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Introduction

During the last Experts group meeting a comparison between Spatial Scalability and Frequency Scalability was made. Software decoding of video streams was one of the items being discussed. During that discussion reference was made to a publication of the University of Berkeley, in which software decoding of MPEG1 streams was discussed.

The document of the University of Berkeley: "Performance of a Software MPEG Video Decoder", *Ketal Patel, Brian C. Smith, and Lawrence A. Rowe* is appended for information to this document.

(Continued from page 8)

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Performance of a Software MPEG Video Decoder*

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Abstract

The design and implementation of a software decoder for MPEG video bitstreams is described. The software has been ported to numerous platforms including PC's, workstations, and mainframe computers. Performance comparisons are given for several different bitstreams and platforms including a unique metric devised to compare price/performance across different platforms (percentage of required bit rate per dollar). We also show that memory bandwidth is the primary limitation in performance of the decoder, not the computational complexity of the inverse discrete cosine transform as is commonly thought.

1. Introduction

The CCITT MPEG group was formed in 1988 to develop a standard for storing video and associated audio on digital media. Their goal was to define a standard that required bit rates less than 1.5 Mbits/sec, a bandwidth achievable by computing networks and digital storage media available today. A draft proposal was agreed upon in September, 1990. Since then, minor changes have been made and released. The work described in this paper is based on the December 1991 committee draft [5].

Many research and commercial groups have developed MPEG decoders. Because of the high stakes involved in commercializing MPEG technology (e.g., videoconferencing, HDTV, and multimedia computing), these groups have been reluctant to release their coders, decoders, and bitstreams. The absence of public domain MPEG utilities has hindered research on MPEG applications.

We implemented an MPEG video decoder for three reasons. First, we wanted to determine whether MPEG video could be decoded in real-time using a software-only implementation on current generation desktop computer. Second, we needed to develop a portable software decoder for inclusion in the Continuous Media Player being developed at U.C. Berkeley [1]. And third, we wanted to contribute public domain source code to the research community.

This paper describes the design and implementation of the decoder. A novel feature of our decoder is the use of a dithering algorithm in YCrCb-space. We also report the performance of playing six anonymous bitstreams that we have acquired on a variety of platforms. Rather than saying "we can play bitstream A on platform P at N frames/second," we have devised a metric that compares the relative price/performance of different platforms. In our analysis, we have found that memory bandwidth is the primary limitation in performance of the decoder, not the computational complexity of the inverse discrete cosine transform (IDCT) as is commonly thought.

The remainder of this paper is organized as follows. Section 2 present a brief introduction to the MPEG video coding standard. Section 3 describes the implementation of our decoder and presents a time/space performance analysis. Section 4 describes optimizations to improve decoder performance. Section 5 describes the bitstreams used in the dithering and cross-platform analyses present in sections 6 and 7, respectively. Lastly, we describe the experience of publishing this software on the Internet.

2. The MPEG Video Coding Model

This section briefly describes the MPEG video coding model. More complete descriptions are given in an introductory paper [4] and the ISO standard [5].

Video data can be represented as a set of images, I_1, I_2, ..., I_N, that are displayed sequentially. Each image is represented as a two dimensional array of *RGB triplets*, where an RGB triplet is a set of three values that give the red, green and blue levels of a pixel in the image.

MPEG video coding uses three techniques to compress video data. The first technique, called *transform coding*, is similar to JPEG image compression [7]. Transform coding exploits two facts: 1) the human eye is relatively insensitive to high frequency visual information and 2) certain mathematical transforms concentrate the energy of an image, which allows the image to be repre-

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I_1 I_2 I_3

Figure 1: Sample sequence.

sented by fewer values. The discrete cosine transform (DCT) is one such transform. The DCT also decomposes the image into frequencies, making it straightforward to take advantage of (1).

In MPEG transform coding, each RGB triplet in an image is transformed into a YCrCb triplet. The Y value represents the *luminance* (black and white) level and Cr/Cb values represent *chrominance* (color information). Since the human eye is less sensitive to chrominance than luminance, the Cr and Cb planes are *subsampled*. In other words, the width and height of the Cr and Cb planes are halved.

