ADAPTIVE WIRELESS VIDEO COMMUNICATIONS: CHALLENGES AND APPROACHES

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ABSTRACT

Error Control in wireless video communications is of primary importance in successful transmission and reception of image/video signals over bandwidth limited wireless networks or fading communication channels. Pre- and post-processing error control mechanisms like error resilience and error concealment have been developed and incorporated in the design of the basic communication structure to make the data more robust to wireless fading channel errors in certain video transmission standards like H.261, MPEG-2, H.263 and MPEG-4. Of high concern is the bridge between the application level QoS requirements of real-time video transmission and the QoS guarantees provided by the wireless channel/networks. This paper reviews the recent advancements in these types of error control techniques for fading limited wireless image/video communications in the standard codecs with regard to the challenges that currently affect them and the approaches taken to overcome these challenges. It then proposes a novel adaptive approach to develop the QoS requirements of wireless video transmission by taking into account the end user perceptual quality rather than network bandwidth parameters. The paper also analyzes the key design issues of the communication system associated with the implementation of these error control tools in some of the standard codecs.

KEY WORDS: Wireless video communications, Error resilience, Error concealment, QoS.

1. INTRODUCTION

The advent of second and higher generation wireless personal communication standards in recent years has made it possible to realize bandwidth efficient video communications. The variation of wireless channel/ network capacity with mobility suggests that substantial performance gains are promised by intelligent multi-mode adaptive transceivers [1] [2] [3]. Specially designed error-resilient, fixed but programmable rate video codecs, which generate a constant number of bits per video irrespective of the video motion activity, provide wireless video communication services [4] over low rate, low latency interactive schemes and multimedia/videophony applications.

Figure 1 describes the basic structure of a digital wireless video communication system which requires five fundamental operations [5]. The process of source encoding involves efficiently converting the input original image/video signal into a sequence of bits and compressing them to near entropy levels. The purpose of channel encoder is to introduce redundancy in the compressed bit-stream which in turn is used at the receiver to overcome the effects of signal transmission through noisy wireless networks/ channels.

The encoded data bit-stream, now segmented into fixed or variable length packets, is modulated and sent over the channel/ network. The channel can be assumed as a physical medium, free space/ atmosphere in case of wireless, that distorts, fades and corrupts any information transmission. At the receiving end of a digital communication system, the channel corrupted received signal is demodulated into a data bit-stream,

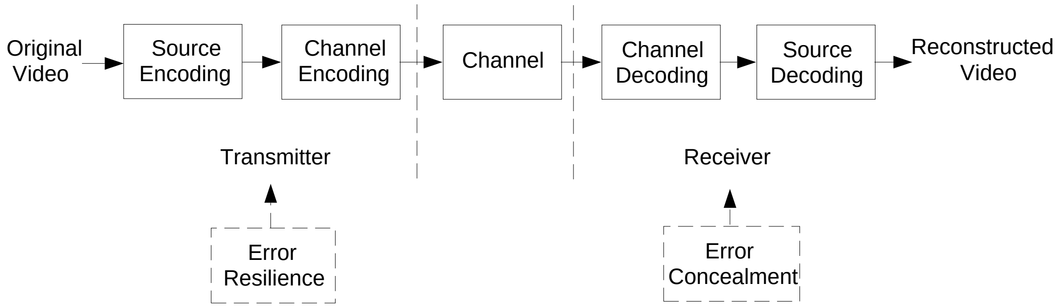

Fig. 1. Basic Structure of a communication system

channel decoded (the decoder attempts to reconstruct the original information sequence from knowledge of the code used by channel encoder and the redundancy contained in the received data) and source decoded to obtain the reconstructed video.

Most of the large variety of errors occurring in the transmission of real-time video are due to its large bandwidth requirements. Even if the channel capacity exceeds the required bit rate (high bandwidth channel/ low bandwidth application), channel errors can severely degrade the system and so, a compression scheme [6] and bit rate must be rigorously chosen in order to match channel characteristics while maximizing the video quality. Error resilience [5] at the encoder, error concealment [7] techniques at the decoder, shown in Figure 1 and their proposed interaction to meet QoS guarantees are key issues in the design of such a compression scheme and are discussed further in detail.

