EXHIBIT C

TO DECLARATION OF S. MERRILL WEISS IN SUPPORT OF PLAINTIFF ACACIA MEDIA TECHNOLOGIES CORPORATION'S MEMORANDUM OF POINTS AND AUTHORITIES IN OPPOSITION TO ROUND 3 DEFENDANTS' MOTION FOR SUMMARY JUDGMENT OF INVALIDITY UNDER 35 U.S.C. § 112 OF THE '992, '863, AND '702 PATENTS; AND SATELLITE DEFENDANTS' MOTION FOR SUMMARY JUDGMENT OF INVALIDITY OF THE '992, '863, AND '720 PATENTS

Pre-Certification

DIGICIPHER™ HDTV SYSTEM

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VIDEOCIPHER DIVISION
6262 Lusk Boulevard
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22 June 1990

Table of Contents

List of Figures

List of Tables

Exhibit C Page 119

1. INTRODUCTION

General Instrument's DigiCipher™ System is an all digital HDTV system that can be transmitted over a single 6 MHz VHF or UHF channel. It provides full HDTV performance with virtually no visible transmission impairments due to noise, multipath, and interference. It offers high picture quality, while the complexity of the decoder is low. Furthermore, low transmitting power can be used, making it ideal for simulcast HDTV transmission using unused or taboo channels.

The DigiCipher™ HDTV System can also be used for cable and satellite transmission of HDTV. There is absolutely no satellite receive dish size penalty (compared to FM-NTSC) in the satellite delivery of DigiCipher™ HDTV. This is important not only for DBS, but for broadcast and cable since broadcast network and cable programming is typically delivered to affiliates via satellite.

To achieve the full HDTV performance in a single 6 MHz bandwidth, a highly efficient unique compression algorithm based on DCT transform coding is used. Through the extensive use of the computer simulation, the compression algorithm has been refined and optimized. Computer simulation results show excellent video quality for a variety of HDTV material. For error free transmission of the digital data, powerful error correction coding combined with adaptive equalization is used. At a carrier-to-noise ratio of above 19 dB, essentially error free reception can be achieved.

The description of the DigiCipher™ System in Chapter 2 presents details of the signal processing and system parameters. The alternate media distribution of Chapter 3 discusses the suitability of the DigiCipher™ system for use in satellite, cable, and other distribution systems.

2. Description of the DigiCipher™ System

2.1 DigiCipher™ System Overview

The DigiCipher™ HDTV System is an integrated system that can provide high definition digital video, CD-quality digital audio, data and text services over a single VHF or UHF channel. Bandwidth for conditional access capability that allows the encryption of video, audio, and data services is also provided.

Figure 2-1 shows the overall system block diagram. At the HDTV station, the encoder accepts one high definition video and four audio signals and transmits one 16-QAM modulated data stream. The control computer can supply program related information such as program name, remaining times, and program rating. At consumer's home, the DigiCipher™ HDTV receiver receives the 16-QAM data stream and provides video, audio, data, and text to the subscriber. On screen display can be used to display the program related information.

Figure 2-2 shows the block diagram of the encoder. The digital video encoder accepts YUV inputs with 16:9 aspect ratio and 1050-line interlaced (1050/2:1) at 59.94 field rate. The YUV signals are obtained from analog RGB inputs by RGB-to-YUV matrix, low pass filtering, and A/D conversion. The sampling frequency is 51.80 MHz for Y,U, and V. The digital video encoder implements the compression algorithm and generates video data stream. The digital audio encoder accepts four audio inputs and generates audio data stream. The data/text processor accepts four data channels at 9600 baud and generates a data stream. The control channel processor interfaces with the control computer and generates control data stream.

The mulitplexer combines the various data streams into one data stream at 15.8 Mbps. The FEC encoder adds error correction overhead bits and provide 19.42 Mbps of data to the 16-QAM modulator. The symbol rate of the 16-QAM signal is 4.86 MHz.

Figure 2-3 shows the block diagram of the decoder. The 16-QAM demodulator receives IF signal from the VHF/UHF tuner and provides the demodulated data at 19.42 Mbps. The demodulator has an adaptive equalizer to effectively combat multipath distortions common in VHF or UHF terrestrial transmission. The FEC decoder corrects virtually all random or burst errors and provides the error-free data to the Sync/Data selector. The Sync/Data Selector maintains overall synchronization and provides video, audio, data/text, and control data streams to appropriate processing blocks.

The control channel processor decodes the program related information. The user microprocessor receives commands from the remote control unit (RCU) and controls various functions of the decoder including the channel selection.

In the following sections, details of the encoder and the decoder are described.

