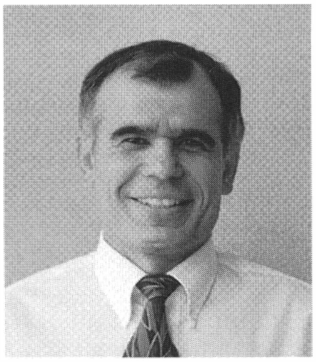

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A Tutorial for Emerging Wireless Medical Video Transmission Systems

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Abstract

The wireless transmission of medical video is expected to see significant growth in the coming years. This is primarily due to the expected availability of significant bandwidth in the next-generation of wireless communications networks, and the emergence of effective wireless video-compression standards. The design of wireless medical video transmission systems presents unique requirements for error-resilience and clinical validation. We present a tutorial introduction to the most important components of a wireless medical video transmission system. We focus on video-encoding requirements for error resilience, quality-of-service transmission, and diagnostic validation. Whenever possible, we give references to open-source software and provide references to the literature. We demonstrate the basic principles through the use of simple examples of stroke ultrasound videos.

Keywords: Medical ultrasound video; error resilience; H.264/AVC; clinical video quality assessment; telemedicine; mobile health systems; mobile; wireless; emergency; wireless video; 3G

1. Introduction

The history of telemedicine systems [1] is tightly coupled with the continuous growth of computing technologies and systems, in general. Driven by significant advances in computational power, we now have a broad spectrum of new mobile health systems and services that were previously unimagined [2, 3]. Health applications include remote diagnosis and care, home monitoring of patients with chronic diseases and the elderly, and assistive technologies. The underlying technologies range from electronic health records to implantable sensors and wearable devices.

Since the publication of [3], we have the emergence of new 3G and 3.5G wireless medical video transmission systems, while deployment of 4G WiMAX and LTE advanced networks is currently planned in the near future. We expect significant growth in this area. This paper is focused on providing a tutorial introduction to available tools and technologies that need to be considered when developing wireless medical video transmission systems. Our approach is to illustrate the general architecture, and to provide more-detailed explanations of the underlying components (see Figure 1).

For medical wireless video transmission systems, the two most significant components include the medical video-compression technology and the wireless infrastructure that will be used for the transmission. Medical video compression needs to address some of the unique requirements associated with the intended diagnostic use. Efficient video-compression systems can be built using the current state-of-the-art video-coding standard H.264/AVC [4], which provides for both an efficient (size-wise) and timely (real-time) encoding. On the other hand, increasingly available bit rates through revolutionary wireless transmission media [5, 6] allow the realization of communications previously only available to wired infrastructures. Coverage is extended across the globe with the latest 3G, 4G, and satellite systems.

Despite the rapid growth of telemedicine systems, wireless channels remain error prone, while the continuous bit-rate and compression efficiency increase is soon met by the rising expectations of the amount of data to be transmitted. In practice, clinical videos are routinely compressed with a limited understanding of the effects of compression on diagnostic quality. Here, it is interesting to note that the topic of video-quality assessment (VQA) is still emerging for general videos [7]. Yet, for medical videos, where crucial clinical information may be lost in compression, there is relatively very little research being done (but see the relevant research in [8]).

This tutorial paper documents some of the most significant factors involved in wireless clinical video encoding, and provides insights into the design considerations, challenges, current and future trends. References to the literature provide for further study in this area.

The rest of the paper is organized as follows. Section 2 highlights wireless-transmission technologies and protocols, while Section 3 provides a brief overview of H.264/AVC error-resilience coding. Scalable video coding is discussed in Section 4. Diagnostic validation is introduced in Section 5, while Section 6 provides a case study incorporating most of the issues raised in this paper. Section 7 provides some concluding remarks.

1.1 Telemedicine System Architecture

We focus in this section on emergency telemedicine, since it provides some of the most-significant challenges. However, several of the underlying technologies are clearly shared by other wireless medical transmission systems.

Figure 1 depicts the system components required for real-time transmission of medical ultrasound (or trauma) video. After the ambulance arrives at the scene, the paramedics need to stabilize the patient(s) following standard medical protocols. The transmission of vital signs (such as electrocardiogram, heart rate, blood pressure) and wireless medical video begins after the patient is transferred into the ambulance. The basic idea is to communicate ultrasound and/or patient (optical) images to an emergency physician, to further assist in the diagnosis and to prepare for the patient's admission to the hospital.

Wireless transmission of medical video is essentially composed of four steps. Following the acquisition, raw medical video is pre-processed so that it is suitable for encoding. This step typically involves resolution and frame-rate adjustments, as well as format conversion. H.264/AVC source encoding is then used to compress the video. The H.264/AVC video-encoding standard is composed of two layers. The first is the video-coding layer (VCL), which is responsible for the compression of the source video. This