The paper is divided into the following sections. Sections 2, 3 and 4 comprehensively discuss the issues in wireless video communications and explain the recent advances in existing approaches to overcome these issues. In particular, Section 2 details about the various error resilient source coding techniques introduced recently, section 3 explains the issues with wireless channel and section 4 describes existing error concealment techniques. Section 5 briefs the problems faced by existing approaches and proposes a new technique that resolves these problems by usinganadaptive integrated end-to-end criteria that focuses on QoS requirements based on visual quality. Section 6 gives an outline of different codecs existing and their current adaptability and variation in efficiency with regard to different error resilience and concealment techniques. Finally, section 7 draws conclusions on the recent advancements in interactive error control mechanisms, their problems and probable solutions for feasibility of image/ video signal transmission over wireless channels.

2. ERROR RESILIENT SOURCE CODING

Error resilience schemes [5] [6] [8] address the issue of compression loss recovery and in specific, they attempt to prevent error propagation by limiting the scope of the damage caused by bit errors and packet losses on the compression layer. The standard error resilient tools include re-synchronization marking, data partitioning, and data recovery. Based on the role that the encoder, the decoder or the network layer plays in the process, these error resilience techniques can be divided into four categories, each of which is described here. A key assumption here is that the video is coded using the block-based hybrid encoding framework.

2.1. Robust Entropy Encoding

The encoder in this approach operates to minimize the adverse effects of the transmission errors of the coded bit-stream on the decoder operation so that unacceptable distortions in the reconstructed video quality can be avoided. Compared to coders that are optimized for coding efficiency, error resilient coders are typically less efficient in that they use more bits to obtain the same video quality in the absence of any transmission errors. These extra bits, or redundancy bits, are introduced to enhance the video quality when the bit stream is corrupted by transmission errors.

The sensitivity of a compressed video stream to transmission errors is mainly due to the fact that a video coder uses Variable Length Coding (VLC) to represent various symbols. Any bit errors or lost bits in the middle of the code word not only makes this code word un-decodable but also makes the following code words un-decodable, even if these bits were received correctly. The design goal in the error resilient coders is to achieve a maximum gain in error resilience with the smallest amount of redundancy. Techniques to introduce such redundancy in the bit-stream include:

- *Re- synchronization markers*: Inserting re- synchronization markers periodically enhances the efficiency of encoder error resilience. These markers are designed to be effectively distinguished from other code words and small perturbations of these code words. Header information regarding the spatial and temporal locations or other in-picture predictive information concerning the subsequent bits is attached immediately after the re-synchronization information. The decoder can then resume proper decoding upon the detection of the re-synchronization marker.

  Synchronous markers' utility interrupts in-picture prediction mechanisms like MV or DC co- efficient prediction, which in turn adds more bits. Longer and more frequently inserted markers would enable the decoder to regain faster synchronization such that the transmission errors affect a smaller region in the reconstructed frame and so long re-synchronization code words are used in the current video coding systems.

- *Reverse Variable length coding (RVLC)*: With RVLC, the decoder can not only decode bits after a synchronization code word, but also decode the bits before the next synchronization code word, from the backward direction and so fewer correctly received bits will be discarded and the effected area by a transmission error will be reduced. RVLC is adopted in both MPEG-4 and H.263 in conjunction with insertion of synchronization markers. For video coding and applications, RVLC can be designed with near perfect entropy coding efficiency in addition to providing error resilience.

- *Error-Resilient Entropy Code (EREC)* is an alternative way of providing synchronization which works by re- arranging variable length blocks into fixed length slots of data prior to transmission. The EREC is applicable to coding schemes where the input signal is split into blocks and these blocks are coded using variable-length codes, each of which is a prefixed code, like the macro-blocks (MBs) in H.263.

