Coding of the Motion-Compensated Residual for an All-Digital HDTV System

by

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Abstract

This thesis presents the research performed toward applying digital signal processing concepts to the compression of High Definition Television (HDTV) signals. The Federal Communications Commission has ruled that any HDTV system to be used in the U.S. must utilize the same 6 MHz channel bandwidth as the current color television system. All-Digital systems relying on digital compression and digital transmission have been proposed as a means of producing very high quality video and audio within the limited channel bandwidth available.

Motion-estimation/motion-compensation is initially applied to the HDTV video in order to predict the current frame from the previous frame. The error in this prediction, or motion-compensated residual (MC-residual), is then coded and transmitted over the channel. The focus of this research is to determine the optimal method to represent and code the MC-residual so that the highest quality video can be produced with the limited available bit rate (approximately .2 to .35 bits/pixel).

To choose the optimal transform/subband filtering scheme for processing the residual, the Block DCT, the Lapped Orthogonal Transform, and the Multi-scale schemes are examined. Both the Block DCT and the Lapped Orthogonal Transform suffer from structured blocking artifacts that are visually degrading. The Multi-scale scheme does not suffer from these artifacts and is judged to have the highest overall performance. The MC-residual is more highpass in nature than a typical image, and to take advantage of this, two new Multi-scale representations are created. Each results in a sharper reconstructed image at the expense of increased ripple artifacts. The precise filterbank chosen for the Multi-scale scheme is critical to its performance. A new filterbank eliminated the detracting artifacts intrinsic to the original filterbank, but at the expense of reduced coding efficiency.

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Chapter 1

Introduction

High Definition Television (HDTV) will most probably be the next generation in the evolution of video media for the home. HDTV will have many improvements over the current color television system. The HDTV set will comprise a larger screen with a greater width/height ratio. The screen will display a clearer and sharper picture, with more spatial detail and better motion rendition. The picture quality will be comparable to that of 35 mm film. This will be coupled with four channels of CD quality digital audio and a variety of special services. From the entertainment point of view, HDTV will give the viewer the feeling of 'actually being there'. HDTV will also have an immense number of applications in fields as diverse as medicine, manufacturing, and the military.

All of these improvements on our current system lead to a great increase in the amount of information that needs to be transmitted over the broadcast channel and to the receiver. This may be done by increasing the bandwidth for each HDTV channel, however, there is only a limited amount of RF spectrum available, and this must be allocated very carefully among the various services that use it.

Recently the Federal Communications Commission ruled that in order to apportion the scarce RF spectrum, any HDTV system to be used in the U.S. must utilize the same 6 MHz channel bandwidth as the current color television system. Toward this goal the Advanced Television and Signal Processing Group at MIT is developing an All-Digital HDTV system that would allow the transmission of high quality digital video and audio within the 6 MHz channel ruled by the FCC.

An HDTV system can be divided into two primary sections, a compression portion to compress the enormous amount of data that needs to be transmitted, usually referred to as *source coding*, and a communication portion that takes care of the over-the-air transmission to the home receivers, usually referred to as *channel coding*. With the current television system in the U.S., both of these portions involve totally analog processing. The proposed All-Digital HDTV system will consist of digital signal processing to compress the data and digital modulation to transmit the data. The term All-Digital comes from the fact that both the source and channel coding will be totally digital.

Since both the compression and transmission will be done digitally, a number of benefits befall the system. Applying complex digital signal processing concepts to the compression can reduce the redundancy of the video and result in the transmission of very high quality video in a much smaller bandwidth. The digital transmission will give everyone within the broadcast area the same received signal, and the fact that the encoder will know exactly what signal every receiver will receive also allows the use of predictive processing for aiding in the compression.

One can get an idea of the amount of processing necessary to compress the HDTV signal into the available bandwidth by examining the amount of information that must be transmitted. Our system will transmit 720 by 1280 pixel frames 60 times a second. Each pixel contains red, green, and blue color components at 8 bits of quantization each. This corresponds to an uncompressed bit rate of

$$\left(\frac{720 \times 1280 pixels}{frame}\right)\left(\frac{60\, frames}{sec}\right)\left(\frac{3\, colors}{pixel}\right)\left(\frac{8\, bits}{color}\right) = 1,327,104,000\,\frac{bits}{sec} \qquad (1.1)$$

This result of 1.3 Gbits/sec (Gbps) is an enormous amount of data to be transmitted. When considering that the proposed digital transmission scheme of 16-point quadrature amplitude modulation (16 QAM), for example, would result in the transmission of only 4 bits/Hz $\cdot$ sec, and with only a little more than 5 MHz of the 6 MHz HDTV channel actually usable, this would result in a transmission rate of approximately

$$\left(\frac{4\,bits}{Hz \cdot sec}\right)(5\,MHz) = 20,000,000\,\frac{bits}{sec} \tag{1.2}$$

After apportioning bits for the audio, error correction codes, and special services, there remains only about 17 Mbps for the video. This corresponds to approximately .32 bits/pixel. Considering that each pixel consists of three colors with 8 bits of resolution for each, the limited bit rate will require a compression ratio of approximately 75.

This extremely-low bit rate per pixel mandates the use of novel and very complex digital signal processing concepts to compress the video while retaining the video's original high resolution and excellent motion rendition. At the Advanced Television and Signal Processing Group at MIT the design of a digital video compression system for HDTV was initiated in 1990. The audio portion for the HDTV system was also designed. The digital communication portion of the system and the actual construction and testing of the system will be performed by the VideoCipher Division of General Instruments Corp., located in San Diego California.

The digital video compression system initially applies motion-estimation/motion-compensation to the video in order to predict the current frame from the previous frame. The error in this motion-compensated prediction, or motion-compensated residual as it is usually called, is then coded and transmitted over the channel. The goal of this thesis was to determine the optimal method to represent and code the motion-compensated residual so that the highest quality video can be reconstructed at the receiver with the limited bit rate available.

This thesis was written so a reader with no previous background in television and only an elementary background in signal processing would be able to gain an understanding of the important issues relevant to designing a very high quality digital video compression system. Chapter 2 begins by describing our proposed HDTV system and comparing it to the current television system. It proceeds by discussing important coding concepts for the system and their application into our overall system. Chapter 3 discusses the research performed in choosing the optimal transform/subband filtering scheme for processing the residual. The most important comparison criterion was the visual appeal of the reconstructed video at the receiver. Chapter 4 explores the idea that image compression techniques are being applied to a motion-compensated residual as opposed to a normal image. A number of new representations for taking advantage of this are created and analyzed. Chapter 5 describes a detracting feature of the video and how it was eliminated by the application of an improved filterbank. Chapter 6 summarizes and provides conclusions for the research performed, and Chapter 7 discusses a number of areas for future research.

Chapter 2

Coding Concepts and Overview of the System

Before attempting to design an HDTV system one should understand the current state of television, the changes or improvements that will encompass HDTV, and the basic digital signal processing concepts that may be applied toward the HDTV system. Deep understanding of the various digital signal processing concepts will enable one to realize the tradeoffs and take advantage of them to create the highest quality HDTV system possible.

This chapter begins by reviewing some important facts about our current television system and how HDTV will compare to it. Then, the basic digital signal processing concepts important for compression of the video are described. The digital processing concepts are then brought together in an overview of our digital video compression system, and in a further description of where the research completed for this thesis falls within the overall system. Finally, a brief description of the video sequences used to test the system is given.

2.1 Television Today and High Definition Television

2.1.1 Television Today

Let us begin by describing some terms. The smallest resolvable "point" in a picture is called a picture element or pixel. Pixels are usually aligned in a rectangular grid to form an image. Video consists of a series of still images or frames that are shown one after the other in rapid succession to produce the illusion of continuous motion.

When a camera records a picture onto film, the entire image is recorded simultaneously. For a TV camera, the image is scanned and the pixels are recorded one at a time. The National Television System Committee (NTSC) television system used in the U.S. utilizes interlaced scanning to record, transmit, and reproduce the individual pixels of each video frame. Each frame of the video is divided into an even and an odd field composed, respectively, of even and odd scan lines of the frame. The even and odd fields can be displayed in succession and interlaced together to reconstruct the original frame, hence the name. In the NTSC television system a frame contains 525 scan lines, of which only about 480 contain video, and 30 frames/sec or 60 fields/sec are transmitted. Interlaced scanning therefore refreshes the screen 60 times a second for only 30 frames/sec being transmitted. In this manner interlaced scanning allows a high refresh rate of the television screen, necessary to reduce the visual perception of flicker, while transmitting at a much lower frame rate.

2.1.2 High Definition Television

HDTV will produce video that is sharper, clearer, and higher in resolution and will incorporate better motion rendition than the current television system. This will be done by utilizing more pixels per frame and more frames per second to give improved spatial and temporal resolution. The current television system has a spatial resolution of approximately 480 by 420 pixels per frame, and transmits 30 interlaced scanned frames/sec. The HDTV system incorporates a spatial resolution of 720 by 1280 pixels per frame and 60 progressively scanned frames/sec. The increase in spatial resolution will produce more detail per frame with the individual frames having approximately the resolution of 35 mm film. The interlaced scanning of the even and odd scan lines for the current system results in visually degrading motion artifacts for some types of motion. The progressive scanning of our proposed HDTV system will scan and reproduce every line of each frame consecutively. No interlaced scanning artifacts will be created and this coupled with the higher total frame rate will result in improved motion rendition.

Current television has a width to height or aspect ratio of 4:3. When we view our surroundings we have a broader viewing angle horizontally than vertically. In order to emulate this, the HDTV system will have an increased aspect ratio of 16:9, in this way adding to the perception with HDTV that one is actually there, living the experience.

In order to conserve bandwidth, the current color television system also grossly bandlimits the chroma components of the video signal. This leads to reduced chroma resolution and resulting visual artifacts. Our proposed HDTV system will retain full chroma resolution and thereby eliminate these visual artifacts.

2.2 Important Coding Concepts

The aforementioned improvements in HDTV over NTSC mean that there will be a massive increase in information needed to be transmitted for HDTV, while there will be no allowable increase in the available channel bandwidth. This mandates the application of very complex digital signal processing concepts to reduce the necessary data rate. This section contains a brief discussion of the most important digital signal processing concepts for the compression or coding of the HDTV video signal. The purpose of this section is to give a reader with a basic background in signal processing an intuitive feel for how and why each of these concepts can be useful. This approach is intended to give an intuitive overview of each concept, noting particularly important properties of each, without becoming immersed in the mathematical details. There are a number of very good references the reader may consult for further details [1, 2, 3]. Furthermore, when important concepts related to this thesis are discussed, the important historical papers are referenced.

Digitizing video: sampling and quantizing

One should begin by realizing the usually ignored, but subtly important fact that the world around us is continuous. What one visually perceives with one's eyes is continuous in time, in spatial dimensions, in amplitude, and in color. The attempt to gather in the world electronically, process, and then prepare it for viewing, intrinsically involves sampling and quantizing. Temporal sampling is performed when creating the individual frames of the video. The individual pixels are spatial samples of each frame. The color of each pixel is usually constrained to the additive combination of three color components, where the amplitude of each color is quantized to a discrete value.

When digitizing video, a continuous signal is represented by a discrete one in time, spatial dimensions, and color, with the amplitudes quantized to discrete levels. This modeling of a continuous signal as a digital one allows the use of bits to describe it, and opens up the power of the digital computer to process and reconstruct it. This brings to mind the sampling theorem, reminding us that the signal must be preprocessed and sampled in an appropriate manner. Of course, the quantization must also be done carefully. The method by which the sampling and quantization is applied is partly an artform, as the purpose of the processing is not necessarily to perfectly reconstruct the world, but to represent it in a manner that is perceptually equivalent, or at least acceptably so.

$$\begin{matrix} \textit{The Continuous} & & \textit{Discrete and Quantized} \\ & \Longleftrightarrow & \\ \textit{World} & & \textit{Representation} \end{matrix}$$

For this research it has been assumed that 60 frames/sec of progressively scanned video at 720 by 1280 pixels/frame where each pixel is composed of red-green-blue color components at 8 bits quantization for each, is appropriate for HDTV applications. All this research presumed that this digitized video contains no demeaning characteristics that should be alleviated, and has as a final goal the reconstruction of this perceptually perfect digitized video at the receiver.

Color Processing

Each pixel of the digitized video contains red, green, and blue (RGB) color components. The human visual system does not resolve fine color detail as well as variations in the luminance. Therefore, by linearly transforming RGB to another color coordinate system, the properties of the human visual system may be exploited. In our NTSC television system, RGB is linearly transformed to the YIQ color coordinate system. Y is the luminance or intensity of the image and contains most of the information. It is precisely the component one sees when viewing a black and white television. I and Q are the chrominance components and they produce the color detail as seen on a color television which utilizes YIQ. The human visual system can not perceive detail in the chrominance as well as it can in the luminance because of its poor spatial frequency response to colored light. A number of methods can take advantage of this, including subsampling the chrominance or quantizing the chrominance more coarsely than the luminance. In this manner, significant bit rate reductions can be achieved by using the YIQ representation instead of RGB.

Spatial and temporal processing

Video contains information along both spatial and temporal dimensions. There is a considerable amount of redundancy among these dimensions, and the reduction of this redundancy will result in significant reductions in the required bit rate. An example of extreme spatial redundancy is the case in which an entire frame has a constant pixel value, then only one pixel value is required to describe it. An example of temporal redundancy is the case in which the same frame is repeated 60 times, then the frame only needs to be coded once. Both of these cases are extreme, but they exemplify how large improvements in the bit rate can be achieved.

Typical video can usually be modeled as stationary for short segments along the spatial and temporal dimensions. The change in pixel value from one pixel to the next or from one frame to the next is usually quite small. The difference between the current pixel and the previous pixel can then be calculated and coded. The idea of quantizing this difference, or residual, instead of the value itself is very important. This is because even though the residual has twice the dynamic range of the original pixel value, it has a much smaller variance (and entropy) and therefore can be coded much more efficiently. Another way to think of this is that the adjacent pixels somehow contain memory, similar to a discrete Markov process. The previous value is like a prediction of the current value. Subtracting the previous value (the prediction) from the current value removes the redundancy between the two values. The residual is then the new information which is not contained in the previous value (the prediction).

Making a prediction and then taking the difference, followed by encoding the prediction error or residual, is the main idea behind all differential pulse code modulation (DPCM) systems. DPCM systems try to reduce the bit rate by eliminating as much redundancy as possible from the previous pixel values. Their coding efficiency is determined by the accuracy of their predictions.

Motion-estimation/motion-compensation is a form of predictive coding usually applied along the temporal dimension that can result in a considerable improvement in the required bit rate. Transform/subband filtering schemes are also extremely effective for low bit rate applications and they can be applied along either the temporal or spatial dimensions. Motion-estimation/motion-compensation and transform/subband filtering will be described further in the following pages.

All of these methods can result in improvements in the necessary bit rate for a given image quality. It is very important to realize that even after the basic processing steps have been chosen, small steps can be taken to greatly improve the performance. All of the aforementioned processing techniques treat the whole image equally. But when viewing a typical image, one notices that some sections of the image contain a large amount of detail while others contain very little detail. By realizing the changing characteristics of different images and of different portions of the same image, and by processing these individual portions while taking advantage of their particular characteristics, significant further improvements can be attained. The idea of *adaptive processing*, realizing changing situations and processing them differently, can be applied to virtually every step of the digital processing. This will result in an increase in complexity of the system, but this may be easily outweighed by the improvements in performance. With the improvements in digital technology that make implementations of complex DSP algorithms easier and easier, adaptive processing will be a key element in creating a high quality digital HDTV system and future high performance digital signal processing systems.

Motion-estimation/motion-compensation

A video sequence is a sequence of still frames that are shown in rapid succession to give the impression of continuous motion. Even though each of the frames is distinct, the high frame rate necessary to achieve proper motion rendition usually results in a great deal of temporal redundancy among the adjacent frames. For example, adjacent frames often contain the same object, albeit possibly at different spatial positions. By realizing that the same object is at different spatial positions in subsequent frames, one does not have to code the object multiple times. Processing and transmission of the object once is sufficient for as long as it is observable in the video sequence. The process of estimating the motion of objects within a video sequence is known as motion estimation. The processing of images compensating for the presense of motion in the sequence is called motion-compensation. The combined processes of motion-estimation and motion-compensation produce a prediction of the current frame from the previous frame. The error in this prediction, usually called the motion-compensated residual, can then be processed and transmitted. In this manner motion-estimation/motion-compensation can be viewed as a form of predictive processing along the temporal direction very similar to DPCM.

Motion estimation methods can be classified into two groups, region matching methods and spatio-temporal constraint methods. In region matching methods, the previous frame is partitioned into separate regions, each of which is examined to determine if it has moved to a new spatial position in the new frame. The name region matching corresponds to the matching of a region in the previous frame to a region in the current frame. The criteria often used for matching include correlation, minimum mean-square-error, and minimum sum of pixel differences. When a match is found, a motion vector is generated which describes the motion of the region from the previously encoded frame to the current frame. This motion vector is then used with the previously encoded frame to produce a prediction of the current frame to be encoded.

The spatio-temporal constraint methods are based on the assumption that small regions within consecutive frames of the video undergo simple translational motion with uniform velocity. Under this assumption, the video signal $f(x, y, t)$ can be modeled as

$$v_x \frac{\partial f(x,y,t)}{\partial x} + v_y \frac{\partial f(x,y,t)}{\partial y} + \frac{\partial f(x,y,t)}{\partial t} = 0 \tag{2.1}$$

where x and y are spatial variables, t is time, and v_x and v_y are the horizontal and vertical velocity components to be determined. These velocity components are once again used to compute a prediction of the current frame from the previous frame.

The assumption of uniform translational motion for the spatial-temporal constraint methods is very restrictive and they often do not achieve the same performance as the block matching methods. The block matching methods, on the other hand, are much more computationally intensive. Other important considerations for motion-estimation/motion-compensation (ME/MC) include the size of the blocks, the search area for the movement, accuracy of the motion vectors, and the method by which the motion vectors are estimated, via full search or a hierarchial or multi-grid approach.