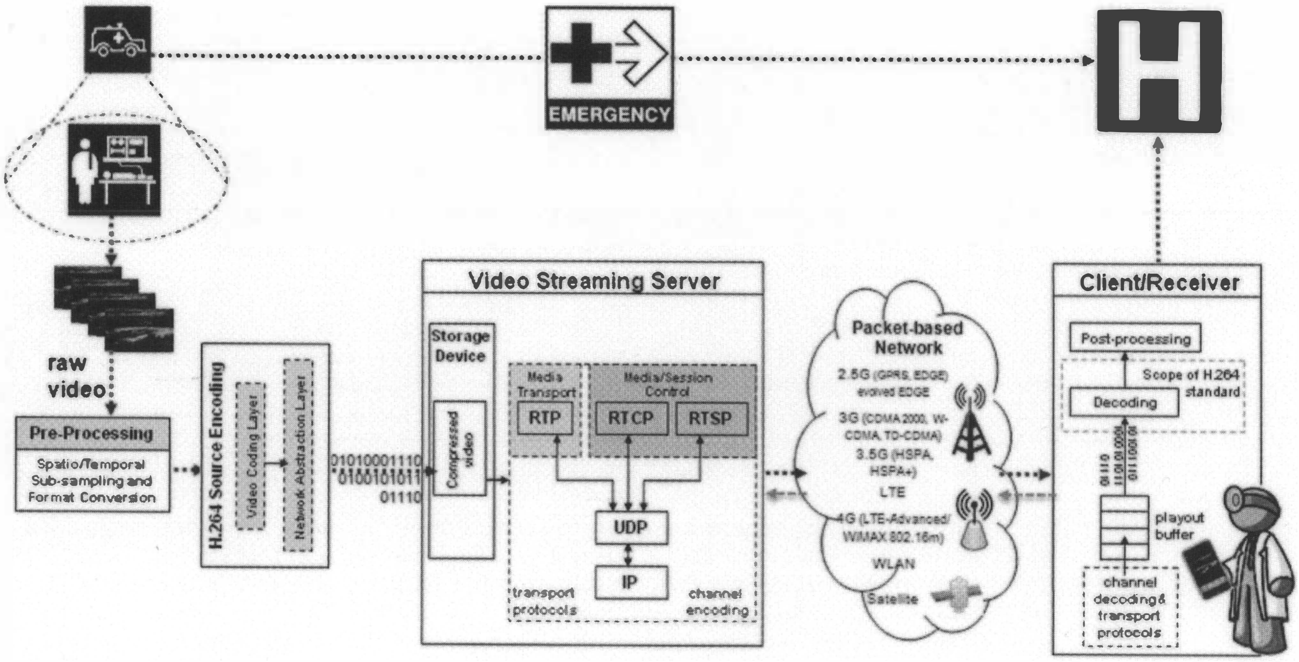


Figure 1. Wireless ultrasound video transmission for emergency telemedicine. The equipment residing in an ambulance captures and transmits the medical video to a remote medical expert and/or hospital premises for assistance with the diagnosis, and to prepare for the patient's admission to the hospital.

is followed by the network-abstraction layer (NAL), which tackles the adaptation of the encoded bit-stream to the underlying transmission medium. In this case, RTP/UDP/IP services are normally adopted to transport H.264/AVC over candidate wireless networks. Given the best-available wireless transmission medium, the medical video is transmitted to the client's side (medical expert), where the reverse procedure is followed for decoding, post-processing, and error recovery. Since the first hour ("golden hour") following a trauma incident is the most important for the patient's recovery and/or survival, such telemedicine systems are most likely to provide a significant time advantage for the treatment of patients involved in emergency incidents.

1.2 Hardware and Software Tools

Acquisition of the ultrasound video is done via a portable device. Source encoding can be performed using the *JM* H.264/AVC reference software [9]. *JM* is the current best-established reference software for H.264/AVC encoding and decoding, and is hugely popular among the video-processing research community. It incorporates most parameters and profiles defined in the standard, is well documented, and frequently updated. It is mostly recommended for research purposes and on-demand streaming, as it is not optimized for real-time encoding. Another popular open-source video codec that implements the earlier MPEG-4 part 2 standard is *Xvid* [10]. *Xvid* is frequently adopted by researchers, as it delivers the best MPEG-4 part 2 performance.

For real-time encoding and streaming, the *FFMPEG* [11] software is widely used. *FFMPEG* is open source, and also incorporates previous coding standards (MPEG-x and H.26x series). It can also be used for several pre-processing tasks, facilitating a large number of multimedia conversion algorithms. Streaming video can be achieved using the cross-platform *VLC* media player (which acts as a streaming server), hosted by VideoLAN [12]. *VLC* incorporates the *FFMPEG* codec, and offers both a graphical user interface as well as command-line-triggered transmission. Here, pre-processing, source encoding, and broadcasting software reside on a laptop in the ambulance.

For research purposes, a controlled environment (compared to actual streaming) is often sought, where parameters involved in video streaming (such as packet loss rate, delay, jitter, available bit rate, background traffic, mobility, end-user devices, etc.) can be easily modified in order to measure different possible aspects that can affect the design of a developed application. This kind of environment can be found in network simulators like *OPNET* [13] or *NS-2* [14] (open-source). To import video traffic in network simulators, video traces can be employed. A detailed study of video-traces utilization can be found in [15], while a video-trace library containing conventional videos is maintained by [16]. A digital medical video library, consisting of both videos and video traces, is not currently available. Trace files for medical video-transmission simulations can be produced using the aforementioned *JM* software and a set of *C* wrapper programs. *JM* incorporates the functionality to produce RTP packets of the encoded video. On the other hand, the *OPNET* software offers the capability of defining both custom and RTP packets. In this fashion, encoded RTP files can be imported into *OPNET*, and used for transmission over the simulated network. In addition to the above-described methods, a network-capture module (such as *Wireshark* [17]) can be triggered to capture real RTP video packets streamed to the network, and to then import them to a network simulator for further experimentation with wireless channels parameters.

2. Wireless Transmission Technologies and Communication Protocols

2.1 Wireless Transmission Technologies for Clinical Signal and Video Transmission

In terms of wireless infrastructure, the Global System for Mobile communications (GSM) [18] is the most popular standard. GSM signified the transition from analog first-generation (1G) to digital second-generation (2G) technology. Despite originally being designed for voice communication, GSM is also capable of data transfer at rates of up to 9.6 kbps. At such low rates, GSM can only be used for still images. It cannot be used for the transmission of medical video.