2.2. Unequal Error Protection (UEP)/ Layered Coding

Layered coding (LC) or scalable coding refers to coding a video into a base layer, which provides a low but acceptable level of quality, and one or several enhancement layers, each of which will incrementally improve the quality. LC is a way to enable users with different bandwidth capacity of decoding powers to access the same video at different quality levels. To serve as an error resilient tool, LC must be paired with UEP in the transport system, so that the base layer is protected more strongly.

There are many ways to divide a video signal into two or more layers in the standard block-based hybrid video coder. A video can be temporally down-sampled, and the base layer can include the bit stream for the low frame-rate video, whereas the enhancement layer(s) can include the error between the original video and that up-sampled from the low frame-rate coded video. The same approach can be applied to the spatial resolution, so that the base layer contains a small frame-size video. The base layer can also encode the DCT coefficients of each block with a coarse quantizer, leaving the fine details (the error between the original and the coarsely quantized value) to be specified in the enhancement layer(s). The base layer may then include the header and motion information, leaving the remaining information for the enhancement layer. In the MPEG and H.263 techniques, the first three options are temporal, spatial, and SNR scalability, respectively, and the last one is data partitioning.

In all the approaches discussed here, including UEP and LC, a local minima is sought and these techniques would operate more effectively when the proposed integrated approach is employed.

3. CHANNEL ISSUES

With highly scalable video compression schemes [9], it is possible to generate one compressed bit-stream such that different subsets of the stream correspond to the compressed version of the same video sequence at different rates. Such a source coding algorithm would not have to be altered with varying wireless network/ channel conditions. This is particularly attractive in heterogeneous multicast networks where the wireless link is only a part of a larger network and the source rate cannot be adapted to the individual receiver at the wireless node.

3.1. Channel Coding

Although channel encoding stage typically uses Forward Error Correction (FEC) codes, the highest coding gain (near entropy level) for Rayleigh fading Additive White Gaussian Noise (AWGN) channels is achieved using Trellis Coded Modulation (TCM). Due to variation in importance of different bits within a bit-stream, protection of source bits using unequal error protection (UEP) schemes is critical to further enhance the performance of channel encoder. TCM schemes proposed for fading mobile channels provide unequal source sensitivity matched error protection when compared to sequential FEC coding and modulation.

Conventional block and convolution codes are successfully used to combat the bursty errors of fading noisy wireless channels/ networks. Rate Compatible Punctured Convolution codecs (RCPCs), which implements UEP, provide bit sensitivity matched FEC protection for sub-band codecs where some of the encoded output bits can be removed or ”punctured” from the bit-stream. A variety of different rate bit protection schemes can be designed using the same decoder and protecting the more error sensitive bits by a stronger low rate code and the more robust source coded bits by a higher rate, less powerful FEC code.

Although both source and channel encoding work towards the goal of making the data more resilient to channel errors, source encoding attempts to compress the information sequence into minimum possible bits (entropy level compressions) while channel encoding introduces redundancy to make the transmission data more robust. A tradeoff needs to be attained in between source and channel coding in order to encompass both maximum compression of data bit-streams and optimum redundancy introduction for successful transmission of error resilient image/video sequences which leads to the following subsection.

3.2. Joint Source/Channel Coding

A common approach for building joint source/ channel codecs [9] is to cascade an existing source codec with a channel codec wherein the key aspect lies in distribution of the source bits and channel bits between the source and channel codecs so that the resulting distortion is minimized.

With bandwidth being the only constraint, solution to the optimal source/ channel bit distribution problem can be approached by first constructing the operational distortion rate curve as function of bits for each sub-band of a wavelet decomposition and then applying one dimensional bit allocation algorithm. The optimal distribution of bits within a sub-band is done by using exhaustive search through all combinations of channel coding rates and quantization step sizes. One common thread among these analyses is that the joint source/ channel codec is adaptive to the channel condition, which is assumed to be known to be estimated correctly.

The joint source/channel coding involves (a) finding an optimal rate allocation between source coding and channel coding for a given channel loss characteristics, (b) designing a source coding scheme, which includes the specification of the quantizer to achieve its target bit rate, and (c) designing/choosing the channel codecs to match the channel loss characteristics and achieve the required robustness.