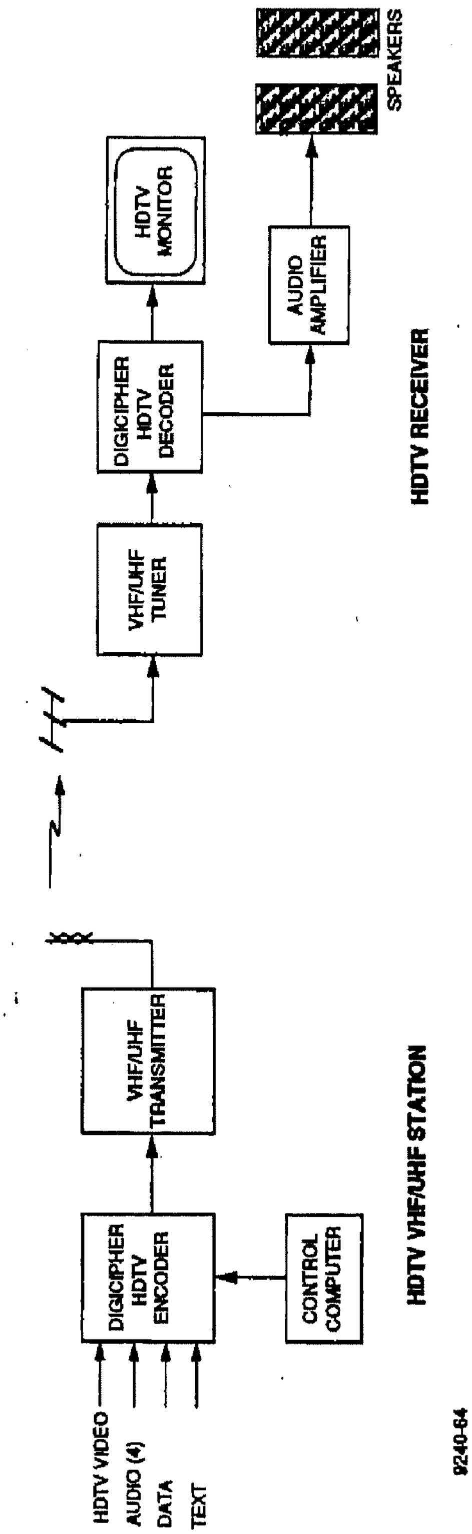

Figure 2-1. System Block Diagram

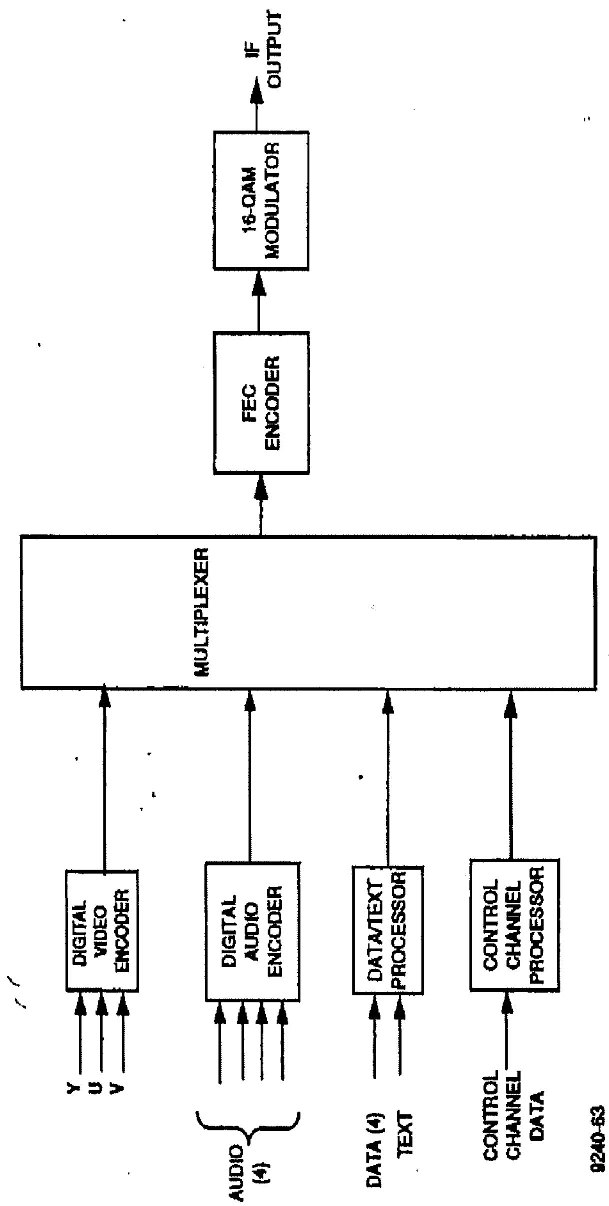

Figure 2-2. Encoder Block Diagram

Exhibit C Page 123

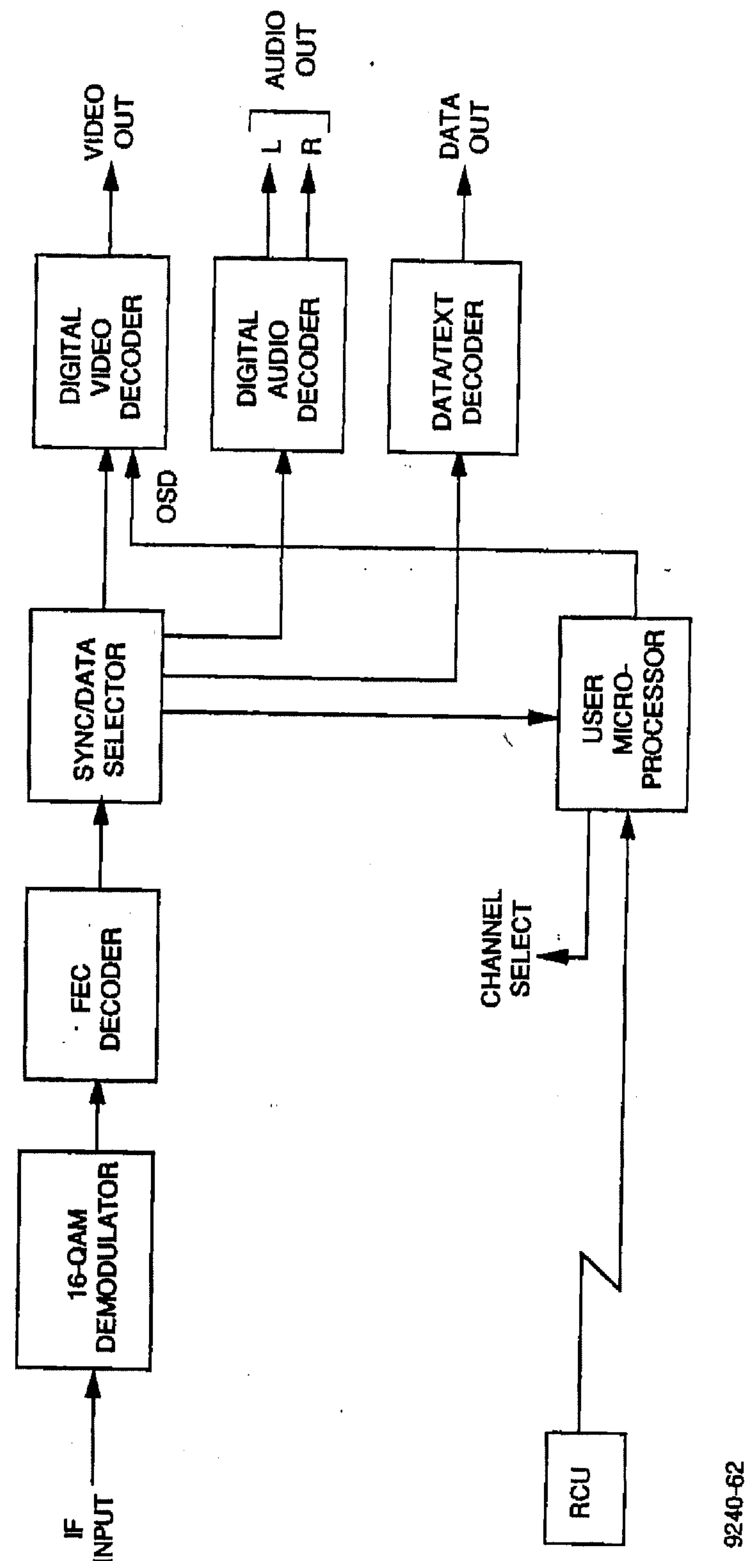

Figure 2-3. Decoder Block Diagram

Exhibit C Page 124

2.2 Digital Video Processing

The compression process can be broken down into five different subprocesses:

1. Chrominance Preprocessor
2. Discrete Cosine Transform (DCT)
3. Coefficient Quantization (Normalization)
4. Huffman (Variable Length) Coding
5. Motion Estimation and Compensation

Basic block diagrams for the encoder and the decoder video processing are shown in Figures 2-4 and 2-5 respectively.

The subsequent discussions refer to certain basic picture processing elements:

- **Pixel:** An 8 bit active video sample (luminance or chrominance). Unless mentioned, the term "pixel" refers to luminance pixels. Representing an image by digitized samples is generally referred to as PCM coding.

- **Block:** An image area 8 pixels horizontally by 8 pixels vertically.

- **Superblock:** An image area 4 luminance blocks horizontally by 2 luminance blocks vertically; associated with 1 chrominance block each for U and V derived from that image area.

- **Macroblock:** An image area 8 superblocks horizontally.

These elements are described further in the appropriate sections.

2.2.1 Chrominance Preprocessor

The resolution of chrominance information can be further reduced relative to luminance resolution with only a slight effect on the perceived image quality. Therefore, to take advantage of this phenomenon, the input signal must first be separated into luminance and chrominance components if it does not already exist in this form. The Y,U,V color space (See CCIR 601) representation has been chosen for this purpose. The U and V chrominance components are decimated horizontally by a factor of 4 and vertically by a factor of 2.

The decimation requires the application of a prefilter prior to subsampling. In this case, simple boxcar filters are used. That is, pixels are averaged in groups of four horizontally, and groups of two vertically. Since the vertical averaging is performed across two different fields, some degradation in motion rendition occurs. In practice, however, this degradation is very difficult to detect. We are not only less sensitive to reductions in chrominance spatial resolution, but in temporal resolution as well.

The luminance signal (Y) bypasses the chrominance preprocessor, and therefore full resolution is maintained. The chrominance components are then multiplexed with the luminance component, one block at a time, and all components are then subjected to the same