The evolution of mobile telecommunication systems from 2G to 2.5G (iDEN, GPRS, EDGE), and subsequently to 3G (W-CDMA, CDMA2000, TD-CDMA), 3.5G (HSPA, HSPA+), mobile WiMAX, and LTE systems has facilitated both an always-on model (as compared with the circuit-switched mode of GSM), as well as the provision of faster data transfer rates and lower delays, thus enabling the development of more-responsive telemedicine systems [3]. Evolving wireless communications networks' theoretical upload data rates range from 50 kbps to 86 Mbps (GPRS: 50 kbps; EDGE: 236.8 kbps; UMTS: 384 kbps; evolved EDGE: 947 kbps; EV-DO Rev A: 1.8 Mbps; mobile WiMAX Rel.1: 4 Mbps; EV-DO Rev B: 5.4 Mbps; HSPA+ Rel. 8: 11.5 Mbps; HSPA+ Rel. 9: 23 Mbps; LTE Rel. 8: 86 Mbps [6]). In practice, typical upload data rates are significantly lower. More specifically, typical upload data rates range from (i) GPRS: 40-50 kbps; (i) EDGE: 70-135 kbps; (iii) evolved EDGE: 150-300 kbps (expected); (iv) UMTS: 200-300 kbps; (v) HSPA: 500 kbps to 2 Mbps; (vi) HSPA+: 1 to 4 Mbps [6].

In terms of bandwidth, both 2.5G and 3G provide sufficient rates for medical image and bio-signal transmission. For medical video transmission, 3G rates are sufficient for QCIF (176 × 144) resolution medical video, as well as specific regions of interest (ROIs). Nevertheless, as described in [8], CIF (352 × 288) resolution video may be transmitted if variable-quality slice encoding is employed, according to the region's diagnostic importance. High-speed packet access (HSPA) and HSPA+ 3.5G technologies enable the transmission of high-quality CIF resolution video, as well as up to 4CIF (704 × 576) resolution video and beyond. The clinical benefits of transmitting high-resolution clinical video are an open area of research.

WiMAX release 2.0 and LTE-advanced networks conforming to the IMT-advanced requirements [19] will constitute the next-generation family of technologies, namely 4G. Low latency, high mobility, high bandwidths (targeting 100 Mbps for high mobility and 1 Gbps for low mobility, in the downlink), and quality-of-service (QoS) provisions, are expected to significantly boost the development of mobile-healthcare systems and services.

Satellite systems provide a variety of data-transfer rates, starting from limited to high-speed data rates of up to 2 × 64 kbps and beyond. The utilization of satellite links in healthcare benefits from world-wide coverage [20], but requires line-of-sight and comparably higher power for similar bit rates. We refer to [3] for a description of healthcare systems that demonstrate wireless transmission over satellite networks.

2.2 Protocols for Real-Time Clinical Video Transmission

For real-time audio and video transmission, RTP [21] provides end-to-end delivery services. Despite being able to provide real-time data delivery, RTP itself does not contain any mechanisms to ensure on-time delivery. To the contrary, it relies on UDP or TCP for doing so. However, it does provide the appropriate functionality for carrying real-time content, such as time-stamping and control mechanisms that enable synchronization of different streams with timing properties. The RTP payload contains the real-time data being transferred, while the RTP header contains information characterizing the payload, such as timestamp, sequence number, source, size, and encoding scheme. RTP distinguishes data delivery and control mechanisms. It basically consists of two parts: the RTP part, which carries the real-time data; and the real-time control protocol (RTCP) part, which is responsible for quality-of-service monitoring and extracting information regarding the participants in an RTP session. For medical video transmission, RTCP information can be used to improve quality-of-service, as it can be supplied as feedback for the encoder to adapt to varying network conditions. As we have already mentioned, RTP packets are usually transferred over UDP. The resulting packets use the Internet protocol (IP) for delivery through the network. As a result, the packets include RTP/UDP/IP headers.

3. Error-Resilience Methods

Our focus here is to discuss and demonstrate some of the most significant error-resilient features of H.264/AVC [4] that are of special interest to medical video transmission. In particular, we focus on new error-resilience techniques defined in the baseline profile, specifically designed for video streaming to mobile devices. A summary of the error-resilience methods found in different video-coding standards appeared in [3]. Here, we provide tutorial examples from clinical ultrasound that can be used to understand the emergence of new applications since the publication of [3]. We also discuss how such features need to be considered in the design of future telemedicine systems.

H.264/AVC can provide for bit-rate reductions of up to 50% for equivalent perceptual quality compared to its predecessors [22]. H.264/AVC defines different profiles and levels. Each profile and level specify restrictions on bit streams, and hence limits on the capabilities needed to decode these bit streams [4]. Baseline, main, extended, and high profiles assume different processing devices, tailored for different applications. They offer incremental level capabilities (and therefore complexity), which are alleviating constraints on bit streams.

Telemedicine systems target end-user devices, such as mobile smart phones. In the context of this study, we consider error-resilience methods found in the baseline profile. Error-resilience techniques can be further distinguished as to where error control actually takes place: error resilience at the encoder, error concealment at the decoder, as well as interactive approaches based on feedback communicated from the receiver. Exploitation of the aforementioned approaches is application specific, and all incorporated aspects should be considered before deploying an error-resilient telemedicine system. A thorough overview of the standard, performance and complexity analysis, error-resilience features and discussion exploiting H.264/AVC in the context of IP-based networks can be found in [22-27].

3.1 Encoding Modes and Frame Types for Clinical Applications

We summarize frame-encoding modes to recognize their impact on both error propagation and video-compression performance. We provide a summary of the different modes, and then discuss how they relate to clinical video encoding.

3.1.1 Intra-Mode

Intra-mode is the procedure where intra-prediction is used for coding a video frame (I-frame). By using intra-mode encoding, we restrict video-frame compression to be based on information contained only within the frame. No prediction from previous or future frames is allowed. As a result, intra-mode-encoded frames will require higher bit rates. On the other hand, we want to use intra-mode coding to significantly limit error-propagation in wireless video transmission networks.

3.1.2 Inter-Mode

Inter-mode is the procedure where inter-prediction is used for coding a video frame.

- P-mode: P-mode uses prediction from previously decoded frames. The use of P-mode will help reduce bandwidth requirements. On the other hand, decoding errors will be propagated, due to the prediction process. The goal in using P-mode in clinical video compression is thus to achieve a balance between the reduction in bandwidth and the loss of error-resilience behavior.