Based on the feedback information system in the design of the joint source/ channel coding scheme, the optimizer makes an optimal rate allocation between the source coding and channel coding mechanisms and conveys this information to the source and the channel encoders. The source encoder, then chooses an appropriate quantizer to achieve its target bit rate and the channel encoder chooses a suitable channel code to match the channel loss characteristics. This results in important low frequency sub-bands of images being shielded heavily using channel codes while higher frequencies are shielded lightly. This UEP technique reduces channel coding overhead, which otherwise is pronounced in bursty wireless channels.

The QoS guarantees of wireless channel models are quite similar to those of differentiated service networks with a major difference being that the parameters involved rapidly change with time. This could be taken care of by allocating bandwidth to VBR in multiple ways that depend on channel variation with time. One of such ways is to allocate the bandwidth equal to the actual packet rate of the video stream and so all the data can be delivered to the destination without any delay. When the application can tolerate large buffering delay at the source and in the network, the bandwidth requirement can be decreased. Apart from bandwidth allocation methods at the channel end, a performance measure, yet to be determined in section 5, is desirable at the transmitter/ receiver end to conduit these guarantees into a deplorable model that fits well with the source Video QoS requirements.

4. ERROR CONCEALED DECODING

Residual errors are inevitable when transmitting a video signal, regardless of the error resilience and channel coding methods used. Decoder error concealment refers to this recovery or estimation of lost information due to transmission/ channel errors. Assuming a motion compensated video coder, there are 3 types of information that may need to be estimated in a damaged MB: the texture information, including the pixel or DCT coefficient values for either an original image block or a prediction error block; the motion information consisting of motion vectors for a MB coded in either P- or B- mode; and finally the coding mode of the MB. For coding mode information, the techniques used are driven by heuristics. The other two approaches are considered in the following sub-sections.

4.1. Recovery of Texture Information

All the techniques that have been developed for recovering texture information make use of the smoothness property of image/ video signals and essentially they all perform the same kind of spatial/ temporal interpolation. The MV field to the lesser extent, also shares the smoothness property and can be recovered by using spatial/ temporal interpretation. The texture information recovery techniques include:

1. *Spatial Interpolation*: It is the technique of interpolating the damaged blocks from pixels in adjacent correctly received blocks. Usually, because all blocks or MBs in the same row are put in the same packet, the only available neighboring blocks are those in the current row and the row above, and typically boundary pixels of the neighboring blocks are used for interpolation purposes. Instead of interpolating the individual pixels, a simpler approach is to estimate the DC coefficient of the damaged block and replace the damaged block by a constant equal to the DC value which can then be estimated by averaging the DC values of the surrounding blocks. One way to facilitate such spatial interpolation is by an interleaved packetization mechanism so that the loss of one packet will damage only every other block. The missing DCT coefficients of the displaced frame difference are estimated by applying

a maximal smoothness constraint at the border pixel of the missing block, where first and second order derivatives are used for quantifying smoothness. Since the DCT transformation is linear, the computation can also be performed in the pixel domain. Another spatial interpolation approach is to use the projection onto convex sets (POCS) technique. The general idea behind POCS based estimation method is to formulate each constraint about the unknown as a convex sets. The optimal solution is the intersection of all the convex sets, which can be obtained by recursively projecting a previous solution onto individual convex sets.

2. *Temporal Interpolation*: MC temporal prediction is an effective approach to recover a damaged MB in the decoder by copying the corresponding MB in the previous decoded frame, based on the MV for this MB. The recovery performance by this approach is critically dependent on the availability of the MV, which must be first estimated if it is also missing. MC temporal error concealment techniques might provide better results than any of the spatial interpolation techniques. To reduce the impact of errors in the estimated MVs, temporal prediction may be combined with spatial interpolation. MPEG-2 provides the capability of temporal error concealment for I-Pictures, since the transmission of additional error concealment motion vectors is allowed in MPEG-2.