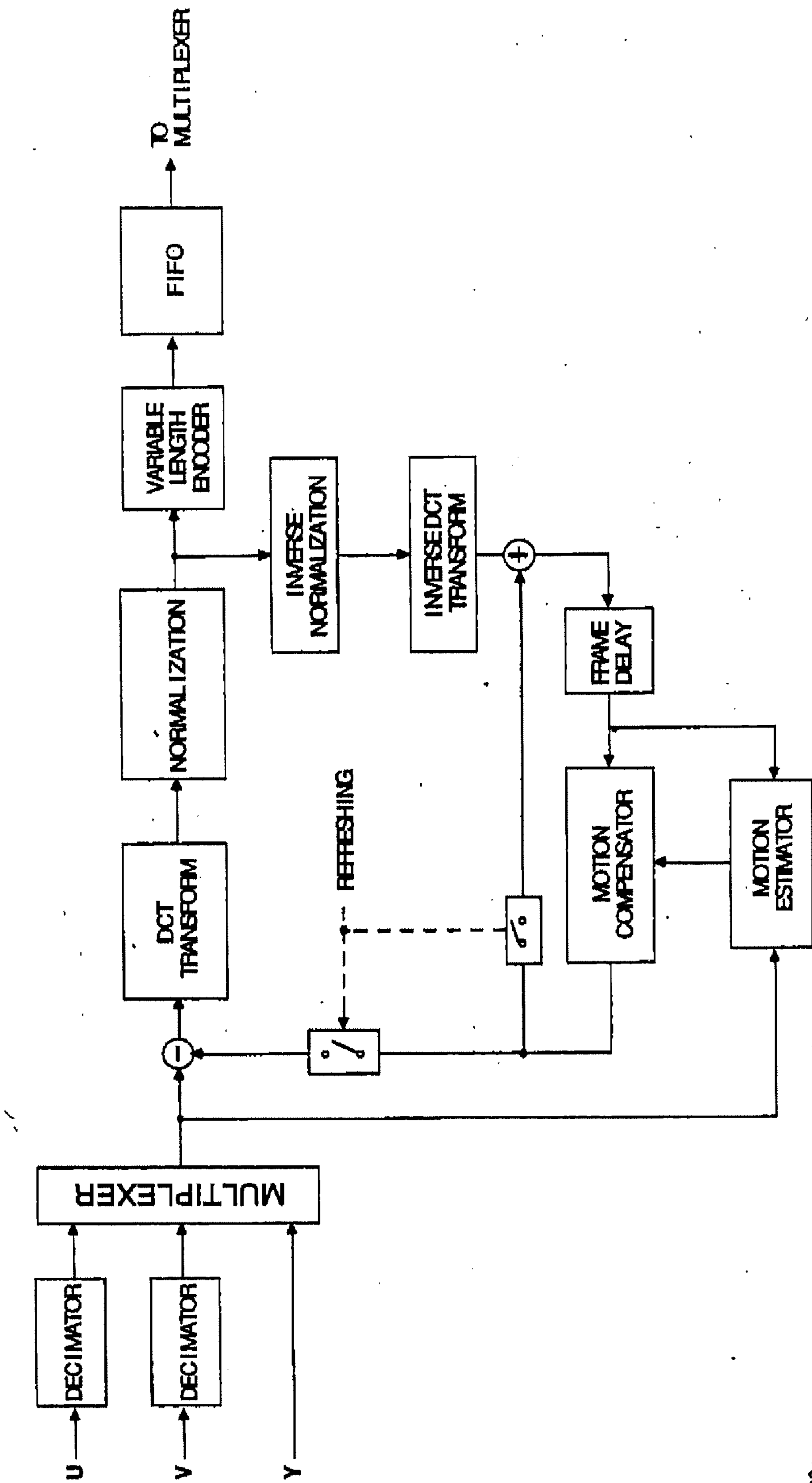

Figure 2-4. Digital Video Encoder Block Diagram

240-20a

Exhibit C Page 126

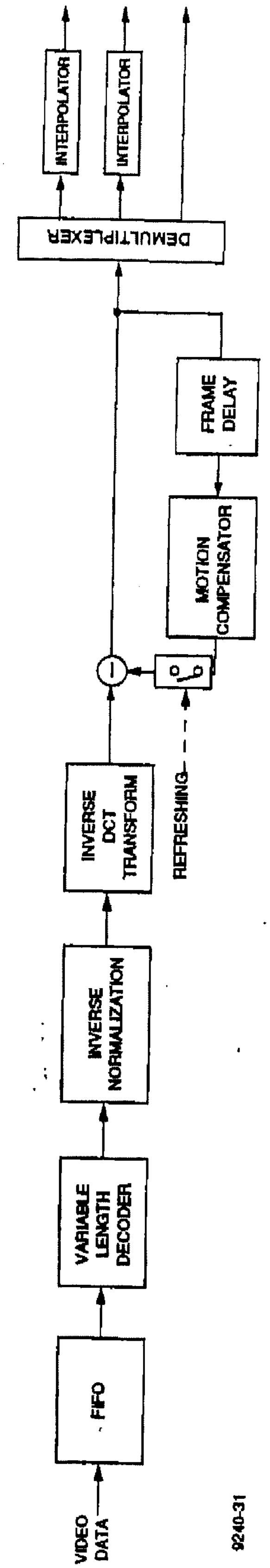

Figure 2-5. Digital Video Decoder Block Diagram

processing. At the decoder, the components are again separated and the chrominance signals are interpolated back to full resolution.

2.2.2 Discrete Cosine Transform

The Discrete Cosine Transform (DCT) transforms a block of pixels into a new block of transform coefficients. A block size of 8 x 8 has been chosen because the efficiency of the transform coding doesn't improve much while the complexity grow substantially beyond the 8 x 8 block size. The transform is applied in turn to each such block until the entire image has been transformed. At the decoder, the inverse transformation is applied to recover the original image.

If $f(i,j)$ represents pixel intensity as a function of horizontal position j and vertical position i, and $F(u,v)$ represents the value of each coefficient after transformation, then the equations for the forward and inverse transformations are

$$F(u,v)=\frac{4C(u)C(v)}{N^2}\sum_{i=0}^{N-1}\sum_{j=0}^{N-1} f(i,j)\cos\frac{(2i+1)u\pi}{2N}\cos\frac{(2j+1)v\pi}{2N}$$

$$f(i,j)=\sum_{u=0}^{N-1}\sum_{v=0}^{N-1} C(u)C(v)F(u,v)\cos\frac{(2i+1)u\pi}{2N}\cos\frac{(2j+1)v\pi}{2N}$$

$$\text{where}\quad C(w)=\begin{cases}1/\sqrt{2} & \text{for } w=0\\ 1 & \text{for } w=1,2,\cdots,N-1\end{cases}$$

where N is the horizontal and vertical dimension of the block.

The DCT merely transforms an image area from a fixed number of pixels to an equal number of transform coefficients. In order to compress the image, it is necessary to take advantage of an important property of the DCT. For typical images, a very large proportion of the signal energy is compacted into a small number of transform coefficients. Consider, for instance, the block of 8 x 8 pixels shown in Figure 2-6. This block is transformed by the DCT into the set of coefficients shown in Figure 2-7. Notice that the first coefficient is significantly larger in magnitude than the others. This particular coefficient is, by definition, twice the average of the 64 pixels. It represents the DC energy of the entire block. The remaining coefficients represent energy levels at increasing horizontal frequencies proceeding from left to right, and at increasing vertical frequencies proceeding from top to bottom. The coefficients at the bottom right corner represent energy levels at high diagonal frequencies. Generally these coefficients tend to be small since images rarely contain significant amounts of diagonal information.

There are instances when the DCT is not effective in compacting the energy into a small number of coefficients. For example, if the input signal was white noise, then the image would be no less random after transformation than it was in the pixel domain. Under such conditions, the image becomes much more difficult to compress, and in fact, cannot be compressed without introducing artifacts of some form or other. Fortunately, under such conditions, artifacts tend to be much less conspicuous than they would be under more quiet conditions. Also, such conditions are not typical of television video. Generally a high degree of horizontal and vertical correlation exists among adjacent pixels. In the next six sections, the procedure for reducing the number of bits required to represent the DCT coefficients and the effect on the appearance of the image is described.

139.	144.	149.	153.	155.	155.	155.	155.
144.	151.	153.	156.	159.	156.	156.	156.
150.	155.	160.	163.	158.	156.	156.	156.
159.	161.	162.	160.	160.	159.	159.	159.
159.	160.	161.	162.	162.	155.	155.	155.
161.	161.	161.	161.	160.	157.	157.	157.
162.	162.	161.	163.	162.	157.	157.	157.
162.	162.	161.	161.	163.	158.	158.	158.

9240-32

Figure 2-6. Original 8 x 8 Block

Exhibit C Page 129

314.91	-0.26	-3.02	-1.30	0.53	-0.42	-0.68	0.33
-5.65	-4.37	-1.56	-0.79	-0.71	-0.02	0.11	-0.30
-2.74	-2.32	-0.39	0.38	0.05	-0.24	-0.14	-0.02
-1.77	-0.48	0.06	0.3[illegible]	0.22	-0.02	-0.01	0.08
-0.16	-0.21	0.37	0.39	-0.03	-0.17	0.15	0.32
0.44	-0.05	0.41	-0.09	-0.19	0.37	0.26	-0.25
-0.32	-0.09	-0.08	-0.37	-0.12	0.43	0.27	-0.19
-0.65	0.39	-0.94	-0.46	0.47	0.30	-0.14	-0.11

9240-33

Figure 2-7. DCT of 8 x 8 Block

2.2.3 Coefficient Quantization (Normalization)

Coefficient quantization, or normalization, is a process that introduces small changes into the image in order to improve coding efficiency. This is done by truncating the DCT coefficients to a fixed number of bits. The truncation is performed by shifting a coefficient from left to right, spilling the least significant bits off the end of its register. In this way, the amplitude of the coefficient is also reduced. The number of bits remaining are pre-assigned individually for each of the 8 x 8 coefficients. However, the number of bits can be further reduced or increased as necessary to maintain a constant bit rate.

In the best case, the encoder will allocate 9 bits (not including sign bit) for each of the transform coefficients. At such times, the system is operating at maximum level on a performance scale ranging from 0 to 9 (the "quantization level"). If the targeted bit rate is exceeded, however, the quantization level is decremented to 8 before encoding the next block. At this level, the low frequency coefficients continue to be represented with 9 bits; however, some of the higher frequency coefficients are now truncated to 8 bits. If the quantization level is then decremented to 7, then those coefficients that were previously truncated to 8 bits will then be truncated to only 7 bits. In addition, a few of the coefficients that were previously 9 bits will now be truncated using 8 bits.

Table 2-1 is used to determined the number of bits assigned to each coefficient of an 8 x 8 block as a function of the quantization level. If the number in Table 2-1 corresponding to a specific coefficient is n, then the number of bits will remain at 9 until the quantization level becomes less than n, after which the number of bits will begin decrementing. In other words, the number of bits is always the minimum of 9 and $9-n+qlevel$.

Table 2-1. Table for Determining the Number of Bits to Allocate for Each of 8 x 8 DCT Coefficients

2	3	4	5	6	7	8	9
3	4	5	6	7	8	9	9
4	5	6	7	8	9	9	9
5	6	7	8	9	9	9	9
6	7	8	9	9	9	9	9
7	8	9	9	9	9	9	9
8	9	9	9	9	9	9	9
9	9	9	9	9	9	9	9

The effect of quantization on the image can be demonstrated using the previous example. In Figure 2-8, the DCT coefficients of Figure 2-7 have been quantized to the nearest integer. The reconstruction in Figure 2-9 was obtained by next applying the inverse DCT transform. The differences between this and the original pixels (Figure 2-6) are shown in Figure 2-10.

The subjective effect of excessively quantizing the DCT coefficients is evident, not so much within the block, but when the image is viewed as a whole. The most objectionable artifact almost always tends to be the blocking effect that arises due to the individual processing of each block. The system has been designed to prevent such artifacts from being visible at normal viewing distances (3 x picture height) and only occasionally visible at very close viewing distances.

315.00	0.00	-3.00	-1.00	1.00	0.00	-1.00	0.00
-6.00	-4.00	-2.00	-1.00	-1.00	0.00	0.00	0.00
-3.00	-2.00	0.00	0.00	0.00	0.00	0.00	0.00
-2.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00
0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00
0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00
0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00
-1.00	0.00	-1.00	0.00	0.00	0.00	0.00	0.00

9240-34

Figure 2-8. Quantized DCT Coefficients

Exhibit C Page 132

139.	145.	150.	154.	154.	153.	154.	153.
145.	150.	154.	157.	157.	155.	156.	156.
150.	155.	158.	161.	160.	157.	157.	155.
159.	161.	161.	163.	161.	158.	159.	158.
159.	160.	161.	163.	161.	157.	156.	155.
163.	162.	160.	162.	161.	157.	157.	158.
162.	161.	159.	162.	161.	157.	157.	157.
164.	162.	160.	163.	162.	158.	159.	160.

