PROGRAM ON APPLICATION OF COMMUNICATIONS SATELLITES
TO EDUCATIONAL DEVELOPMENT

WASHINGTON UNIVERSITY

Report No. T-71/2 May, 1971

STILL-PICTURE TELEVISION (SPTV) TRANSMISSION

Gulab Sharma

WASHINGTON UNIVERSITY / SAINT LOUIS / MISSOURI 63130

PROGRAM ON APPLICATION OF COMMUNICATIONS SATELLITES
TO EDUCATIONAL DEVELOPMENT

WASHINGTON UNIVERSITY

Report No T-71/2 May, 1971

STILL-PICTURE TELEVISION (SPTV) TRANSMISSION

Gulab Sharma

This research is supported by the National Aeronautics and Space Administration under Grant No Y/NGL-26-008-054 and it does not necessarily represent the views of either the research team as a whole or NASA

WASHINGTON UNIVERSITY
SEVER INSTITUTE OF TECHNOLOGY

ABSTRACT

STILL-PICTURE TELEVISION TRANSMISSION

by Gulab Sharma

ADVISOR: Professor D.L. Snyder

June, 1971

Saint Louis, Missouri

To produce a diversity of program material in a limited frequency spectrum, various multichannel, continuous-audio still-video, television transmission-systems, compatible to the existing systems, have been suggested and investigated. In this report, we categorize and describe these alternative systems and identify some of the system parameters and constraints. The issues explored are: the number of still picture channels that can be realized in a limited spectrum, the interrelation of various parameters with system constraints, and general system considerations.

TABLE OF CONTENTS

No.		Page
1.	Introduction	1
	1.1 Main Objective and Scope	2
	1.2 Television Broadcast Standards	3
	1.3 System Performance Objectives	4
	1.4 Subjective Picture Quality	7
	1.5 Satellite Power and Picture Quality	11
2.	Still-Picture Television (SPTV) Transmission	17
	2.1 Slow Scan	17
	2.1.1 General System Concept	17
	2.1.2 Effect of Scanning Speed on Camera Tube SNR	21
	2.2 Time-Shared SPTV Transmission	23
	2.2.1 Classification	24
	2.2.2 Time-Shared-Video, Frequency-Shared-Audio	24
	2.2.3 General System Considerations	26
	2.2.3.1 Base Bandwidth Fixed and Equivalent to one Television Channel	30
	2.2.4 Transmission and Receiving	31
	2.2.5 Frame Sequence Synchronization	37
	2.2.6 Frame Repeating System	39
3.	Time-Shared-Video, Time-Shared, Time-Compressed Audio SPTV Transmission	43
	3.1 Relative Merits	44
	3.2 Composite Frame and Time Allocations	45
	3.3 General Transmission and Reception	49
	3.3.1 Synchronization	55
	3.3.2 Audio Compression and Expansion	57
	3.3.3 Video Multiplexing	66
	3.4 General System Considerations	66

TABLE OF CONTENTS
(continued)

No.		Page
3.4.1	Audio-Video Crosstalk.......	68
3.4.2	Relation Between Audio Bandwidth and Number of Channels............	70
3.4.3	Switching Noise with Common Storage Elements..................	71
4.	Summary and Conclusions............	76
5.	Acknowledgements...............................	79
6.	Bibliography..	80
7.	Vita...	84