A shortcoming with spatial interpolation approach is that it ignores the received DCT coefficients. This could be resolved by requiring the recovered pixels in a damaged block to be smoothly connected with its neighboring pixels both spatially in the same frame and temporally in the previous/following frames. If some of the DCT coefficients are received for this block, then the estimation should be such that the recovered block be as smooth as possible, subject to the constraint that the DCT on the recovered block would produce the same values for the received coefficients. The above objectives can be formulated as an unconstrained optimization problem and the solutions under different interpolation filter in the spatial, temporal and frequency domains.

4.2. Coding modes and Motion Vectors

Inter frames are reconstructed using the motion vectors and the DCT coefficients of the prediction error and therefore, the loss of the motion vectors seriously degrades the decoded video. This degradation propagates to the subsequent inter frames [6] until an intra frame is encountered. In case of H.263, the loss of a MB motion vector propagates to the remaining MBs in the frame. In other standards including H.261, the previous motion vector is used for the encoding rather than the median of the neighboring vectors.

To facilitate decoder error concealment, the encoder may perform data partition to pack the mode and MV information separate partition and transmit them with more error protection. This is the error resilient mode for both H.263 and MPEG-4. Even after this error resilience implementation, there is a fair chance of the mode and MV information being damaged. One way to estimate the coding mode for a damaged MB is by collecting the statistics of the coding mode pattern of the adjacent MBs and find a most likely mode given the modes of surrounding MBs. An simple approach is to assume that the MB is coded in the intra-mode and use only spatial interpolation for recovering the underlying blocks.

For estimating the lost MVs, there are several operations: (a) The lost MVs can be assumed to be 0s, which performs well for video sequences with relatively small motion, (b) The MVs of the corresponding block may be used in the previous frame (c) The average of the MVs from spatially adjacent blocks may be used (d) The median of MVs from the spatially adjacent blocks can also be used; and (e) MVs could be re-estimated. It was observed that the last two methods produced the best reconstruction results. Different MVs can be used for different pixel regions in the MB, instead of estimating one MV for a damaged MB to obtain a better result.

5. ADAPTIVE VISUAL QUALITY ORIENTED QOS: AN END-TO-END APPROACH

A QoS guarantee is performed in each operational unit of a video communication system shown in Figure 1 and in all the error control mechanisms discussed above since it requires QoS guarantees to achieve its effective predetermined quality. Usually, these requirements [10] vary with respect to the application, perceptual quality, bandwidth and time. User related QoS is subjective and can be related in terms of spatial and temporal resolution scalability, SNR and resolution scalability. The spatial resolution of the perceived video is a measure of the number of pixels in each frame while the temporal resolution is the number of received frames in unit time (frames per second, fps or sometimes also referred to as bit-rate in terms of bandwidth criteria). SNR resolution scalability is the allowable loss to the visual quality of the video and is realized by adjusting the degree of quantization during the video coding process. When a larger quantizer scale is applied, the quality of the decoded block reduces, which leads to degraded SNR values. However, the coded block size can become smaller, which has a positive effect from a view point of effective resource usage within the network.

As seen above, the QoS requirements of the source video are predominantly different from the QoS guarantees of the wireless channel/network described in section 3, more so in case of real-time video communication where the QoS requirements of the application also change with time. This problem of relating these variations and bridging them is yet to be addressed. One way to consider this would be to statistically model the QoS requirements with variation in time and bandwidth of the channel and come up with a measure of the perceptual quality at the receiver end. An added constraint would be to force this measure to be a constant throughout the time and bandwidth variation for a reliable and robust real-time video communication. However, this measure is subjective and is varied with varying applications and end users.

The problems with the existing solutions described in sections 2, 3 and 4 are that firstly, they are fragmented and not integrated. In each step of the video communication system in Figure 1, a local optimum is sought rather than a global one in that the efforts to minimize channel errors are independent at the transmitter and receiver ends. Secondly, these efforts are open ended giving way to un-required overhead when each of the techniques considers its best optimum performance. In such a scenario, the performance of the integrated system is jeopardized to match the independent individual units' optima. And lastly, these techniques are non-adaptive with regard to bandwidth, display and viewing conditions and application types.