Processing continues by dividing the image into *macroblocks*. Each macroblock corresponds to a 16 by 16 pixel area of the original image. A macroblock is composed of a set of six 8 by 8 pixel *blocks*, four from the Y plane and one from each of the (subsampled) Cr and Cb planes. Each of these blocks is processed in the same manner as JPEG: the blocks are transformed using the DCT and the resulting coefficients quantized, run length encoded to remove zeros, and entropy coded. The details can be found in [5], but the important facts for this paper are that: 1) the frame is structured as a set of macroblocks, 2) each block in the macroblock is processed using the DCT, and 3) each block, after quantization, contains many zeros.

The second technique MPEG uses to compress video, called *motion compensation*, exploits the fact that a frame I_x is likely to be similar to its predecessor I_{x-1}, and so can be nearly constructed from it. For example, consider the sequence of frames in Figure 1, which might be taken by a camera in a car driving on a country road. Many of the macroblocks in frame I_2 can be approximated by pieces of I_1, which is called the *reference frame*. By *pieces* we mean any 16 by 16 pixel area in the reference frame. Similarly, many macroblocks in I_3 can be approximated by pieces of either I_2 or I_1. The vector indicating the appropriate piece of the reference frame requires fewer bits to encode than the original pixels. This coding results in significant data compression.

Note, however, that the right edge of I_2 (and I_3) can not be obtained from a preceding frame. Nor can the portion of the background blocked by the tree in I_1; these areas contain new information not present in the reference frame. When such macroblocks are found, they are encoded without motion compensation, using transform coding.

Further compression can be obtained if, at the time I_2 is coded, both I_1 and I_3 are available as reference frames[1]. I_2 can then be built using both I_1 and I_3. When a larger pool of reference frames is available, motion compensation can be used to construct more of the frame being encoded, reducing the number of bits required to encode the frame. A frame built from a earlier frame is called a *P* (forward or predicted) frame, and a frame built from both a preceding frame and a subsequent frame, is called a *B* (bidirectional) frame. A frame coded without motion compensation, that is, using only transform coding, is called an *I* (intracoded) frame.

Motion compensation in P and B frames is done for each macroblock in the frame. When a macroblock in a P or B frame is encoded, the best matching[2] macroblock in the available reference frames is found, and the amount of x and y translation, called the *motion vector* for the macroblock, is encoded. The motion vector is in units of integral or half integral pixels. When the motion vector is on a half pixel boundary, the nearest pixels are

[1] This will, of course, require additional buffering and introduce delay in both encoding and decoding.

[2] The criteria for "best matching" is determined by the encoder.

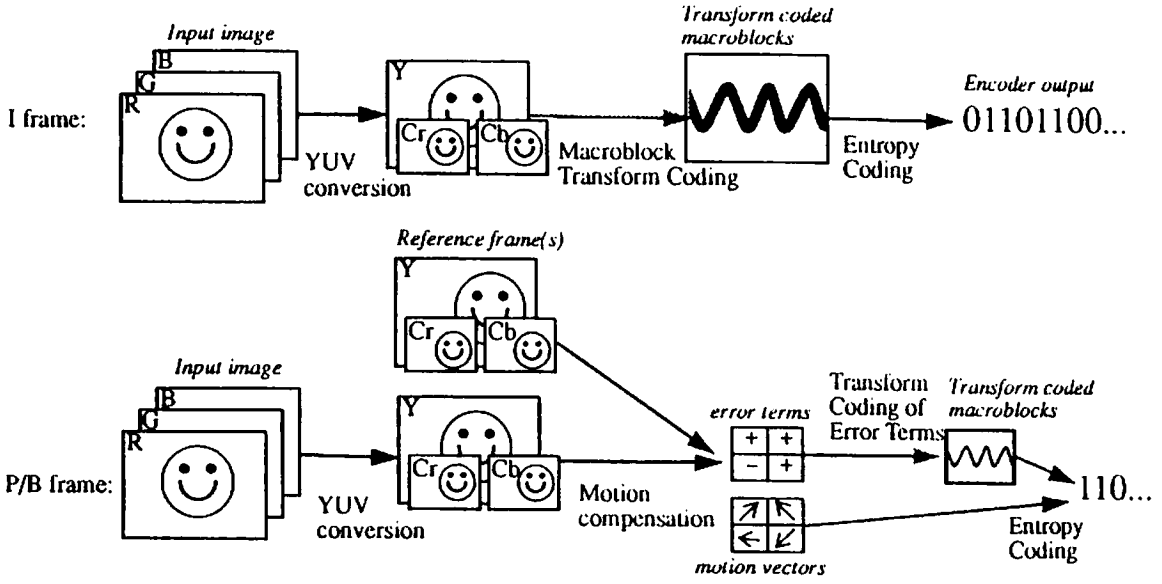

Figure 2: MPEG video coding procedure.

averaged. The match between the predicted and actual macroblocks is often not exact, so the difference between the macroblocks, called an *error term*, is encoded using transform coding.