LIST OF TABLES

No. Page

1.1 Television Broadcast Standards for the U.S.A.... 5

1.2 Subjective Assessment of the Signal to Noise Ratio for Television........................10

LIST OF FIGURES

No.		Page
1.1	Satellite Power Requirements for TASO Picture Grades with VSB/AM at 0.860GHz	13
1.2	Satellite Power Requirements for TASO Picture Grades with Frequency Modulation at 0.860GHz	14
1.3	Satellite Power Requirements for TASO Picture Grades with Frequency Modulation at 12GHz	15
2.1	Block Diagram of a Possible Slow Scan Transreceiver	20
2.2	Timing Diagram for Time-Shared SPTV Video Information	25
2.3	Relation Between the Number of SPTV Channels and Channel Frame Updating Time for a Fixed Video Base Bandwidth of 4.5MHz	32
2.4	Base Bandwidth Configuration of Composite Signal	33
2.5	Block Diagram of a Time-Shared-Video, Frequency-Shared Audio Transmitter	34
2.6	Block Diagram of a Central Receiving Station	35
2.7	A Possible Decoder	38
2.8	Basic Frame Repetition Sequence	40
2.9	Frame Repeat System	40
3.1	Channel Allocations for Time-Sharing of Video and Audio of Each Channel	46
3.2	Plot of Audio Compression Ratio for a Number of Channels	48
3.3	Plot of a Number of Channels vs Television Frame Update Time	50
3.4	Timing Diagram of Time-Shared Video, Time-Shared-Compressed Audio SPTV Transmission	51
3.5	Block Diagram of Transmitter	52
3.6	Block Diagram of Receiver	53
3.7	A Typical Synchronizer	56

LIST OF FIGURES
(continued)

No. Page

3.8 Elementary Audio Compression Expansion Unit.........59

3.9 Transmission Reception with Common Storage Elements...60

3.10 Timing Diagrams for Common Storage Elements Arrangement.......................................61

3.11 Rearrangement of Samples After Storage..............62

3.12 Scanning Pattern for Storage..................... ..65

3.13 Video Multiplexing Assembly.........................67

3.14 Distortion in Time-Compressed Audio Information with Common Storage Elements........................73

STILL-PICTURE TELEVISION TRANSMISSION

1. INTRODUCTION

Multi-channel television transmission with continuous audio and continuous video gives a wide choice of program selection. With a satellite transmission system, where the cost per channel is high, a limited number of channels may be available for these purposes. To have a diversity of program material for such a case, and where motion is not an important factor for the video information, a multi-channel continuous audio still video format can be considered as an alternative to the standard multi-channel, continuous audio-video format. The effectiveness of this format for educational or for any other purposes has yet to be investigated, but some research done (1,2)* is encouraging.

A continuous audio, still video format is called the Still-Picture Format here. The transmission scheme for this, when a standard television receiver is used for display, is called the Still-Picture Television (SPTV) transmission system.

---

*The numbers in parentheses in the text indicate references in the Bibliography.

SPTV transmission through satellite involves the transmission of information needed for Still-Picture Format to a receiving point, which converts this information into the still-picture format compatible to the display receiver. If, for example, a conventional television receiver is used for signal display, then the information received from satellite has to be processed to form a compatible conventional broadcast television format. This processing has to be done either remotely from display equipment, many of which may be connected by cable to the centrally located processor, or processors may be located with a few or each of them, depending upon the various transmission and receiving system considerations.

1.1 MAIN OBJECTIVE AND SCOPE

In the design and construction of any communication system, there are several important factors which must be comsidered; some are: (1) cost, (2) reliability, (3) simplicity and (4) versatility. The main objective of the study reported here is to investigate alternative multi-channel, continuous audio, still-video television transmission systems compatible with existing television transmission systems. This was considered in light of the above and other requirements. The aim of such a system is to produce a diversity of program material in a limited frequency spectrum. The scope of this report is two-fold: (i) to categorize and describe some alternative systems, (ii) to identify some of the system constraints and parameters.

The basic transmission systems are assigned three categories: (i) slow-scan transmission system; (ii) time-shared-video, frequency-shared-audio transmission system; and (iii) time-shared-video with time-shared time-compressed audio transmission system. The system concept for each has been described. Relations between such parameters as video frame updating time, number sub-channels, audio bandwidth, and total bandwidth are derived and plotted. Suggestions for the solutions of various technical problems encountered are made. Each system is considered with a view to making it compatible with the existing conventional television display system. Since the system compatibility to the existing system is one of the important parameters, a brief discussion of existing television broadcast standards is given in the next section.

1.2 TELEVISION BROADCASTING STANDARDS

By television broadcasting standards we mean the picture and transmission standards in use. The United States picture standards define the method by which luminance, chrominance, and synchronization information are formed into a signal suitable for transmission. The transmission standard defines the modulation method and frequency of transmission. A receiving installation must be compatible with both picture and transmission standards of the broadcast being received.