The above described problems in existing approaches lead way to an integrated adaptive approach which requires an end-to-end QoS criteria based on visual quality. This approach aims at having maximum possible perceptual quality at the viewers' ends with a pre-determined latency and a specified channel bandwidth. The QoS measure would then suffice the visual quality requirements for a given latency at the cost of required channel bandwidth, power consumption and algorithm complexity.

An effective way to approach this goal is by the notion of effective channel bandwidth which can be defined as the average number of packets that can be transmitted without error in a unit time multiplied by the number of bits of source video per packet (this bandwidth being independent of the packet size). An objective, intrinsic perceptual quality metric needs to be developed that can quantify this visual quality. The proposed approach would then aim at maximizing the perceptual quality by jointly considering error resilience, channel characterization, concealment/ post processing techniques for a given latency, fixed effective channel bandwidth and display/ viewing conditions. The implementation of such a codec strategy would involve dynamically monitoring effectively channel bandwidth and adapting to its variations.

In effect, the proposed integrated adaptive wireless video codec should aim at achieving a Constant Perceptual Quality (CPQ) by using an integrated end-to-end QoS oriented performance criteria. The design of this codec involves:

- defining QoS metrics for wireless video (although QoS requirements exist for both MAC-layer wireless networks and video quality, they are independent of one another, and so there exists a need to develop an integrated ”optimal” QoS requirements for wireless digital video transmission)

- defining a new subjective and/or objective perceptual quality performance metric that measures the parameters involved in maintaining CPQ at the end user by using the concepts of 'Effective Bandwidth' and 'Effective Capacity' [11] of physical and link layer channel modeling

- optimizing the metric parameters by implementing an adaptive feedback loop that involves network conditions, efficiency of scalable coding, error control and QoS requirements

- specification of the required formats (different for different network conditions) over varied range of implementations based on the optimized performance metric to achieve CPQ, and

- lastly, developing the required post-processing algorithms as given in [12] and [13] and error control mechanisms as discussed in sections 2, 3 and 4 to accentuate the performance of the designed codec.

A key technical barrier for designing and implementing a codec in this fashion is the problem of optimization in integration of all the steps: error control, parameter estimation, joint source-channel coding, scalable coding, compression and QoS requirements for maintaining CPQ. If the control parameters are not efficiently handled, then the codec presents a risk of giving almost a linear performance with respect to bit rate. In the next section, the design details of currently existing standard codecs are discussed with regard to the error control.

6. CODECS - INTERACTIVE ERROR CONTROL

The H.261, H.263, MPEG-1,MPEG-2 and MPEG-4 standards are all based on hybrid motion compensated Discrete Cosine Transform (DCT) coding algorithms. They work on the principle of first removing the temporal correlation and then removing spatial correlation. Hence, single bit errors in the compressed information can result in enormous errors in decoded video. Furthermore, due to predictive coding implemented in the above standards, errors will propagate from one frame to another until uncorrupted reference is re-established. Error resilience in these standards is provided precisely for these reasons [14]. Recently introduced codec standards, their characteristics, channel usage and applications are described in greater detail in [15].

6.1. H.263

H.263 [16] is the second generation video coding standard following H.261. It involves a more powerful 1/2 pixel motion estimation and compensation algorithm, overlapped block motion compensation and Signal to Noise Ratio (SNR), temporal and spatial scalability when compared to H.261. H.263 has a good error resilience structure [17] [18] [19] which includes the Group of Blocks (GOB) feature from H.261 and a number of annexes, which include the following:

- Group of Blocks (GOB): A GOB is defined as a spatially localized group of MBs within a frame that are coded independent of one another. GOB restricts the errors that propagate to the end of the frame to only those MB errors within the same GOB.

- Forward Error Correction (FEC) for coded video Signal (Annex H): This allows for 492 bits of the coded data to be appended with 2 bits of framing information and 18 bits of parity information to form a frame. The framing information bits are such that these bits form a specific frame alignment pattern over 8 frames. The FEC adds error correction capabilities to the encoder.

