmitter until it is monitored in the receiver. It depends mainly on the frame resolution, 0.64 s, and T_B that is determined for the application.

In order to limit the delay we have established that T_B is 1.5 seconds, hence the delay is approximately 2.2 seconds.

3. Results and discussion

The retransmissions mechanism is sensitive to the fragment size, the higher the fragment size, the lower the percentage of time with guaranteed clinical quality and the higher the used bandwidth. This is because with the lowest fragment size, fewer bits are retransmitted (the fragment size) for each erroneous bit. However, with the lowest fragment size more overhead is introduced, because more packets for the same fragment size are needed. Since header compression is used, this problem has been reduced, but fragment sizes lower than 200 bytes are not recommended. This is because for these sizes a lot of overhead is introduced and in low error channels an excessive bandwidth is used. Thus, henceforth the blocks are fragmented into packets of 200 bytes ($X = 200$ bytes) having a total of of 80 packets ($N = 80$ packets) per block [(16 000 bytes/block) / 200 bytes/fragment].

3.1. Proposed protocol

An important parameter is the correction code use. A correction of 26.25 % has been chosen ($K = 101$ packets). The correction code barely affects the percentage of time with guaranteed clinical quality, but with correction codecs lower than 26.25 % there are some channel conditions for which the clinical quality is not achieved. However, the bandwidth is quite different for different correction codes, being higher for the higher correction codes because with lower correction less correction packets are transmitted. As can be seen in Fig. 1, suitable clinical quality (100% of time with guaranteed clinical quality) is achieved for almost all the simulated channel conditions. There is suitable clinical quality for all the F values and mobile speed up to 10 km/h. For higher mobile speeds, suitable clinical quality is achieved for F values up to -10.

3.2. Protocol comparison

The proposed protocol, which uses retransmission combined with FEC techniques, is compared with these techniques working separately. Fig. 2 shows the percentage of time with guaranteed clinical quality for these techniques with a mobile speed of 60 km/h. Only one mobile speed is represented, but the other mobile speeds have the same tendency. As can be seen, the proposed technique achieves guaranteed clinical quality for BER values higher than with the other protocols. With retransmissions mechanism, the delay limitations is the cause of that the minimal quality has not being achieved for some channel conditions, because more than one retransmission may be required for the same packet, and with the FEC technique, higher error correction code has to be used to recover more channel errors. However, the proposed technique achieves

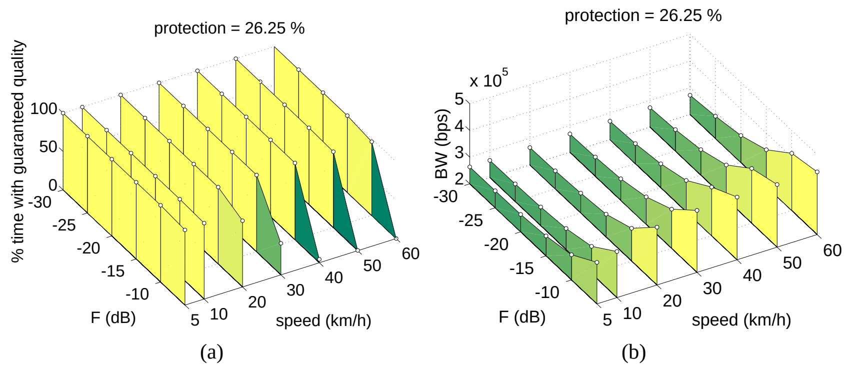

Figure 1: Percentage of time with guaranteed clinical quality and bandwidth used for all the channel conditions.

the best results in terms of clinical quality because it uses a combination of both techniques. The retransmissions mechanism adapts the used bandwidth to the channel conditions and the error correction code is applied in the case that more than one retransmission is required, correcting the lost retransmitted packets.

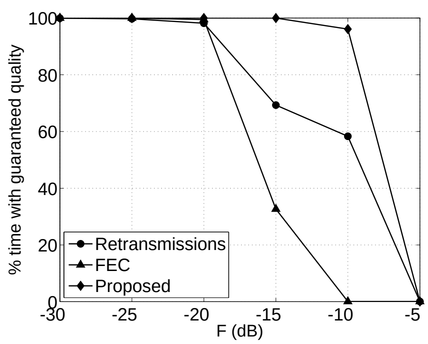

Figure 2: Percentage of time with guaranteed clinical quality for different techniques.

4. Conclusion

This paper presents a methodology for real-time transmission of echocardiograms over WiMAX networks. In order to guarantee suitable clinical quality over a wide range of channel conditions, a reliable application layer protocol is proposed using retransmissions and FEC techniques combined. In order to have suitable clinical quality for the compressed 2D mode of an echocardiogram video, 200 kbps of transmission rate is required. Thus, if the receiver receives video at 200 kbps or more, the clinical quality is guaranteed. The proposed technique achieves under the conditions of 373 kbps of transmission (200 kbps plus retransmissions and correction codes), speed of 10 km/h and a visualization delay about 2.2 seconds a guaranteed diagnosis for a BER of 0.254 in the WiMAX network for echocardiogram transmissions.

Acknowledgements

This research work has been partially supported by projects TIN-2011-23792 from the Ministerio de Ciencia e Innovación (MICINN) and the European Regional Development Fund (ERDF).

References

[1] Cavero EM, Alesanco A, García J. A new approach for echocardiogram compression based on display modes, 10th IEEE Int Conf on Information Technology and Applications in Biomedicine (ITAB) 2010:1-4.

[2] Alesanco A, Hernandez C, Portoles A, Ramos L, Aured C, Garcia M, Serrano P, Garcia J. A clinical distortion index for compressed echocardiogram evaluation: recommendations for Xvid codec. Physiological Measurement 2009;30(5):429-440.

[3] Martini MG, Hewage C T E R. Flexible Macroblock Ordering for Context-Aware Ultrasound Video Transmission over Mobile WiMAX. International Journal of Telemedicine and Applications 2010, Article ID 127519, 14 p.

[4] Pelletier G. Sandlund K. RObust Header Compression Version 2 (ROHCv2): Profiles for RTP, UDP, IP, ESP and UDP-Lite. IETF RFC 5225 2008; http://tools.ietf.org/html/rfc5225

[5] Alesanco A. Garcia J. Clinical Assessment of Wireless ECG Transmission in Real-Time Cardiac Telemonitoring. IEEE Transactions on Information Technology in Biomedicine 2010;14(5):1144-1152.

Address for correspondence:

Eva Cavero
Dpto. Ing. Electrónica y Comunicaciones.
Universidad de Zaragoza. Edif. Ada Byron.
C/ María de Luna, 1. 50018 - ZARAGOZA. SPAIN
email: racaj@unizar.es