At least twelve different television standards are in use in the world. All of these standards were originally established for monochromatic broadcasting. Later, a number of methods were developed for expanding the monochromatic systems to color systems compatible with existing monochrome broadcast facilities and receivers. This compatibility means that a color receiver can receive monochrome broadcast while a monochrone receiver can receive the color broadcast. The color broadcasting uses the same Radio Frequency allocations previously assigned for monochrome. The existing color methods meet these compatibility requirements by adding a chrominance signal to monochrome luminance signal.

There are three standard systems for color television NTSC, PAL and SECAM. The existing standards either use 405, 525, 625, or 819 lines per television frame. The 525 and 625 line standards are the most important ones. This is because of the total number of receivers in the world and present plans for expansion of television broadcasting services for 525 and 625 line systems. In the United States and Canada, the 525 line system is used.

Table 1.1 shows video and audio signal characteristics of a standard 525 line television broadcasting system. This system uses amplitude modulation with vestigial side-band (AM/VSB). Like most standards, it uses video modulation with negative polarization, i.e., a larger RF amplitude corresponds to a lower luminance. The amplitude reaches a maximum during the synchronization pulses and is lowest for white level of the luminance signal. Frequency modulation is used for audio information with the characteristics stated in the table.

1.3 SYSTEM PERFORMANCE OBJECTIVES

By system performance objectives we mean the grade of service and the quality of picture desired. The International Radio Consultative Committee (CCIR) study (3) proposed

Table 1.1 Television Broadcast Standards

Video Signal Characteristics:

Number of lines per field		525
Nominal video bandwidth, MHz		4.2
Frame frequency, Frame/sec		30
Field frequency, Fields/sec		60
Line frequency, Lines/sec		15,750
Color subcarrier frequency MHz		3.58
Relative video voltages	White level	0
	Blank level, color burst bias	0.71
	Syne pulse top level	1.0
	Color burst amplitude	0.143
Signal components durations μsec	Line period	63.5
	Line blanking monochrome	10.8
	Line blanking color	10.95
	Line syne pulse, monochrome	4.95
	Line syne pulse, color	4.65
	Color burst NTSC	2.3-3.4
Rise times (10-90%) μsec	Blanking signal, monochrome	≤ 0.64
	Blanking signal, color	≤ 0.48
	Line syne pulse	≤ 0.25

Audio Signal Characteristics:

Audio bandwidth	kHz	15
Maximum frequency swing	kHz	±25
Time constant of pre-emphasis	μsec	75
Test tone frequency	Hz	400
Pre-emphasis test tone frequency	db	±0.2
Pre-detection bandwidth	kHz	200

definitions of three broadcasting satellite services: principal, rural, and community. Slightly different definitions have been proposed by the study group IV (4). These classifications and proposed definitions have been considered adequate by some of the papers (5) submitted to the United Nations Working Group on Direct Broadcast Satellite. This report takes an approach similar to that being pursued within the CCIR in discussing various grades of service to principal, rural, and community installations. These definitions are discussed in the following paragraphs.

Primary (Principal) Grade of Service is a grade of service with a power flux density of sufficient magnitude to enable the general public to receive transmissions directly from satellites by means of individual installations and with a quality comparable to that provided by a terrestrial transmitter to its primary service area. It is assumed to be offered to urban areas where man-made noise level is high and the receiver population is or has the potential of being extremely high. A field strength of 70 dbu (relative to one microvolt per meter) is considered to be a reasonable estimate (5) for this grade of service This is equal to the CCIR recommendation (6) and is about midway between the FCC Grade A and B (5).

Secondary (Rural) Grade of Service is a grade of service with a lower power-flux density than that required for a primary grade of service. The signals are intended for direct public reception from satellites by means of individual installations and with an acceptable quality in sparcely populated areas which are not served, or are inadequately served, by other means and where satellite reception conditions are favorable.

Community Grade of Service is a grade of broadcasting service from satellites with a limited power flux density. The signals are intended for group viewing or listening or for reception by a master receiver installation. This grade of service could provide a quality of picture about equivalent to that of primary grade although the signal strength may be considerably lower. This grade of service may be applied for educational and national development purposes (5).