Transform/subband filtering

Transform/subband filtering schemes take a different approach to increase the coding efficiency. Transform schemes are based on the idea that an image can be linearly transformed into another domain where most of the energy of the image is contained within a small fraction of the transform coefficients. The coding and trans-

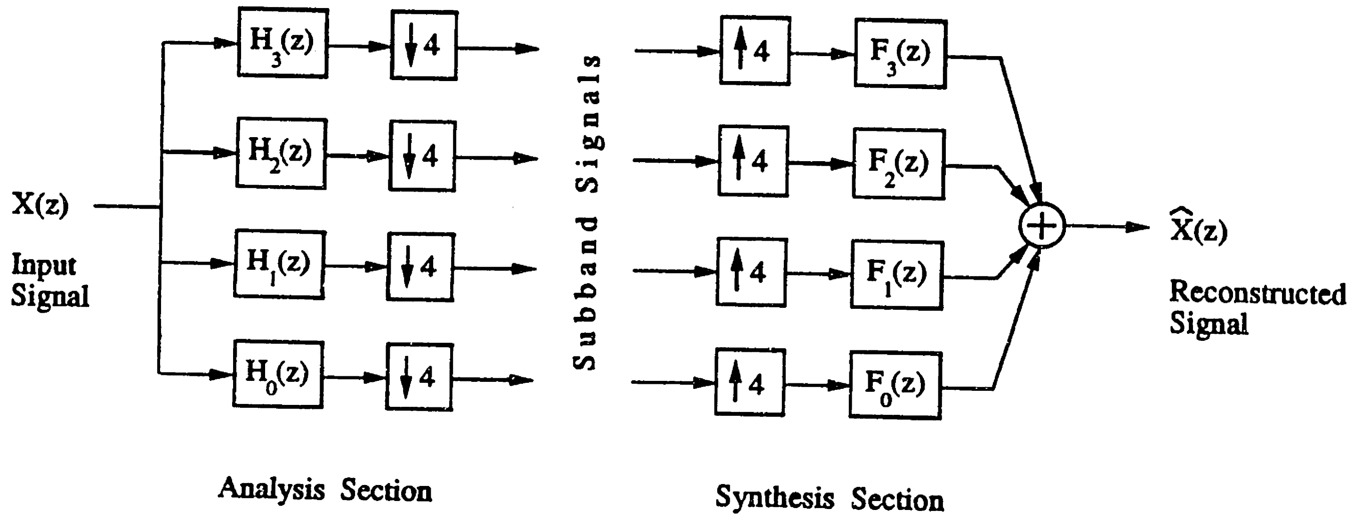


Figure 2-1: A one-dimensional subband filtering scheme.

mission of these few energetic coefficients may then result in the reconstruction of a high-quality image with minimal distortion. Some possible transforms that may be used include the Discrete Fourier Transform (DFT) and the Discrete Cosine Transform (DCT). In order to take advantage of the varying characteristics of an image over its spatial extent, the image to be processed is typically partitioned into 8×8 or 16×16 blocks which are then independently transformed and adaptively processed. The partitioning of the image into small blocks before taking the transform not only allows spatially adaptive processing, but also reduces the computational and memory requirements necessary for performing the transform [1].

Subband filtering schemes process an image by filtering it into various frequency bands or subbands. A one-dimensional four-band analysis/synthesis subband filtering scheme is shown in Figure 2-1. Once the signal is divided into many subbands, each of the subbands can then be coded with its own coder, which is specifically designed to take advantage of the particular characteristics of its subband. This idea of dividing a signal into subbands with subsequent adaptive processing of each subband has been applied to speech coding [4] and image coding [5, 6] with very good results.

There is an important point about most subband filtering schemes that one should notice. When taking the transform of a signal, such as the DFT, the number of transform coefficients is equal to the number of signal samples. On the other hand, if one attempts to divide a signal into subbands simply by applying an M-band filterbank, the number of samples resulting from the filterbank will be M times the number of original signal samples. This would not be a good first step for a coding algorithm. Because each subband signal is a bandpassed version of the original signal, and therefore does not occupy the full bandwidth of the original signal, it may seem possible to decimate the subband signals without losing information. Subband filtering schemes that decimate the subband signals so that the number of subband samples are equal to the number of original signal samples are called maximally decimated or critically sampled filterbanks [7]. Figure 2-1 is an example of this type of filterbank. Also, as the subband filters are not ideal brickwall filters, there will be aliasing resulting from the decimation process. Fortunately this aliasing can be eliminated in the reconstruction by the careful choice of the synthesis filters. The design of these filters will be described further in Chapter 3.

An image is a large 2-dimensional signal and computing the 2-D transform or performing a 2-D convolution with it can be computationally very expensive. For this reason only transform/subband filtering schemes that can be applied separably to the rows and columns of the image are usually considered. Also the application of fast algorithms wherever possible is very important. Transform/subband filtering schemes also require a large number of samples to code efficiently. For example, a 2 or 3-point transform would not result in much reduction in bit rate. For this reason motion-estimation/motion-compensation is usually utilized for temporal processing instead of a transform/subband filtering scheme. ME/MC would only require 2 frame stores for processing, while a useful transform/subband filtering scheme would require a minimum of 3 or 4 frame stores, making the decoder more expensive for consumers to buy.

Transform and subband filtering schemes can both be viewed as processes that partition an image into its frequency or subband components. The two types of schemes are therefore conceptually very similar. This similarity between the two schemes will be examined in detail in Chapter 3.

Zonal coding vs adaptive selection

The main concept behind transform/subband filtering schemes lies in the observation that not all the transform/subband coefficients need to be coded and transmitted for acceptable signal reconstruction. Only a small fraction of the most energetic coefficients need to be coded and transmitted in order to obtain very high quality video. The question that then arises is which coefficients should be transmitted?

One approach, known as *Zonal Coding*, is to always code and transmit the same group or zone of transform/subband coefficients. But this method would process the whole image in the same manner, without accounting for the local variations in image characteristics that may occur. A more adaptive approach would result in higher performance. One possibility is to group the transform/subband coefficients into zones and then determine and transmit appropriate zones of coefficients for each region of the image. In this manner the selection of coefficients can be made adaptive to the individual characteristics of the local regions of the image. This method is often referred to as *Adaptive Zonal Coding.* One problem with determining and transmitting zones of coefficients as opposed to always transmitting the same coefficients is that the decoder must know which zones of coefficients were transmitted. Therefore, extra side information describing which zones were picked must be sent, increasing the bit rate. This problem of extra side information is inherent to most adaptive schemes where there is a decision to be made as to the method of processing. The decoder needs to know how the encoder processed the data. This side information is not necessary if the decoder can determine from the previously transmitted data what the adaptive processing will be, but this is not usually the case for most adaptive processing schemes.

Adaptive zonal coding is more adaptive than simply selecting the same coefficients all the time, but it is possible to achieve even higher adaptivity. Instead of selecting groups of coefficients at a time, one can select single coefficients at a time. Selecting single coefficients at a time is referred to as Threshold Coding, Pel-Adaptive Selection, or *Adaptive Selection.* Adaptive selection allows higher adaptivity than zonal coding, but there is also a great amount of side information necessary to describe which coefficients were selected. This side information is often referred to as the *Selection Information.* The amount of selection information required for adaptive selection is much greater than the side information required for zonal coding, but the visual improvements resulting from adaptive selection have been found to far outweigh its increase in bit rate and complexity.

How does one determine which coefficients should be selected? A good starting point here is to select the transform/subband coefficients based on energy. The assumption being made is that the coefficients containing a large amount of energy have a greater impact on the reconstructed video than the coefficients containing a smaller amount of energy. Coefficients with the highest amount of energy may be selected until the required number of bits necessary to encode them approaches the total available number of bits.

Coding the selection information

The selection information is simply the information stating which coefficients were selected to be coded. The selection information needs to be transmitted and therefore should be coded to reduce the necessary bit rate. It can be viewed as an image of 1's and 0's stating whether each coefficient was selected (1) or not (0). In this way one can view the selection information as a 1-bit image to be coded, and all the coding tools developed toward coding 1-bit images, such as used in facsimile machines, can be brought to bear.

Depending on whether the selected coefficients are in a transform or subband filterbank arrangement, they will be grouped together in particular arrangements that one may take advantage of when coding the selection information. For example, if all the DC coefficients are grouped together and one wants to select all the DC coefficients, one simply says to the receiver "this group is selected". With these coefficient arrangements, the joint correlation among selected coefficients can be taken into account to improve the coding performance.

Another very useful method of coding the selection information is runlength coding. Runlength coding is the process of coding runs of coefficients. For example, a run of length 5 of selected coefficients or a run of length 10 of unselected coefficients. The longer the runlengths, in general, the higher the possible performance. Different scanning methods are utilized to increase the runlengths. Zigzag or serpentine scan runlength coding within appropriate blocks of coefficients in either the subband or transform domain representation can be very beneficial in terms of coding the selection information at a very low bit rate.

Quantization

A subband/transform coefficient, for example, is an analog value with infinite possible resolution. A digital computer has a finite number of bits and can not replicate this infinite resolution. Much more importantly, when attempting to transmit an analog value with a digital transmission scheme the accuracy of the representation is limited by the available number of bits, which is usually very few. To represent an analog value with a finite number of bits, only a finite number of reconstruction or quantization levels can be used. Quantization is the noninvertible *mapping* of an analog value to its quantized representation so that it can be described by a digital codeword.

There are many issues involved with quantization. First of all, should all the analog variables be quantized separately, or should they be quantized together? How should the quantization levels and the decision levels be chosen? The quantization levels may be chosen so that the distance between quantization levels is always the same, usually referred to as uniform quantization, or so that the distances differ, usually referred to as nonuniform quantization. In either case the distance or stepsize between quantization levels is very important. Also, the treatment of analog values around zero is very important. There may or may not be a reconstruction level assigned to zero. Also a dead zone may be utilized around zero to perform coring for noise reduction or bit rate reduction.

The chosen quantization method directly affects the other processing steps taken. For example, the analog transform/subband coefficient values may be quantized separately or they may be quantized jointly with their selection information. Jointly quantizing variables is called *vector quantization* or *block quantization* and it takes advantage of the statistical (linear and nonlinear) dependency among the variables. With this method the selection information and amplitude information are tied together in a single processing step. Another example in which quantizing affects the other processing steps is when the quantization levels are chosen so that each of them has an equal probability of occuring. This is a logical method of assigning quantization levels, but it obviates the benefit of applying entropy coding to the quantization levels. Entropy coding will be discussed shortly.

Taking the human visual system into account

It is very beneficial when considering adaptive selection, quantization, and other processing methods, to realize that the quality which we perceive the HDTV video to be is directly related to how our human visual system responds to the video.

For example, some transform/subband coefficients are more important than others. The human visual system responds differently to the various coefficients and by taking this into account the performance of the system can be greatly improved. Toward this goal, the transform/subband coefficients may be weighed based on frequency and luminance/chrominance component in order to exploit the variation in sensitivity of the human visual system. The weighing may be done to affect the adaptive selection of the coefficients and/or to affect the relative coarseness or fineness of their quantization.

Another example is that when there is a scene change or motion in the video, the spatial resolution of the human visual system decreases while the temporal resolution is what is critical. On the other hand, when a scene is relatively constant it is the spatial resolution that is more important. Performing the digital processing to take advantage of the changing spatio-temporal resolution of the human visual system can result in significant perceptual improvements in the video.

Researchers are beginning to have an understanding of the human visual system, but there are still many things that are unknown. Also, modeling of the human visual system is something that is not easily numerically quantifiable. This makes adapting the system to take advantage of the human visual system very much of an artform, requiring many tests with minor changes and careful viewing of the results of each test. It is very important to note that even if one processing method results in a lower mean-square-error (or any other objective criterion) than another method, that does not necessarily mean that the resulting video is perceptually more appealing. It is not the mean-square-error that we "see" when we view a video.

Entropy coding

The various quantized coefficient values may be directly transmitted or may be assigned specialized codewords which are actually transmitted. The assigning of special (custom) codewords to the quantized values allows one to take advantage of the statistical properties of the data. This is because the data to be transmitted has a certain amount of information, or entropy, based on the probability of the different possible values or events occuring. For example, an event that occurs very infrequently conveys much more new information than an event that occurs very often. By realizing that some events occur more frequently than others, the average bit rate may be decreased.

If the various events are transmitted directly, their digital codewords will be of the same length independent of their probabilities of occurring. This is referred to as constant length or *uniform* codeword assignment. By assigning *variable* length codewords to the different events based on their probabilities of occurring, one can take advantage of the statistical properties of the data. The events that are more likely to occur may be assigned shorter length codewords while the events that are less likely to occur will be assigned longer codewords. This will result in a variable bit rate per event and a reduction in the *average* bit rate of the system.

Adaptively assigning codewords based on the statistical properties of the data is referred to as Entropy Coding. Huffman and arithmetic coding are two very important coding methods which take advantage of the statistical properties of the data.

Allocation of bits

Entropy coding accounts for the statistical properties of the data and thereby reduces the average bit rate, but it also produces a variable bit rate output from the encoder. Because the channel has a fixed transmission rate, the allocation of bits must be regulated between the variable bit rate encoder and the fixed bit rate channel. One possible method is to have a buffer which collects and holds the bits within the encoder before passing them on to the transmitter. The buffer would have a feedback mechanism to the adaptive selecter, quantizer, and/or entropy coder to vary their efforts depending on the buffer's fullness. For example, if the buffer begins to fill, the quantizer may be made more coarse. This would reduce the bit rate and avoid possible overflow of the buffer. As the buffer begins to empty, the quantizer may be made finer. This would increase the bit rate and fill up the buffer, preventing possible underflow of the buffer. Optimally, the buffer should be held in a half full state all the time. Careful design of the buffer control mechanism is required to keep the buffer in a proper state of fullness, to ensure stability, and to ensure that the buffer never overflows or underflows.

Other important considerations

When designing our digital HDTV compression system, we must carefully consider the scenario our system is being designed for. The intention of the system is to produce the highest quality video possible with the limited bit rate available in the terrestrial television broadcast scenario. Toward this goal, we must find and eliminate all detracting artifacts that our processing produces. In the broadcast scenario, there will be few transmitters, but very many receivers. There are physical and economical limitations on memory and processing power for economical receivers. A great deal of complex digital signal processing may be performed at the transmitter, but the receiver should be kept as simple, from a cost point of view, as possible. The system must be designed for rapid channel acquisition and image buildup once a receiver is

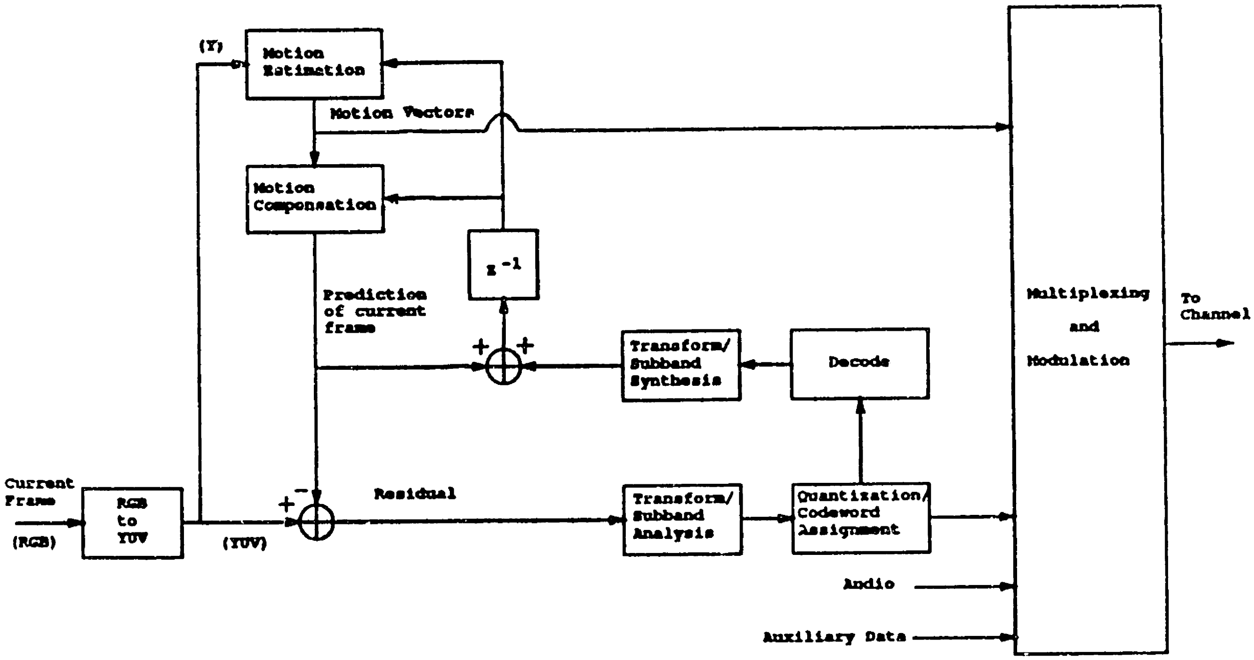


Figure 2-2: The encoder portion of the All-Digital HDTV system.

turned on, a channel is changed, or a scene change occurs. It is very important for the system to be made as robust as possible to minimize the effects of channel errors. Also, if the system could be made to vary gracefully with the bit rate, then the same system could be used at various bit rates and applications.

2.3 Overview of the system

In order to produce the highest visual quality possible for our digital video compression system, all of the previously discussed digital signal processing concepts have been brought together. The encoder portion of the system is shown in Figure 2-2.

Progressive scanning is utilized to circumvent the motion artifacts that result from interlaced scanning and also to simplify the signal processing. The system linearly transforms the red, green, and blue color components to the YUV color space. Y is the luminance and U and V are the chrominance. This color space is part of the Society of Motion Picture and Television Engineers (SMPTE) 240M production standard and results in higher color processing performance than the normal YIQ color space used with today's color television.

As previously discussed, the human visual system can not perceive detail in the chrominance as well as it can in the luminance. Subsampling of the chrominance is usually done to take advantage of this reduced resolution. But this subsampling can lead to color bleeding and other visually unpleasant effects, especially for computer generated images and other video material specifically designed to take "advantage" of HDTV resolution. For this reason our system retains full chrominance resolution.

