Advanced Telecommunications and Signal Processing Program

Academic and Research Staff
Prof. Jae S. Lim

Research Affiliate
Carlos Kennedy

Visiting Scientists
Dr. Cheul-hee Hahm
Mr. Gi-Byoung Park

Graduate Students
David M. Baylon
Brian Heng
Eric C. Reed
Wade Wan

Support Staff
Cindy LeBlanc

Introduction

The present television system was designed nearly 50 years ago. Since then, there have been significant developments in technology, which are highly relevant to the television industries. For example, advances in the very large scale integration (VLSI) technology and signal processing theories make it feasible to incorporate frame-store memory and sophisticated signal processing capabilities in a television receiver at a reasonable cost. To exploit this new technology in developing future television systems, the research areas of the program focused on a number of issues related to digital television design. As a result of this effort, significant advances have already been made and these advances have been included in the U.S. digital television standard. Specifically, the ATSP group represented MIT in MIT's participation in the Grand Alliance, which consisted of MIT, AT&T, Zenith Electronics Corporation, General Instrument Corporation, David Sarnoff Research Center, Philips Laboratories, and Thomson Consumer Electronics. The Grand Alliance digital television system served as the basis for the U.S. Digital Television (DTV) standard, which was formally adopted by the U.S. Federal Communications Commission in December 1996. The standard imposes substantial constraints on the way the digital television signal is transmitted and received. The standard also leaves considerable room for future improvements through technological advances. Current research focuses on making these future improvements.

In addition to research on issues related to the design of digital television system, the research program also includes research on signal processing for telecommunications applications.

1. Video Compression with Complete Information for Pre-Recorded Sources

Sponsor: Advanced Telecommunications Research Program

Project Staff: David M. Baylon

Traditional video compression algorithms focus on causal processing of the video data. The causal constraint is necessary for real-time video programs where delay is an important consideration. However, many video programs such as movies are pre-recorded and can be processed offline prior to compression. This study proposes using pre-processing for pre-recorded video programs to obtain useful information for compression. By exploiting information obtained about a program prior to compression, better video quality can be achieved.

An application of this general approach to a buffer-constrained quantization problem using MPEG-2 intraframe video compression is studied. Intraframe video compression is useful for noisy channels or when lower complexity is desired, although the approach can be applied to interframe compression as well. The issues addressed include what a priori information should be extracted, how this information should be used, and what performance gains can be achieved. In particular, rate-distortion curves are proposed as a set of a priori sufficient statistics for intraframe compression, and a noncausally computed target quality level is introduced into the distortion function. A new noncausal iterative algorithm is developed to reduce the number of quantization changes, and it is shown to be optimal under certain conditions. It is demonstrated that by using this algorithm, improved and more constant video quality can be delivered using the noncausal approach relative to a traditional causal approach. Gains of up to 1.0 dB in PSNR were observed in multiscene sequences, corresponding to savings of up to 10% in bit rate.

Since there exist many applications involving the transmission and storage of pre-recorded video programs, such as video-on-demand, digital versatile disk (DVD), etc., there are many potential applications of the proposed approach. In addition, the approach can be applied to processing and compression of other sources known a priori such as images and audio.

2. Study on Duplicate Video System

Sponsors: Samsung Electronics

Advanced Telecomunications Research Program

Project Staff: Cheul-hee Hahm

In video transmission with high quality and reliability, such as contribution and primary distribution of video for Olympic broadcasting, a duplicate redundant system based on private channels is often set up in order to prevent transmission failure. The basic idea of the duplicate redundant system is to decrease the risk of transmission failure by distributing the video signal into several streams. Generally, a regular channel and a backup channel are used in the duplicate video system. The regular channel is used for the normal periods without any transmission failure, and when it has some troubles, a backup channel is used. The requirements for the duplicate video system are to transmit the encoded video stream through separate channels and to reconstruct the picture from any single stream.

To satisfy these requirements, a simulcast scheme and a flat multi-scalable scheme using a shifted picture have been used. In the simulcast scheme, the same video bitstream is transmitted through each channel and the picture is reconstructed from each stream. Because the regular channel is used for most of time, the capacity of the backup channel of this scheme is usually wasted. In the flat, multi-scalable scheme using a shifted picture, the preprocessor shifts the pixel positions of the picture for one channel before encoding and the decoder restores the original positions of the pixels in one channel by inverse shifting and averages the two decoded pictures. However, the utilization of the backup channel of this scheme is low. So, to increase the utilization of the backup channel for failure-free video transmission, an alternate temporal sub-sampling scheme is currently being examined. In this scheme, the encoded even pictures and the encoded odd pictures are transmitted through separate channels and the pictures at the decoder are chosen from each decoded picture of each channel alternatively. By this method, we can use the full backup channel for failure-free video transmission.