- B-mode: B-mode provides the ability to make use of two motion-compensated signals for the prediction of a picture. B-mode will thus allow for further reductions in the required bandwidth. However, error-resilience can be compromised by errors in either one of two reference frames. Compared to P-mode, the use of B-mode will have to balance higher reductions in bandwidth reduction to larger losses in error-resilience behavior. B-mode is not supported in the baseline profile.

For clinical videos, efficient intra-mode encoding is only possible when the anatomy contains similar regions. For example, compression of relatively straight arterial walls can benefit from repetition of similar regions. On the other hand, high compression of thin, circular cardiac boundaries will produce edge artifacts. To see this, note that circular boundaries cannot be effectively represented by translations of small image blocks. They require block rotations that are not available in the baseline mode.

There are also significant clinical limitations associated with inter-mode encoding. Efficient prediction relies almost exclusively on block-matching methods. These block-matching methods assume translational (strain-free) motions. The assumption is clearly inappropriate for visualizing cardiac strain imaging. To see this, note that strain accounts for either an expansion or a contraction of image regions. Such changes cannot be described by translation-only approaches. Here, these non-translational effects are encoded as the difference between the predicted and actual image blocks. Effective inter-mode compression will thus only be possible if the prediction residuals remain small for relatively small deformations. We provided an example of atherosclerotic plaque ultrasound video encoding in [28]. In [28], effective bandwidth utilization of bidirectional prediction was depicted. However, as the noise level increases, the use of single directional prediction provides for a more robust setting. Overall, the ratio of dropped P-frames to B-frames was shown to be a good predictor of plaque ultrasound video quality. Here, the corruption of P-frames will lead to significant degradation of atherosclerotic plaque ultrasound video quality, as opposed to the corruption of B-frames.

3.2 Intra-Updating Example for Atherosclerotic Plaque Ultrasound

We provide an example of three different intra-update schemes in Figure 2. To depict error-resilience, we plotted the decoded video quality as a function of the packet-loss rates (PLR). In pre-defined and random intra-refresh, 22 macroblocks per frame were intra-coded, giving a completely intra-coded frame every 18 frames (CIF resolution video). In intra-frame, an I-frame was encoded every 18 frames. The latter technique outperformed aforementioned techniques since error propagation was completely stopped at each I-frame, whereas with intra-macroblock techniques, error propagation was limited (but not stopped). On the other hand, note that I-frames require additional bandwidth, as demonstrated by the significant peaks in Figure 2b. When comparing video encoding at fixed bandwidths, the introduction of I-frames will translate into additional latency to be handled by the rate-control algorithm. Intra-macroblock refreshing can prove particularly efficient in the presence of a low end-to-end delay feedback channel [29]. It provides information as to which part of the picture is affected by losses and needs to be intra-coded in order to limit error propagation.

Here, results were obtained using *JM version 16.2* [9] for encoding, and a modified version of the RTP packet-loss simulator included in the *JM* software, enhanced to support significantly random performance [30]. Results were averaged over 10 consecutive runs. Quality was measured over the luma component, which contains the most-significant grayscale information (Y-PSNR).

3.3 Flexible Macroblock Ordering, Redundant Slices, and Arbitrary Slice Ordering

We will next discuss the use of flexible macroblock ordering (FMO), redundant slices (RS), and arbitrary slice ordering (ASO). We begin with an example of the use of flexible macroblock ordering in clinical ultrasound.

Figure 3 depicts the use of flexible-macroblock-ordering type 2 for atherosclerotic plaque ultrasound video encoding [8]. Here, the basic idea is to partition the video frame into regions of interest that represent different anatomical features. In Figure 3b, we have four different regions of interest. They were encoded using different quantization parameters (QP): (i) plaque ($QP = 28$), (ii) carotid artery walls ($QP = 30$), (iii) ECG signal ($QP = 30$), and (iv) background ($QP = 38$). Each region of interest was broken into macroblocks, as shown in the quantization parameter-allocation map (QPA map) of Figure 3b. Here, the plaque represented the most-significant region of interest. It had the lowest quantization parameter of 28 (see [8] for details). This resulted in the least amount of compression and a better quality reconstruction, as clearly seen in Figure 3c. We thus had a spatially-varying encoding scheme, where quantization levels were functions of diagnostic significance. Diagnostic regions of interest can be derived using segmentation algorithms, which can then be used as input for flexible macroblock ordering slice encoding. Alternatively, diagnostic regions of interest can be specified by the medical expert. Each region of interest is represented by a slice that can be independently encoded and decoded. We refer to [31, 32] for more details on the flexible macroblock ordering.

For noisy channels, we use *redundant slices* to recover slices that may be lost in transmission. Furthermore, *arbitrary slice ordering* [22] can be used to allow them to be transmitted and decoded independently of their order within a picture. This allows reducing decoding delay in environments where out-of-order delivery of a packet is possible (e.g., Internet, wireless networks).

4. Scalable Video Coding for Medical Video Applications

Scalable video coding (SVC) [33] can be used to allow for a single medical video bit stream to be displayed on different devices with different requirements for spatial resolution and frame rates. The basic idea is to first encode video into different layers. The base layer is to be decoded by every device. Additional (enhancement) layers then need to be decoded in order to provide for higher spatial resolutions and frame rates.

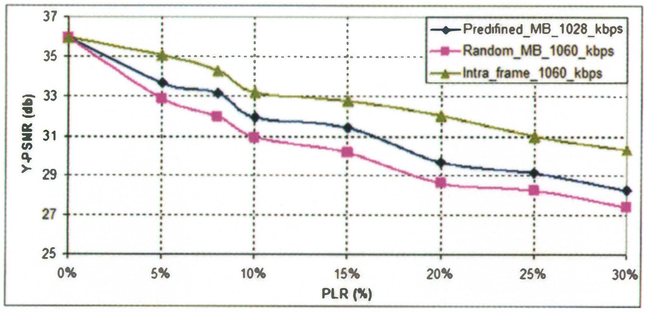


Figure 2a. An atherosclerotic plaque ultrasound video compression example using intra-updating. Three intra-update schemes are demonstrated as a function of packet-loss rate (PLR): predefined macroblock (MB) intra-update, random intra-update, and intra-frame update. Shown is the peak signal-to-noise ratio quality as a function of packet-loss rates.