Motion-estimation/motion-compensation is applied along the temporal direction to reduce the temporal redundancy in the video. The motion-estimation is performed by partitioning the frame into 16 by 16 blocks which are then used with a block matching algorithm. The search range for the movement is $\pm$ 16 pixels horizontally and vertically with half pixel accuracy for the motion vectors. The motion vectors are estimated only from the luminance (Y) component and the same motion vectors are applied to the luminance (Y) and chrominance components (U, V).

Motion-compensation is applied to the previously encoded frame and the motion vectors to generate a prediction of the current frame. The error in this motion-compensated prediction, or motion-compensated residual as it is usually called, is then adaptively transformed/subband filtered to reduce the spatial redundancy of the residual.

The purpose of this research was to determine the optimal transform/subband filtering scheme for representing and encoding the motion-compensated residual. The results of the research will be discussed in the subsequent chapters.

Not all the transform/subband coefficients need to be coded and transmitted. Eliminating the low energy coefficients significantly reduces the required data rate without adversely affecting the visual quality of the system. The transform/subband coefficients are weighed based on the human visual system response and then adaptively selected based on their weighed energy. The selected coefficients are then quantized with the quantizers designed specifically to take advantage of the human visual system's response for each coefficient. Entropy coding is then applied to the selection information and the quantized values through the generation of Huffman codes. Because the distribution of the quantized values differs for each frequency band, it is beneficial to code each frequency band differently. For this reason, frequency adaptive codebooks were designed to take advantage of the different statistical properties of the different coefficients.

For coupling the variable bit rate entropy coder with the constant bit rate transmitter, a global buffer control was implemented. The global buffer control is more computationally intensive than a local buffer control and also requires greater memory, but it results in more precise control of the allocation of bits. The local buffer control is only locally adaptive, it allocates bits only based on what it sees in a given portion of the frame, without being aware of the rest of the frame. The global buffer control iteratively interacts with the complete frame before deciding on the allocation of the bits. This allows the buffer to optimally allocate the bits on a global basis instead of a local basis, significantly improving the performance of the system. The feedback mechanism for the buffer control operates jointly on the adaptive selection and quantization to ensure that the total number of allocated bits is within the capacity of the transmission system.

Many forms of source material do not contain 60 frames to be transmitted per second. Many of them, including film, contain 24 or 30 frames per second and the common practice is to frame repeat to achieve the 60 frames necessary for transmission. Our system is designed to realize when the source material is of this form and to allocate bits to take advantage of this. This form of processing is referred to as adaptive source coding or *source-adaptive encoding*, and it can result in significant increases in performance for very small changes in complexity.

Probably, the key element for the performance of the whole system was the *adaptivity* designed into nearly every aspect of the system. The initial encoding is adaptive to the type of source material to be encoded. The motion-estimation/motion-compensation is adaptively suppressed based on if the residual is more difficult to code then the original frame (as in a scene change). The transform/subband filtering process is applied adaptively to small regions of the residual. The transform/subband

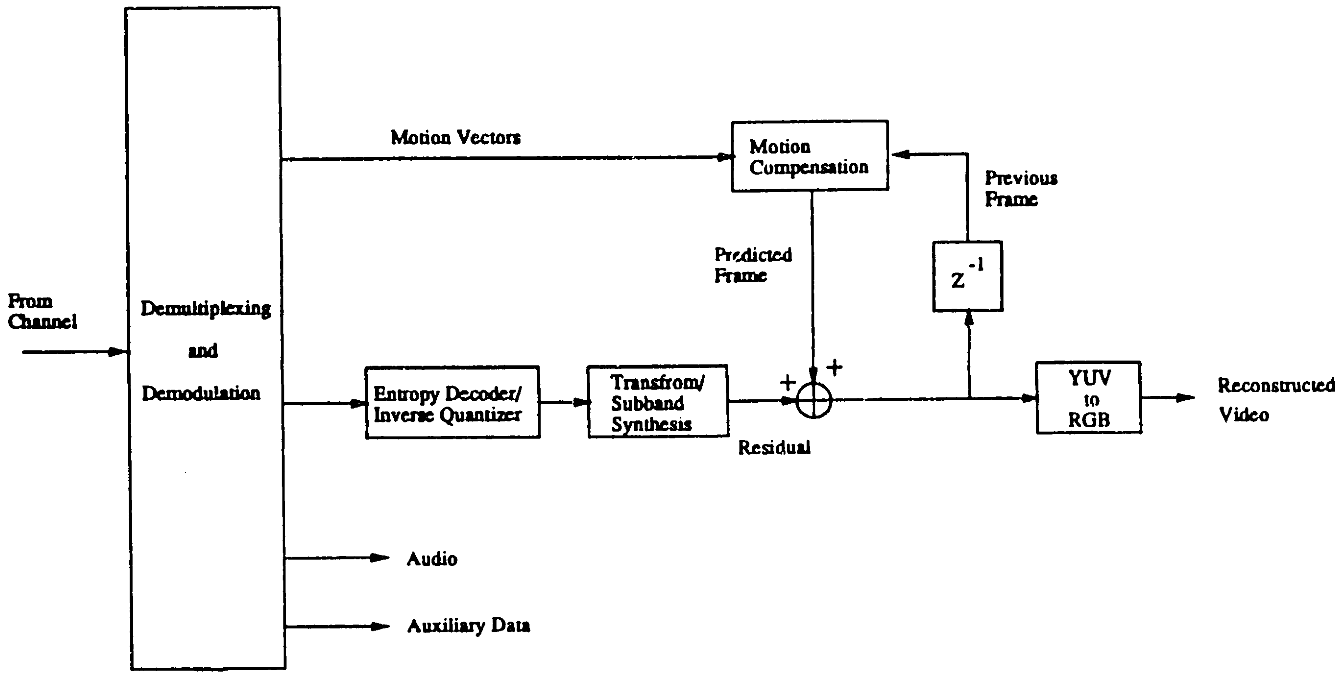


Figure 2-3: The decoder portion of the All-Digital HDTV system.

coefficients to be coded are adaptively selected. The weighing factors for the selection and the quantization stepsizes are adaptively set based on luminance/chrominance and frequency components. The Huffman codebooks for the quantization values were designed in a frequency adaptive manner. And the allocation of the bits is adaptive to the whole frame to be processed. In one word, *adaptivity* is the key.

The decoder at the receiver performs the inverse of the encoding process and this is shown in Figure 2-3. A further discussion of the digital HDTV video compression system and the relevant issues may be found in Monta's Ph.D. thesis [8]. Monta was the principle designer of the digital HDTV video compression system.

2.4 Generating test sequences

Digital video test sequences were required to test the digital HDTV system. To generate the testing material, a number of high resolution single frame images were processed with a form of cubic spline interpolation to create video with zooms and pans. Most of the test sequences were comprised of these synthetically generated sequences. These artificial sequences were in some ways easier to code and in some ways more difficult to code than normal video. They were easier in that they did not contain some of the noise inherent to many low to mid performance video cameras today. This noise corresponds to high frequency components in the video that the system attempts to code as best as it can. They were more difficult to code because they contained a great deal of detail and there was also very fast motion in ways that the motion-estimation/motion-compensation could not perform well.

Four of the most important test sequences all shown within this thesis are:

balloon	$\Longrightarrow$	synthetic video sequence
girl	$\Longrightarrow$	synthetic video sequence
picnic	$\Longrightarrow$	synthetic video sequence
toy-table	$\Longrightarrow$	real video sequence

In the future a large portion of the video shown may be artificially created with computer animation/graphics, and therefore artificial video may be a very appropriate test subject.

Chapter 3

Choosing the Transform/Subband Filtering Scheme

3.1 Main Ideas

The first step in creating a coding algorithm for the residual is to find the optimal method to transform/subband filter the residual. This will result in reducing the spatial redundancy of the residual and thus concentrating most of its energy within a small fraction of the coefficients. To first order, the greater the energy compaction the fewer coefficients needed to be transmitted in order to reconstruct a high quality image at the receiver. The optimal transform/subband filtering scheme will be chosen based on the quality of the reconstructed video at the receiver with the limited bit rate available. Before attempting to analyze various transform/subband filtering schemes, we should examine the relationship between transform and subband filtering.

3.2 Transform vs Subband Filtering

Transform and subband filtering have historically been classified as separate entities. This is primarily due to the methods by which each is computed. Transform schemes usually partition a signal into non-overlapping blocks and then decompose the blocks into basis functions. This may also be referred to as block transforms. Subband filtering schemes convolve the signal with a set of bandpass filters and then decimate the results. These two schemes are actually conceptually equivalent. Subband filtering can be viewed as a block transform, and a block transform can be viewed as a subband filtering process. A block transform groups an image first in terms of spatial location and then in terms of frequency content. Subband filtering groups an image first in terms of frequency content and then in terms of spatial location.

If an image is grouped first by frequency content (subband) and then by spatial position, the coefficients of each subband form a filtered and decimated image, and the filtered coefficients are said to be arranged in a spatial-frequency or subband representation. Each subband is a "picture," and has features of the original image. If the image is grouped first by spatial location and then by frequency content, the coefficients within each spatial block form a spectrum of the frequency content of that particular block. The coefficients are then arranged in transform representation.

The transform scheme can be viewed as a special case of subband filtering where the impulse responses of the synthesis filters are the transform basis functions, the impulse responses of the analysis filters are equal to the time-reversed basis functions, and the decimation factor in each subband is equal to the transform length or block size. The two schemes then result in simple rearrangements of the same coefficients. That is, both transform and subband representations can be simple rearrangements of each other. Even though the two representations contain the same coefficients, these arrangements give further insight into the image being analyzed. The subband representation shows the spatial correlation among subbands. As an example, one can see a vertical edge in many of the horizontal high-frequency subbands. On the other hand, the transform representation allows spatial localization for determining the frequency content of the image. This is illustrated in Figure 3-1. An image is divided into 8×8 blocks and the 2-dimensional Discrete Cosine Transfrom (DCT) is applied to each block. This is often referred to as the Block DCT. The figure shows the DCT coefficients arranged in both the transform and subband representations. As is easily seen, each representation stresses particular information about the image. Both of these representations can be beneficial for image coding [9, 10].

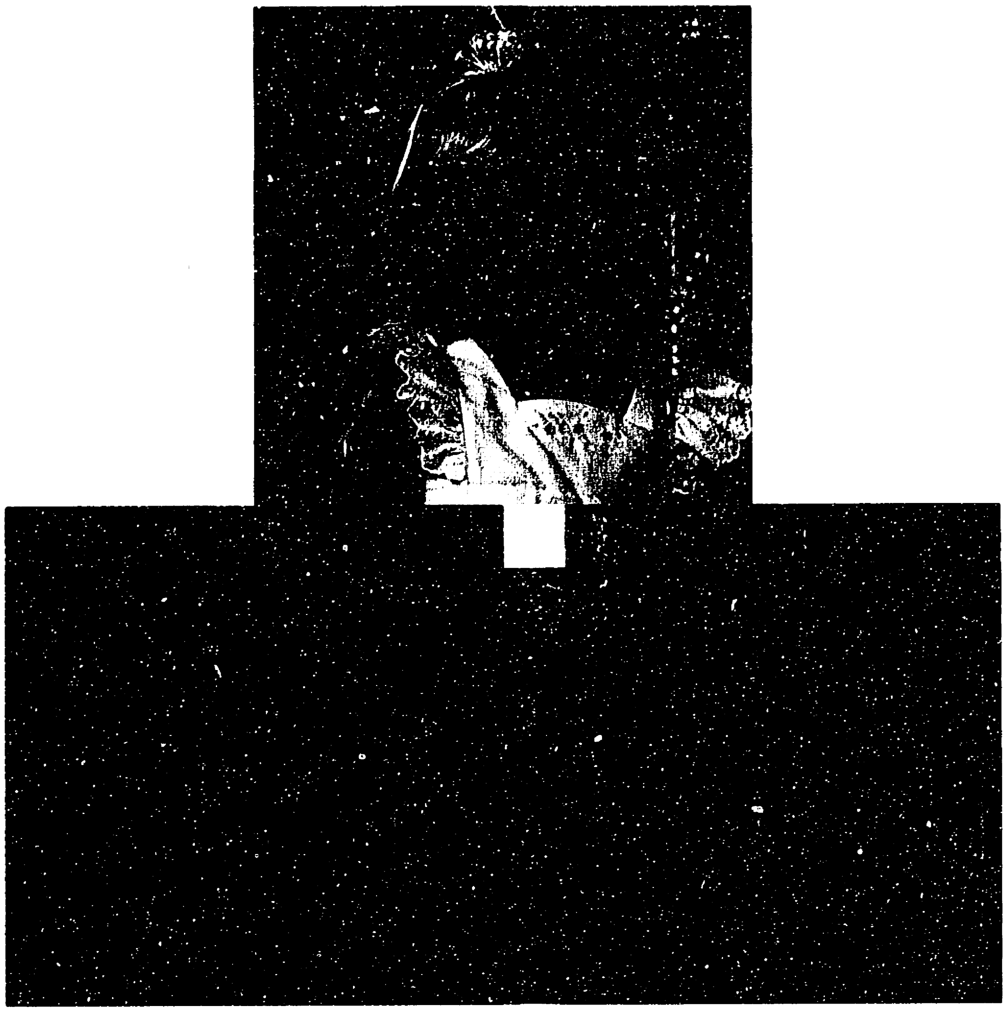


Figure 3-1: The top image is partitioned into 8x8 blocks each of which is processed using the 2-D DCT. The DCT coefficients are arranged in the transform representation on the bottom left, and in the subband representation on the bottom right.

As the transform and subband schemes are conceptually equivalent, we will sometimes use terms pertaining to each of them interchangeably, for example, transform basis functions and subband filter impulse responses. It is also useful to note that invertibility of a transform scheme corresponds to perfect reconstruction for a filter bank, and orthogonality of a transform scheme corresponds to losslessness of the filter bank.

3.3 The Block DCT

When one thinks about transforming a signal, the Discrete Fourier Transform is usually the first thing that comes to mind. The DFT has good energy compaction and decorrelation properties and a fast computational implementation, all crucial for application toward HDTV. The DFT also has some disadvantages including complex coefficients and an inherent inefficiency in its energy compaction. It is instructive to see where this inefficiency in energy compaction originates, because this directly leads us to a much improved transform, the DCT.

When the N-point DFT is taken of a discrete-space, non-periodic signal as shown in Figure 3-2 (a), what is actually calculated is the Discrete Fourier Series of the periodic signal shown in Figure 3-2 (b). The periodic signal in (b) has sharp discontinuities at the boundaries between the replications of signal (a). These sharp artificial discontinuities contribute energy to the high frequency coefficients and reduce the DFT's efficiency in energy compaction. This inefficiency may not be very harmful when transforming a single 512 by 512 image, but when an image is partitioned into 8×8 or 16×16 blocks and the transform is taken of each block there is a discontinuity every 8 or 16 pixels.

The Discrete Cosine Transform (DCT) has properties similar to the DFT, but also has improved energy compaction. The DCT has real coefficients, a fast computational implementation, and eliminates the artificial discontinuity inherent to the DFT, thereby resulting in improved energy compaction. The DCT eliminates this artificial discontinuity by making the signal to be transformed symmetric, by reflect-

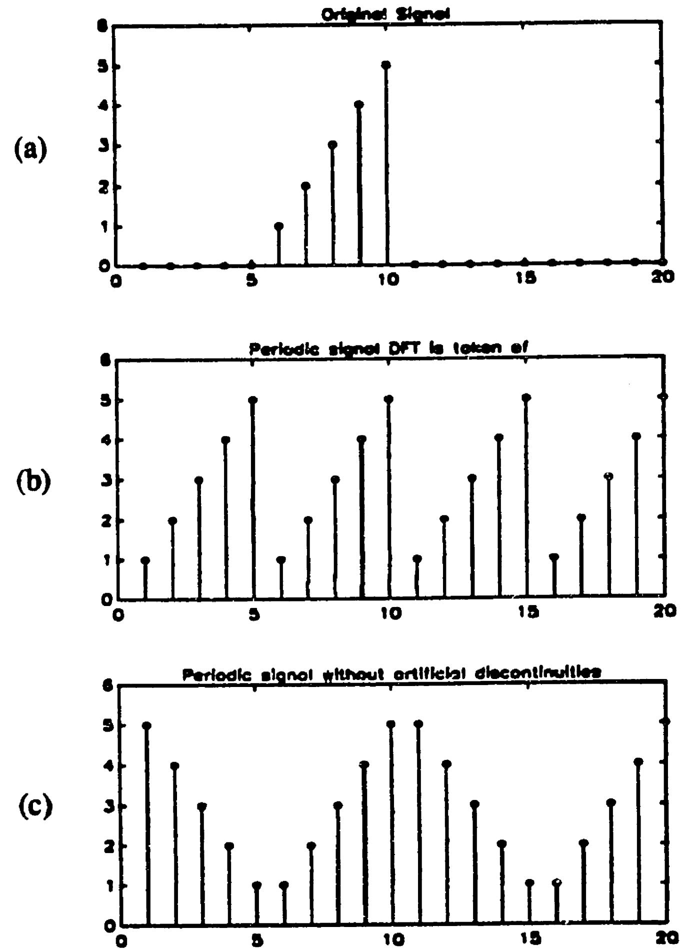


Figure 3-2: (a) The discrete-space, non-periodic signal to be analyzed. (b) Performing the N-point DFT of signal (a) is equivalent to performing the Discrete Fourier Series (DFS) of this periodic signal. Note the artificial discontinuities which contribute energy to the high frequencies. (c) Transforming signal (a) with the DCT is equivalent to performing the DFS on this discrete-space, periodic signal. Note the elimination of the artificial discontinuities from part (b).

over its ends as shown in Figure 3-2 (c). Now if a $2N$-point DFT is taken of ; (c) there will be no artificial discontinuities to spoil the energy compaction of)FT. The DCT of signal (a) is defined in terms of the $2N$-point DFT of signal [he $2N$-point DFT of (c) reduces to N DCT coefficients and requires essentially ame computational requirements as an N-point DFT because of the symmetry ,nal (c). The definition of the N-point DCT of an N-point signal $x[n]$ is

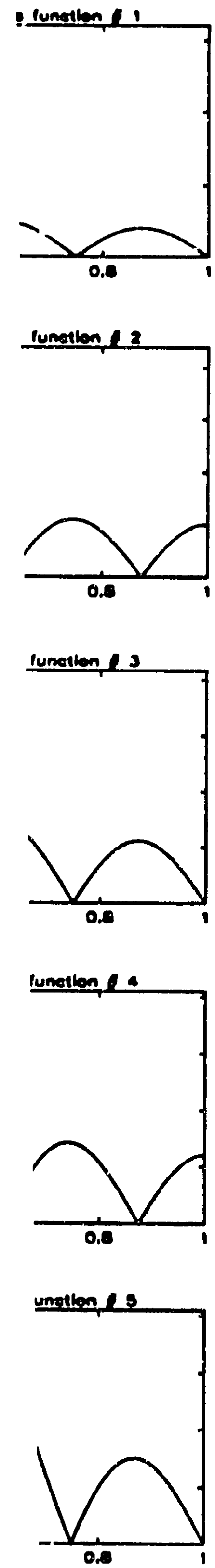
$$C_x[k] = \begin{cases} \sum_{n=0}^{N-1} 2\,x[n] \cos \frac{\pi}{2N}\, k\,(2n+1), & \text{for } 0 \le k \le N-1 \\ 0, & \text{otherwise} \end{cases} \tag{3.1}$$

first five basis functions of an 8-point DCT and their Fourier Transforms are n in Figure 3-3.