9240-35

Figure 2-9. Reconstructed 8 x 8 Block

0.	-1.	-1.	-1.	1.	2.	1.	2.
-1.	1.	-1.	-1.	2.	1.	0.	0.
0.	0.	2.	2.	-2.	-1.	-1.	1.
0.	0.	1.	-3.	-1.	1.	0.	1.
0.	0.	0.	-1.	1.	-2.	-1.	0.
-2.	-1.	1.	-1.	-1.	0.	0.	-1.
0.	1.	2.	1.	1.	0.	0.	0.
-2.	0.	1.	-2.	1.	0.	-1.	-2.

9240-36

Figure 2-10. Error (Original-Reconstructed)

Exhibit C Page 134

2.2.4 Huffman Coding

Normalization improves the compressibility of an image by reducing the amplitude of the transform coefficients. In order to take advantage of the result, an algorithm for assigning a variable number of bits to these coefficients is required. At this stage, a statistical coding technique is used, which unlike the normalization process, is information preserving, and therefore, does not degrade the image.

Huffman coding is an optimum statistical coding procedure capable of approaching the theoretical entropy limit, given a priori knowledge of the probability of all possible events. The encoder can generate such probability distributions and send them to the decoder prior to the transmission of a given frame. This table is then used to derive Huffman code words where relatively short code words are assigned to events with the highest probability of occurrence. The decoder maintains an identical code book and is able to match each code word with the actual event.

In order to apply Huffman coding for this application, the 8 x 8 DCT coefficients are serialized into a sequence of 64, and "amplitude/runlength" coded. Scanning the sequence of 64, an event is defined to occur each time a coefficient is encountered with an amplitude not equal to zero. A code word is then assigned indicating the amplitude of the coefficient and the number of zeros preceding it (runlength). Table 2-2 shows the length of each code word in bits. It does not include the sign bit which must be also included with each code word.

When the coefficient amplitude is greater than 16 or the number of preceding zeros is more than 15, a special code word is used to tell the decoder not to use the code book to interpret the bits that follow. Instead, the runlength is sent uncoded. The coefficient amplitude is also sent uncoded with the number of bits determined by the normalization process described previously. In addition, it is sometimes more efficient to directly code the amplitude and runlength even if it can be coded through the use of the two-dimensional table. The encoder detects these occasions and will switch to direct coding if necessary to shorten the length of the code word. A special code word is also reserved to indicate the end of a block. It is always inserted after the last non-zero coefficient.

The efficiency of this coding process is heavily dependent on the order in which the coefficients are scanned. By scanning from high amplitude to low amplitude, it is possible to reduce the number of runs of zero coefficients typically to a single long run at the end of the block. As defined above, any long run at the end of the block would be represented efficiently by the "end of block" code word.

2.2.5 Motion Compensation

There is a limit to the amount of compression possible by spatial processing alone. An interframe coder, however, can benefit from temporal correlation as well as spatial correlation. A very high degree of temporal correlation exists whenever there is little movement from one frame to the next. Even if there is movement, high temporal correlation may still exist depending on the spatial characteristics of the image. If there is little spatial detail, then frame-to-frame correlation remains high even at high velocities. If the image is highly detailed,

Exhibit C Page 135

Table 2-2. Number of Bits Used for Each code word of Two-Dimensional Huffman code book

	AMPLITUDE															
RUNLENGTH	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
0	2	3	5	5	6	7	8	8	9	9	9	10	10	11	11	11
1	4	5	7	8	9	10	10	11	12	12	13	14	14	15	15	16
2	4	7	8	10	11	12	13	14	15	16	16	16	18	18	19	19
3	5	8	10	11	13	14	15	16	17	18	18	19	19	19	21	21
4	6	9	12	14	15	17	18	18	20	21	20	22	28	29	29	29
5	7	10	13	16	18	19	22	21	21	29	29	29	29	29	29	29
6	7	11	14	17	18	19	19	17	20	21	28	28	28	28	28	28
7	8	12	16	18	19	22	20	28	28	28	28	28	28	28	28	28
8	9	14	17	21	28	28	28	28	28	28	28	28	28	28	28	28
9	9	15	19	28	28	28	28	28	28	28	28	28	28	28	28	28
10	10	16	20	28	28	28	28	28	28	28	28	28	28	28	28	28
11	11	18	28	22	28	28	28	28	28	28	28	28	28	28	28	28
12	11	17	28	22	28	28	28	28	28	28	28	28	28	28	28	28
13	11	17	28	22	28	28	28	28	28	28	28	28	28	28	28	28
14	12	20	28	22	28	28	28	28	28	28	28	28	28	28	28	28
15	13	20	28	22	28	28	28	28	28	28	28	28	28	28	28	28

however, and contains high spatial frequencies, then even slight displacements of one pixel or less can significantly reduce the amount of correlation that exists.

In the DigiCipher™ system, we compress the signal by first predicting how the next frame will appear and then sending the difference between the prediction and the actual image. A reasonable predictor is simply the previous frame. This sort of temporal differential encoding (DPCM) will perform very well if little movement occurs or if there is little spatial detail. At other times, it will be less effective and occasionally worse than if the next frame had simply been encoded without prediction (PCM).

Motion compensation is a means of improving the performance of any temporal compression scheme when movement occurs. In order to apply motion compensation, it is first necessary to determine what has moved since the previous frame and where it has moved to. If this information is known at the decoder site, then the previous frame can be shifted or displaced in order to obtain a more accurate prediction of the next frame that has yet to be transmitted. The encoder would reproduce the same prediction as the decoder and then determine the difference between the prediction and the actual image. If the movements match the model used to estimate motion and if the motion estimates are accurate and the signal is free of noise, then this error signal would, in fact, be zero.

Exhibit C Page 136

Displacement of the previous frame can be performed on a frame, partial frame, or pixel basis. That is, a unique displacement (motion vector) could be generated for every frame, part of a frame, or pixel respectively. The usefulness of generating a single motion vector per frame, however, is limited since it can only model simple panning of the entire image. Ideally, a unique motion vector would be generated for each pixel. However, since motion estimation is a complex process and requires knowledge of the next frame, it can only be performed at the encoder, and the overhead involved in making this per-pixel motion information available to the decoder would be excessive. Therefore, the motion estimation is performed on a partial frame basis with the area of the portion chosen to equal a superblock. The superblock has a horizontal dimension equal to 4 DCT blocks and a vertical dimension equal to 2 DCT blocks. This sizing is compatible with the 4 times horizontal subsampling and 2 times vertical subsampling of the chrominance components, thus allowing the same motion vector to be used to displace a single chrominance DCT block.

The process of displacing portions of the previous frame in order to better predict the next frame is illustrated in Figure 2-11.

2.2.6 Motion Estimation

Before describing the integration of motion compensation with the overall compression process, an explanation of how motion vectors are derived is necessary.

Motion estimation algorithms can be divided into two classes - those which focus primarily on extracting three dimensional motion parameters from a sequence of two dimensional images or projections, and those which estimate velocities on a point-by-point or region-by-region basis without any consideration of how the object(s) is (are) moving as a whole. In the first case, a common formulation of the problem is to assume rigid three-dimensional bodies with movement patterns that can be described by a translation component, a rotation component, a zooming component, and a center about which the rotation or zooming is occurring. It is clear, however, that typical television imagery is far too complex to be satisfactorily characterized by this simple motion description. Therefore, we restrict ourselves to the second class of algorithms. Unlike the first class, these algorithms are unintelligent; they have no understanding of the higher level motion events that are occurring. These algorithms are usually based on translational models. However, if the region over which each estimation is performed is small, then more complex (large area) movements can be modeled satisfactorily. The selected method determines a good match between superblock regions in the current and the previous frames. The overhead required to send a single motion vector to the decoder is 9 bits per superblock (approximately 0.018 bits/pixel).