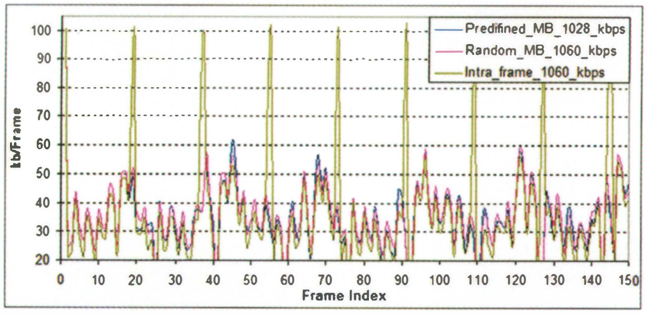


Figure 2b. The single-frame encoding requirements for each method described in the caption to Figure 2a. Rate control was used to smooth the variable frame-rate requirements (not discussed here).

We present a typical example of spatio-tempral scalability using atherosclerotic plaque ultrasound video in Table 1. In Table 1, we can see that the base layer required the lowest bandwidth at the lowest possible spatial resolution and frame rates. The first of enhancement layers provided higher frame rates at increased bandwidth requirements. The highest enhancement layers then provided higher spatial resolutions at the even higher bandwidth requirements. The highest layer provided the highest spatial resolution at the highest possible bit rate. It is clear that scalable video coding can also be used to control bandwidth requirements. It can thus be used to adapt the video compression bit stream to available bandwidth. An overview of the scalable video coding standard can be found in [34], *JSVM* reference software can be found in [35], while scalable video coding for mobile applications was exploited in [36].

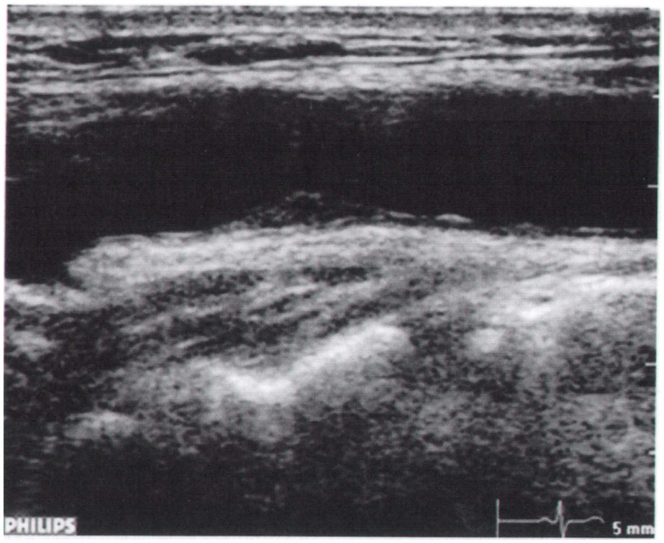
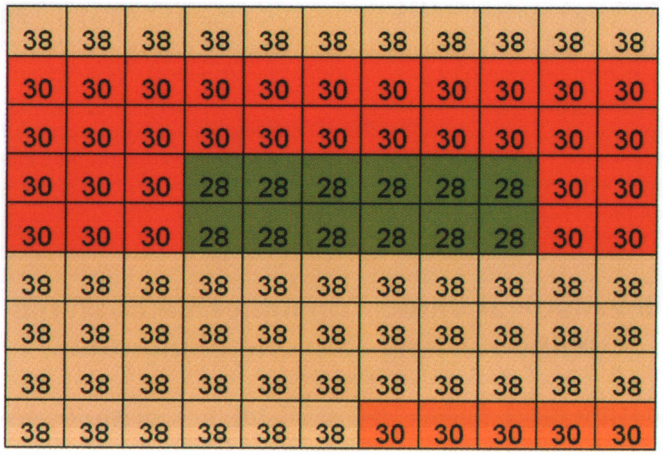


Figure 3a. A variable quality slice-encoding example. In this example, we show the captured carotid plaque ultrasound video frame, with quantization parameters (QPs) of 38/30/28. Diagnostically, the focus was on the reliable encoding of the plaque. The plaque region was encoded using $QP = 28$ as in Figure 3b (example based on [8]).



38	38	38	38	38	38	38	38	38	38	38
30	30	30	30	30	30	30	30	30	30	30
30	30	30	30	30	30	30	30	30	30	30
30	30	30	28	28	28	28	28	28	30	30
30	30	30	28	28	28	28	28	28	30	30
38	38	38	38	38	38	38	38	38	38	38
38	38	38	38	38	38	38	38	38	38	38
38	38	38	38	38	38	38	38	38	38	38
38	38	38	38	38	38	30	30	30	30	30

Figure 3b. Flexible macroblock ordering type 2 with diagnostic regions of interest derived after segmentation and the corresponding QP allocation map (QPA map).

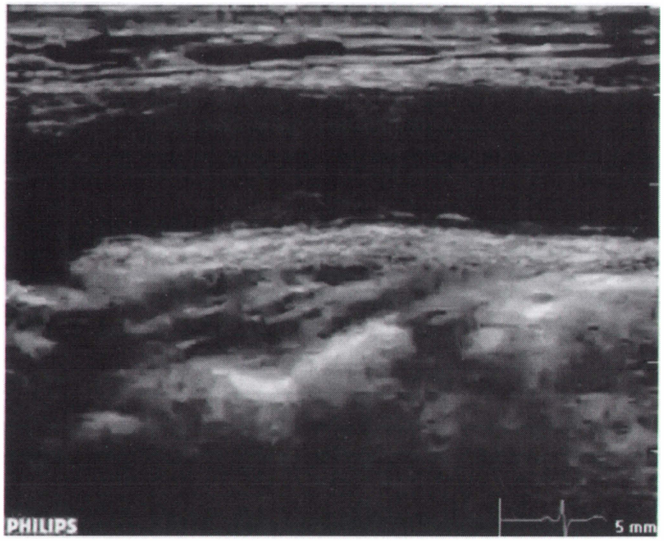


Figure 3c. The resulting decoded video after variable quality slice encoding.