ιs already stated, the DCT achieves higher energy compaction than the DFT. The ', under certain conditions, also approaches the optimum performance possible ny linear transform. The Karhunen-Loève (KL) transform is the optimum of all ιr transforms in terms of energy compaction and decorrelation of the transform icients. However, the KL transform requires knowledge of the statistics of the al in order to derive the transformation and the basis functions. This makes implementation of the KL transform impractical, as it requires knowledge of the ;e statistics for every image to be transformed. The DCT, on the other hand, has ormance very close to the KL transform for most images and has deterministic s functions which do not depend on the statistics of the images.

The near-optimal energy compaction and decorrelation properties of the DCT pled with its fast computational implementation have resulted in its extensive ly and application toward image compression. Partitioning an image into 16x16 x8 blocks which are then adaptively transform coded using the DCT is the basis most cutting-edge image compression algorithms today. The MPEG and JPEG ndards for image compression both utilize this Block DCT scheme as do many eo conferencing and HDTV video compression algorithms.

An image is more stationary on a local basis than over the entire image. Therefore rtitioning an image into blocks allows for adaptive processing on a block by block



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basis which can take advantage of the local image characteristics and result in much improved visual quality. This sort of locally adaptive processing has been the key for many major improvements in image coding over the last few years. However, this adaptive block coding usually results in a visual blocking effect in the decoded image. This blocking effect is a structured artifact that the human visual system can easily detect, making the image very unpleasant to the viewer. Blocking effects are the greatest detractor of image quality for a low bit rate blocked transform coding system. For HDTV, it is desirable to have the highest video quality possible. We would like to have the energy compaction properties of the DCT and the block by block adaptivity possible with the block DCT, while eliminating the unpleasant structured block artifacts.

The blocking effect is produced because adjacent blocks are coded independently, resulting in discontinuities along the block boundaries of the decoded image. This will occur whenever an image is partitioned into blocks which are coded independently in a low bit rate system. One possible method to reduce the blocking effects is by lowpass filtering the image along the block boundaries. However, this will result in reducing the image quality and therefore is not an acceptable procedure for an HDTV system. Since the blocking effects are intrinsic to coding an image with a low bit rate blocked DCT, or for that matter any blocked transform, a different approach must be taken to eliminate the blocking artifacts.

One method to reduce the blocking effects is to code adjacent blocks together rather than independently. The DCT uses basis functions that are local to the individual blocks. One way to code adjacent blocks together would be to use basis functions that overlap adjacent blocks. This is the basis for the development of the Lapped Orthogonal Transform.

3.4 The Lapped Orthogonal Transform

The term Lapped Orthogonal Transform (LOT) in its most general sense corresponds to any transform/subband filtering operation where the basis functions or impulse responses of the filters have length greater than the block size. There have been many LOTs developed, but most exhibit the property that their basis functions decay slowly towards zero after the block boundaries so that a smooth transition is obtained in the reconstructed signal from one block to the next. As a result, there should be a significant reduction in the blocking effects.

The selected LOT has high coding efficiency, has fast computational algorithms for analysis and synthesis, and is relatively simple to design. It is called the Modulated Filter Bank or the Modulated Lapped Transform. The use of both these terms to describe it stresses the fact that transform and subband filtering schemes are conceptually the same.

For this LOT, an 8-band, one-dimensional modulated filter bank is designed and then separably applied to the two spatial dimensions of the residual. The filter bank is composed of modulated versions of a single lowpass filter. This method reduces the filter design problem to the design of a single filter. The impulse responses for the analysis filters are shown below. M is the number of bands and equals 8 for our scheme.

$$h_k[n] = h[n]\cos\left(\frac{\pi}{M}\left(k+\frac{1}{2}\right)\left(n-\frac{2M-1}{2}\right)+\frac{\pi}{2}\left(k+\frac{1}{2}\right)\right) \tag{3.2}$$

The baseband filter is given by

$$h_0[n] = h[n]\cos\left(\frac{\pi}{2M}\left(n+\frac{1}{2}\right)-\frac{\pi}{4}\right) \tag{3.3}$$

where h[n] is the so-called "window" to be designed. For high coding efficiency there should be no DC leakage into the higher subbands within the analysis filter bank. The utility of this can be seen by realizing that only the DC subband is then necessary to code a constant background region. Designing the filterbank with no DC leakage into the higher subbands results in the following unique solution for the window:

$$h[n] = \sin\left(\frac{\pi}{2M}\left(n+\frac{1}{2}\right)\right) \tag{3.4}$$

The synthesis filter impulse responses are the time-reversals of the corresponding analysis filter impulse responses. The analysis filters and synthesis filters each have nonlinear phase, but as one is the time-reversal of the other, their combined response has linear phase. This LOT has a fast implementation, allows for perfect reconstruction, and is orthogonal. Malvar has written two very good articles describing the general LOT and the particular version tested here [11, 12]. It may also be noted that the LOT tested is a special case of the oddly stacked time-domain aliasing cancellation filter bank (TDAC)[13].

The first five basis functions of the LOT and their Fourier Transforms are shown in Figure 3-4 . The basis functions of the LOT are tapered at the edges, as opposed to the basis functions of the DCT which abruptly begin and end. Looking back at the Fourier Transforms of the DCT basis function in Figure 3-3 and viewing these frequency plots as the subbands of the DCT we see that there is a severe amount of aliasing among the subbands. The LOT contains much less aliasing. This better localization in frequency results in improved coding performance. Most importantly, though, the tapered basis functions of the LOT considerably reduce the previously mentioned block artifacts, resulting in a visually more appealing reconstructed image than the Block DCT.

With a Digital HDTV system, the bit rate is so low that only very few of the coefficients can be coded. When coding very few LOT coefficients, blocking effects are still observed. These are not the hard blocks of the DCT, but instead a sort of soft blocking. The soft blocking effects are visually detracting and are still the prominent degradation of the system. Can the blocking effects be reduced even further, or possibly totally eliminated?

One possible method to eliminate the blocking artifacts is rather than having all of the basis functions of the same length, to have basis functions of different lengths, or different scales. This would produce a multi-scale representation of the residual. A study of Multi-scale representation may also be beneficial because various researchers studying the human visual system have theorized that it may be how humans view the world, seeing detail at many scales.

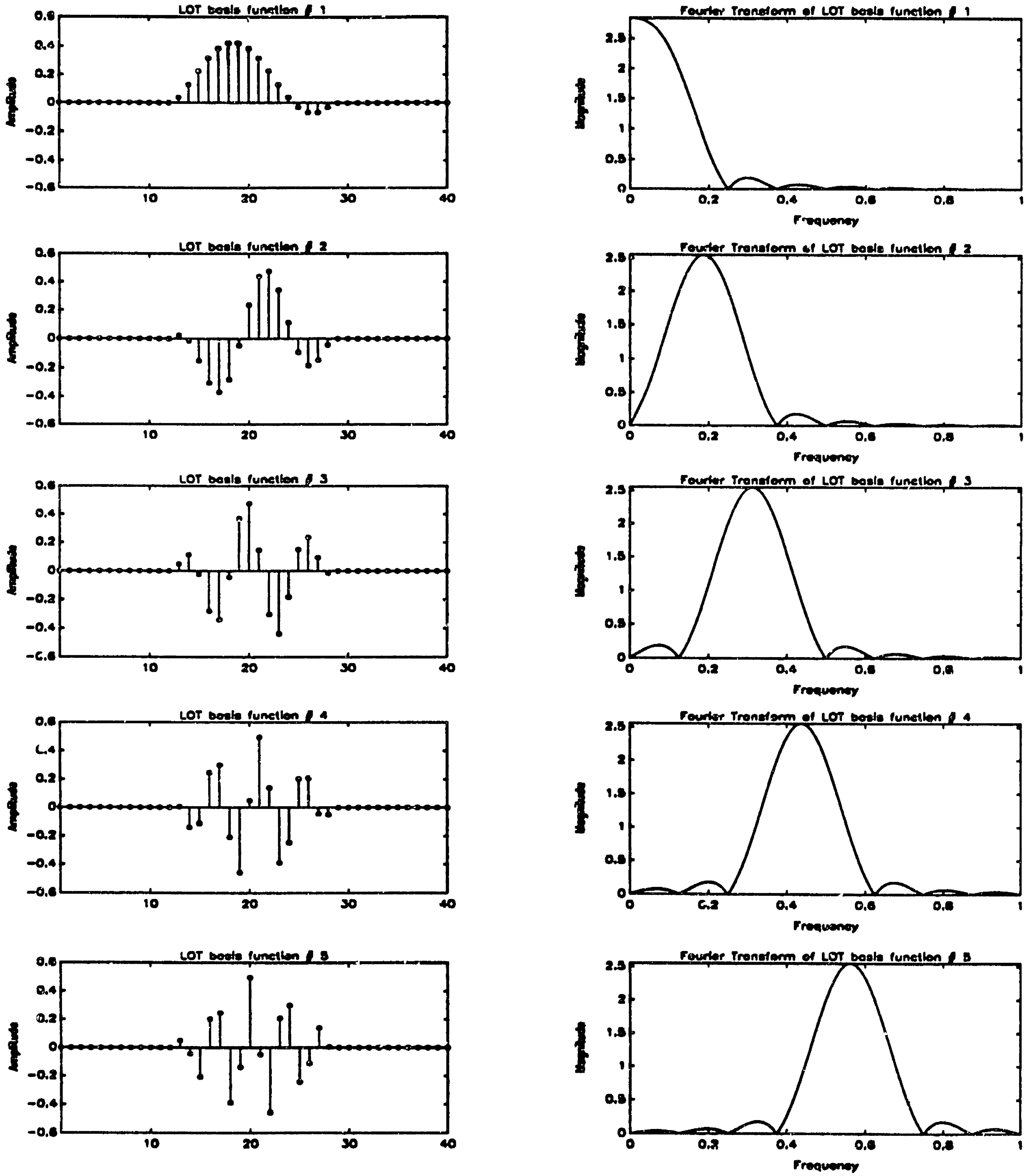

Figure 3-4: The first five basis functions of the LOT and their respective Fourier Transforms.

3.5 The Multi-scale scheme

Multi-scale representations come under many names: Wavelets, Pyramids, Quadrature Mirror Filterbanks (QMF), Multi-resolution. All of them are conceptually very similar. They decompose a signal into basis functions that are of different lengths. An early and very interesting representation that utilized multiple length basis functions was the Laplacian pyramid [14]. This scheme allowed for perfect reconstruction, but the basis functions were not orthogonal and the representation was overcomplete, as there were 4/3 as many coefficients as original image pixels. Although the nonorthogonality of the basis functions and overcompleteness of the representation makes this representation inefficient for coding, the representation exemplifies how multiple scale basis functions can be used to encode an image.

The initial tested scheme was based on a number of papers by Adelson and Simoncelli [15, 16, 17]. They utilized a hierarchy of bandsplitting quadrature mirror filters in a tree structure to produce an octave subband division of the frequency domain. This results in basis functions whose lengths are multiples of two of each other.

In order to better understand the Multi-scale scheme, we examine a simple one-dimensional two-band analysis/synthesis subband filtering scheme as shown in Figure 3-5. We would like to design the analysis filters to split the frequency domain into two equal parts which can then be coded/decoded to perfectly reconstruct our original signal. In the ideal case the analysis filters would split the frequency domain as shown in Figure 3-6 (a). But because ideal brick-wall (infinite cutoff) filters do not exist, this can not be done. This leaves two choices, either the analysis filters can be designed so that their frequency responses do not overlap as shown in Figure 3-6 (b), or they can be designed so that their frequency responses overlap as shown in Figure 3-6 (c). If the frequency responses of the analysis filters are disjoint as shown in (b) then there will be in affect a spectral hole in the frequency response of the system. This results in a loss of information and does not allow for perfect reconstruction of an input signal that contains frequency components within that hole. If the frequency responses of the analysis filters overlap as shown in (c), that is each filter has

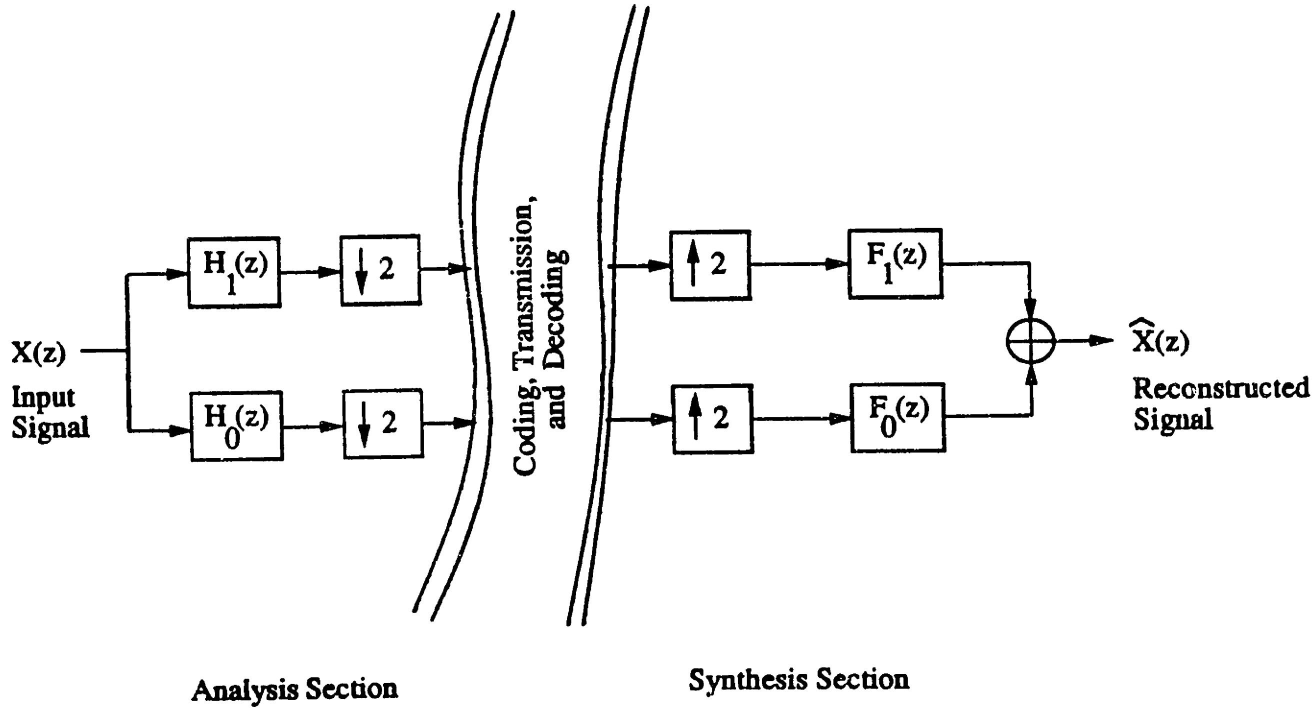

Figure 3-5: A two-band analysis/synthesis subband filtering scheme.

bandwidth greater than $\pi/2$, then aliasing will occur as a result of the decimation.

The QMF filterbank approaches this problem in a very elegant way. The QMF bank allows the analysis filter responses to overlap as shown in Figure 3-6 (c), resulting in aliasing at the output of the decimators, but then chooses the synthesis filters so that the imaging produced by the interpolators exactly cancels the aliasing. This can be easily shown mathematically. The reconstructed signal $\hat{X}(z)$ can be expressed as

$$\begin{aligned}\hat{X}(z) &= \frac{1}{2}\left[H_0(z)F_0(z) + H_1(z)F_1(z)\right]X(z)\\ &\quad + \frac{1}{2}\left[H_0(-z)F_0(z) + H_1(-z)F_1(z)\right]X(-z) \qquad (3.5)\end{aligned}$$

Notice that the first term is a function of $X(z)$ while the second is a function of $X(-z)$. The first term is the linear shift-invariant term and the second is the troublesome term produced by the aliasing and imaging. The second term in (3.5) can be made to disappear by choosing the synthesis filters so that the imaging cancels the aliasing.