3. Application of Deinterlacing for the Enhancement of Surveillance Video

Sponsor: Advanced Telecommunications Research Program

Project Staff: Brian Heng

Surveillance cameras have become an integral part of security systems worldwide. However, their usefulness to law enforcement officials is often limited by a lack of clarity. These sequences are typically recorded in interlaced format due to the fact that standard television displays use interlaced scanning. Interlaced video sequences contain only the even or the odd lines of a given frame. For a given bandwidth, this increases the display rate without reducing the apparent resolution. However, it also introduces a number of artifacts, such as line scroll and interline flicker. On many occasions, it is also necessary to view a single frame for printing or analysis, which requires some form of interpolation since only half of each frame is recorded. In an attempt to improve the quality of surveillance data, the process of deinterlacing can be used to interpolate the missing scan lines. However, the low frame rates used in these recordings, typically 4 or 5 frames per second, increases the difficulty of the task. This low frame rate decreases the temporal correlation between adjacent frames, which increases the interlacing artifacts making it very difficult to identify individuals in these sequences. This thesis investigates a number of deinterlacing techniques to determine which method demonstrates the highest performance for this type of application.

4. Multi-Dimensional Bit Rate Control for Video Communication

Sponsors: Charles Stark Draper Laboratory
Advanced Telecommunications Research Program

Project Staff: Eric Reed

In digital video communications, buffering is required to absorb variations between the source bit rate and the channel transmission rate. Hence, a bit rate control strategy is necessary to maintain the buffer level. In conventional bit rate control, the buffer level is maintained by adapting the quantization stepsize while the frame rate and spatial resolution remain fixed at levels chosen a priori. We investigate a Multi-Dimensional (M-D) bit rate control where the buffer level is maintained by jointly adapting the frame rate, spatial resolution and quantization stepsize. In this approach, the frame rate and spatial resolution are chosen automatically and can adapt to a nonstationary source.

We introduce a fundamental framework to formalize the description of the M-D buffer-constrained allocation problem. Given a set of operating points on a M-D grid to code a nonstationary source in a buffer-constrained environment, we formulate the optimal solution. Our formulation allows a skipped frame to be reconstructed from one coded frame using any temporal interpolation method and is shown to be a generalization of formulations considered in the literature. In the case of intraframe coding, a dynamic programming algorithm is introduced to find the optimal solution. The algorithm allows us to compare operational rate-distortion (R-D) bounds of the M-D and conventional approaches. We also discuss how a solution can be obtained for the case of interframe coding using the optimal dynamic programming algorithm for intraframe coding by making an independent allocation approximation.

We experiment with zero-order hold and global motion-compensated temporal interpolation and illustrate that the M-D approach provides bit rate reductions up to 50 %. We also show that the M-D approach with limited lookahead provides a slightly suboptimal solution that consistently outperforms the conventional approach with full lookahead. While our algorithm is computationally expensive, it can be directly used for nonreal-time applications and can be used to benchmark performance of limited lookahead strategies for real-time applications.

Publications

E. Reed and F. Dufaux. “Constrained Bit Rate Control for Very Low Bit Rate Streaming Video Applications,” To appear in IEEE Transactions on Circuits and Systems for Video Technology.

E. Reed and J. Lim, “Optimal Multi-Dimensional Bit Rate Control for Video Communication,” IEEE Transactions on Image Processing, submitted October, 2000.

E. Reed and J. Lim, “Multidimensional Bit Rate Control for Video Communication,” Proc. of SPIE: Applications of Digital Image Processing XXIII, vol. 4115, San Diego, CA, July, 2000.

5. Migration to Higher Resolution Digital Television Systems

Sponsors: Advanced Telecommunications Research Program

Project Staff: Wade Wan

The new U.S. digital television standard that recently went into commercial service incorporate many technological advances made over the past few decades. As a result, digital televisio systems are significantly better than their analog counterparts. Despite the substantial improvements, the digital television standard has a significant limitation in its video resolution. The 1080P format (1080 lines with progressive scanning at 60 frames per second) is not allowed in the standard. The need to migrate to this resolution (and other higher resolutions) in the future has already been recognized and desired by terrestrial broadcasters. Scalable coding techniques can be used to migrate to formats beyond the current standard. This involves transmitting a single independently coded base layer and one or more dependently coded enhancement layers. The base layer would be transmitted at a resolution format allowed by the current standard. An advanced receiver would receive the base layer and convert the decoded video to the desired 1080P format with the assistance of an enhancement layer. Standard receivers would ignore the enhancement layers and decode video according to the current standard. This backward-compatible approach is highly desirable so as not to render earlier receivers obsolete.

An important issue with any scalable coding scheme is the information that is encoded in the enhancement layer to efficiently aid the migration from the standard format to the desired 1080P format. The additional resolution of enhancement layers is provided by converting a decoded lower resolution video sequence to a higher resolution format and then encoding the residual, the difference between this sequence and the original high resolution sequence. Many scalable coding algorithms use a fixed method of format conversion for the entire video sequence. However, since the transmitter has access to both the original and decoded high resolution sequences, it can be used to adaptively select different format conversion methods for different regions in an intelligent manner.

This thesis is aimed at investigating the potential of adaptive format conversion (AFC) for providing video scalability. Since the parameters needed for AFC are small compared to residual coding, AFC can provide video scalability at low enhancement bitrates that are not possible with residual coding. Previous results have shown that a significant improvement in video quality can be gained by using only AFC (without any residual coding) in these regions. AFC is well matched to the migration path problem for digital television since the additional bandwidth for migration is expected to be low which prevents residual coding.