Unlike the principal grade of service, no specific signal strength requirements exist for the other two grades of services. Hence, the performance objectives are established with signal to noise ratio (SNR) as a parameter.

1.4 SUBJECTIVE PICTURE QUALITY

A commonly used picture quality measure is the receiver Signal to Noise Ratio (SNR) and is defined as

$$\left(\frac{S}{N}\right)_p = \left(\frac{\text{blank-to-white video voltage}}{\text{RMS voltage of video noise}}\right)^2$$

This quantity is known as "picture SNR", as it compares the noise voltage with the voltage range of picture signal. Some other definitions of SNR include the synchronization pulse too, which increases the picture SNR by about 3db.

These definitions do not give a meaningful measure of the effect of noise on picture quality as subjectively experienced by the viewers unless qualified by the video noise spectrum because the noise at the upper end of the video spectrum is less objectionable than equal noise power at the lower end. Weighting networks are used to account for this effect by spectrally weighting the noise according to the perception of an average viewer. The power transfer characteristic of the filter used for 525 line television can be found in the literature (7,9). Thus the new weighted SNR can be defined as:

$$\left(\frac{S}{N}\right)_{p,w} = \left(\frac{\text{blank-to-white video voltage}}{\text{weighted RMS voltage of video noise}}\right)^2$$

where the subscripts p and w refer to power ratio with weighting. The weighting factor, i.e., ratio by which weighting increases the picture-SNR is:

$$W = 10 \log \frac{\int_0^{B_v} n(f_v)\, df_v}{\int_0^{B_v} n(f_v)\, W(f_v)\, df_v}$$

where

B_v = upper frequency limit of video band

f_v = video frequency

$n(f_v)$ = one sided power spectral density of video noise

$W(fv)$ = power transfer characteristic of the weighting network.

A source of information on the subjective effect of random noise on viewer satisfaction with monochrome and color television pictures is the study (8) done for the Federal Communications Commission by the Television Allocations Study

Organization (TASO) during the 1950's. Table 1.2 gives the results of this taken from reference 5. These data differ from the CCIR data in two ways: first, they are the signal to noise ratio at the input of the receiver, while the CCIR data are signal to noise ratio in the video channel. Second, the TASO numbers result from tests with both picture and noise present, while the CCIR data refer to noise measurements performed in the absence of signal.

The conversion of TASO's SNR to the weighted SNR has been discussed in the literature (7,9). There is a slight variation in the results obtained by various authors, however the relation derived in reference 7 appears reasonable and is used here for conversion purposes. The relation is:

$$\left(\frac{S}{N}\right)_{p,w} = \left(\frac{S}{N}\right)_{T} + 0.9 \text{ db}$$

where $\left(\frac{S}{N}\right)_{p,w}$ = weighted picture-SNR, in db

$\left(\frac{S}{N}\right)_{T}$ = picture-SNR used by TASO to express its test results, in db.

The values of carrier to noise ratio stated by TASO relate to the controlled R F noise injected at the test receiver input. Consequently, these figures do not account for camera noise, which contributed to the interference rated by TASO's viewer panel. Accounting for camera noise (7) in the TASO picture-SNR, the last column in the Table 1.2 gives the weighted picture-SNR for the desired TASO grade.

TASO reports that color television requires a slightly lower signal-to-noise ratio than monochrome for equal

Table 1.2 Subjective Assessment of Signal to Noise Ratio for Television

TASO GRADE	NAME	DESCRIPTION	MEDIAN OBSERVER (db)	MEAN OBSERVER (db)	WEIGHTED SNR (db)
1	Excellent	Extremely high quality, as good as could be desired	43	42	45.5
2	Fine	High quality providing enjoyable viewing, perceptible interference	33	38	40.2
3	Passable	Acceptable quality, interference not objectionable	27	31	32.2
4	Marginal	Poor quality; improvement desired, interference somewhat objectionable	23	25	25.9
5	Inferior	Very poor quality but could be watched, definitely objectionable interference	17	19	19.9

subjective quality (Reference 8, page 532 to 534, Figure 40), but opposite results have been reported by Barstow and Christopher (10).

