Chapter 3. Advanced Television Research Program

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3.1 Advanced Television Research Program

The present television system was designed nearly 35 years ago. Since then, there have been significant technological developments 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 techology in developing future television systems, Japan and Europe established large laboratories, funded by government or industry-wide consortia. The lack of this type of organization in the U.S. was considered detrimental to the broadcasting and equipment manufacturing industries, and, in 1983, a consortium of U.S. companies established the Advanced Television Research Program (ATRP) at MIT.

Currently, the consortium members include ABC, Ampex, General Instruments, Kodak, Motorola, NBC, NBC Affiliates, PBS, Tektronix and Zenith. The major objectives of ATRP are:

- To develop the theoretical and the empirical basis for the improvement of existing television systems, as well as the design of future television systems.
- To educate students through television-related research and development and to motivate them to enter careers in television-related industries.
- To facilitate continuing education of scientists and engineers already working in the industry.
- To establish a resource center for discussion and detailed study of problems and proposals
- To transfer the technology developed from this program to its industrial sponsors.

The research areas of the program include the development of transcoding methods and the design of (1) a channel-compatible advanced television (ATV) system, (2) a receiver-compatible ATV system, and (3) a digital ATV system. We have already made significant advances in some of these research areas. We have designed a channel-compatible ATV system, scheduled for testing in 1991 by the FCC for its possible adaption as the U.S. HDTV standard for terrestrial broadcasting.

3.1.1 ATRP Computer Facilities

The main ATRP computer facility currently is a VAX 11/785 minicomputer with approximately 2.4 GBytes of online disk space. Attached to the VAX is a DATARAM Wide Word Storage system which provides 320 MBytes of RAM. The high speed interface to the Wide Word system drives a three-dimensional interpolator that was constructed by graduate students in the lab. The three-dimensional interpolator can perform separable spatiotemporal interpolation. The output of the interpolator feeds a custom built data concentrator which drives a Sony 2k by 2k monitor, running at 60 frames/sec.

In addition to displaying high resolution real time sequences, the ATRP facilities include a 512 by 512 Rastertek frame buffer and an NTSC encoder. The Rastertek frame buffer feeds static images to nearly a dozen monitors that are distributed throughout the lab. The NTSC encoder allows us to record the results of sequence processing onto either 3/4 inch or VHS tape. For hard copy output, the lab uses an Autokon 8400 graphics printer for generating high resolution black and white images directly onto photographic paper.

For preparing presentations, ATRP also employs a Macintosh SE30 microcomputer and a Mac IIx, feeding an Apple LaserWriter.

To support the growing computation needs of the group, three Sun-4 workstations will be installed in the near future. They will have 24-bit color displays, local disk storage, and DSP boards to assist with computation-intensive image processing.

A fast network (FDDI) is under consideration to link the machines and to display devices such as the Dataram. The workstations will also support Ethernet and will be connected to Internet through the building subnet.

3.1.2 Receiver-Compatible Adaptive Modulation for Television

Sponsors

National Science Foundation
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Project Staff

Matthew M. Bace, Professor Jae S. Lim

There have been numerous proposals for developing methods to improve the quality of the current NTSC television picture. Most of these proposals have concentrated on methods for increasing either the spatial or the temporal resolution of the television picture. While these proposals promise significant improvements in picture quality, until an effective scheme to combat channel noise has been introduced, these improvements will never be fully realized. Degradations such as random noise ("snow"), echo, and intersymbol interference (channel crosstalk) are still the greatest barrier to high-quality television.

This research will attempt to develop a receiver-compatible scheme to reduce the effects of channel imperfections on the received television picture. In particular, the method of adaptive modulation will be employed to make more efficient use of the currently under-utilized bandwidth and dynamic range of the NTSC signal. By concentrating more power in the higher spatial frequencies and using digital modulation to send additional information in the vertical and horizontal blanking periods, we can make existing television signals more robust in the presence of high frequency disturbances. Furthermore, we can adjust the parameters of this scheme so that the modulated signal may be received intelligibly even on a standard receiver (although an improved receiver will be required to realize the full benefits of adaptive modulation).