2.2.7 Motion Picture Processing

Almost all movies developed for the cinema and a significant amount of program material developed for television are initially shot on film. Except for a few special cases, the display rate used for film is 24 frames/second, and therefore the motion rendition is degraded in comparison to normal television video. Eventually, when this program material is converted to the NTSC or HDTV television standard, a process called three-two pulldown is used. As shown in Figure

Exhibit C Page 137

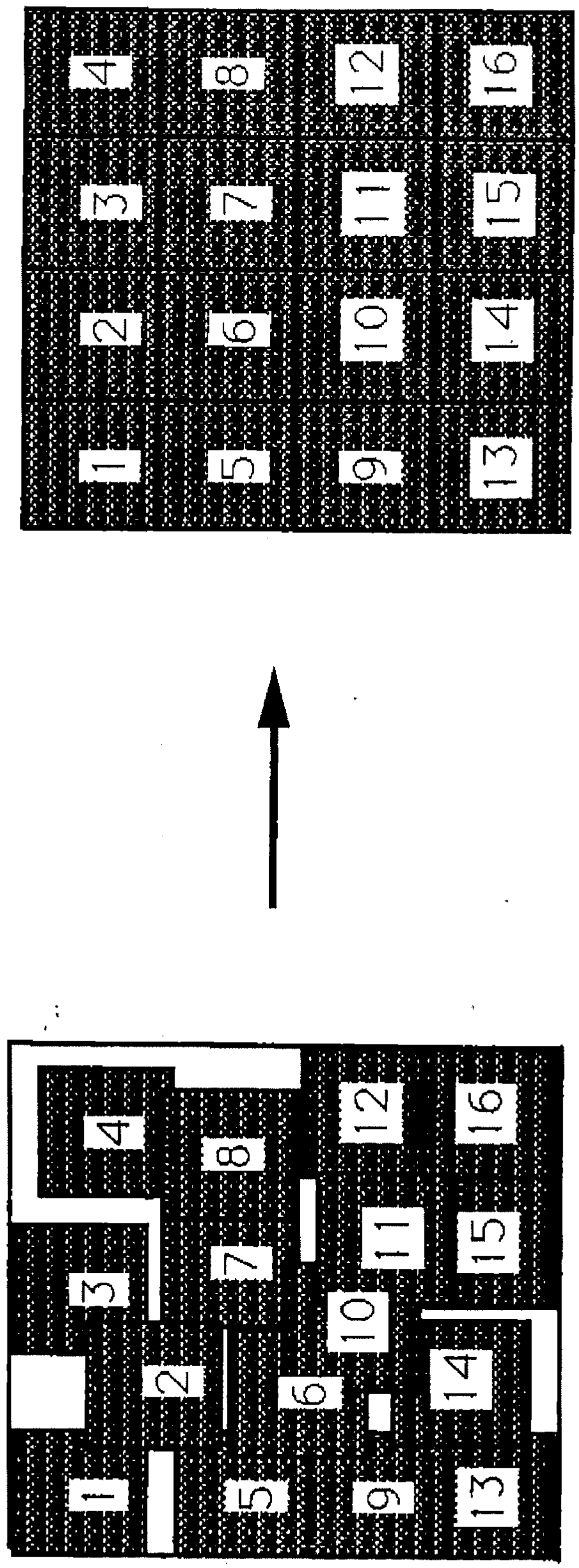

Figure 2-11. Using Motion Compensation to Predict Next Frame

2-12, it involves alternating between three repetitions and two repetitions of each frame of the film.

Since the three-two pulldown process increases the number of video frames from 24 to 30 without increasing the amount of information in the signal, the first step in the source coding process is to restore the signal to its original state. Since one of every five fields is redundant, it can either be discarded, or averaged with the other identical field as shown in Figure 2-12. In the latter case, a 3 dB noise reduction results; however, its significance is questionable, since it will benefit only one of every four fields.

After the 24 frame/second signal is reconstructed at the decoder, it must be converted back to 60 fields/second before it can be displayed. This is easily accomplished by applying the three-two pulldown process once again.

The DigiCipher™ System processes material shot on film in this manner to further improve the performance. Other video source material is, of course, handled without going through this process.

2.2.8 Rate Buffer Control

Each single channel video processing section in the encoder requires a rate buffer in order to match the variable rate of the Huffman-coded data to the fixed output rate necessary for channel transmission. This rate buffer is implemented as a one frame FIFO located after the Huffman encoder. The total storage size is large enough to handle variations of plus and minus one field.

In order to prevent the video output buffer FIFO from overflowing or underflowing, the FIFO input rate must be continuously adjusted. This is the purpose of the multi-quantization level coding structure. As the quantization level is decremented, quantization is increased, blocks are shortened, and an increase in the FIFO input block rate results. As the quantization level is incremented to a maximum level of 9, finer quantization results in longer blocks, and a reduced FIFO input block rate. This adjustment has the required effect of keeping the bit rate into the FIFO relatively constant. The status of the buffer is continuously monitored, and as long as the number of stored blocks remains within a predetermined window, the quantization level will remain unchanged. If the buffer level drops below the lower threshold or rises above the higher threshold, then the quantization level will decrement or increment respectively. Fill bits are inserted into the channel in order to prevent underflows during the transmission of very simple images.

2.3 Digital Audio Processing

Exhibit C Page 139

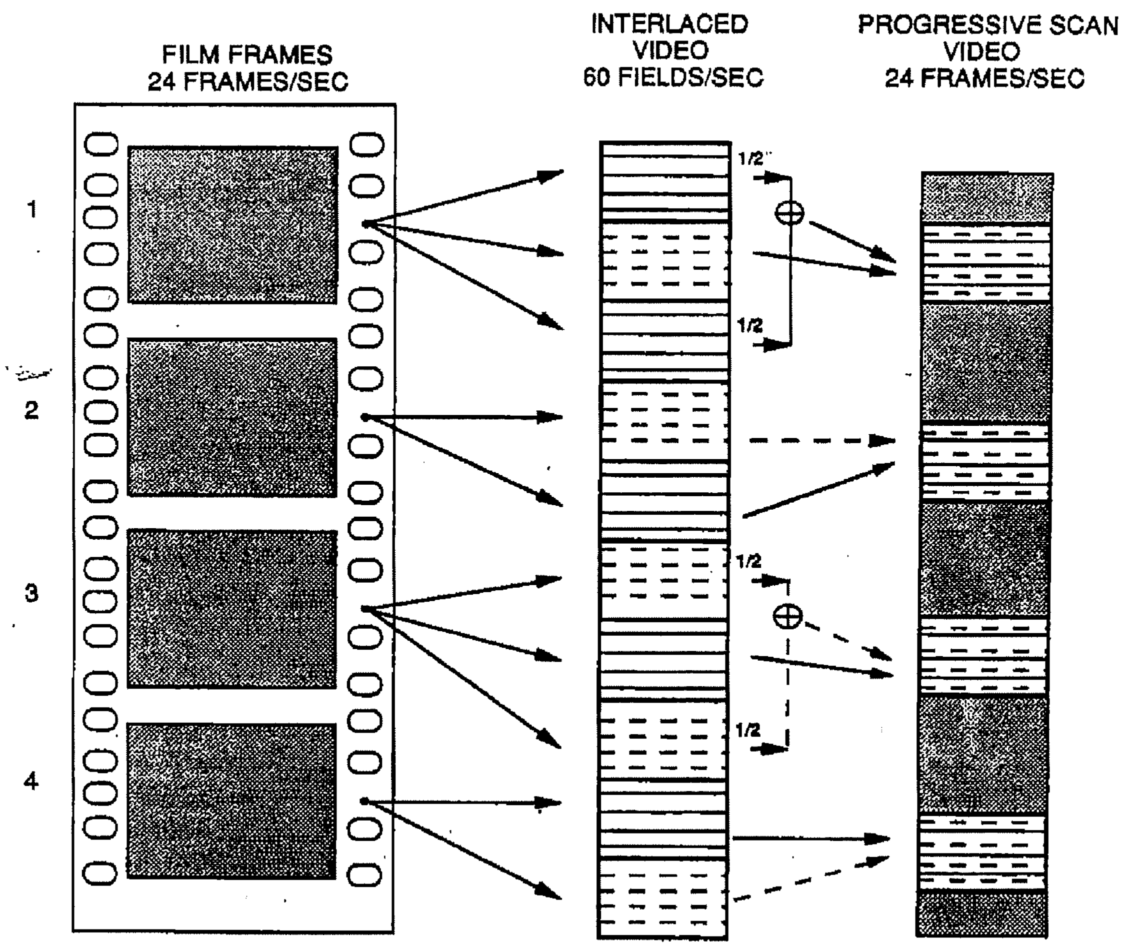

Figure 2-12. Conversion of Film to Interlaced Video and Restoration of Progressive Scan