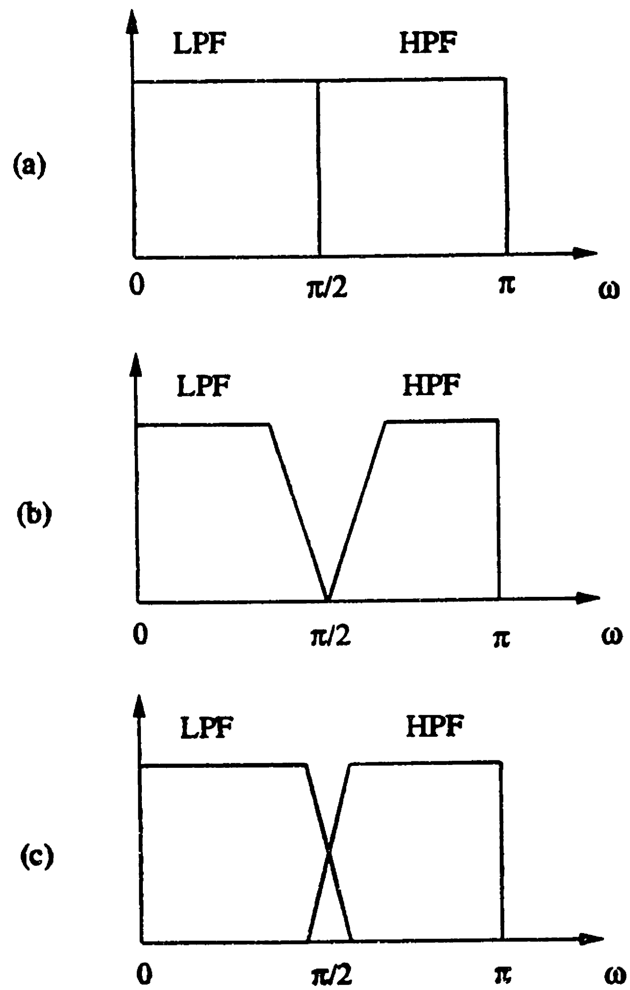

Figure 3-6: Division of the frequency domain in the ideal case (a), without overlap (b), and with overlap (c).

This is done by choosing

$$F_0(z) = H_1(-z), \qquad F_1(z) = -H_0(-z) \tag{3.6}$$

The aliasing is canceled irrespective of the choice of $H_0(z)$ and $H_1(z)$.

$$\hat{X}(z) = \frac{1}{2}\left[H_0(z)H_1(-z) - H_1(z)H_0(-z) \right] X(z) \tag{3.7}$$

With the canceled aliasing, the system becomes a linear shift-invariant system. The term quadrature mirror filters (QMF) comes from the fact that the original QMF filterbank was designed with the analysis filters related as

$$H_1(z) = H_0(-z) \tag{3.8}$$

If $H_0(z)$ were a lowpass filter then $H_1(z)$ would be a highpass filter. They would be *mirror-images* of each other, with respect to the frequency $\pi/2$ which is a *quarter* of the sampling frequency. Note that using (3.6) to eliminate the aliasing and (3.8) to relate the lowpass and highpass filters results in requiring the design of only a single filter for the complete two-band QMF system. Also, if $H_0(z)$ is a linear-phase FIR filter then the complete system would be a linear-phase FIR system.

A two-subband FIR QMF filterbank whose filters are constrained by (3.6) and (3.8) can achieve perfect reconstruction if and only if $H_0(z)$ is a trivial transfer function. In practice, it may be beneficial to sacrifice perfect reconstruction for improved frequency selectivity of the subbands. Vaidyanathan has written two very comprehensive papers discussing these issues [18, 19].

For a two-band subband filterbank with perfect reconstruction one can take each of the decimated intermediate signals produced by the analysis portion of the filterbank and send it through another complete two-band subband filterbank and still achieve perfect reconstruction. By cascading filterbanks in this manner, it is possible to construct a hierarchical (tree structure) filterbank which will separate the frequency domain into the desired subbands. If the cascading is done uniformly to all the intermediate signals then the filterbank is called a uniform cascade system. This

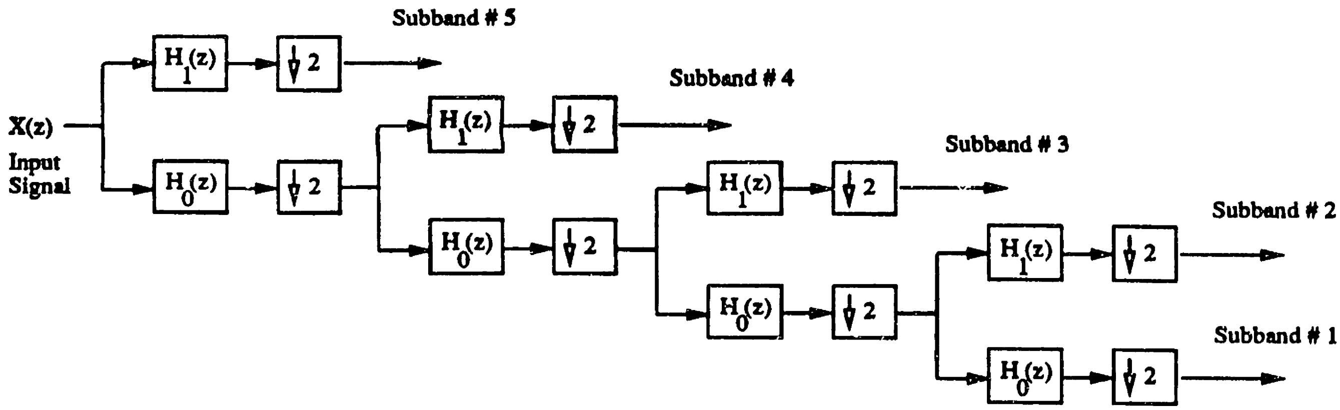

Figure 3-7: The analysis portion of the four-level QMF filterbank which produced the Multi-scale representation.

results in a partitioning of the frequency domain into subbands of equal bandwidth.

The subband filterbanks may also be cascaded in a nonuniform manner, usually referred to as a pyramid. This structure results in a nonuniform decomposition of the frequency domain. A four-level pyramid structure of band-splitting QMF filterbanks is implemented in order to create the Multi-scale representation. The analysis portion of the filterbank structure in 1-D is shown in Figure 3-7. The synthesis portion is symmetric to the analysis portion.

It is instructive to analyze the way this filterbank partitions the frequency domain into subbands. The first level of the filterbank divides the frequency domain into lowpass and highpass frequency regions as shown in Figure 3-8 (a). The second level of the filterbank divides the previous lowpass region into respective lowpass and highpass regions as shown in Figure 3-8 (b). The hierarchial structure of the pyramid recursively subdivides the lowpass region into lowpass and highpass regions. The complete filterbank results in a decomposition of the frequency domain into five

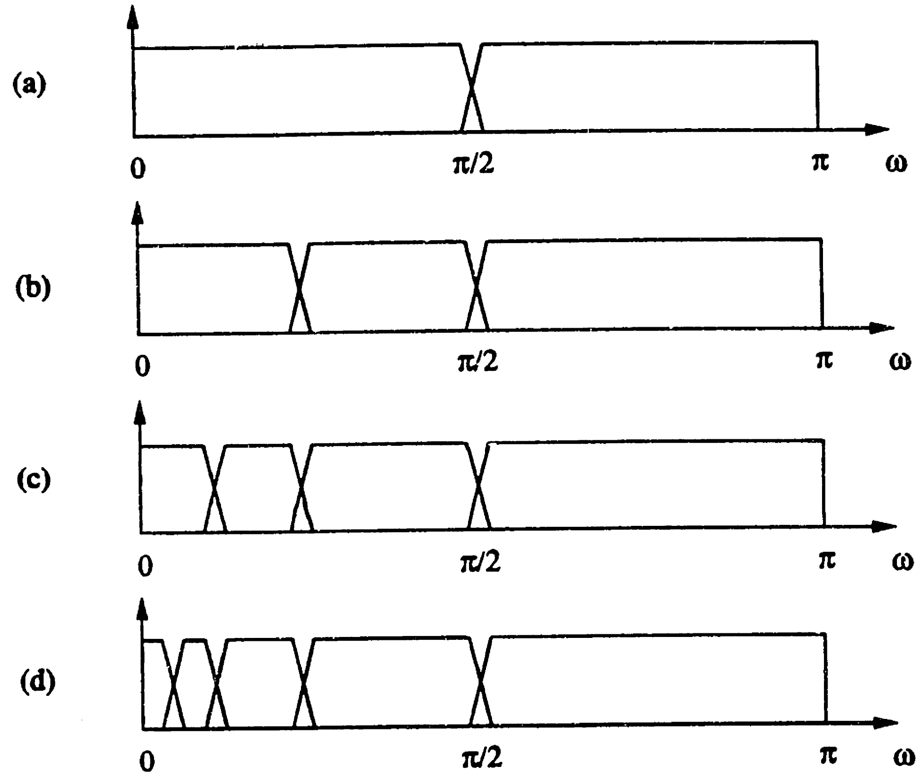

Figure 3-8: The partitioning of the frequency domain by the four-level pyramid filterbank. Each subsequent level of the filterbank divides its respective frequency range in two. In this way the pyramid filterbank partitions the frequency domain into octave-width subbands.

octave-width subbands as shown in Figure 3-8 (d).

As the QMF filterbank is recursively applied to the lowpass subband and the resulting subbands shrink in bandwidth by factors of two, the respective basis functions for these subbands increase in size by factors of two. Therefore, as the signal is partitioned in the frequency domain into five octave-width subbands, it is similarly decomposed into five basis functions whose lengths increase by multiples of two. In this way, the Multi-scale representation of a signal is created.

The first tested QMF filterbank utilized a filter impulse response (or kernel) designed by Simoncelli [16]. Again, because of the QMF filterbank structure, only a single kernel was required to define the complete filterbank. The basis functions of the representation are the original kernel and its recursively upsampled and filtered versions. The five basis functions for the 1-D implementation of the four-level QMF pyramid are shown in Figure 3-9 with their corresponding frequency plots.

This Multi-scale scheme has basis functions which are scaled versions of each other, and achieves close to perfect reconstruction of the image and orthogonality among the basis functions. Also the Multi-scale scheme has no DC leakage into the higher subbands, which is very important for high coding efficiency. Comparing the basis functions for the Block DCT, LOT, and Multi-scale schemes, one sees that the Multi-scale basis functions do not have the abrupt beginnings and ending of the Block DCT basis functions, and have even more gradual transitions than the LOT basis functions. For both the Block DCT and the LOT, all the basis functions begin and end at the same spatial locations. This causes boundary effects at these locations that produce the block artifacts. As the Multi-scale scheme has different length basis functions which do not all begin and end at the same spatial locations, there should not be the same boundary effects as the Block DCT and the LOT and the blocking artifacts should be reduced.

In order to apply the Multi-scale scheme to a two-dimensional residual, the 1-D QMF filterbank is applied separably along the two spatial dimensions of the residual. This results in a partitioning of the residual into subbands as shown in Figure 3-10. In the top right corner is the horizontal-high frequency vertical-low frequency subband (HL). In the bottom left corner is the horizontal-low frequency vertical-high frequency subband (LH). In the bottom right corner is the horizontal-high frequency vertical-high frequency (HH) or diagonal-high frequency subband. And in the top left corner is the horizontal-low frequency vertical-low frequency subband which is recursively partitioned by the filterbank into further subbands. A four-level QMF filterbank was chosen because it is the highest level filterbank that could evenly decompose the 720 by 1280 residual into subbands. This is because each level of the filterbank divides its input signal into subbands whose dimensions are one-half of the length of its input signal, and the dimensions of the residual, 720 by 1280, are each divisible by 2 four times. One applies the largest level filterbank possible in order to achieve the greatest energy compaction of the residual, and hence, higher coding efficiency.

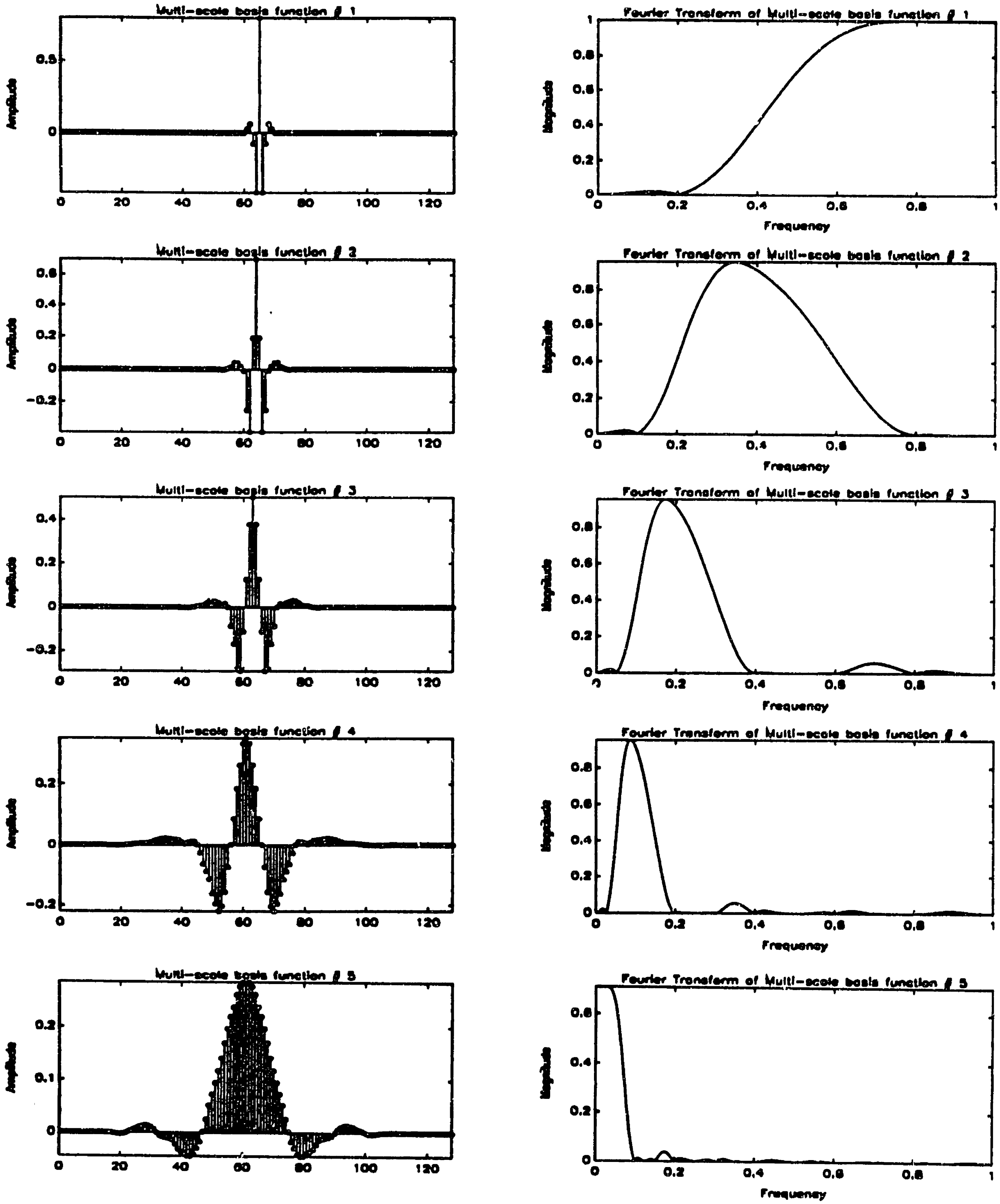

Figure 3-9: The five basis functions of the Multi-scale scheme and their respective frequency plots.

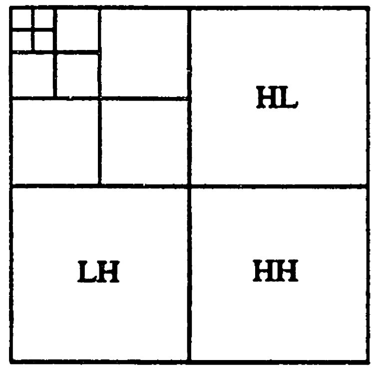

Figure 3-10: The subband decomposition performed by applying the Multi-scale scheme to the residual.

3.6 Comparing the three transform/subband filtering schemes

3.6.1 Comparison criteria for the different schemes

The DCT, LOT, and Multi-scale transform/subband filtering schemes were compared in order to determine which scheme achieves the highest performance in encoding the residual. Important considerations in comparing the three schemes were (1) visual appeal, (2) energy compaction, (3) amount of selection information necessary, and (4) quantization effects. The visual appeal was of course the most important comparison criterion, with the amount of energy compaction being a good quantitative measure close behind. The amount of selection information necessary was hypothesized to be rather similar for all three schemes, and the effects of quantization was thought to be of even lesser importance in comparing the schemes.

The most appropriate comparison test would be to allocate the same number of bits to each of the three schemes and test how well each could perform. However, this requires a great deal of time optimizing the encoding for each one. Instead, a good first order test is to examine how well each scheme represents a residual when only a limited number of coefficients is retained. The first test in the comparison was to apply each of the three transform/subband schemes to the residual of the digital system while retaining only a given percentage of the most energetic coefficients, keeping these energetic coefficients with full amplitude resolution and setting the others to zero. In this way, we examined the visual and energy compaction qualities of each scheme without requiring the optimization of the selection and quantization methods for each. Then, after the initial comparison was made, we could examine the amount of selection information necessary for reconstruction and the effects of quantization on each scheme to ensure that they would have only minor effects on the coding of the residual.

It is very important to realize that when our HDTV system processes a video sequence, the first frame has no previous frames to make a prediction from, and therefore the residual is exactly the first frame, a normal image to be coded. For the following frames, motion-compensated predictions are made so the residuals are more similar to pure residuals than normal images. Also, and more importantly, whenever there is a scene change, or a large amount of new imagery in a frame, the prediction based on ME/MC will be drastically in error and the residual will resemble a typical image, i.e. it will be more lowpass in nature. When there is not a scene change and there are only small changes in the imagery, the residual will be more like the pure residual we expect it to be, a difference signal, and more highpass in nature. Because the HDTV system will process video that includes scene changes, new imagery, fast motion, etc., both of these kinds of residuals will be created, and the system must perform very well for both.

3.6.2 Comparison Test Number One

For the first test, four frames of the picnic sequence were coded using the digital system with each of the three schemes, the Block DCT, the LOT, and the Multiscale. The precise comparison criteria for the tests were, in order of importance: (1) visual appeal and (2) normalized mean-square-error (NMSE). The energy compaction property has been discussed as a very important property of a transform/subband filtering scheme, but the normalized mean-square-error measure was used for the comparison criteria instead of a pure energy compaction measure because it gives a better indication of how each scheme represents the residual. The mean-square-error was normalized by the energy of the residual in order to make the measure independent of the actual energy of the residual.