The final technique MPEG uses to compress video data is *entry coding*. After motion compensation and transform coding, a final pass is made over the data using Huffman coding. Figure 2 summarizes the MPEG video coding process.

To rebuild the YCrCb frame, the following operations are needed:

(1) the entropy coding must be inverted,
(2) for P and B frames, the motion vectors must be reconstructed and the appropriate parts of the reference frame copied, and
(3) the error terms must be decoded and incorporated (which includes an application of the IDCT).

Once the YCrCb frame has been built, the frame is converted to a representation appropriate for display. This last step is called *dithering*, and is discussed further in section 6.

In summary, MPEG uses three techniques to compress video data: motion compensation, transform coding, and entropy coding. MPEG defines three types of frames, called I, P and B frames. These frames use zero, one, and two reference frames for motion compensation, respectively. Frames are represented as an array of macroblocks, and both motion compensation and transform coding operate on macroblocks.

3. Implementation

The decoder is structured to process a small, implementation dependent *quantum* of macroblocks at a time so it can be suspended between any quantum of macroblocks. We envision using the decoder as a part of a larger system (the CM Player [1]) for delivering video data over local networks. This architecture is required by the player for timesharing CPU resources and servicing other tasks required in a multimedia system (e.g., handling user input or playing other media).

The decoder was implemented in C using the X Windowing System. It is composed of 12K lines of code. Our intent was to create a program that would be portable across a variety of UNIX platforms. To date, the decoder has been ported to over 10 platforms.

Preliminary analysis of the run-time performance indicated that dithering accounted for 60% to 80% of the time, depending on the architecture. Consequently, we focussed our attention on speeding up this part of the code. We also optimized other decoding procedures using standard optimization methods: 1) in-line procedure expansion, 2) caching frequently accessed values, and 3) custom coding frequent bit twiddling operations. Altogether, these changes to the decoder reduced the time by 50% from our initial implementation. The specific optimizations are discussed in the next section. More significant improvements (over a factor of 15) were made by improving the dither algorithm. Dithering is discussed in detail in section 5.

The fastest color dithering algorithm with reasonable quality is an ordered dither that maps a 24-bit YCrCb image to a 7-bit color space (i.e., 128 colors) using a fixed color map. We analyzed the performance of the decoder using this technique. The following table shows the results:

Function	% Time
Parsing	17.4%
IDCT	14.2%
Reconstruction	31.5%
Dithering	24.3%
Misc. Arithmetic	9.9%
Misc.	2.7%

Parsing includes all functions involved in bitstream parsing, entropy and motion vector decoding, and coefficient reconstruction. The IDCT code, which is called up to six times per macroblock, is a slightly modified version of the fastest public domain IDCT available [2]. The algorithm applies a 1 dimensional IDCT to each row and then to each column. Zero coefficients are detected and used to avoid unnecessary calculation. Functions that perform predictive pixel reconstruction, including copying and averaging relevant pixels from reference frames, are grouped under the category *reconstruction*. Finally, dithering converts the reconstructed YCrCb image into a representation appropriate for display.

The table shows that over half the time is spent in reconstruction and dithering. Parsing and IDCT each require about 15% of the time. The reason reconstruction and dithering are so expensive is that they are memory intensive operation. On general purpose computers with RISC processors, memory references take significantly longer than arithmetic operations, since the arithmetic operations are performed on registers. Even though steps such as parsing and IDCT are CPU intensive, their operands stay in registers, and are therefore faster than the memory intensive operations of reconstruction and dithering. While improving the IDCT is important, our decoder could be sped up most significantly by finding a scheme to reduce memory traffic in reconstruction and dithering.

4. Optimizations

This section describes some of the low-level optimizations used to improve the basic decoder. Three kinds of improvements are discussed: general coding techniques, IDCT optimizations, and average cheating.