Exhibit C Page 140

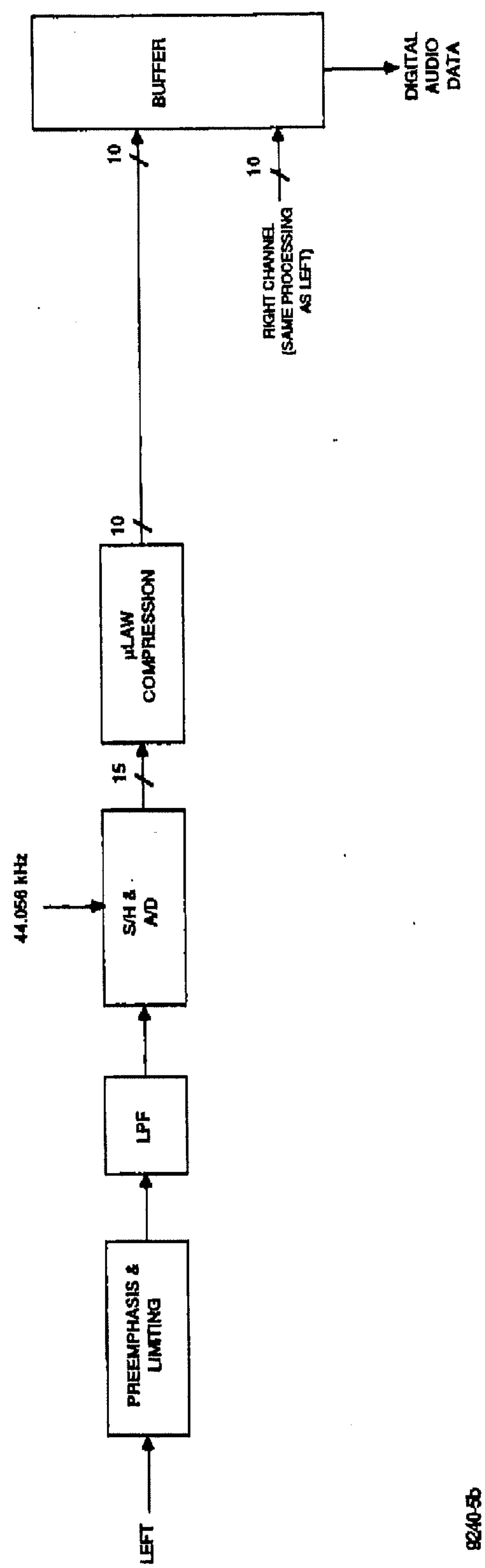

Figure 2-13. Digital Audio Encoder

2.3.1 Digital Audio Encoding

Figure 2-13 shows the digital audio processing at the encoder. The system uses the emphasis characteristics specified by the EIAJ standard for PCM VTR Adapters *(T_1 = 50 µsec; T_2 = 15 µsec.)* Digital audio sample pairs are processed into compressed floating point notation, as used in the VideoCipher™II Plus system. A 15 to 10 bit instantaneous µ-law compression technique was selected, which produces 1 sign bit, 3 exponent bits, and 6 mantissa bits per audio sample.

The sampling rate is 44.056 kHz, which is also common to the VideoCipher™II Plus system.

Error correction for all DigiCipher™ transmitted bits is performed in a separate decoder block, and is *not* integrated with the audio processing. Detected eerors can be used to mute the audio. Furthermore, error concealment is not used since the bit error rate performance curve is very steep.

Overall, audio compression produces a 10 bit coded representation of each audio sample. Given the 44.056 kHz sampling rate, 56 audio bits must be transmitted per line time to support 4 audio channels.

2.3.2 Digital Audio Decoding

Figure 2-14 shows the digital audio processing at the decoder. Received audio bits are decrypted along with the received video bits. After buffering and µ-law expansion, precision audio reconstruction at the decoder is accomplished with a switched capacitor filter followed by analog deemphasis and a 18 kHz lowpass filter.

2.4 Data Channel Processing

The DigiCipher™ transmission format has 4 bits per line time assigned to data channel capacity. This allocation of 125.87 kbps is sufficient capacity for 13 9600 baud data streams. The initial DigiCipher™ design will support 4 such data streams, with the remaining capacity reserved.

2.5 Control Channel Processing

The DigiCipher™ transmission format has 4 bits per line time, which amounts to 125.87 kbps. There is great flexibility allowed in message mixing within the control channel on a bit stream.

2.6 Data Multiplex Format

This section defines the data multiplex for video, audio, data, text, and control channel.

Prior to forward error coding at the encoder, each video line time includes 504 information bits. Figure 2-15 shows the data transmission format.

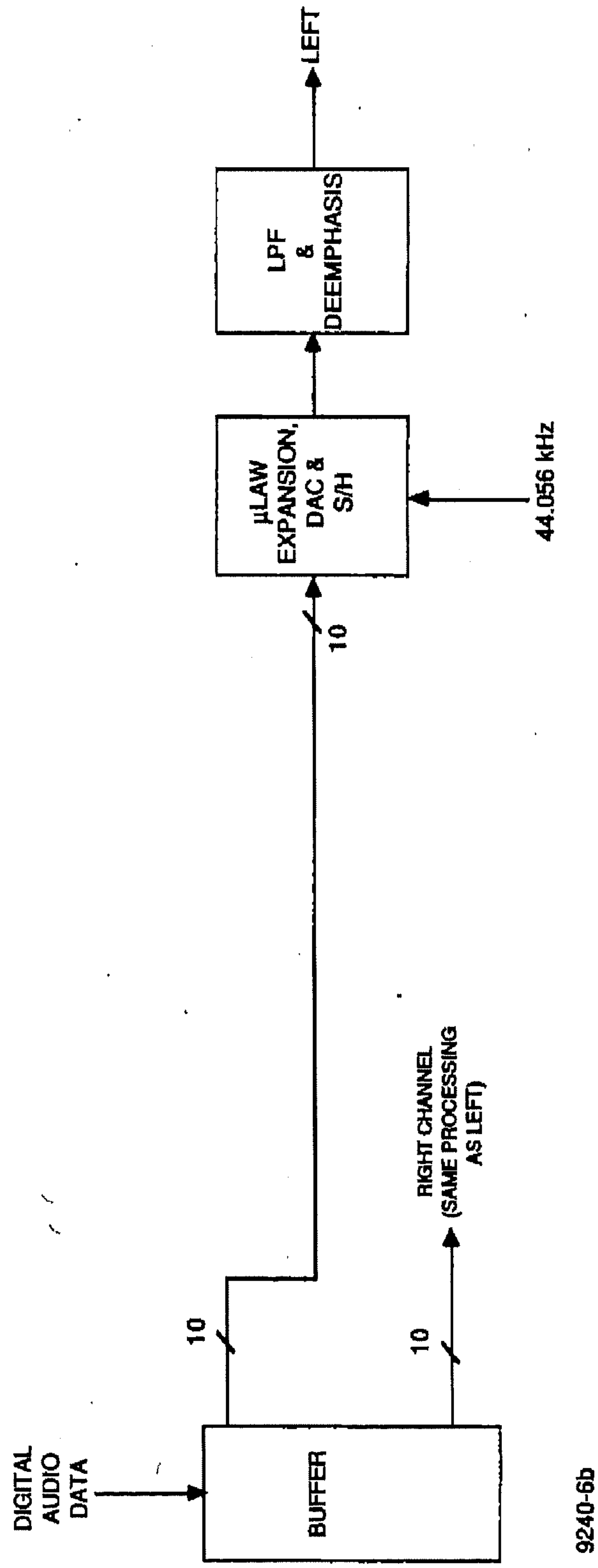

Figure 2-14. Digital Audio Decoder

Exhibit C Page 143

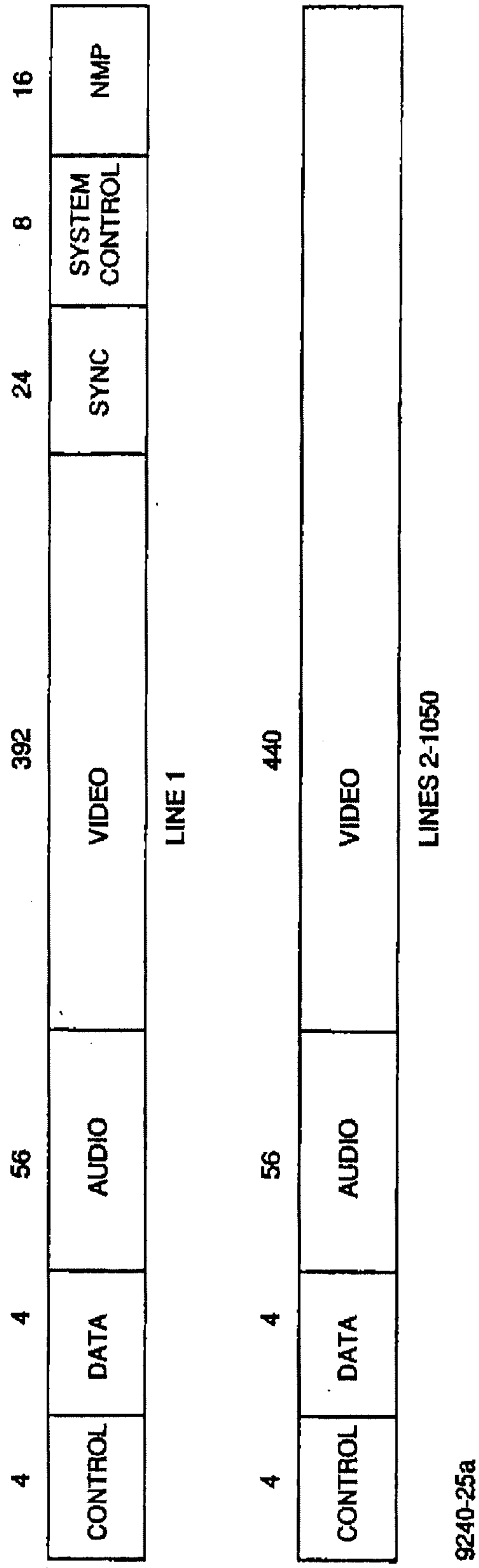

Figure 2-15. Data Multiplex Format

Exhibit C Page 144

Each line has a fixed allocation of 4 control channel bits followed by 4 allocated data channel bit positions. The next 56 bits represent audio data. The remainder of the 504 bits are dedicated to video for lines 2 through 1050. Line 1 differs in that the 48 bit positions 457 through 504 represent 3 specific fields, as listed below.

- 457 through 480: The 24-bit **sync pattern** provides frame synchronization for the decoder. All the decoder timing signals are derived from this per-frame sync.

- 481 through 488: The **system control** field contains LSBits of the nominal frame count number as well as other system control information.