The first of the four coded residuals models a scene change, while the following are more like pure residuals, in that they model high-frequency detail and motion. The three schemes were applied while retaining the following percentages of the most energetic coefficients: 5, 10, 15, 20, 25, 30, 40, 50, 60, 70, 80, 90. and 100 percent. Smaller percentage intervals were tested for the lower values because the required video compression will most likely allow for only a limited number of coded coefficients.

Visually, the Multi-scale scheme performed better than either the Block DCT or the LOT when only a small percentage of the coefficients were retained. As the percentages of retained coefficients increased, all three schemes performed nearly flawlessly in reconstructing the image and it was very difficult to decipher any differences between them. When low percentages of coefficients were retained, the Block DCT had the problem of uncanceled aliasing errors being visually localized at the block boundaries, thereby creating block artifacts. At the same percentages, the LOT did not have the hard blocking effects of the Block DCT, but did exhibit a "soft" blocking effect. For the Multi-scale scheme, the uncanceled aliasing was spread over the entire reconstructed image, and no blocking artifacts were perceived. As a result, over the tested percentage range, the Multi-scale scheme performed visually better than or equal to the other two schemes.

The plots shown in Figure 3-11 were generated when comparing the normalized mean-square-error between the three schemes. The results of these tests indicated that when considering normalized mean-square-error, all three schemes achieved approximately the same performance.

In these tests, the most energetic coefficients were retained with full amplitude resolution. This is not a true comparison of how the three schemes would perform

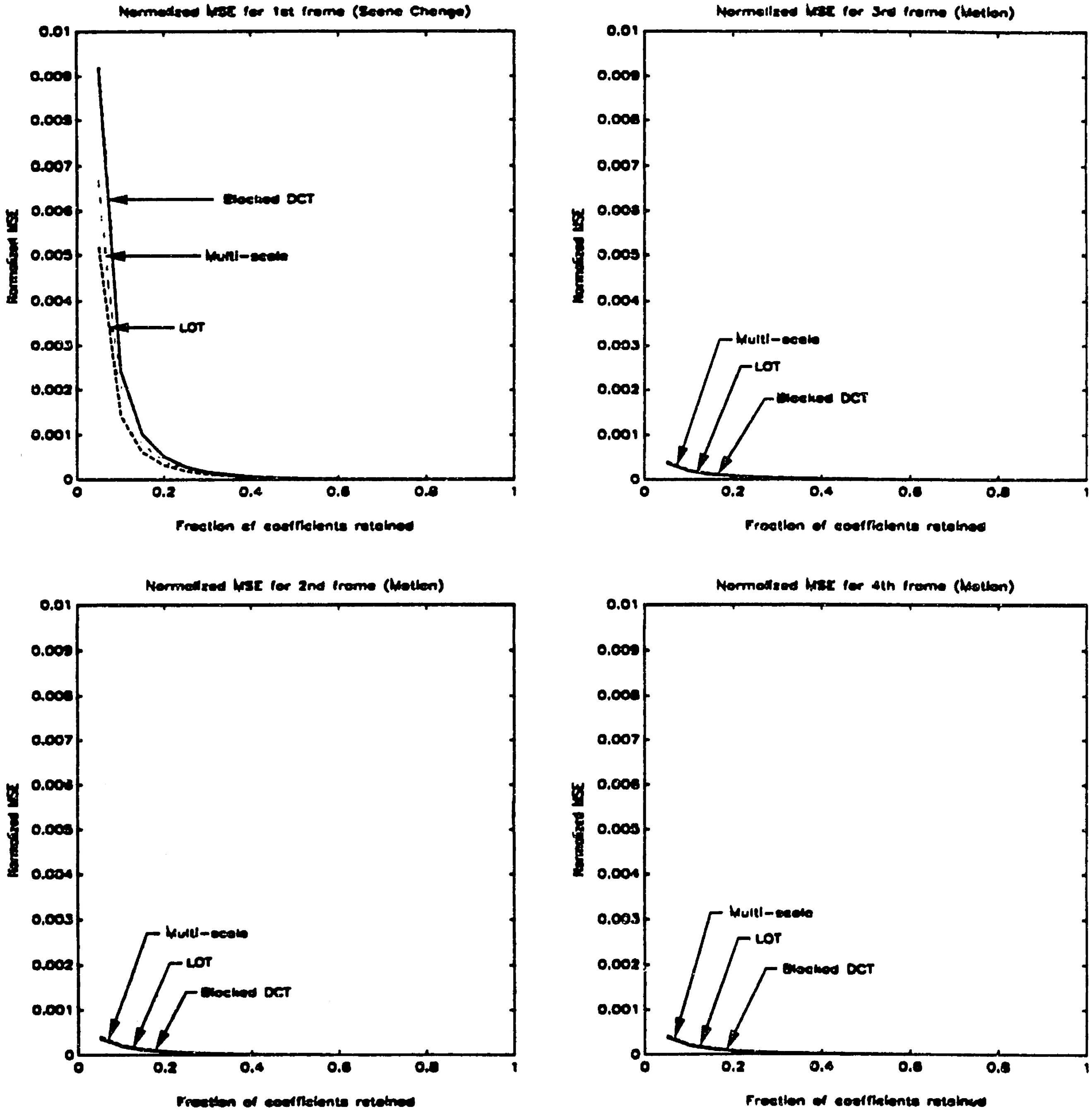

Figure 3-11: The normalized mean-square-error for the first four frames of the picnic sequence for each of the three transform/subband filtering schemes.

when actually being coded. To ensure that the results would be similar when coding the schemes, we compared the amount of selection information necessary for each scheme and the effects of quantization on each scheme.

To compare the amount of selection information necessary for each of the three schemes, a multitude of tests were performed using various coding techniques. For each technique, the entropy of the selection information necessary for each of the three schemes was found. The scheme that achieved the lowest entropy was judged to have the highest performance.

The selected coefficients were arranged in both subband and transform representations, since each of these representations can be used to aid in the coding of the selection information. The subband representation allows one to take advantage of the spatial correlation among selected coefficients, while the transform representation allows one to take advantage of the location and frequency correlation among the selected coefficients. Zigzag and serpentine scan runlength coding of appropriate blocks of selection information was performed in both subband and transform representations. Vector quantization of 2×2 and 4×4 blocks of selected coefficients was also performed in both the subband and transform domain representations. Basic FAX runlength coding techniques were also applied. All the tests showed that the entropy of the selection information was approximately equal for all three schemes. If one scheme had to be judged to perform better than the others in terms of the amount of selection information necessary, it would be the Multi-scale scheme.

The possibility of taking advantage of the joint correlation among the coefficient amplitudes to be coded and the subsequent runlength of uncoded coefficients was also investigated. Toward this goal the amplitude information was jointly coded with the runlength information. But tests performed by other members of the research group showed that this was not the case.

In order to test the effects of quantization upon the different schemes, the coarseness/fineness of the quantization (the stepsize), for the different coefficients depending on the frequency band and color component was varied. Also compared were midtrend (zero is one of the reconstruction levels) and midriser (zero is not one of the reconstruction levels) quantizer types. A deadband around the zero reconstruction level was applied to examine if coring of the coefficients would lead to any improvement. A completely comprehensive analysis of the quantization effects was not done, nor was any nonuniform quantization performed except for the coring (strictly uniform quantization was tested), but the results did not show any dramatic advantages or disadvantages for one transform/subband filtering scheme over the other two. The different quantization methods resulted in changes in the reconstructed images, but it was not evident that one method was perceptually superior to the others.

The conclusion drawn from the first test was that for the cases in which the low percentages of coefficients were retained, the Multi-scale scheme was visually more appealing than either the Block DCT or the LOT schemes. The normalized mean-square-error was comparable for all three schemes. Tests done comparing the amount of selection information required for each scheme and the effects of quantization on each scheme showed that all three schemes appeared to have approximately the same performance.

Following this work, performance tests of the Digital HDTV system showed that only 1.5 to 3 % of the coefficients were being coded. This came as a surprise as to how few coefficients were actually being coded. The encoding algorithms were still relatively simple, but even with very complex algorithms it became apparent that we could realistically code only about 3 % of the coefficients. This led to the second set of tests.

3.6.3 Comparison Test Number Two

The second set of tests was similar to the first set except that only 3 % of the most energetic coefficients were retained. Also the sequence to be tested was composed of ten frames in the middle of the picnic sequence which contains a great deal of motion. These changes resulted in a more appropriate test for comparing the three schemes for the Digital HDTV system since first, they more accurately modeled the number of coefficients that could be coded, and second, the new video test sequence not only contained a scene change but also residuals which resulted from a great deal

Original	**Block DCT**
LOT	**Multi-scale**

Figure 3-12: Placement of the original image and the reconstructed images processed with each of the three transform/subband filtering schemes.

of motion. This sequence heavily taxed the motion estimation/motion compensation portion of the system and therefore resulted in residuals which were more difficult to encode.

Visually, once again, the Multi-scale scheme performed considerably better than either the Block DCT or the LOT. The first and fourth reconstructed frames are shown in Figure 3-13 and Figure 3-14 respectively where Figure 3-12 shows the original image and also which reconstructed image was processed with each scheme. The Block DCT and the LOT both experienced blocking artifacts that were visually persistent and degrading. The block artifacts slowly diminished as the reconstructed sequence went from the first to the tenth frame, but remained evident. The Multi-scale scheme also looked visually displeasing for the first few frames, but then quickly improved. The normalized mean-square-error for the Multi-scale scheme was lower than that achieved by either of the other two schemes for all ten frames in the test. This is shown in Figure 3-15.

Comparing selection information and quantization effects once again among the three schemes showed results very similar to before. One significant observation was that the number of coded coefficients was *extremely* important. The system was at a

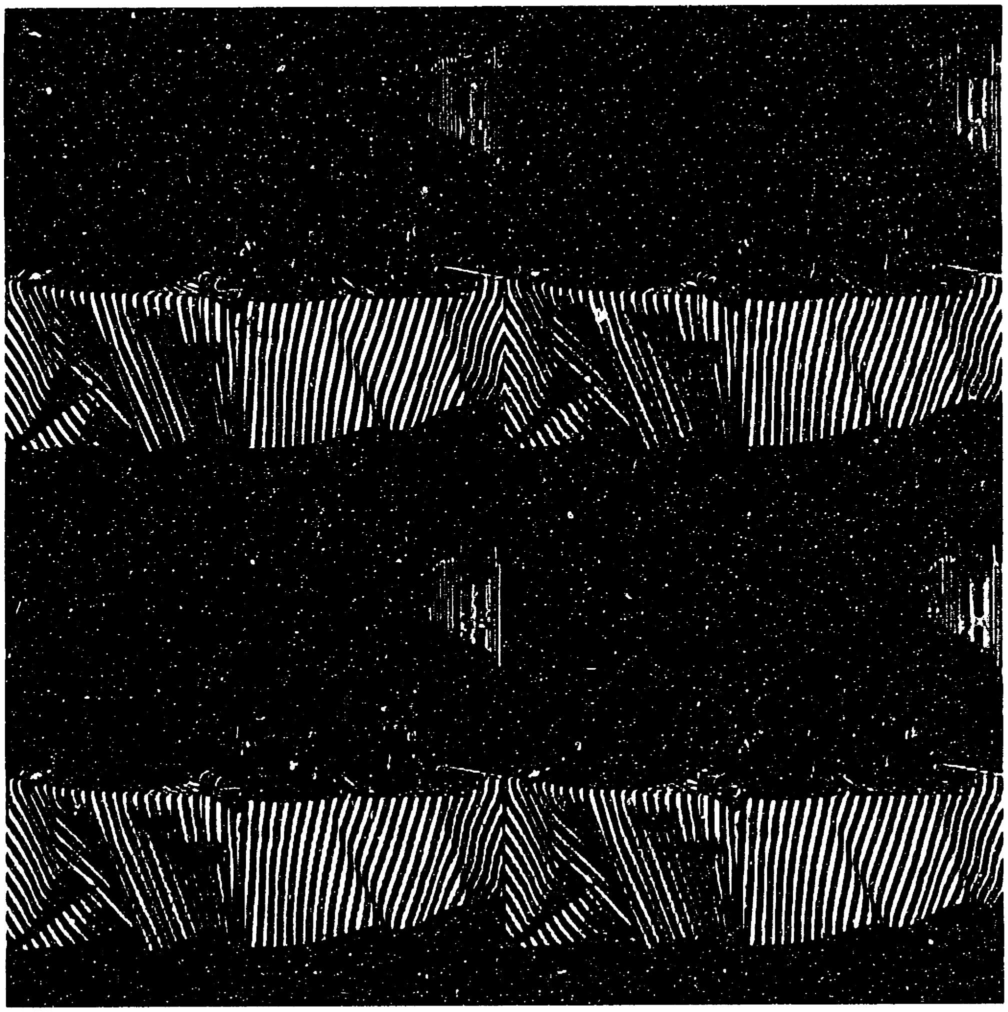

Figure 3-13: The first frame processed with each transform/subband filtering scheme when only the 3 % of the most energetic coefficients are retained.

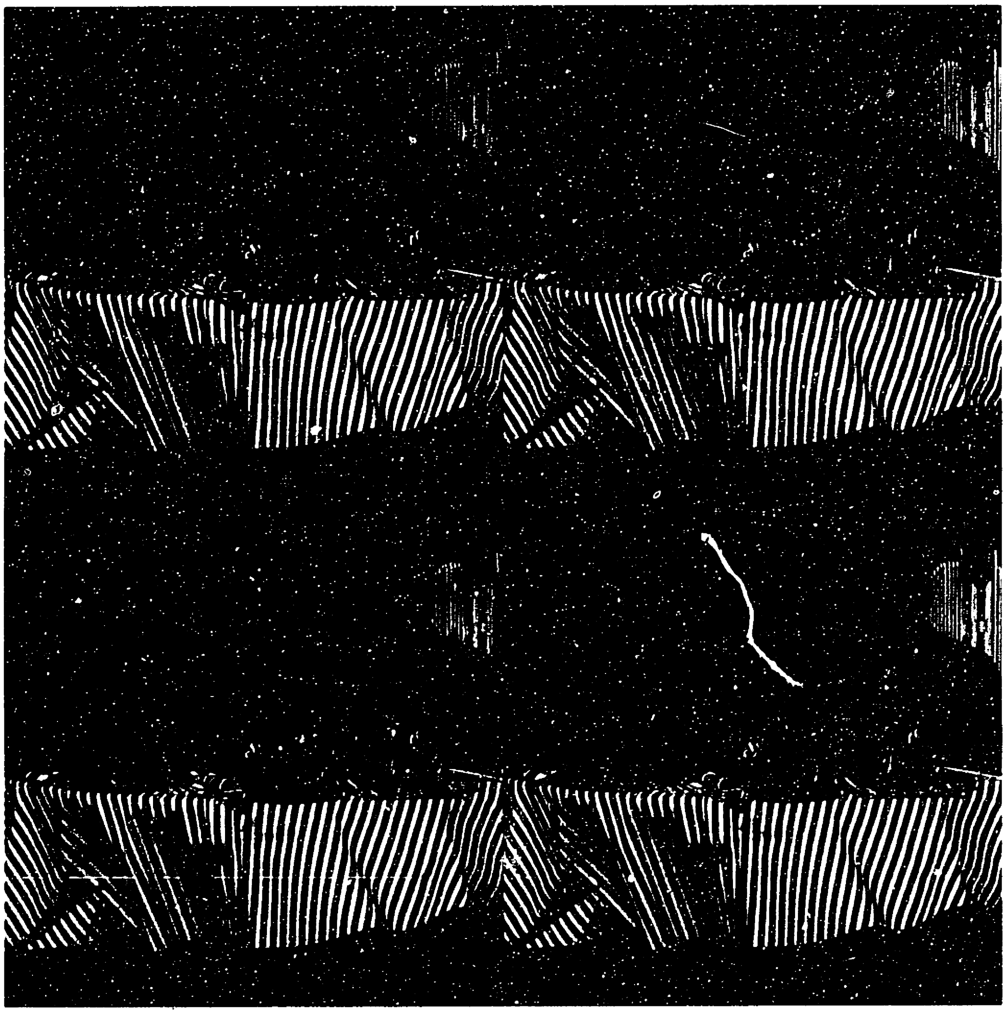

Figure 3-14: The fourth frame processed with each transform/subband filtering scheme when only the 3 % of the most energetic coefficients are retained.