Before we can conclude which adaptive modulation schemes are optimal, we must consider many details. Among the parameters which can be varied are: (1) the control over the adaptation and compression factors, (2) the form of the input low-pass filters, (3) the interpolation scheme to be used at both the transmitter and receiver, and (4) the encoding of the digital data. We will adjust these parameters to optimize the performance of the modulation scheme with repect to two fundamental performance criteria. These criteria are the degree to which (1) the channel degradations are removed when the signal is received on an improved receiver and (2) the signal is distorted when received on a standard receiver.

3.2 Adaptive Amplitude Modulation for Transform Coefficients

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Project Staff

David M. Baylon, Professor Jae S. Lim

In this project, we have shown that adaptive amplitude modulation/demodulation (AM/DM) is an effective noise reduction technique. However, this technique requires transmitting the adaptation factors as side information. It is important to minimize the required side information in systems that have limited transmission bandwidth. Our research will focus on representing the adaptation factors by a few parameters exploiting properties of the signal in the transform (frequency) domain.

Previous investigations of adaptive amplitude modulation have been based on time domain methods. Specifically, in two-dimensional subband filtering, an image is decomposed into a set of spatial frequency subbands that are adaptively modulated. Similarities among subbands are exploited by reducing the number of adaptation factors to about one-sixth the number of data points. Nevertheless, further reduction in the amount of side information is desirable.

This research will take a different approach to reducing the amount of side information by adaptively modulating the transform of the signal. Transform coefficients of typical images tend to decrease in energy away from DC. By exploiting this property, we can model the transform coefficients and the adaptation factors with a few parameters (for example, an exponential model). Consequently, we can significantly reduce the amount of side information compared with that required by previous methods.

Research will focus on determining the best way to model the adaptation factors with a few parameters in systems that are bandwidth and peak power constrained. Performance criteria of the various AM/DM schemes will include signal-to-noise ratios and overall image quality. Among the many ways of obtaining the coefficients (such as using subband filtering or the lapped orthogonal transformation (LOT)), the discrete cosine transformation (DCT) will be used because of its many desirable properties, including coefficient uncorrelation, energy compaction, and efficient computation using the fast Fourier transform (FFT). Issues that we will address include choosing the appropriate block size and determining the best AM/DM method with an adaptive coefficient selection scheme (such as used in image coding systems). We will study both two-dimensional images and three-dimensional video.

3.3 Transform Coding for High Definition Television

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Ibrahim A. Hajjahmad, Professor Jae S. Lim

Image coding has many useful applications. One important application is for compressing channel bandwidth for image transmission systems such as HDTV, video conferencing, and facsimile. Reducing storage requirements for tasks such as digital video recording is another important application of image coding.

Image coding can be divided into a number of classes, depending on which aspects of the image are being coded. One class is the transform image coder,[1] in which an image is transformed from the spatial domain to a different domain more suitable for coding. Then, the transform coefficients are quantized and coded. When received, the coded coefficients are decoded and then inverse transformed to obtain the reconstructed image.

To perform transform coding one must select an appropriate transform. In particular, the

[1] J.S. Lim, *Two-Dimensional Signal and Image Processing* (Englewood Cliffs, New Jersey: Prentice Hall, 1990); R.J. Clarke, *Transform Coding of Images* (London: Academic Press, 1985).

Discrete Cosine Transform (DCT)[2] is very useful because of two important properties. First, the energy compaction property states that a large amount of energy is concentrated in a small fraction of the transform coefficients (typically the low frequency components). Because of this property, only a small fraction of the transform coefficients need to be coded, while little is sacrificed in terms of the quality and intelligibility of the coded images. Second, the correlation reduction property reduces the high correlation among pixel intensities in the spatial domain. In effect, the redundant spatial information is not coded.