- 489 through 504: The **Next Macroblock Position** (NMP) field indicates the number of video multiplex data bits from the end of the NMP field to the beginning of the next macro block. This 15 bit parameter is necessary to support the acquisition process. Additionally, if errors cause a loss of sync within the decoder, the unit can always recover through a reacquisition. Note that there is a 16th bit in the field, reserved for future use.

2.7 Digital Transmission

Modulation and channel coding are key elements of the DigiCipher™ system. The modulation technique must be efficient in order to send the required number of information bits through a single VHF or UHF channel. The channel coding technique must be powerful in order to maintain a very low error rate; the more compression (source coding), the more serious the effect of a single error.

Figure 2-16 shows the basic communication system blocks, including coding, modulation, pulse shaping (transmit filtering), receive filtering, demodulation, tracking, and decoding.

The modulation selected for digital transmission over the VHF or UHF channel is 16-QAM at 4.86 Msps. The 16-QAM provides two times the data rate with moderate penalty (approximately 5 dB) in power compared to QPSK. Figure 2-17 shows the BER performance of uncoded 16-QAM.

The implementation of the modulator and the demodulator for the 16-QAM is similar to the QPSK design, except that the I channel and the Q channel have 4-level signals present, rather than two-level signals.

Adaptive equalization is employed to handle the reflections (multipath) found typical VHF or UHF reception. Forward error correction using Reed Solomon coding of rate 130/154 (t=12) is used to correct transmission errors caused by noise and/or interference. The system threshold is 19 dB C/N including 2.5 dB of implementation margin. At 19 dB C/N there will be one undetected error event per day. The threshold C/N is much lower than C/N required for satisfactory reception of analog VHF/UHF signals. Proper signal filtering (a 15% roll-off raised cosine, for example) will be provided to prevent adjacent channel interference (See Figure 2-18).