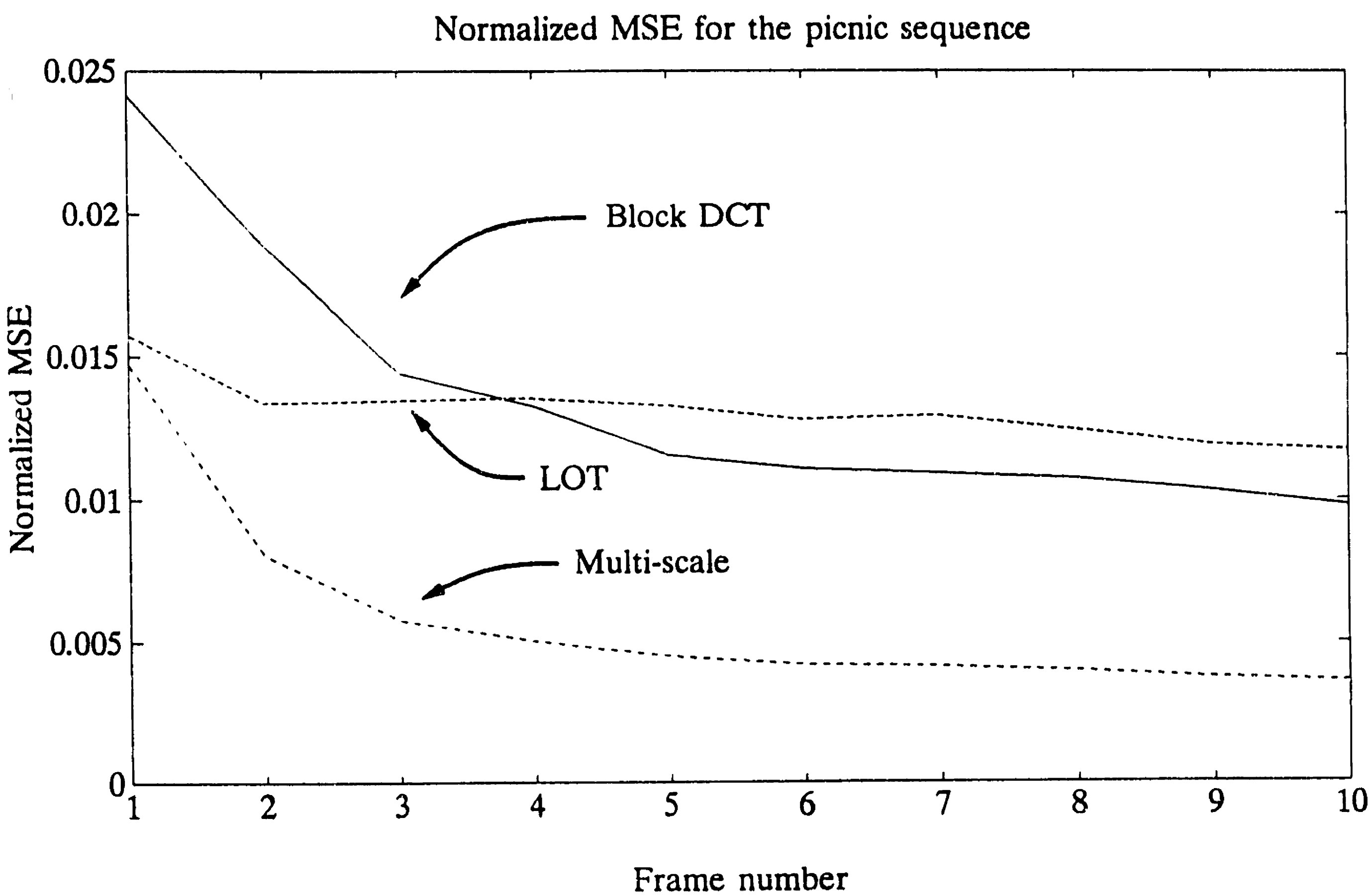

Figure 3-15: Normalized mean-square-error for a ten frame cut of the picnic sequence where there is a great deal of motion.

point in performance where every small increase in the number of selected coefficients resulted in significant visual improvements in the video. Steps were taken toward this goal including quantizing the medium and high frequency coefficients more coarsely in order to have more bits available to select more coefficients.

3.7 Conclusions

The Block DCT, LOT, and Multi-scale transform/subband filtering schemes all perform very well when a large percentage of the coefficients are coded. In the scenario of Digital HDTV, the limited available bit rate restricts the number of selected coefficients to be a very small percent, probably only about 3 %, with very complex coding algorithms. When coding only 3 % of the coefficients, all three schemes achieve poorer performance. At this level, the dominant degradations are the structured blocking artifacts noticeable in the reconstructed images from the Block DCT and LOT schemes. The human visual system easily detects these structured artifacts and finds them very unpleasant. The Block DCT has hard blocking effects along its block boundaries. The LOT has soft blocking effects which are still easily perceptible. The Multi-scale scheme, on the other hand, does not produce any blocking artifacts. The Multi-scale scheme has superior visual performance, lower normalized mean-square-error, and approximately the same required selection information and quantization effects. As a result, the Multi-scale scheme is the optimal choice of the three schemes tested for coding the motion-compensated residual for our All-Digital HDTV system.

Chapter 4

New Representations

4.1 Idea

In our Digital HDTV system, motion-estimation/motion-compensation is applied along the temporal dimension of the video to make a prediction of the current frame from the previous frame. It is important to realize that after this temporal processing the digital compression techniques will be applied to a motion-compensated residual as opposed to a typical image. The motion-compensated residual will sometimes exhibit the characteristics of a typical image (e.g. during a scene change), but it will usually be much more high pass in nature than a typical image. By taking this difference into account, it may be possible to improve the visual quality of the system.

4.2 Representations

Because most of the frames to be coded will be pure residuals, which are more high frequency in nature than normal images, it would appear to be beneficial to code the high frequencies in an improved manner. In this chapter, this approach is taken toward improved coding of the residual.

The original Multi-scale frequency decomposition, or Multi-scale representation # 1, is shown in Figure 4-1. Multi-scale representation # 2 for the residual was based on an idea directly related to transform coding. Transform coding is when the coefficients

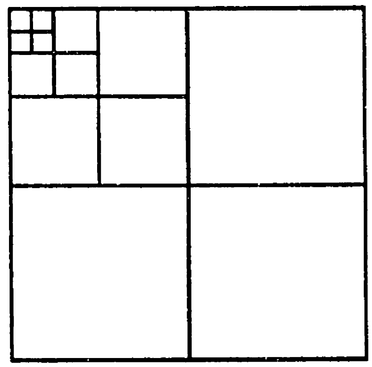

Figure 4-1: The original Multi-scale frequency decomposition.

for some transform, usually the Block DCT, are arranged and coded in the transform representation. The most important frequencies for representing an image are the low-frequencies and the horizontal-high vertical-low and horizontal-low vertical-high frequencies. For high performance, these frequencies are usually treated somewhat equally. Figure 4-2 shows the coefficients of an 8×8 block of DCT coefficients that are treated equally in coding.

For the original Multi-scale frequency decomposition a band-splitting QMF filterbank is recursively applied to successive lowpass subbands. In this manner the low frequencies (LL) are processed differently from the horizontal-high vertical-low (HL) and the horizontal-low vertical-high (LH) frequencies. Treating these frequencies equally may lead to higher performance. One method to treat them equally is to apply the band-splitting QMF filterbank not only recursively to the LL subband, but also for a single recursion to each HL and LH subband generated. This results in Multi-scale representation # 2 which is shown in Figure 4-3. It is interesting to realize that by applying the bandsplitting QMF filterbank to the HL and LH subbands, the energy in each of these subbands is compacted, resulting in greater energy compaction of the residual. Multi-scale representation # 2 produced a sharper image than the original representation, but it also contained an "artificialness" and a larger

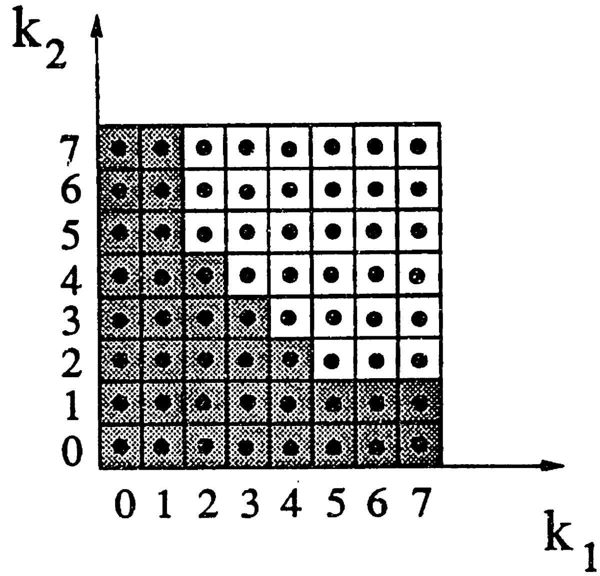

Figure 4-2: An 8×8 block of DCT coefficients where the shaded coefficients are treated equally.

amount of ripple along the edge boundaries.

The increased energy compaction resulting from applying the QMF filterbank for a single recursion to the HL and LH subbands, gives the impression that further energy compaction and improved performance may be obtained by still further applications of the filterbank. The QMF filterbank may also be applied to the HH subband and the number of recursions within each subband may be varied. Experiments done to test this hypothesis concluded that further energy compaction was obtained but the resulting pictures were of visually inferior quality. The images produced through these tests exhibited large amounts of ripple and a very "artificial" feeling.

Multi-scale representation # 3 was conceived through a number of ideas. 1) The residual is more high pass in nature and these high frequencies should be processed better. 2) The residual is composed of high frequency detail, like edges, which are predominantly aligned either horizontally or vertically. 3) A vertical edge is composed of horizontal high frequencies and would result in a vertical line in the horizontal-high vertical-low subband. Since an edge contains high frequencies perpendicular to its alignment and not along its alignment, we only need to filter along the perpendicular direction to better decompose the high frequencies within the edge. This implies that

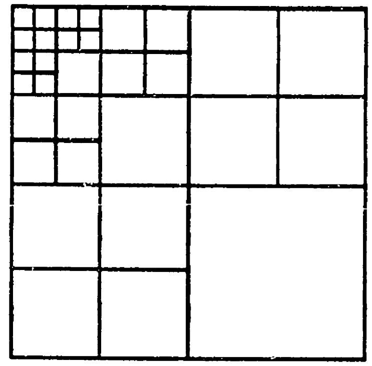

Figure 4-3: Frequency decomposition for Multi-scale representation #2.

the HL and LH subbands only need to be filtered along one spatial direction, and not along both.

For each level of filtering performed the length of the basis functions is doubled. Therefore filtering along both horizontal and vertical directions results in the respective basis functions being twice as long along each direction as before the filtering. Doubling along each direction reduces the localization of the basis functions and results in the errors in coding the basis functions being spread over a greater portion of the residual. This may be why Multi-scale representation # 2 exhibited a feeling of artificialness and extra ripple. As improved decomposition of the frequencies is desired only along one direction, filtering and doubling of the basis function's length along both directions is unnecessary.

Multi-scale representation # 3 is created by filtering all the HL and LH subbands along only a single spatial direction for each. Each horizontal-high vertical-low subband is filtered using a 1-D band-splitting QMF filterbank along the horizontal direction. Each horizontal-low vertical-high subband is filtered using a 1-D band-splitting QMF filterbank along the vertical direction. The resulting frequency decomposition for Multi-scale representation # 3 is shown in Figure 4-4.

Experiments were performed comparing the three Multi-scale representations.

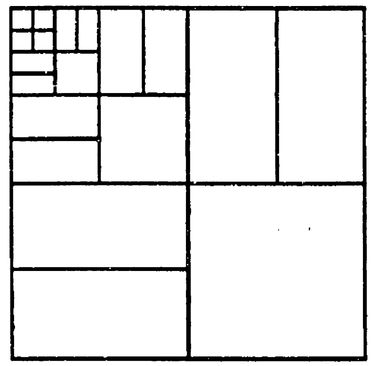

Figure 4-4: Frequency decomposition for Multi-scale representation # 3.

The different representations were implemented in the digital system and a number of sequences were processed. In order to compare the three representations without individually optimizing the coding algorithms for each, the same approach was taken as was done for comparing the various transform/subband filtering schemes. The top 3 % most energetic coefficients were retained with full amplitude resolution. The results for the 5th frame of the girl sequence is shown in Figure 4-6 where the representation corresponding to each image is shown in Figure 4-5.

4.3 Conclusions

The visual performance of all three representations was very similar. Representations # 2 and # 3 resulted in slightly sharper images, but at the expense of a slight increase in ripple. Overall, it was very difficult to distinguish among the three representations. Since representations # 2 and # 3 require a large increase in computations over the original Multi-scale scheme, without any resulting performance improvement, the original Multi-scale scheme was chosen to code the residual.

Original	**Multi-scale Representation # 1**
Multi-scale Representation # 2	**Multi-scale Representation # 3**

Figure 4-5: The representations corresponding to the processed images.

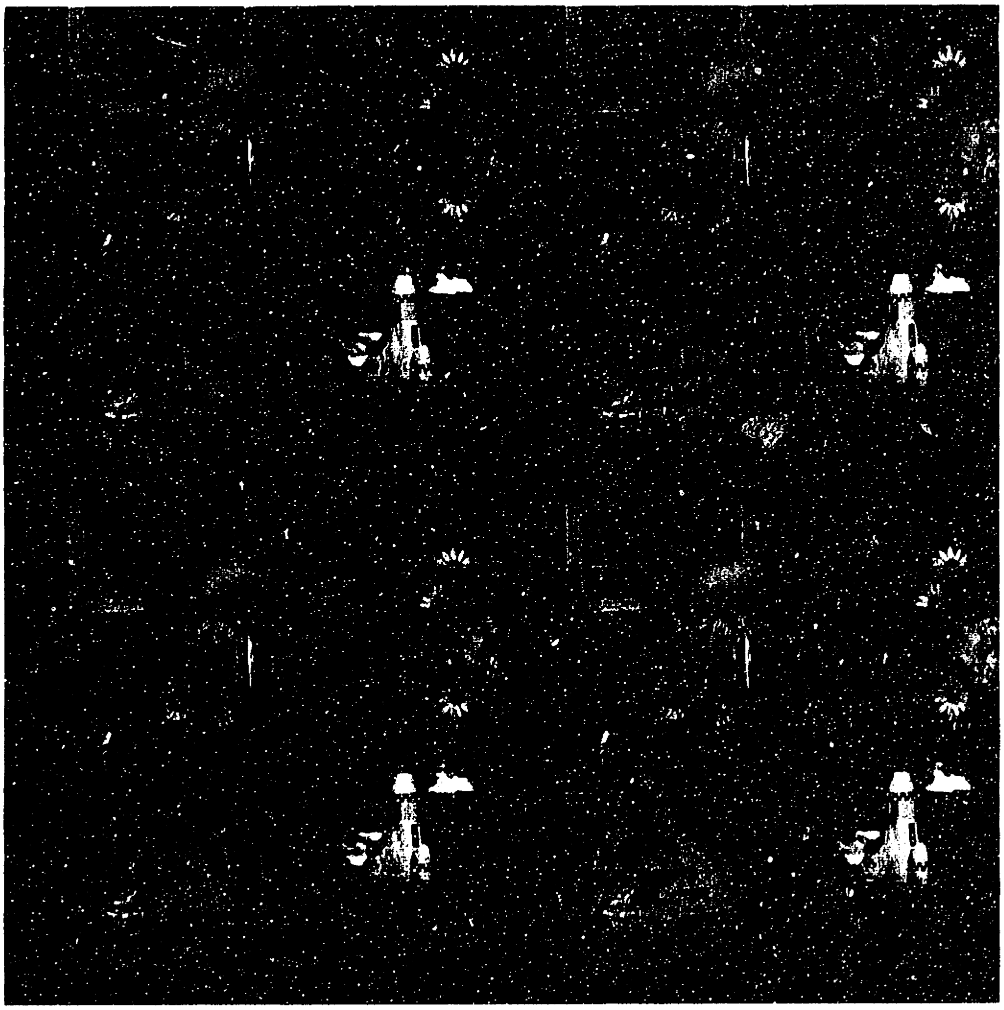

Figure 4-6: The fifth frame of the girl sequence processed with the three different Multi-scale representations.

Chapter 5

Improved Filters

5.1 Current Status of the System

The Multi-scale scheme resulted in eliminating the blocking artifacts, the dominant degradation for low bit rate block transform/subband filtering schemes today. The reconstructed video was perceptually superior for the Multi-scale scheme over either the Block DCT or the LOT schemes. At this point, after eliminating the blocking artifacts, there is one further degradation that is visually unpleasant and should be eliminated. This is the ripple effect, or Gibbs phenomena, that occurs around sharp edges in the images.

5.2 Eliminating the Ripple

The ripple effect can be easily perceived on hard-to-code video sequences where there are sharp edges adjacent to constant background regions. The ripple is directly related to the filters applied. Another problem with the current filters is that they allow close to perfect (but not perfect) reconstruction. It may be beneficial to have a filter bank that does allow perfect reconstruction.

In order to eliminate the ripple and produce a filter set that would allow perfect reconstruction, an attempt was made to design a new set of filters. The filter set would be of the form of a band-splitting QMF filterbank. The filters should (1) be

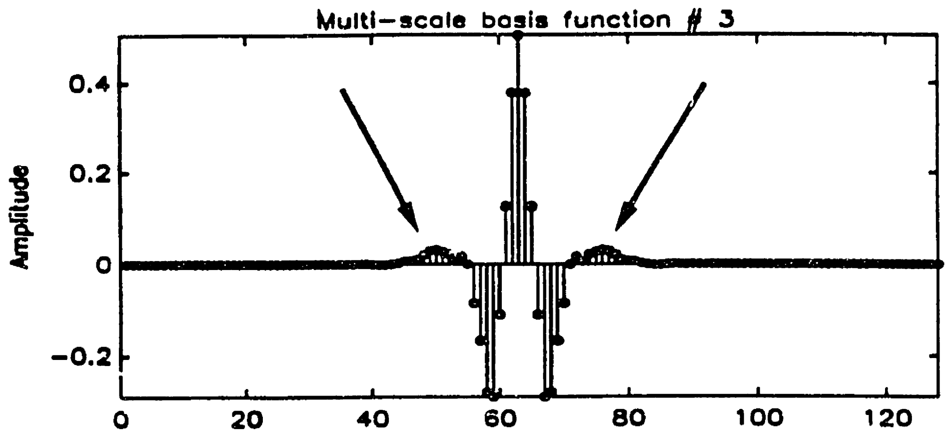

Figure 5-1: An intermediate frequency basis function of the Multi-scale scheme with arrows indicating the secondary lobes that produce the ripple effect.

localized in frequency, (2) allow perfect reconstruction, (3) eliminate aliasing resulting from the decimation, (4) be orthogonal, (5) reduce ripple, (6) be spatially localized, and (7) have linear phase for the complete analysis/synthesis system. If the filters are spatially localized (have short length) then the ripple may be less perceptible and possibly visually masked by the edge.

When analyzing the ripple, one realizes that it is produced because the basis functions of the filterbank contain secondary lobes[20]. Figure 5-1 shows one of the intermediate frequency basis functions for the original Multi-scale scheme with arrows indicating the secondary lobes that produce the ripple effect. The ripple can be eliminated by designing a filterbank that does not have these secondary lobes. Le Gall and Tabatabai designed such a filter set [21].

Le Gall's filter set has as the lowpass synthesis filter kernel the sequence {1 2 1}. This kernel has no secondary lobes and the lowpass synthesis process can simply be viewed as a linear interpolation process. Linear interpolation does not produce any ripple and therefore the use of these filters in the Multi-scale scheme should produce a reconstructed image without ripple. Le Gall's paper does not mention ripple effects or the possibility of reducing them with this filter set.