Table 1. An atherosclerotic plaque ultrasound example of spatiotemporal scalability.

Layer	Resolution	Frame Rate	Bit Rate (kbps)
0	176 × 144	1.875	108.2
1	176 × 144	3.75	150.6
2	176 × 144	7.5	194.3
3	176 × 144	15	226.9
4	176 × 144	30	243.5
5	352 × 288	1.875	320.1
6	352 × 288	3.75	470.5
7	352 × 288	7.5	643.1
8	352 × 288	15	801.5
9	352 × 288	30	890.2

Input Videos: {352 × 288, 176 × 144}@30 fps,
QP: 28, software version: *JSVM 9.19.7.*

5. Diagnostic Validation

Diagnostic validation is the most significant requirement for emerging medical video transmission systems. Diagnostic validation requires an accurate assessment of the diagnostic capacity of the transmitted medical video.

5.1 Objective Video-Quality Assessment

Objective video-quality assessment (VQA) differs largely from image-quality assessment, which has seen significant growth over the last five years. However, today's most established video-quality assessment metrics are extensions of algorithms originally designed for image-quality assessment. Such examples are the peak signal-to-noise ratio (PSNR), which utilizes the mean square error (MSE) between the original and transmitted videos on a frame basis, and the average structure similarity index (SSIM) [37]. As documented in [7], there are significant problems associated with the use of peak signal-to-noise ratio for video-quality evaluation purposes, as it fails to adequately correlate with perceived video quality.

In general, there is a growing demand for the development of video-quality metrics that are significantly different from image-quality metrics. Aspects such as perceived motion, quality-of-service, and quality of experience (QoE) [38] need to be considered during the design of new video-evaluation techniques. Of particular interest is the new motion-based video integrity evaluation (MOVIE) metric described in [39], which claims to outperform all video-quality assessment algorithms to date [7]. Toward this direction, the video-quality metric (VQM) [40], developed by the national telecommunication and information administration (NTIA), is also capable of addressing some of the issues. A thorough study of objective and subjective video-quality assessment methods was found in [7]. Having said this, there is also a strong need for new, clinically-driven video-quality metrics.

5.2 Clinical Quality Assessment

There are unique challenges associated with both objective and subjective clinical video-quality assessment. In addition to the motion and quality-of-service aspects of conventional video-quality assessment, unique clinical criteria, often different for each medical modality, need to be properly assessed. These clinical criteria often correspond to specific video portions that are of diagnostic interest. These regions of diagnostic interest are much more sensitive to compression and error impairments, given their significant clinical contribution to the diagnostic yield of the particular medical video. On the other hand, diagnostic region-of-interest encoding may lead to diagnostically lossless medical videos. Clearly, both schemes are not adequately assessed by current objective video-quality assessment algorithms.

For subjective clinical evaluation, while we do expect that the basic features of the subjective quality-assessment criteria described in [41] will also play a role in emerging medical video-assessment standards, unique clinical criteria will also need to be adequately modeled. The diagnostic yield of transmitted medical video is restricted by a number of factors, including resolution, frame rate, and end-user equipment. The diagnostic capacity of a QCIF-resolution medical video at 15 fps displayed on a PDA differs significantly from a 4CIF resolution at 30 fps displayed on a laptop.

Examples of the use of diagnostic regions of interest can be found in [8, 42-44]. Diagnostic validation examples can be found in [8, 44-46]. An example of the use of clinical criteria for assessing diagnostic performance at different bit rates was presented in [8]. To demonstrate clinical validation, we refer back to the diagnostic regions of interest of Figure 3b. These diagnostic regions of interest correspond to clinical criteria of (i) plaque presence, (ii) plaque type, and (iii) degree of stenosis. For each clinical criterion, a mean opinion score (MOS) was computed, based on evaluations by medical experts. For assessing the decoded regions of interest quality, video-quality assessment algorithms were also applied over the segmented regions (e.g., peak signal-to-noise ratio over regions of interest). To then validate the video-quality assessment algorithms, correlations to the corresponding mean-opinion-score values were computed (see [8] for details). Ultimately, video-quality assessment algorithms that produce high correlations to mean opinion score may be used to predict clinical quality.

Similar approaches as to the use of clinical ratings for different medical video modalities were presented for (i) pediatric respiratory distressed medical video in [44], and (ii) echocardiogram ultrasound video in [47]. Recommendations for diagnostically lossless encoding bit rates were provided for specific clinical situations and incorporated parameters (such as codec, resolution, and frame rate).

6. Case Studies: Wireless Medical Video Transmission Systems Using 3G

We provide a summary of some recent wireless medical video transmission studies in Table 2. These systems were further categorized as regions-of-interest and no-regions-of-interest-based systems. An overview of earlier m-health systems and services, classified by wireless transmission technologies, appeared in [3].

6.1. Diagnostic Region-of-Interest-Based Systems

In this section, we present studies that required the identification of a region of interest that was of diagnostic interest. The systems are summarized in Table 2.

We have two studies associated with atherosclerotic plaque ultrasound video transmission [8, 42]. In both cases, automatic plaque segmentation was used for specifying the most significant region(s) of interest. In [42], the regions of interest were extended based on a visual attention model of what human readers would find interesting. Non-diagnostic regions were then blurred using a smoothing filter. The smoothed regions were also easier to compress. This resulted in significant bandwidth savings. In [8], several regions of interest were defined as described in Section 3.3 and Figure 3. Minimum-bandwidth recommendations were provided for the clinical evaluation of atherosclerotic plaque presence, type, and degree of stenosis.