Numerous coding optimizations were applied throughout the code. One strategy was to use local copies of variables to avoid memory references. For example, since the addition of the error term to a pixel value often causes underflow or overflow (i.e., values less than 0 or greater than 255), bounds checking was required. This implementation results in three operations: the addition of the error term to the pixel, an underflow check, and an overflow check. Instead of accessing the pixel in memory three times, a local copy is made, the three operations are performed, and the result is stored back into memory. Since the compiler allocates the local copy to a register, we found the operations themselves to be about four times faster.

We also applied this technique to the bit parsing operations by keeping a copy of the next 32 bits in a global variable. The actual input bitstream is only accessed when the number of bits required is greater than the number of bits left in the copy. In critical segments of the code, particularly macroblock parsing, the global copy is again copied into a local register. These optimizations resulted in 10-15% increases in performance.

Other optimizations applied included: 1) loop unrolling, 2) math optimizations (i.e., replacing multiplications and divisions with left and right shifts), and 3) in-line expansion of bit parsing and Huffman decoding functions.

The IDCT code was also heavily optimized. The input array to the IDCT is typically sparse. Analysis showed that 30%-40% of the blocks contained less than five coefficients in our sample data, and frequently only one coefficient exists. These special cases are detected during macroblock parsing and passed to the IDCT code which is optimized for them.

Finally, we found a way to cheat on computation of pixel averaging in interframes (i.e., P- and B-frames). The MPEG standard specifies that predictive pixel values for interframes are constructed from copying areas in past or future frames based on a transmitted set of motion vectors. These motion vectors can be in half-pixel increments which means pixel values must be averaged.

The worst case occurs when both the horizontal and vertical vectors lie on half-pixel boundaries. Here, each result pixel is an average of four pixels. The increased precision achieved by doing the pixel averaging is lost, however, in the dithering process. We optimize these functions in three ways. First, if both horizontal and vertical vectors lie on whole pixel boundaries, no averaging is required and the reconstruction is implemented as a memory copy.

Second, if only one motion vector lies on a half-pixel boundary, the average is done correctly. And finally, if both vectors lie on half-pixel boundaries, the average is computed with only 2 of the 4 values. We average the

value in the upper left quadrant with the value in the lower right quadrant, rather than averaging all four values. Although this method produces pixels that are not exactly correct, dithering hides the error.

5. Sample Bitstreams

This section describes the bitstreams used for the performance comparisons presented in the next two sections.

Public domain MPEG data is scarce. We selected six bitstreams available to us that we believe gives a reasonable approximation of a random sample. Table 1 presents the characteristics of the bitstreams. Four distinct coders were used to generate the data. Bitstreams A and B and bitstreams D and E were generated by the same coders. The video sequences are completely different except the sequences encoded in bitstreams B and C which use different sequences from the same raw footage.

The variation in frame rates, frame size, and compression ratios makes analysis with these bitstreams difficult to compare. We believe the best metric to judge the performance of the decoder is to measure the percentage of the required bit rate achieved by the decoder. For example, if a bitstream must be decoded at a rate of 1 Mbit/sec to play it at the appropriate frame rate, a decoder that plays at a rate of 0.5 Mbit/sec is able to achieve 50% of the required bit rate. Given a set of bitstreams, two decoders running on the same platform can be compared by calculating the percentage of bitstreams each decoder can play in real-time (i.e., at the required bit rate).

6. Dithering Performance

This section describes the performance improvements made to the dithering algorithm(s) used in the decoder. In this context, dithering is the process of converting a 24 bit YCrCb image into a representation appropriate for display. In principle, the YCrCb image is first converted to an RGB representation and the dithering algorithm is applied. Virtually all dithering algorithms, however, can be applied directly to the YCrCb image. This approach avoids the memory traffic and arithmetic computation associated with RGB conversion, and further reduces memory accesses since the Cr and Cb planes are subsampled.

The decoder supports monochrome, full (24 bit) color, gray scale and color mapped display devices. Dithering to full color devices simply requires RGB conversion. Dithering to gray scale devices is done by using only the luminance plane of the image. For color mapped devices, two dithering techniques are used: error diffusion (sometime called Floyd-Steinberg) and ordered dither. Both are discussed in [3].