To test this hypothesis about the ripple, both Le Gall's and Simoncelli's filters were implemented in a 1-D Multi-scale filterbank scheme and used to decompose and then reconstruct a signal. The results are shown in Figure 5-2. (a) is the original signal that was tested. The signal contains two sharp transitions and will result in large ripples at each transition. The signal was decomposed using both sets of filters and then reconstructed while retaining fewer and fewer subbands, with the coefficients in the retained subbands at full amplitude resolution. The plots on the left are the results when using Simoncelli's filters and on the right when using Le Gall's filters. Plots (b) and (c) exhibits the results when all but the highest subband for each filter set are retained. Plots (d) and (e) exhibit the results when all but the highest two subbands are retained. This process continues until the bottom pair of plots, (h) and (i), show the result of only keeping the lowest frequency subband. Plot (h) has overshoots which result in the ripple that is seen in the images. Plot (i) does not exhibit this overshoot and hence one should not see the same ripple effect in the images. The sharpness of plot (i) is of concern, though, as it may lead to the Mach band effect, a visual sharpening of the image at transition points.

Le Gall's set of filters reduce the ripple effects and achieve perfect reconstruction. On the bad side, there is considerable overlap among the lowpass and highpass subbands, the subbands are not orthogonal, and there is the possibility of visual degradations because of the Mach band effect. The considerable overlap of subbands and their nonorthogonality will result in a reduction of coding efficiency for the coefficients. Because of these advantages and disadvantages of the new filter set, the only way to determine if they would improve the visual quality of the system was through direct application to the digital system. The results are shown in Figure 5-3.

The old filter set exhibits very pronounced ripple in uniform background regions bordering on sharp edges. An example of this is the uniform background region adjacent to the color grid. The new filter set results in considerably reduced ripple and visually more pleasant reconstructed images than the old filter set. This is when the images are viewed as still images, without motion. However, when the images are viewed in full motion video, there are some detracting features. There is what seems

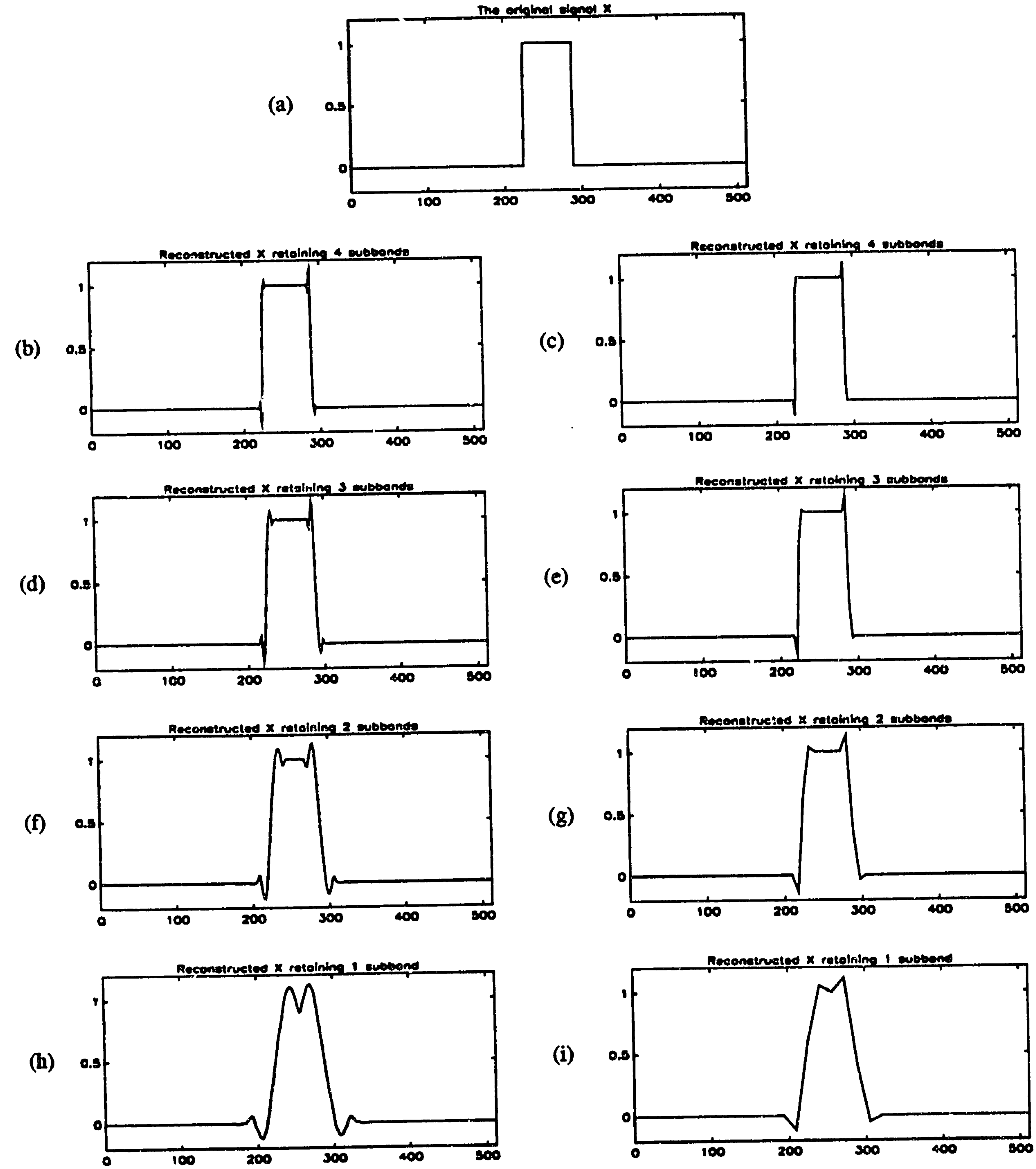

Figure 5-2: The original signal on the top and the results of filtering and reconstructing the signal while retaining fewer and fewer subbands. The original filter set in the Multi-scale scheme is used on the left and a filter set that is hypothesized to reduce the ripple is shown on the right.

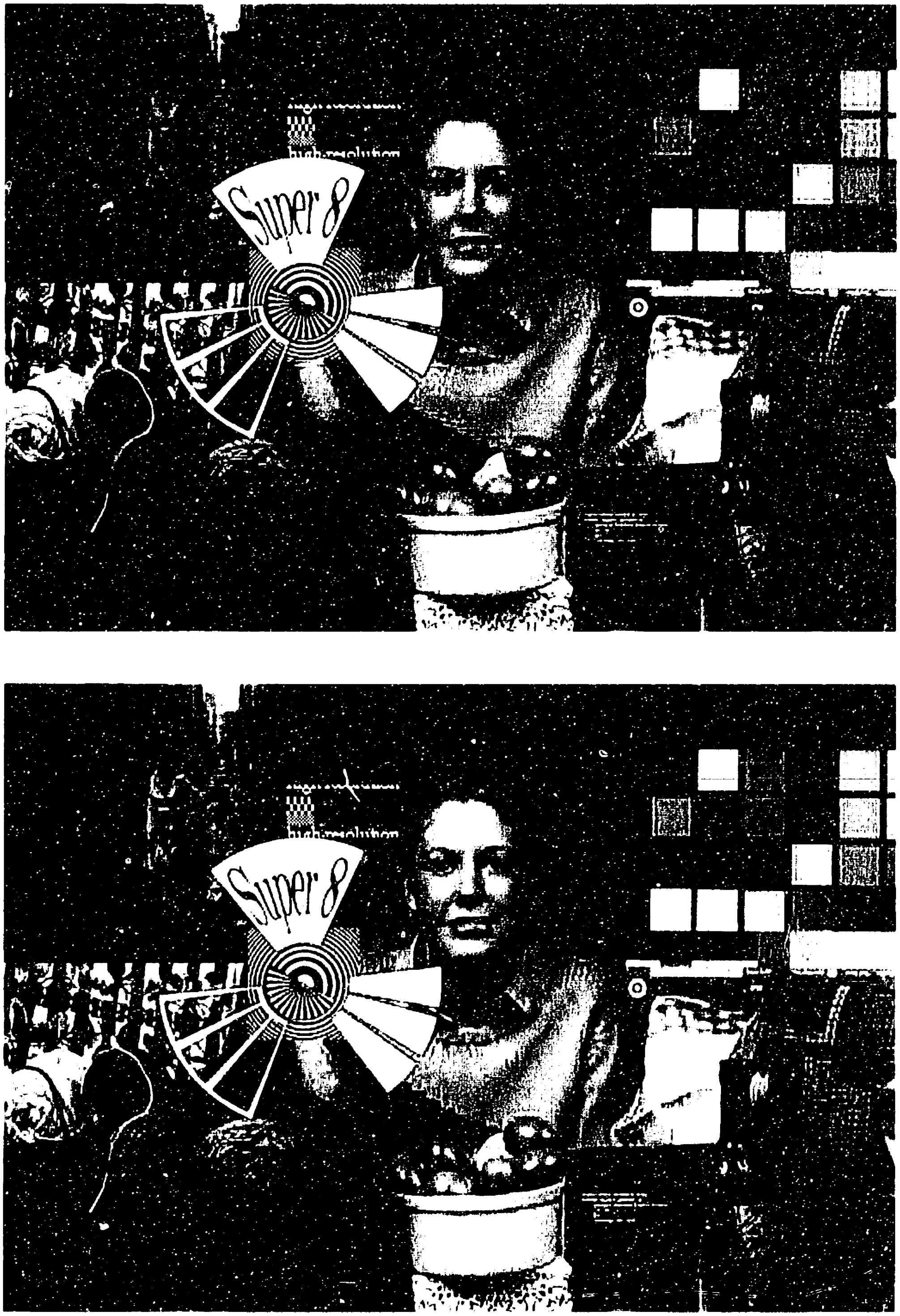

Figure 5-3: Eighth frame of the girl sequence processed using the old filters, on the top, and the new filters, on the bottom.

to be a high frequency coding error. The error manifests itself in terms of a sort of buzyness or moving film, or more specifically, small variations in local pixel intensity that do not move with the picture. This error is only noticeable, and detracting when viewing a sequence. When viewing a single still frame it is unperceivable. The error has spatial-temporal characteristics that make it visually unpleasant only when viewing a sequence. Initial thought was that the error was a function of the temporal processing, possibly the motion-estimation/motion-compensation was being done incorrectly, with an error in the high frequencies. But experiments performed with the DCT and the Multi-scale scheme using the original filters resulted in no buzyness or moving film.

Another possibility for the error is uncanceled aliasing. Uncanceled aliasing is the leakage or overlap of one subband onto another. Uncanceled aliasing occurs for all non-ideal filtering schemes, but it may be much more pronounced for Le Gall's filter set. Le Gall's filter set has a large overlap among subbands and with few coefficients being selected, both low and high frequencies are coded badly. The original Multi-scale filter set has less uncanceled aliasing than Le Gall's and that may be the reason why it does not exhibit the buzyness or film. Comparing Simoncelli's filter set verses Le Gall's, the ripple is more unpleasant than the buzyness or film, and therefore Le Gall's filter set was chosen among the two.

5.3 Conclusions

The choose of the filterbank utilized is critical to the visual performance of the system. Simoncelli's filterbank resulted in very good visual performance except for the ripple effects around sharp edges. Le Gall's filterbank was found to eliminate the detracting ripples, but at the expense of nonorthogonality of the subbands and a visual busyness or film. Simoncelli's filterbank has a number of very important properties and so does Le Gall's. There was a tradeoff among the two. Since the Le Gall's filterbank has considerably reduced the ripple at the expense of the buzyness and nonorthogonality, it seems possible that a different filter set can be designed that may also reduce the ripple and be orthogonal, but without adding the busyness.

Chapter 6

Conclusions

The scarce RF spectrum has mandated that very complex and novel digital signal processing techniques be applied to an HDTV signal in order to compress it to fit in the available bandwidth, while still achieving the highest visual quality possible. The Advanced Television and Signal Processing Group at MIT has designed a digital HDTV system utilizing both temporal and spatial processing to compress the HDTV video while retaining its high fidelity.

Motion estimation/motion compensation is initially applied to the video in order to predict the current frame from the previous frame. The error in this prediction, or motion-compensated residual, is then coded and transmitted over the channel. This thesis presents the work performed in order to determine the optimal method to represent and code the motion-compensated residual so that the highest quality video can be reconstructed at the receiver with the limited bit rate available (approximately .2 to .35 bits/pixel).

Most cutting-edge image compression algorithms today partition an image into 8×8 or 16×16 blocks and then adaptively transform code these blocks with the DCT. These schemes achieve high compression, and allow easy spatial adaptivity, but also have intrinsic blocking effects for low-bit-rate applications that are unacceptable for HDTV. The first step in this research was an attempt to find a transform/subband filtering scheme that would have the compression capability and spatial adaptivity of the Block DCT, but without its inherent blocking effects. The three schemes studied were (1) the Block DCT, which uses basis functions local to the individual blocks, (2) the LOT, which uses basis functions that overlap adjacent blocks, and (3) the Multi-scale scheme which uses basis functions of different lengths.

Qualitative and quantitative tests have determined that the Multi-scale scheme performed as good as or better than the other two schemes. The Multi-scale scheme did not contain the block artifacts that were prevalent in and detracting from the other schemes. The initial Multi-scale scheme contained ripple effects for some images which were visually detracting. A new filter set (Le Gall) was found which reduced the ripple at the expense of reduced coding efficiency and a buzyness or film in the video. When viewing only a still frame, the Multi-scale scheme performed far superior to the others in terms of visual quality. This effect was predominantly produced by the total elimination of blocking artifacts by the Multi-scale scheme. When viewing full motion video the Multi-scale scheme also eliminated the blocking artifacts, but the buzyness or film created detracted from the visual quality of the system. The Multi-scale scheme still outperformed the others for the video test, but the difference was not as great because of the produced buzyness or film.

In our HDTV system, image compression techniques are applied to a motion-compensated residual as opposed to a typical image. As the motion-compensated residual is more highpass in nature than a typical image, it was thought that taking this difference into account may improve the visual quality of the system. Toward this goal two new Multi-scale representations were created. Each resulted in a sharper reconstructed image at the expense of increased ripple artifacts.

Previous research in image compression primarily encompassed encoding moderate resolution images at medium to low bit rates, where minor artifacts in the decoded image were sometimes acceptable. This research was aimed at encoding a high definition motion-compensated residual at a very low bit rate with the decoded HDTV video as artifact-free as possible. The resulting Digital HDTV system produces high quality video without any blocking artifacts and within the limited bit rate available for HDTV in the U.S.

Chapter 7

Future Research

There are a number of areas where future research may result in significant improvements in the HDTV system. The first area is filter design. The choice of filters was found to be critical to the performance of the system. The original filter set performed quite satisfactorily, except for the ripple around sharp edges. The current filter set was chosen because it eliminated most of the detracting ripple, but with a tradeoff. The current filters are not orthogonal, resulting in an inefficiency in coding. Also there is significant leakage among the subbands and the magnitude response of some of the subbands is greater than one. Possibly these last two items may be the cause of the detracting film when viewing the decoded video sequences. Both filter sets had a number of very important properties. Through further work in filter design, it may be possible to produce a filterbank that exhibits the properties of orthogonality and reduced ripple, without the detracting film. Another, more basic issue, is the question of whether an octave-band subdivision of the frequency domain is the optimal choice for the Multi-scale scheme. This is particularly important since the motion-compensated residual is usually more highpass in nature than a typical image. Different decompositions of the frequency domain may produce improved visual effects and therefore should be examined.

Another important issue is whether the residual should always be coded, or are there instances when the original image should be coded. When using motion-compensated prediction, or any other form of temporal prediction, also known as interframe processing, there are times when the prediction error is so great, (e.g. during a scene change or new imagery), that coding the residual may be more difficult than coding the original image. In those cases, the motion-compensated prediction should be suppressed and intraframe processing, where the original image itself is coded, should be performed. This "inter/intra" decision making process should be performed in a spatially adaptive manner to achieve maximum performance gain. The Block DCT allows easy spatially adaptive processing because it partitions the image into separate blocks which are processed independently. This allows inter/intra decisions on local regions of the image to be performed independently and easily. Initially, inter/intra processing may seem more difficult to perform with the Multi-scale scheme whose basis functions are of different lengths and overlap different regions of the image. In order to perform spatially adaptive processing for the Multi-scale scheme a new spatially localized representation for the Multi-scale scheme must be created. The main concept here is that even though the Multi-scale scheme's basis functions have different lengths and overlap different regions of the residual, they can still be arranged so that they correspond to a particular region of the residual. Inter/intra processing may eliminate errors in the temporal prediction and thereby result in improved video quality. Furthermore, it may be utilized for performing periodic refreshes of the video and thereby assist in channel acquisition and also limiting the length of time that transmission errors would affect the video. The concepts related to inter/intra processing of the motion-compensated residual are being studied by members of the Advanced Television and Signal Processing Group and will be considered in depth in Monta's Ph.D. thesis [8].

The human visual system has only begun to be considered. Currently, the human visual system is taken into account when performing adaptive selection among the various subbands and also with the coarseness/fineness of the quantization. A further improvement in the visual quality of the video may be achieved by choosing methods of adaptive selection and quantization based on utilizing the spatial and temporal masking effects of the human visual system.

Another very important consideration is the implementation of the system. The economical issues of performance versus cost must be studied for the building of the transmitters and receivers. The transmission issues must be examined and the system must be made as robust as possible to minimize the effects of transmission errors.

Furthermore, the possible future uses of the system should be examined. For example, the Multi-scale processing of the residual allows the system to be utilized as a progressive or scalable system. This is because the Multi-scale representation inherently allows for video reconstruction at multiple resolutions, since the different scale basis functions correspond to different resolutions of the residual. A low-cost, low-resolution HDTV receiver may therefore be built which would only use the low and mid frequency subbands to reconstruct the video. This may also lead to other applications aside of direct broadcast. There may be one basic standard while various grades of encoders and decoders may be built for use in areas such as cable, medicine, the military, and manufacturing, which may have different performance requirements.

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