In [43], the authors proposed context-aware medical image and video transmission. Context was defined based on patient status (normal or urgent). The urgent state was decided based on the ECG signal, blood pressure, pulse rate, heart rate, and oxygen level. Region-of-interest encoding was used to encode trauma in optical images or skin lesions for transmission in low-bandwidth systems. In the urgent state, data prioritization was achieved through scalable video encoding using a base layer and enhancement layers (see Section 4).

In [44], the authors assigned different quantization levels to the foreground and background regions of interest (given in bits per pixel). Bit allocations in the two regions of interest were varied to match different bit rates. Six different encoder states were considered. According to bandwidth availability, a switch was triggered to the most relevant state. A state switch was considered for every group of pictures (GOP). The system was validated for diagnostically lossless performance based on a variety of clinical criteria for pediatric respiratory-distress-related videos.

6.2 Medical Video Transmission Systems without Using Regions of Interest

The use of regions of interest provides for very efficient systems that target specific diseases. While the regions of interest provide for a direct way to address clinical criteria, defining regions of interest can be difficult. In this case, other criteria need to be explored.

Table 2a. End-to-end wireless medical video transmission systems using 3G: ROI-based systems.

Author	Year	Resolution, Frame Rate, Bit Rate[4]	Encoding Standard	Medical Video Modality	Comments
Tsapatsoullis et al. [42][1]	2007	352 × 288 @ 10fps 10 videos average: 507.2 Kbps	MPEG-2/ MPEG-4	Common carotid artery (CCA) plaque ultrasound video	Saliency-based visual attention ROI coding for low bit-rate medical video transmission.
Doukas et al. [43][1]	2008	Resolution: N/A @ 25 fps 128-256 Kbps	H.264/ AVC (Scalable)	Skin legion and MRI images, trauma video snapshots	Scalable ROI encoding. Adaptive transmission based on context awareness (patient status and network state).
Rao et al. [44][1,3]	2009	360 × 240 @ 30 fps 500 Kbps	MPEG-2	Pediatric respiratory distress related videos	ROI coding incorporated different quantization levels for ROI and non-ROI, targeting diagnostically lossless encoding over bit rate limited wireless channels utilizing physician expert feedback.
Panayides et al. [8][1,3]	2011	352 × 288 @ 15fps 10 videos: 197-421 Kbps (median: 253 Kbps)	H.264/ AVC	CCA plaque ultrasound video	Diagnostically-driven ROI quantization (FMO) and redundant slices for error-resilient encoding. Clinical validation of diagnostic quality.

[1]Simulation, [2]Real-time, [3]Clinical evaluation by medical experts (presented video parameters achieve diagnostically lossless encoding for preset clinical criteria), [4]For real-time transmission, both encoded video bit rates and available channel bit rates are provided.

Table 2b. End-to-end wireless medical video transmission systems using 3G: non-ROI-based systems.

Author	Year	Resolution, Frame Rate, Bit Rate[4]	Encoding Standard	Medical Video Modality	Comments
Chu et al. [48][2]	2004	{320×240 and 160×120} <5fps Channel bit rate: 50-80 Kbps	M-JPEG	Trauma video	Real time trauma video transmission. Network adaptation enabled through media transformation, data prioritization, and application-level congestion control.
Garawi et al. [49][2,3]	2006	176 × 144 @ 5fps 18.5-60 Kbps Channel bit rate: 64 Kbps	H.263	Echocardiogram	A performance analysis of an end-to-end mObile Tele-Echography using an ultra-Light rObot (OTELO)
Pedersen et al.[45] [2,3]	2009	320 × 240 @ 10fps 349 Kbps Channel bit rate: 380 Kbps	H.264/ AVC (Scalable)	Echocardiogram	Spatiotemporal scalability over different wireless networks. Diagnostic quality and how it's affected by network parameters.
Istepanian et al. [46][2,3]	2009	176 × 144 @ 8-10fps 50-130 Kbps Channel bit rate: 360 Kbps	H.264/ AVC	Abdomen	QoS Ultrasound Steaming Rate Control (Q-USR) algorithm based on reinforcement learning that satisfies a medical QoS criterion.

[1]Simulation, [2]Real-time, [3]Clinical evaluation by medical experts (presented video parameters achieve diagnostically lossless encoding for preset clinical criteria), [4]For real-time transmission, both encoded video bit rates and available channel bit rates are provided.

A portable tele-trauma system that provided for simultaneous transmission of trauma video, medical images, and ECG signals was presented in [48]. Media transformation, data prioritization, and application-level congestion control provided adaptation to network conditions. Data transformation tackled image and video compression using JPEG and M-JPEG standards, respectively. For image transmission, following manual region-of-interest segmentation on the first transmitted image, only the region-of-interest part was sent in subsequent transmissions. Priority classes were set to high for ECG transmission, medium for image transmission, and low for video transmission. During simultaneous transmission of the aforementioned priority data classes, when congestion occurred, a congestion-control algorithm that adjusted the trauma video's transmitting frame rate was triggered to alleviate congestion effects that led to packet losses. System-performance evaluation included effective resolution and frame-rate investigations for which transmission of simultaneous streams was possible.

Scalable video coding, employing spatiotemporal scalability for a number of ultrasound videos over different wireless networks, was described in [45]. Here, the transmission of different resolutions and frame rates over different wireless channels was investigated based on the wireless-transmission-medium's parameters and physician's feedback. Data rate, packet losses, delay, and jitter were measured for different scalability layers streamed over wireless local-area networks (WLANs) and 3G channels. Based on the parameters examined, and also in agreement with the diagnostic quality provided by medical experts, the highest scalable video-coding layer supported by each wireless channel was derived. Clinical assessment included evaluation of different clinical criteria.

A performance analysis of an end-to-end mObile Tele-Echography using an ultra-Light rObot (OTELO) system was presented in [49]. Low-bit-rate echography ultrasound medical video transmission over 3G channels was investigated, and functional bounds were provided. Based on remote clinical evaluation, the authors made recommendations in terms of quality (peak signal-to-noise ratio), resolution, frame rate, and maximum delay that provided for acceptable diagnostic performance over the tested 3G link.