Currently, we are investigating the use of the DCT for bandwidth compression. In addition, we are studying new adaptive techniques for quantization and bit allocation to reduce further the bit rate without sacrificing image quality or intelligibility.

3.4 Filter Design for Multirate Filter Banks

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Steven H. Isabelle, Professor Jae S. Lim

Multirate filter banks have wide application in the areas of subband coding of speech and images. In this application, the signal is passed through a bank of bandpass filters. The resulting bandpass signals are decimated and then coded. At the receiver, the signals are interpolated and added to form the reconstructed signal. In this scheme, the coding technique can be adjusted on a frequency dependent basis to match the observer's perceptual characteristics. Clearly, the performance of this kind of analysis/synthesis system depends strongly on the properties of the filters in the analysis and synthesis filter banks. For example, there are two desirable properties: (1) in the absence of coding, the reconstructed signal should be nearly identical to the original, and (2) there should be little interaction between the different subband signals. One goal of this research is to develop improved analysis and design tools for multirate filter banks. To this end, we have developed a design algorithm with improved convergence behavior over existing methods for the special case of a two-channel perfect reconstruction filter bank.

The subband decomposition of an image is typically performed using separable filters along its horizontal and vertical axis. The use of nonseparable filters allows for directional selectivity in the subband decomposition and a potential improvement in the subjective quality of encoded images. A second goal of this research is to develop design techniques for general multidimensional, multirate filter banks.

3.5 Adaptive Spatiotemporal Filtering

Sponsors

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David Kuo, Professor William F. Schreiber

The current NTSC television standard specifies a frame rate of 60 fields/sec throughout the transmission chain. The purpose of this frame rate is for minimizing the visibility of annoying flicker at the display. However, to eliminate flicker, only the display must operate at the high frame rate; the channel does not need to be constrained to operating at 60 frames/sec. Because there is a great deal of correlation between neighboring frames of an image sequence, a high frame rate through the channel seems bandwidth inefficient.

One way to take advantage of the correlation between neighboring frames is to transmit

[2] N. Ahmed, T. Natarajan, and K.R. Rao, "Discrete Cosine Transform," *IEEE Trans. Comput.* C-23:90-93 (1974).

only a temporally subsampled version of the original sequence, relying on the receiver to recover the inbetween frames. However, our prior work suggests that the receiver must have more information than simply the subsampled frames. This research focuses on using motion vectors as part of the image sequence representation.

There are three main areas of focus in this research. First, we consider the use of adaptive spatiotemporal prefiltering as a means of reducing the aliasing that arises from temporal subsampling. Secondly, we explore the characteristics of the motion vectors. Finally, we consider how to use multiple frames of data to improve the motion estimation process.

3.6 Signal Processing for Advanced Television Systems

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Project Staff

Peter A. Monta, Professor Jae S. Lim

Digital signal processing will play a large role in future advanced television systems. Source coding to reduce the channel capacity necessary to transmit a television signal and display processing such as spatial and temporal interpolation are the major applications. Present-day television standards will also benefit significantly from signal processing designed to remove transmission and display artifacts. This research will focus on developing algorithms and signal models to (1) enhance current standards (both compatibly and with some degree of cooperative processing at both transmitter and receiver) and (2) improve proposed HDTV systems.

Using a receiver with a high-quality display and significant computation and memory, we can improve the American television standard, NTSC, in several ways. We can remove interlace artifacts, such as line visibility and flicker by converting the signal to a progressive format prior to display. We can greatly reduce color cross-effects with accurate color demodulators implemented with digital signal processing techniques. We have tested and implemented an algorithm for an advanced receiver that can recover a much improved image by exploiting structure in the film-NTSC transcoding process if film is the original source material.