It should, however, be noted that the above picture ratings are for conventional television frame rates For still-picture television, where the frame repeating system is used (Section 2.2.6), the noise pattern associated with each frame is also frame repeated, thus producing the "frozen" noise effect. Some research (32) done for a small number of repetitions indicates that the noise level increases rapidly as the number of repetitions are increased followed by a general flattening out or saturation above 60 to 100 millisecond. This corresponds roughly to the integration period or critical duration of the eye. Below the critical duration, the eye sums "frozen" noise frames and sees increasing granularity with increasing frame repetition. Above the critical duration the granularity stays constant, but the apparent spatial movement of the noise becomes slightly more noticeable with larger numbers of repetitions. For frame repetition up to 0.1 second, 2 to 3 db apparent increase in the noise level has been reported (32). In the absence of any data for large numbers of repetitions, a series of psychophysical experiments are recommended to get quality ratings for still-picture television. Until then we will use the standard scale.

1.5 SATELLITE POWER AND PICTURE QUALITY

The satellite power requirements depend on. (i) the grade of service desired; (ii) the picture quality desired; (iii) the frequency band assigned for the given channel; (iv) the area covered or antenna gain; and (v) the modulation scheme used. The grade of service at the receiving installation has been characterized (5) by the ratio of its antenna gain to system noise temperature (G/T). The ratios $10 \log_{10} \frac{G}{T}$ for three grades of service, viz., primary, secondary, and community, are 27, 11, and 3.85 dbu, respectively (5). Picture quality has already been discussed in terms of TASO grades and the corresponding signal to noise ratios have been given. Now the three other important parameters: frequency band, antenna beamwidth, and modulation scheme have yet to be decided. We shall keep them as parameters and consider for various values of frequency, different modulation schemes (vestigial side band amplitude modulation and frequency modulation), and a set of beamwidths.

Three curves have been drawn [Figures 1.1, 1.2, and 1.3]. Figure 1.1 is for vestigial side band-amplitude-modulation (VSB-AM) 0.86 GHz television transmission scheme. It gives the values of satellite effective radiated power (ERP) required for a given picture quality, grade of service and antenna size (antenna beam-width) for a VSB/AM television transmission. The satellite borne antenna is characterized by the width of beam in two orthogonal planes. These beams do not have to be equal, but they have been taken so for convenience.

Figures 1.2 and 1.3 give the same information for the frequency modulated 0.86 GHz and 12 GHz carrier respectively. These curves have been derived from the nomograms in reference