In [46], the authors developed a quality-of-service ultrasound streaming-rate-control (Q-USR) algorithm. Based on reinforcement learning, the frame rate was varied as a function of the state of the network, while at the same time conforming to a pre-defined medical quality-of-service criterion. That is, the varied frame rate and resulting quality indices were monitored not to fall below what was diagnostically acceptable with respect to the bounds described in [49]. A robotic tele-ultrasonography system OTELO was also used in this study, for abdominal ultrasound video. Simulations and real-time experiments over 3.5G wireless networks validated the efficiency of the proposed mechanism with respect to the default rate-control algorithm of the *JM* H.264/AVC reference software. This was also verified by clinical video quality assessment.

Ultrasound video acquired using the OTELO system was also used in [50], where multilayer control was employed to optimally tune source and channel-encoding parameters for enhanced streaming of medical video. Frame rate, quantization step, intra-refreshing period, and average code-rate channel protection were the key parameters that were modified in these experiments to construct different encoder states. A trigger between states was considered on a per-second basis, based on the medical video's quality index, channel indications such as average SNR, and network parameters including packet-loss rates, delay, and jitter. Experiments were performed over WLANs.

6.3 Current Status and Future Directions

The basic source-encoding features that directly impact a video's clinical capacity are quantization levels, resolution, and frame rates. High quality corresponds to increased clinical capacity, but also accounts for increased bandwidth demands, a limiting factor in bandwidth-limited wireless networks. Resolution and frame rates are also bounded by the wireless channels' upload data-transfer speeds, as well as the end-user devices. Higher resolutions, close to medical video's acquired resolution, allows addressing more clinical criteria. For example, as documented in [8], clinical assessment of plaque type could not be efficiently addressed in QCIF resolution as opposed to CIF resolution. Moreover, clinical motion assessment benefits from higher frame rates. Hence, according to the underlying transmission medium and display sizes, minimum bandwidths, resolutions, and frame rates that provide for diagnostically lossless systems for specific clinical criteria should be sought.

We refer to Table 2 for parameters considered for typical upload data rates of 3G systems (and slightly higher, up to 510 kbps). Resolutions were typically varied between QCIF and CIF, frame rates between 10 and 15, while the clinical quality threshold for diagnostically lossless encoding was above peak signal-to-noise ratio ratings of 35 db [8, 46, 49]. Slight variations are possible.

Wireless video transmission quality of service parameters include packet-loss rate, delay, and jitter. Signal attenuation due to mobility, fading channel, and distance from the base station, congestion, background traffic, modulation and coding schemes, and packet sizes are some of the parameters contributing to quality-of-service deterioration. While some packet losses are tolerable, clinical quality cannot be allowed to lead to misdiagnosis. Moreover, increased delays usually result in additional dropped packets.

To address these issues, error-resilient encoding, adaptation to varying network conditions by modifying encoding setting, as well as combination of the latter two schemes is employed. Efficient error-resilient encoding for high diagnostic performance in noisy wireless channels was undertaken in [8]. Clearly, error-resilient encoding needs to be further exploited in the design of new medical video telemedicine systems, given the error-prone nature of wireless channels. The tradeoff between coding efficiency and error resiliency should be thoroughly investigated with respect to available bit rates.

Network monitoring for adaptation to network status by triggering a switch to a different encoder state was used in [44, 46, 50]. This typically involved a combination of quality, resolution, and frame-rate reduction, which accounted for reduced bandwidth demands and hence channel load. Efficient utilization of the network's resources has also been achieved through diagnostically relevant region-of-interest encoding [8, 44]. Scalable-video-coding encoding has been used for encoding different, states as discussed in Section 4 [43, 45]. Network adaptation was also achieved via data prioritization, as illustrated in [43, 48]. While the aforementioned techniques efficiently addressed packet losses due to channel load, they did not account for corrupted packets or burst errors, typical in wireless channels. Given that packet losses are inevitable in wireless transmission, medical video transmission systems should be equipped with appropriate error-resilient and concealment mechanisms. Next-generation 4G channels promise quality-of-service provision through data-prioritization classes for low-delay fixed-bandwidth allocation. This will enable the development of telemedicine systems that will exploit *a priori* channel states, and provide for a diagnostically robust system at different channel bit rates.

7. Concluding Remarks

Wireless transmission of medical video is likely to see significant growth in the coming years. Wide deployment of such systems and services in the near future could have a significant, positive impact on the patient's quality of life. It is foreseen that different medical modalities will be exploited, utilizing the already gained knowledge from current and completed research activities, leading to new telemedicine applications.

In the design of an efficient telemedicine system, several different criteria need to be considered. Most importantly, these systems require efficient encoding, and adaptation to channel bandwidth and end-user equipment, while being resilient to data losses introduced by error-prone transmission media. For the physicians to reach a confident diagnosis, minimum clinical-evaluation criteria must be determined.

This paper provided guidance on the most-important system components and design considerations, highlighting emerging trends. The paper was focused on H.264/AVC error-resilient methods and scalable video coding, wireless-transmission technologies, and objective and subjective clinical-quality assessment. Examples using carotid ultrasound video were used to demonstrate basic concepts. Case studies highlighting medical video-streaming systems over 3G channels were presented. We also provided references to the literature and open-source software standards to help further research in this area.

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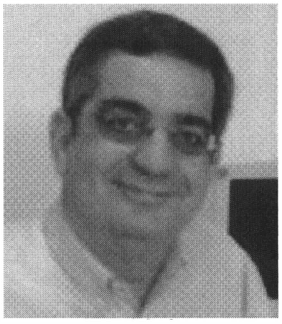

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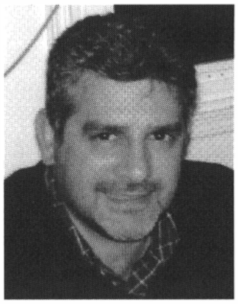

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