Similar ideas apply to HDTV systems. For example, film will be a major source material well into the next century, and HDTV source coders should recognize film as a special case, trading off the inherent reduced temporal bandwidth for better spatial resolution. The MIT-CC channel-compatible HDTV system will adapt to film in this way.

3.7 Adaptive Frequency Modulation for Satellite Television Systems

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Advanced Television Research Program

Project Staff

Julien Piot, Professor William F. Schreiber

Frequency modulation is the first choice coding scheme in many existing applications such as satellite television transmission.

A simple model of image formation predicts large variations in the short-time bandwidth of the modulated signal. Based on this model, we adjust the frequency deviation in a small block of the picture to keep the bandwidth constant. We show that the resulting noise improvement is significant when we use a subjective measure of the transmission error. This measure, based on noise masking, has an average intensity related to the block statistics.

In some applications the modulated signal is bandlimited, resulting in envelope and phase distortion. Both terms generate artifacts in the recovered picture, mostly when noise is present in the link. We show through measurements that the peak short-time bandwidth is related to the severity of the distortion, hence justifying the prior approach to adaptation. We introduced improved algorithms that minimize the transmission noise while maintaining negligible distortion.

When the information is transmitted in the form of multiple components, we present a technique based on a sequential transmission such as subbands. This technique can also be used to transmit some side information, such as the adaptation function of the modulator. By adjusting the rate of transmission of the various components, we can minimize the subjective impairment. For example, we show that a vertical subband decomposition is very effective in reducing transmission noise, with a larger improvement than preemphasis techniques. Alternatively, adaptive modulation can be combined with this technique.

Finally, we applied the principle of adaptive frequency modulation to broadcasting and distributing television signals by satellite. We demonstrated a dual-in-one system, where two NTSC video signals are transmitted through one transponder. Another system proposes direct broadcasting by satellite of high definition television using subband coding and adaptive frequency modulation. A simulation of the two systems demonstrates that high quality transmission is possible in noisy narrow-band channels, using analog modulation.

This study was completed in December 1989.

3.8 Subband Coding for Channel-Compatible Transmission of High-Definition Television

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Ashok C. Popat, Professor William F. Schreiber

In recent years, subband coding has received considerable attention from the image coding community as a simple and effective means of efficiently representing image and image-sequence data.[3] We have proposed a three-dimensional (horizontal, vertical, and temporal) subband coding technique for application in a 6-MHz channel-compatible high-definition television (HDTV) distribution system.[4] Although preliminary "proof-of-principle" tests have demonstrated that the technique is effective, the tests have also shown that there is considerable room for improvement. The technique can be improved by adjusting various parameters in the system; these parameters include the degree of data compression, the number of subbands in each dimension, the type and length of the subband analysis/synthesis filters, and the means of selecting subband pixels to be retained and transmitted. We have observed a high degree of interdependency among many of the system parameters which has complicated the process of identifying the particular combination of parameters that is best suited to the present application. In particular, the strong interdependency seems to eliminate the possibility of finding the best choice for each parameter separately. A major objective of the present research is to search through the vast parameter space by judicious choice of parameters for computer simulation, and by objective and subjective evaluation of the simulation results.

One critical set of system parameters is the set of coefficients used in the subband analysis/synthesis filter banks. We have developed a novel approach to designing such filters based on time-domain numerical search; the approach is fairly general and has resulted in critically-sampled filter banks that

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3 J.W. Woods and S.D. Oneil, "Subband Coding of Images," *IEEE Trans. Acoustics, Speech, and Signal Processing.* 34:1278-1288 (1986); H. Gharavy and A. Tabatabai, "Subband Coding of Monochrome and Color Images," *IEEE Trans. Circuits Syst.* 35:207-214 (1988).