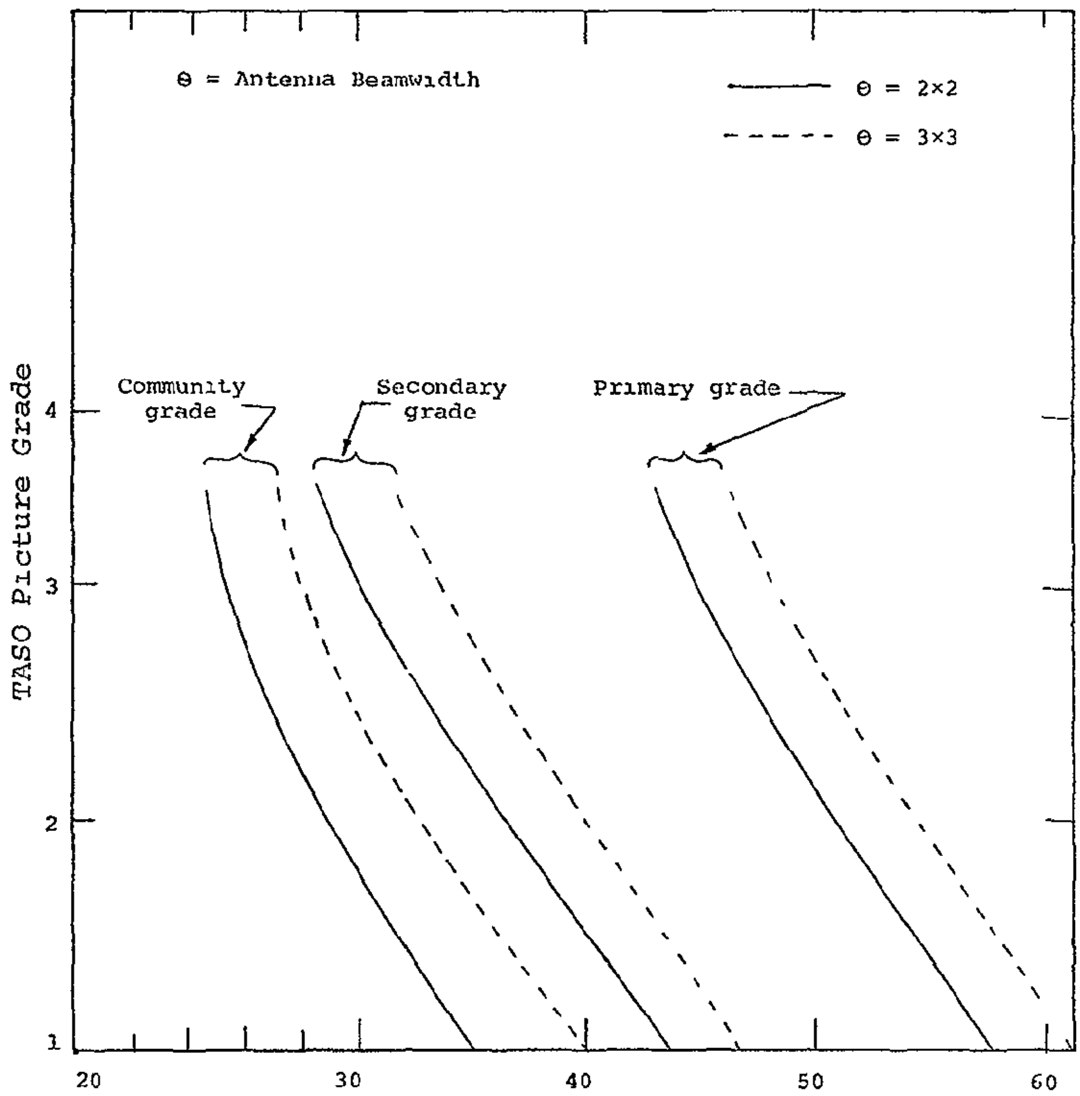


Figure 1 1

Satellite power requirements for TASO picture grades with VSB/AM at 0 860 GHz.