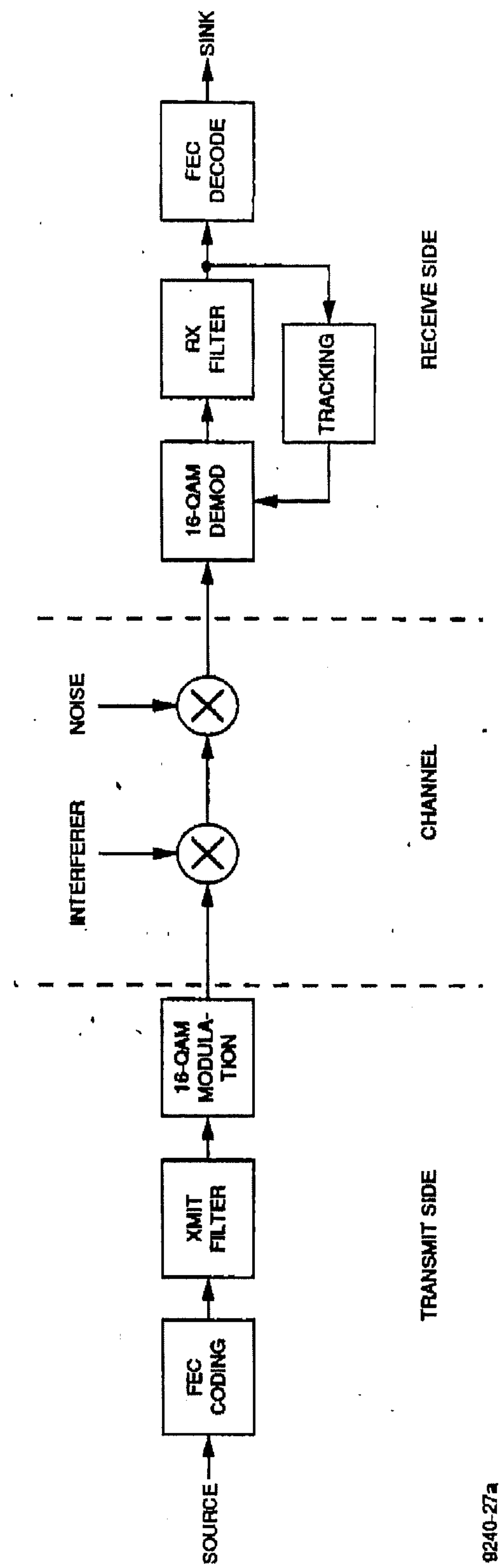

Figure 2-16. Communication System Blocks

Exhibit C Page 146

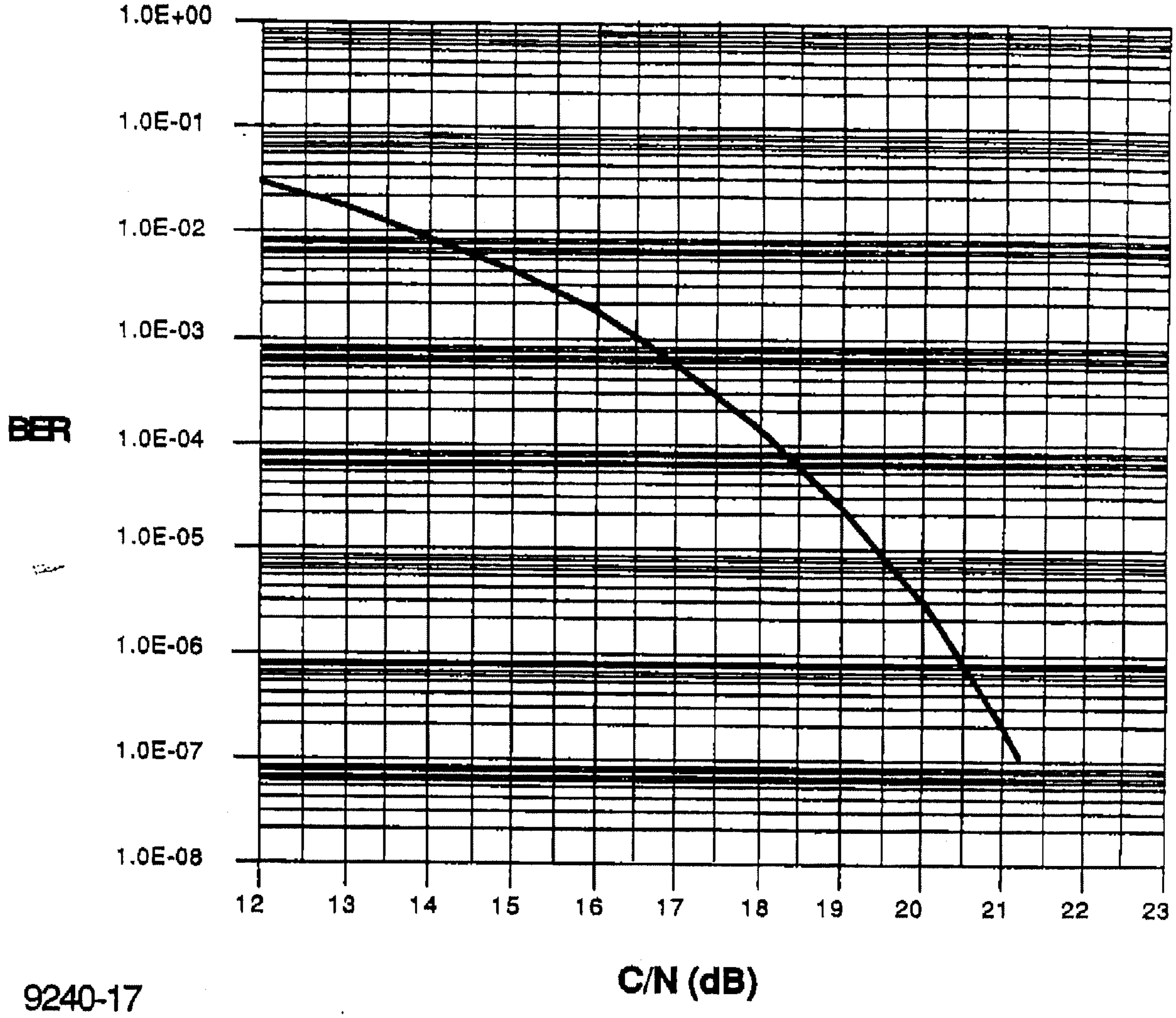

Figure 2-17. BER Performance of 16-QAM

Exhibit C Page 147

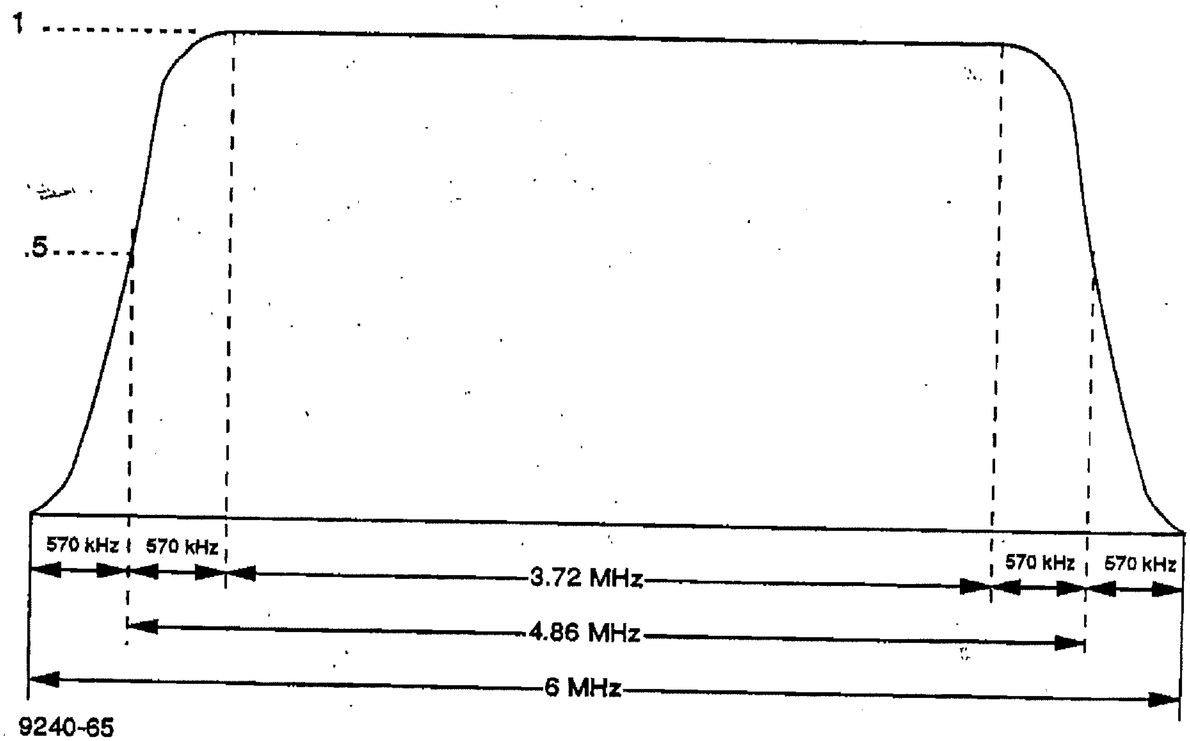

Figure 2-18. IF Output Spectrum

Exhibit C Page 148

2.8 Synchronization

2.8.1 Frame Synchronization

Once clock synchronization is achieved, frame synchronization is achieved by using the twenty-four bit sync sequence transmitted every frame (33.4 msec). The sync detection algorithm is similar to the one used in VideoCipher II Plus for detecting its twenty-four bit frame sync sequence. The overall acquisition time of the decoder through to the start of the video decompression process should be less than 100 msec after a channel change.

2.8.2 Digital Video Acquisition and Recovery

When a decoder tunes to a channel, its video data FIFO must be flushed. Once 24 bit frame sync is established, the decoder must monitor the macroblock stream for the FIRST macroblock. Once identified, data can be stored in the FIFO, and subsequent sections of the decoder can be notified as to the early or late status. All decoder processing from the Huffman decoder output side through to the filtered decompressed output video has a static timing alignment to the 24 bit pattern location. This alignment is computed based on nominal FIFO operation (half full), plus all the processing delays and buffering from decoder input to output. Likely alignment values are between 0.7 and 0.9 frames, as measured from the 24 bit pattern received *before* a late FIRST macroblock, or from the 24 bit pattern received *after* an early FIRST macroblock. Although processing can commence based on this static alignment, the video output will remain blanked for a full second more, while the decoder builds up the PCM blocks received during the one second long interval.

Uncorrected error events, although rare, must be recoverable. That is, the decoder may lose the picture momentarily, but not indefinitely. Currently, a decoder can detect an error in two ways. First, the Reed-Solomon decoder outputs a flag that is a robust, but not a guaranteed, error detector. Secondly, the Huffman decoder counts pixel flow, and expects an end-of-block code word for pixel 64 of every block. If this code word is not detected, or is detected early, an error has occurred.

2.9 Summary of DigiCipher™ System Parameters

Key parameters of the DigiCipher™ HDTV System are listed in Table 2-3. The sampling frequency has been chosen to optimize the video performance as well as to reduce the hardware complexity of the decoder. The video data includes both luminance and chrominance data as well as various overhead information.

Table 2-3. DigiCipher™ System Parameters

Parameters	Value
VIDEO	
Aspect Ratio	16:9
Raster Format	1050/2:1 Interlaced
Frame Rate	29.97 Hz
Bandwidth	
Luminance	22 MHz
Chrominance	5.5 MHz
Horizontal Resolution	
Static	660 Lines per Picture Height
Dynamic	660 Lines per Picture Height
Horizontal Line Time	
Active	27.18 μsec
Blanking	4.63 μsec
Sampling Frequency	51.8 MHz
Active Pixels	
Luminance	960(V) x 1408(H)
Chrominance	480(V) x 352(H)
AUDIO	
Bandwidth	15 kHz
Sampling Frequency	44.05 kHz
Dynamic Range	85 dB
DATA	
Video Data	13.83 Mbps
Audio Data	1.76 Mbps
Async Data and Text	126 Kbps
Control Data	126 Kbps
Total Data Rate	15.84 Mbps
TRANSMISSION	
FEC Rate	130/154
Data Transmission Rate	19.43 Mbps
16-QAM Symbol Rate	4.86 MHz

3. Alternate Media Distribution

3.1 Cable Transmission

One of the main features of the DigiCipher™ HDTV System is its potential to provide digital transmission starting at the satellite uplink all the way to the cable subscriber. This cable pass-through approach offers the following advantages:

- Subscribers can enjoy HDTV services free from signal degradations caused by the cable.
- Addressing and authorization for cable subscribers can be inserted centrally although local (cable headed) control would still be provided.
- Subscribers can enjoy the quality of digital audio.

Since the DigiCipher™ HDTV System requires substantially lower C/N, it helps reduce intermodulation problems and the power loading of the cable system. Also, extra channels can be placed above usable channels where VSB-AM signals cannot be placed due to low C/N or FCC emission requirements.

3.2 Satellite Transmission

The DigiCipher™ HDTV System can be transmitted over C-band or Ku-band satellite channels using QPSK modulation.

The system can support both FSS and BSS satellite transponders. A satellite channel can carry two HDTV signals. The threshold C/N is 8 dB measured over a 24 MHz bandwidth, therefore the DigiCipher™ HDTV System allows the use of smaller dish size compared to other analog or hybrid HDTV systems.

3.3 Other Terrestrial Distribution

Since the DigiCipher™ HDTV System is an all-digital system, it can be readily applied to other transmission media such as microwave distribution service (MDS), multi-channel MDS (MMDS) and fiberoptic cables (FO).

An inherent characteristic of the all-digital system is that the HDTV service is free from transmission artifacts caused by various transmission media. Also, the complexity of the interface equipment between various transmission media is substantially lower.

3.4 VCR and Video Disc Recorders

All-digital recording and playback of HDTV signals using the signal format of the DigiCipher™ HDTV System is within the reach of current technology for consumer use since the total data rate is less than 20 Mbps. The cost and performance benefits of digital recording will be significant compared to analog recording.

Appendix A. ATV Interface Questionnaire

A.1 Sync Interfacing between ATTC and Proponents

A.1.1 Encoder Sync. Specification

The input video format is 1050/2:1/59.94. The active position of each line is 27.18 μsec and the blanking portion of each line is 4.63 μsec.

A.1.2 Decoder Sync. Specification

The decoder video and sync signals are compatible with the input video and sync signals.

A.2 RF Interfacing between ATTC and Proponents

A.2.1 Modulator Carrier Frequency

The modulator IF output will be 44 $\pm$ 3 MHz.

A.2.2 Up-Conversion Frequency

The IF output can be upconverted to any VHF or UHF channels. For the VHF test channel 11, a 245 MHz local oscillator will be used. For the UHF test channel 23, a 527 MHz local oscillator will be used.

A.2.3 Receiver Tuner Frequency

The receiver can receive any VHF or UHF channels. For the VHF test channel, a local oscillator of 245 MHz will be utilized and the spectrum inversion will be corrected.

A.2.4 Multi-path Cancellation

The system will have an adaptive equalizer that can cancel complex multipath of up to 2 μsec. The system will also have a single echo cancellation equalizer for a long multipath of up to a line time (32 μsec).

A.2.5 Signal Peak Power

The peak power will be 9 dB higher than the average power.

A.2.6 System Carrier Frequencies

The system does not employ any subcarriers.

A.2.7 UHF Taboos

Since the system requires substantially lower C/N and lower peak power, it is ideally suited for use of UHF taboo channels.

A.2.8 Time Sharing of Augmentation Channel

Not applicable.

A.2.9 Augmentation Channel Energy Density

Not applicable.

A.2.10 Maximum Energy Density Image

The energy of the transmitted signal is independent of the image.

A.2.11 Digital Input Port

The system can accept four 9600 baud data inputs.

Appendix B. References

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5. D. Hatfield, "Report on the Potential for Extreme Bandwidth Compression of Digitalized HDTV Signals," Hatfield Associates, Inc., March 20, 1989.

6. R. C. Reininger and J. D. Gibson, "Distribution of Two-Dimensional DCT Coefficients for Images, "IEEE Trans. on Comm. Vol. COM-31, No. 6, June 1983.

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8. M. Barbero, R. Del Pero, M. Muratori and M. Stroppiana," Bit-Rate Reduction Techniques Based on DCT for HDTV Transmission," IEEE Globecom Proceedings, 343.2.2 - 343.3.5, 1990.

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