HIGH ESTIMATION CODING TECHNIQUE IN MPEG-4 VIDEO PROCESSING

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Abstract: The processing of video coding is increasing rapidly. Various coding techniques were proposed for the improvement of estimation accuracy in progressive video coding. The estimation accuracy in conventional video coding standards is needed to be improved as the availed conventional coding approaches are limited to a defined noise level. In current scenario the channel effects are dynamically variant in nature. This paper presents an approach for the improvement of error free coding in video processing.

Keywords: Estimation accuracy, MPEG-4 Coding, progressive streaming.

I. INTRODUCTION

The development of digital video technology in the 1980s has made it possible to use digital video compression for a variety of telecommunication applications: teleconferencing, digital broadcast codec and video telephony. Standardization of video compression techniques has become a high priority because only a standard can reduce the high cost of video compression codecs and resolve the critical problem of interoperability of equipment from different manufacturers. The existence of a standard is often the trigger to the volume production of integrated circuits necessary for significant cost reductions. An example of such a phenomenon where a standard has stimulated the growth of an industry is the spectacular growth of the facsimile market in the wake of the standardization of the Group 3 facsimile compression algorithm by the CCITT. Standardization of compression algorithms for video was first initiated by the CCITT for teleconferencing and videotelephony [7]. Standardization of video compression techniques for transmission of contribution-quality television signals has been addressed in the CCIR (more precisely in CMTT/2, a joint committee between the CCIR and the CCITT). Digital transmission is of prime importance for telecommunication, particularly in the telephone network, but there is a lot more to digital video than teleconferencing and visual telephony. The computer industry, the telecommunications industry and the consumer electronics industry are increasingly sharing the same technology--there is much talk of a convergence, which does not mean that a computer workstation and a television receiver are about to become the same thing, but certainly, the technology is converging and includes digital video compression. In the view of shared technology between different segments of the information processing industry, the International Organization for Standardization (ISO) has undertaken an effort to develop a standard for video and associated audio on digital storage media [4] [6] [7][11][12][13], where the concept of digital storage medium includes conventional storage devices CD-ROM, DAT, tape drives, winchesters, writable optical drives, as well as ISDNs, and local area networks. This effort is known by the name of the expert group that started it: MPEG--Moving Picture Experts Group--and is currently part of the ISO-IEC/JTC1/SC2/WG11[6][7][11][12][13]. The MPEG activities cover more than video compression, since the compression of the associated audio and the issue of audio-visual synchronization cannot be worked independently of the video compression: MPEG-Video is addressing the compression of video signals at about 1.5 Mbits, MPEG-Audio is addressing the compression of a digital audio signal at the rates of 64, 128 and 192 kbit/s per channel, MPEG-System is addressing the issue of synchronization and multiplexing of multiple compressed audio and video bit streams. This article focuses on the activities of MPEG-Video. The premise of MPEG is that a video signal and its associated audio can be compressed to a bit rate of about 1.5 Mbits/s with an acceptable quality.
II. MPEG-VIDEO CODEC
Because of the various segments of the information processing industry represented in the ISO committee, a representation for video on digital storage media has to support many applications. This is expressed by saying that the MPEG standard is a generic standard. Generic means that the standard is independent of a particular application; it does not mean however, that it ignores the requirements of the applications. A generic standard possesses features that make it somewhat universal eg., it follows the toolkit approach; it does not mean that all the features are used all the time for all applications, which would result in dramatic inefficiency. In MPEG, the requirements on the video compression algorithm have been derived directly from the likely applications of the standard. Many applications have been proposed based on the assumption that an acceptable quality of video can be obtained for a bandwidth of about 1.5 Mbits/second (including audio). We shall review some of these applications because they put constraints on the compression technique that go beyond those required of a videotelephone or a videocassette recorder (VCR). The challenge of MPEG was to identify those constraints and to design an algorithm that can flexibly accommodate them.