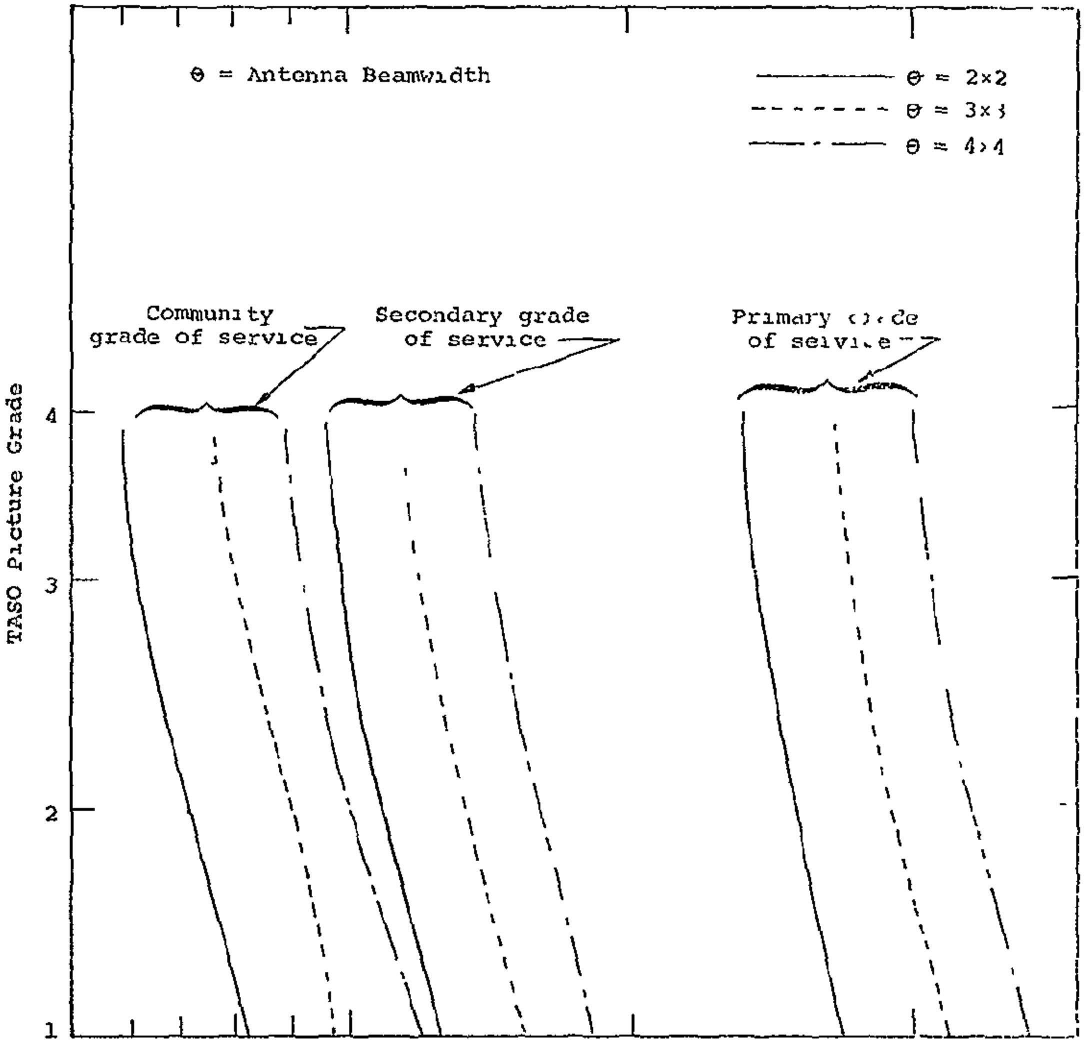


Figure 1 2

Satellite power requirements for TASO picture grades with frequency modulation at 0 860 GHz