In error diffusion dithering, each image pixel is mapped to the closest pixel in a fixed-size color map. In our decoder, the color map has 128 entries, with 3 bits allocated for luminance, and 2 bits for each chrominance channel. The difference, expressed as a YCrCb triplet, between the image pixel and the colormap pixel is called the *error*. This error is distributed to neighboring pixels. For example, half the error might be added to the pixel below and half to the pixel to the right of the current pixel. The next (possibly modified) pixel is then processed. Processing is often done in a serpentine scan order: odd number rows are processed left to right and even number rows are processed right to left.

Ordered dithering is a form of threshold dithering. In threshold dithering, any pixel below a certain *threshold* of luminance is mapped to black, and all other values are mapped to white. Ordered dithering uses the pixel's (x,y) coordinate in the image to determine the threshold value. An N by N *dithering matrix* D determines the threshold: D(x mod N, y mod N) is the threshold at position (x,y). A four by four dithering matrix is used in our decoder. The matrix is chosen so that, over any N by N region of the image with the same pixel value, the mean of the

Stream	Stream Size	Frame Size	Avg. Size I frame	Avg. Size P Frame	Avg. Size B Frame	Frames/ second	Bits/ pixel	Bits/ second	I:P:B
A	690 K	320x240	18.9K	10.6K	0.8K	30	.488 (50:1)	1.12M	10:40:98
B	1102 K	352x240	11.2K	8.8K	6.3K	30	.701 (34:1)	1.78M	11:40:98
C	736 K	352x288	23.2K	8.8K	2.5K	25	.469 (51:1)	1.19M	11:31:82
D	559 K	352x240	8.1K	5.5K	4.1K	6	.445 (54:1)	0.23M	6:25:88
E	884 K	352x240	12.4K	9.1K	6.5K	6	.698 (34:1)	0.35M	6:25:89
F	315 K	160x128	2.8K	N/A	N/A	30	1.09 (20:1)	0.67M	113:0:0

Table 1: Sample bitstreams.

dithered pixels in the region is equal to the original pixel value. This scheme can be easily extended to dither color images. For further details, the interested reader is referred to [3] and [6].

When implementing the decoder, we started with a straightforward implementation of the error diffusion algorithm with propagation of 4 error values (*FS4*). The first improvement we tried was to implement a error diffusion algorithm with only 2 propagated error values (*FS2*). This change improved run-time performance at a small, and insignificant, reduction in quality.

The second improvement we implemented was to use an ordered dither. We map directly from YCrCb space to an seven bit color map value by using the pixel position, three bits of luminance, and two bits each of Cr and Cb chrominance as a key to a lookup table of pixel values. We call this dither *ORDERED*.

Table 2 shows the relative performance of these dithers along with a grayscale (*GRAY*) and 24-bit color dither (*24BIT*). The table also shows the performance of the decoder without dithering. These tests were run on an HP 750 which is the fastest machine currently available to us. The results are expressed as percentages of required bit rates to factor out the differences in bitstreams.

Several observations can be made. First, notice that only 3 bitstreams (D, E, and F) were playable at the required bit rate (i.e., the percentage was over 100%) using the fast ORDERED dither. These bitstreams have low required bit rates because streams D and E were coded at 6 frames per second (*fps*), and stream F is only 160x128 pixels.

The remaining bitstreams can be played at approximately 35% of the required bit rate which implies that current generation workstations can play 8-10 fps.

Second, notice that even without dithering, which is shown in the column labeled NONE, our decoder can achieve only 50% of the required bit rate.

Notwithstanding this pessimistic result, private communications with other groups working on decoders optimized for particular platforms say that their decoders operate 2-7 times faster than our portable implementation.[3] The implication is that we are close to being able to decode and play reasonable-sized videos. Indeed, videos with small images and low frame rates can be played on PC's (e.g., Macintosh QuickTime).

7. Cross-Platform Performance

In evaluating the decoder on different platforms, we cannot use the percentage of required bit rate metric to rate the platform because price/performance is important. For example, running a decoder on two platforms where one platform is 4 times more expensive does not really tell you much. A better metric would factor in the cost of the hardware.

The metric we propose is the *percentage of required bit rate per second per thousand dollars*. We will call this metric *PBSD*. For example, suppose two machines M1 and M2 that cost $15K and $12K respectively play a bitstream at 100% and 50% of the required bit rate. The PBSD metrics for the two machines are 6.7 and 4.2. Higher numbers correspond to better price performance, so machine M1 is better than M2.