Many storage media and telecommunication channels are perfectly suited to a video compression technique targeted at the rate of 1 to 1.5 Mbits/s. CD-ROM is a very important storage medium because of its large capacity and low cost. Digital audio tape (DAT) is also perfectly suitable to compressed video; the recordability of the medium is a plus, but its sequential nature is a major drawback when random access is required. Winchester-type computer disks provide a maximum of flexibility (recordability, random access) but at a significantly higher cost and limited portability. Writable optical disks are expected to play a significant role in the future because they have the potential to combine the advantages of the other media (recordability, random accessibility, portability and low cost). The compressed bit rate of 1.5 Mbits is also perfectly suitable to computer and telecommunication networks and the combination of digital storage and networking can be at the origin of many new applications from video on Local area networks (LANs) to distribution of video over telephone lines [1].

Asymmetric Applications. In order to find taxonomy of applications of digital video compression, the distinction between symmetric and asymmetric applications is most useful. Asymmetric applications are those that require frequent use of the decompression process, but for which the compression process is performed once and for all at the production of the program. Among asymmetric applications, one could find an additional subdivision into electronic publishing, video games and delivery of movies. Symmetric applications. Symmetric applications require essentially equal use of the compression and the decompression process. In symmetric applications there is always production of video information either via a camera (video mail, video telephone) or by editing prerecorded material. One major class of symmetric application is the generation of material for playback-only applications; (desktop video publishing); another class involves the use of telecommunication either in the form of electronic mail or in the form of interactive face-to-face applications.

III. COMPRESSION APPROACH
The difficult challenge in the design of the MPEG algorithm is the following: on one hand the quality requirements demand a very high compression not achievable with intraframe coding alone; on the other hand, the random access requirement is best satisfied with pure intraframe coding. The algorithm can satisfy all the requirements only insofar as it achieves the high compression associated with interframe coding, while not compromising random access for those applications that demand it. This requires a delicate balance between intra- and interframe coding, and between recursive and non-recursive temporal redundancy reduction. In order to answer this challenge, the members of MPEG have resorted to using two interframe coding techniques: predictive and interpolative. The MPEG video compression algorithm relies on two basic techniques [8]: block-based motion compensation [1] for the reduction of the temporal redundancy and transform domain-(DCT) based compression for the reduction of spatial redundancy. Motion-compensated techniques are applied with both causal (pure predictive coding) and non-causal predictors (interpolative coding). The remaining signal (prediction error) is further compressed with spatial redundancy reduction (DCT). The information relative to motion is based on 16 X 16 blocks and is transmitted together with the spatial information. The motion information is compressed using variable-length codes to achieve maximum efficiency.

IV. RECURRENT COEFFICIENT ESTIMATION
Because of the importance of random access for stored video and the significant bit-rate reduction afforded by motion-compensated interpolation [9], three types of pictures are considered in MPEG. Intrapictures (I), Predicted pictures (P) and Interpolated pictures (B-for bidirectional prediction). Intrapictures provide access points for random access but only with moderate compression; predicted pictures are coded with reference to a past picture...
and will in general be used as a reference for future predicted pictures; bidirectional pictures provide the highest amount of compression but require both a past and a future reference for prediction; in addition, bidirectional pictures are never used as reference. In all cases when a picture is coded with respect to a reference, motion compensation is used to improve the coding efficiency. The organization of the pictures in MPEG is quite flexible and will depend on application-specific parameters such as random accessibility and coding delay.

V. ERROR FREE CODING APPROACH
A number of features of the codec design are designed to enable recovery of video fidelity in the presence of network transmission errors or losses. For example, the NAL design, with its highly robust treatment of sequence and picture header content (more properly called sequence and picture parameter sets in the standard), establishes a high degree of robustness. The basic slice structure design adds further robustness, as each slice is designed to be completely independent of all other slices of a picture in the basic decoding process (prior to application of the deblocking filter, which can also be made independent by the encoder if desired). No content of any slice of a picture is used for the prediction of syntax elements or sample values used in the decoding process of other slices in the picture. Additionally, the encoder can select to specify that the prediction of intra macroblock sample values in P and B slices will not use spatial neighbors that were not also coded in intra modes – adding further robustness against temporal error propagation. The multiple-reference-picture support can also be used by an encoder to enable further resilience against data losses and errors (basically by avoiding the use of any pictures as reference pictures in the prediction process if the fidelity of those pictures may have been adversely affected by transmission errors or losses). Going beyond these basic feature that are an inherent part of the design, there are essentially four additional tools that are specified in the standard for further protecting the video bitstream from network transmission problems, which may occur for example as a result of congestion overloads on wired networks or due to channel errors in wireless networks. These tools are: 1) Flexible Macroblock Order (FMO), 2) Arbitrary Slide Order (ASO), and 3) Redundant Slices (RS), and 4) Data Partitioning (DP). FMO can work to randomize the data prior to transmission, so that if a segment of data is lost (e.g. a packet or several packets), the errors are distributed more randomly over the video pictures, rather than causing corruption of a complete regions, making it more likely that relevant neighboring data is available for concealment of lost content. (FMO also has a wide variety of other uses, which we do not discuss here for the sake of brevity.) ASO allows slices of a picture to appear in any order for delay reduction (particularly for use on networks that can deliver data packets out of order). RS offers more protection by reducing the severity of loss using redundant representations of pictures. DP separates the coded slice data into separately-decodable sections according to how important each type of data is to the resulting picture fidelity.