- Reference Picture Selection (RPS) (Annex N and Annex U): This enables the decoder to inform the encoder through the acknowledgment channel that an error has occurred. The encoder can then change the frame used for prediction of the current frame to the decoded last successful frame. This results in truncating the error propagation.

- Temporal, SNR and Spatial Scalability (Annex O): This annex defines independent layers that offer increasing levels of either SNR, temporal or spatial quality. Given a base layer with a base level quality, independent enhancement layers can be added to enhance the quality resulting in error resilience. Therefore, the base layer can be well protected bit stream providing a baseline quality of service, while enhancement layers can be less protected, since they only serve the QOS feature.

H.263 is well designed to facilitate interactive error resilience and concealment of a given bit-stream by structured detection and coding techniques. Most of these tools are already incorporated in the ITU-T recommendation H.263 version 3 (also known as H.263++) used currently and in the most recent test model version 2.0 (known as tmn2.0).

6.2. MPEG-2 and MPEG-4

MPEG-2 standard is most widely recognized video coding standard and has a large number of practical applications. Similar to the H.xxx series, MPEG-2 defines the standard [20] for source coding and protocols for the system level. Nevertheless, there are certain features that makes MPEG-2 different with regard to its impact on the error concealment of the bit-streams.

The video coding layer of MPEG-2 consists of a number of tools that can be used for error resilience and concealment. The more important ones include improved slice structured mode, intra motion vectors, data partitioning and scalability.

Video coding aspects of MPEG-4, however are based on the hybrid motion compensated DCT techniques which keep it's error resilience capabilities intact. MPEG-4 error resilience characteristics [21] can be divided into four broad categories - re-synchronization, data recovery and containment, and data concealment.

- Re-synchronization: The motivation, importance and basic methods for re-synchronization have been stated in the sub-sections of section 3 and remain the same in MPEG-4. The use of a Slice Structured Mode remains from MPEG-2 and is again based on the number of bits rather than the absolute MB address. Another approach to re-synchronization is the Fixed Interval Synchronization that requires start and resynchronizes codes to appear only at valid fixed interval locations within the bit-stream.

- Data Recovery and Containment: Data recovery has been greatly facilitated by the use of RVLCs. These are VLCs that can be decoded either backwards or forwards. The use of these codes assist in recovering some contaminated data. The use of re-synchronization information dispersed throughout the bit stream is an important containment technique.

- Concealment: Concealment techniques are more a function of the decoder than that of the standard and is based on the distinguishability criterion of different decoders choosing to conceal lost or corrupted information differently. Errors in the entire MBs can be concealed by replacing them at the decoder with the same MB from the previous decoded frame. This technique more effectively depends on the robustness of the decoders and their ability to handle errors efficiently.

These features of interactive error control with resilient and concealment techniques enable MPEG-4 to produce an acceptable video quality even at highly error prone environments of wireless transmissions.

7. CONCLUSION

The paper reviewed the most recent advancements in adaptive and interactive wireless video communications with primary focus on the error control tools and mechanisms of different video coding standards. Due to the effects of noisy fading wireless channel, it was found that error resilience and error concealment were the most important aspects of current research for successful realization of transmission and reception of image/video signals over bandwidth limited fading wireless networks/channels.

The paper provided an overview of the source and channel encoding implementations at the transmission end and specified the significance of the encoder error resilience by reviewing different error resilient mechanisms that have already been adopted in recent coding standards like H.263 and MPEG-4. The channel characteristics and its effects with regard to QoS requirements have been studied and a review of the differences between the QoS of video and channel is presented. In particular, the paper proposed a novel technique to resolve most of the problems associated with the current approaches by substantiating a need for an adaptive integrated QoS criteria based on visual quality and stressing the basis for such a realization.

The paper also discussed different error concealment techniques that can be implemented in the present day standards to take care of the channel/transmission errors in the obtained signal at the receiving end to obtain a near-original reconstruction of the transmitted information. Finally, it elaborated on the approaches the current state-of-the-art video coding standards like H.263 and MPEG could implement in order to achieve error control interactively both at the transmission and reception with regard to wireless video communication services over high error prone environments.

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