On the other hand, suppose that M1 played only 60% of the required bit rate. The PBSD metrics would be 4.0 and 4.2, which implies that M2 has better price performance. Finally, suppose both machines can play the bitstream at 100% of the required bit rate. Here, M2 is clearly better since it is less expensive, and the metrics confirm this comparison because they are 6.7 and 8.3.

[3] For example, machines with parallel processors can pipeline the decoding process, and bit parsing can be optimized using special processor instructions.

Stream	FS4	FS2	ORDERED	GRAY	24BIT	NONE
A	12.1%	24.6%	39.1%	43.2%	30.2%	52.7%
B	10.6%	20.9%	31.8%	35.2%	25.1%	41.2%
C	11.1%	23.0%	37.0%	42.0%	27.9%	49.6%
D	54.5%	109.6%	174.0%	196.4%	131.6%	230.6%
E	52.1%	104.7%	161.3%	180.2%	124.2%	204.1%
F	41.1%	76.9%	110.8%	121.5%	88.8%	139.5%

Table 2: Relative performance of different dithering algorithms.

The following table shows the PBSB metric for playing the sample bitstreams using ordered dithering on three workstations available to our research group. The tests were run with image display accomplished using shared memory to copy to the frame buffer.

Stream	HP 750	SUN Sparc 1+	DECstation 5000/125
Cost	$43 K	$7 K	$10 K
A	0.91	1.4	1.2
B	0.74	1.1	1.0
C	0.86	1.3	1.1
D	4.0	6.1	5.1
E	3.7	5.6	4.8
F	2.6	4.0	3.4

From the table we conclude that the HP is roughly a factor of three more efficient on a price/performance basis. This result is expected, since the Sparc 1+ and DECstation 5000 are previous generation workstations compared to the HP 750. If we extrapolate from these two data points, we can expect the next generation of workstations to be able to support video at the quality of streams A, B and C (i.e., 320 by 240 pixel video at 30 frames per second) using a software-only solution.

Many factors will influence this comparison, including whether the X shared memory option is available to reduce copies between the decoder and the X server, whether the X server is local or across a network, and whether the file containing the compressed data is local or NSF mounted. All these changes can significantly affect the results.

8. Internet Distribution

This section describes our experiences distributing the decoder. We were amazed at the response when we distributed this code on the Internet.[4] Within 6 weeks of announcing the availability of a portable software decoder for MPEG video on several newsgroups (e.g., alt.graphics.pixutils and comp.compression), over 500 people had FTP'd the software. They have reported numerous bugs and suggestions for improvement, and they have contributed code to add features, fix bugs, and support new platforms.

We also received our first video mail when a user sent us an MPEG bitstream in a message.[5] We played the mail message with our decoder. It would be easy to add definitions for playing MPEG components to an extensible mail system like MIME [8] using our decoder.

9. Conclusions

Several conclusions can be drawn from this work. First, while IDCT performance is important, it is not the most critical process in a software decoder. Data structure organization and bit-level manipulations are critical.

Second, memory bandwidth is critical on RISC processors. We suspect hardware implementations of MPEG will use fast static RAM and pipeline key operations (i.e., parsing, IDCT, reconstruction, etc.) to avoid this memory bandwidth problem.

Lastly, current generation workstations, like the HP 750, can decode 320 by 240 video sequences at *10-15* frames per seconds, which means they are roughly a factor of two away from real-time performance. We cannot wait to try the new generation workstations that will soon be available.

Acknowledgments

We want to thank the numerous people who ported the decoder to new platforms, supplied ideas and code to improve performance of the software, and provided bug fixes and extensions. While we do not have room to name all the people who have helped, important contributions were made by Todd Brunhoff of North Valley Research, Reid Judd of Sun Microsystems, Toshihiko Kawai of Sony, and Tom Lane of the Independent JPEG Group.

References

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[2] T. Lane, "JPEG Software," Independent JPEG Group (Dec. 1992).

[3] R. Ulichney, *Digital Halftoning* MIT Press, Cambridge, Mass. 1987.

[4] The software is available via anonymous FTP from toe.cs.berkeley.edu [128.32.149.117] in the directory pub/multimedia/mpeg.

[5] The message was *uuencode*'d which converts a binary file to ASCII. *Uudecode* is a companion program that converts the ASCII back to binary. It works well with saved mail messages because it ignores message headers.