VI. ERROR ESTIMATION CODING MPEG-4 VIDEO CODING
MPEG-4 video coding [3] adopts DCT transform and quantization on a block-by-block basis to reduce the spatial redundancy within one frame. In order to make use of the temporal dependence between adjacent frames, MB based motion estimation and compensation are deployed. Motion estimation searches for the best reference MB for the current MB yielding a motion vector for each MB that gives the position of this best match. Motion compensation computes the difference (residual error) between the predicted MB and the original MB. MBs that are coded using motion compensation are named inter MBs and MBs coded without motion compensation are named intra MBs. Both of them can be encoded with VLCs. A number of tools have been incorporated into the MPEG-4 video encoder to make it more error-resilient [2][5][10]. These error resilience tools include resynchronization, data partitioning, and reversible variable length codes. Resynchronization tools attempt to keep the synchronization between the decoder and the bitstream after errors have been detected. MPEG-4 inserts resynchronization markers into the bitstream periodically. Header bits, carrying information about the spatial location of the following MBs, are added immediately after these resynchronization markers. When synchronization is lost, these header bits can help the decoder to regain it. Data partitioning divides the data within a video packet into motion vector(MV) part and texture part(coded using a DCT based scheme), separated by a unique motion marker, which can help the decoder to prevent inter-field error propagation. When errors are detected solely in the DCT field of a MB, only MV is used to reconstruct the current frame, which will result in better quality than simple replacement from the previous frame. Illustrating the bitstream organization with resynchronization and data partitioning. The parameters; MB number, QP and HEC denote the number of MBs, the quantization and the header extension code. Reversible variable-length codes (RVLC) are a class of entropy codes that can be uniquely decoded in both the forward and reverse directions3, 4 such that, when the decoder detects an error while decoding the bit stream in the forward direction, it decodes the bit stream in the reverse direction until it encounters an error. Therefore, the decoder can recover some of the data that would otherwise have been discarded. The process
presented is evaluated on a basic MPEG-4 codec architecture on video sequence and the obtained observations are as illustrated below.

VII. SIMULATION OBSERVATION
A simulative analysis for the proposed video codec is carried out and computed on a Matlab tool. The Sequences of the video frames are been captured and processed per plane for video processing samples and the received sample are at 1:2 scale ratio with variable Gaussian noise level. The noise affected samples are then processed for the estimation of original samples. The simulation observations are as outlined below,

Case study I:

![Figure 1 Original video sequence considered](image1)

![Figure 2 per Plane video processing samples](image2)

![Figure 3 received sample at 1:2 scale ratio with noise effect](image3)

![Figure 4 retrieved sample sequence after estimation](image4)

Case study II:

![Figure 5 Original video sequence considered](image5)

![Figure 6 per Plane video processing samples](image6)
Figure 7 received sample at 1:2 scale ratio with noise effect

Figure 8 retrieved sample sequence after estimation

Figure 9 performance plot of the suggested approach over conventional approach at variable Gaussian noise variance level.

VIII. CONCLUSION
This paper presents an error free coding technique in MPEG-4 coding approach. A coding approach for maximization of error estimation at different Gaussian noise level is proposed. The estimation accuracy for various samples of captured video frames is carried out. The effect of noise Margin at different scaling factors for the captured video sequence is analyzed and evaluated. The obtained PSNR on the variation of Gaussian noise at different mean and variance level. This approach illustrates a basic coding approach for the noise free coding in video streaming at different scaling factors.

IX. REFERENCES