4 W.F. Schreiber, et al., *Channel-Compatible 6-MHzHDTV Distribution Systems,* CIPG Technical Report ATRP-T-79, MIT, January 1988.

are extremely well-suited to image coding applications.[5]

A seemingly basic principle of image subband/transform coding has emerged from the present study. In particular, the best choice for the length of the analysis/synthesis filters depends only weakly on the number of subbands, depending more strongly on the spatial extent over which the image can be well-modeled as stationary. Thus, the nonstationarity of images inevitably leads to an uncertainty-principle based tradeoff in the selection of the number of subbands and lengths of filters.

We also found that it is extremely important that the allocation of channel capacity is spatially varying. It is essential to be able to increase the number and/or fidelity of samples used in representing action regions of the image at the expense of more poorly representing inactive regions. Currently, we are devising a fixed-rate, practicable means of exploiting this principle.

3.9 Channel Equalization and Interference Reduction Using Adaptive Amplitude Modulation and Scrambling

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Project Staff

Adam S. Tom, Professor William F. Schreiber

Terrestrial broadcast channels and cable channels are imperfect. Random noise, multipath (ghosts), adjacent and co-channel interference, and an imperfect frequency response degrade these transmission channels so that the quality of the signal at the receiver is significantly below that at the transmitter. To appreciate the increased resolution of high definition images, degradation due to channel defects needs to be reduced. Conventional methods of channel equalization, which use adaptive filters, are limited by convergence time, length of filters, and computational complexity. We are researching a new method of channel equalization and interference reduction based upon the ideas of adaptive amplitude modulation and pseudo-random scanning (scrambling). This new method is not bound by the above limitations; however, it is limited by the energy of the channel degradations produced.

Adaptive modulation is a noise reduction technique that is applied only to high frequency components of signals. Prior to transmission, a set of adaptation factors are multiplied with the input signal to raise the amplitude of the signal according to the strength of the signal. At the receiver, the signal is divided by these same adaptation factors. In this manner, the random noise added in the channel is reduced by a factor equal to the adaptation factor. The noise is reduced more in the blank areas relative to the busy areas.

Scrambling is a technique for reducing the effects of multipath, adjacent and co-channel interference and also imperfect frequency response. Prior to transmission, we pseudorandomly scan the input signal, scrambling the signal so that it appears as random noise. Then we transmit this scrambled signal through the channel and perform the reverse of the scrambling at the receiver. Consequently, any degradations in the channel are themselves scrambled at the receiver and, thus, have the appearance of pseudo-random noise in the decoded signal, while the desired signal remains sharp and in full bandwidth.

Since the resultant signal in the receiver now has a noisy appearance, we apply the noise reduction technique of adaptive modulation to the input signal. In our scheme, we apply adaptive modulation to the input signal before scrambling. The coded signal is transmitted through the imperfect channel and decoded at the receiver. The decoding con-

[5] A.C. Popat, "A Note of QMF Design," unpublished memo, Advanced Television Research Program, MIT, December 1988; A.C. Popat, "Time-Domain Numerical Design of Critically Sampled Filter Banks," presentation viewgraphs, Advanced Television Research Program, MIT, October 1988.

sists of doing the reverse of the scrambling and then the reverse of the adaptive modulation. Scrambling causes any channel degradations to have a noiselike appearance, and adaptive modulation reduces the appearance of this pseudo-random noise. In this manner, the degradations to a transmitted signal are reduced and the channel is equalized.

3.10 A Novel QMF Design Algorithm

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Kambiz C. Zangi, Professor William F. Schreiber

In this research, we presented a new algorithm for designing two-band quadrature mirror filter (QMF) banks. We show that this algorithm is computationally more efficient by a factor of eight than existing algorithms. Moreover, because this algorithm is not sensitive to the choice of the initial guess, by using it we can also design two-band QMF banks in which one of the filters is predescribed.

This work was completed in December 1989.

Professor William F. Schreiber shown with his graduate students. From the left: David Kuo, Adam S. Tom, Professor Schreiber, and Peter A. Monta.