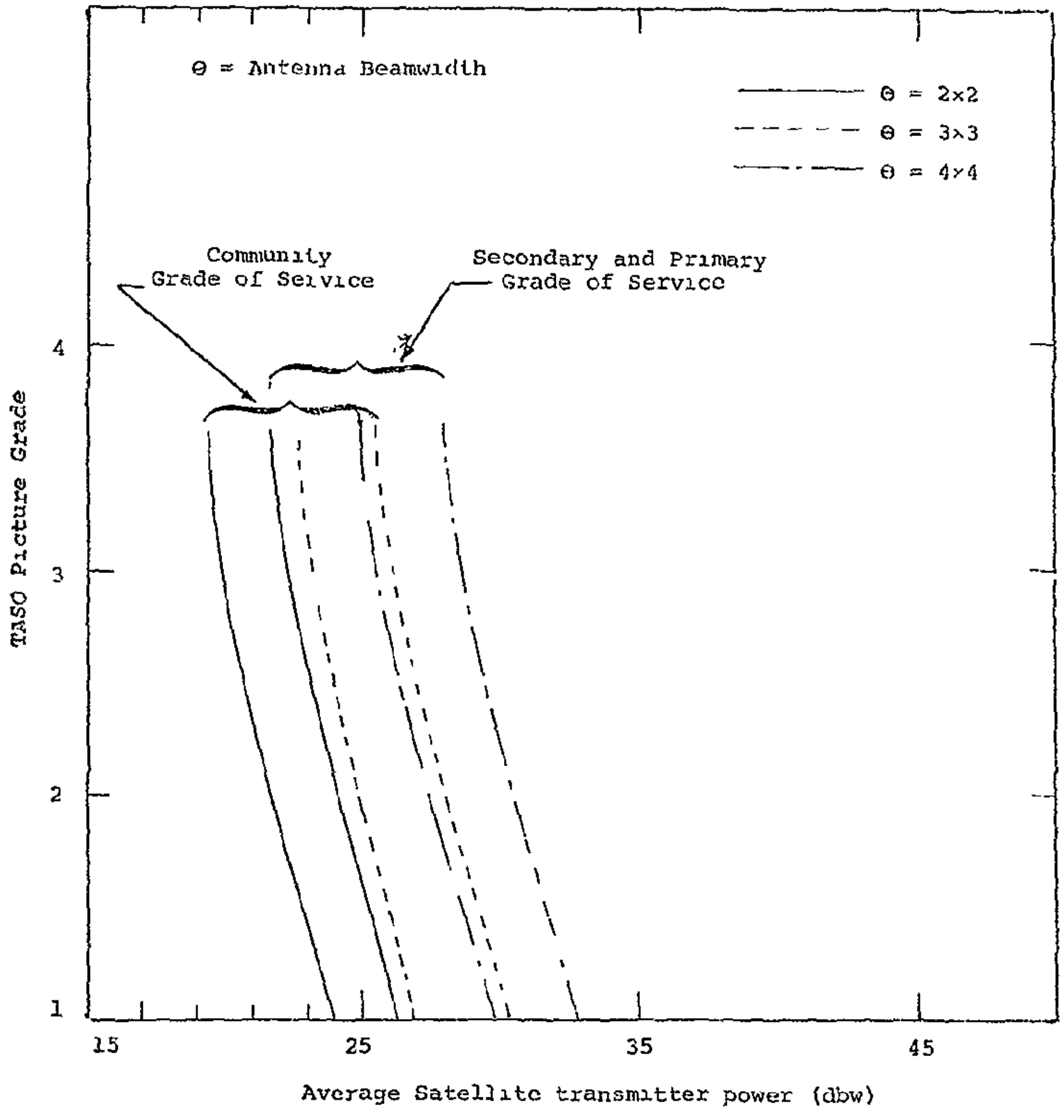


Figure 1 3

Satellite power requirements for TASO picture grades with frequency modulation at 12 GHz

5 with the following assumptions: (i) the receiving antenna is mounted outside. This arrangement overcomes building attenuation losses which could raise the required satellite power by up to several orders of magnitude; (ii) a variety of transmission losses, which exist in the practical system, have been considered. These losses include a pointing loss due to imperfect alignment of the receiving antenna, polarization mismatch losses, ionospheric absorption losses, cloud attenuation, refraction and tropospheric loss, fading, and precipitation loss. A factor of 2 db has been used for these losses and a margin of 3 db has been assumed.

As an example of the use of these curves, let us find the satellite power requirements for community grade of service with TASO grade 2, given that the satellite antenna beamwidth is 2°×2°. From Figure 1.1 we find that for the above requirements, a peak transmitter power of 21 dbw is required for VSB/AM at 0.86 GHz and for the same requirements with frequency modulation at frequencies 0.86 GHz and 12 GHz the average transmitter power is approximately 6 dbw and 15 dbw respectively. As another example, if an excellent picture is desired for a primary grade of service, then the power requirement, at 0.86 GHz frequency modulated system with antenna beamwidth 3°×3°, is approximately 36.5 dbw.

In the above power considerations, the audio channel power has not been included, which can be considered about 10% of the video power (11), per audio channel.

2. STILL-PICTURE TELEVISION (SPTV) TRANSMISSION

2.1 SLOW-SCAN

Basically, slow-scan is a method for reducing the video information rate to a value lower than that used for conventional television transmission. Slow-scan television is not new (12,13); however, in recent years there has been an increasing interest in it for applications in various fields including educational and commercial television broadcast (14,15).

2.1.1 General System Concept

In a television system, there is a fixed relationship between the number of lines per field, the number of fields per unit time, the resolution across the line, and the video bandwidth (for a given value of aspect ratio and blanking time ratios). This is as follows (14).

$$\frac{W}{F} = 2AR_H N_F / 2B_L$$

where A = aspect ratio (width/height of active picture area)

R_H = horizontal resolution in number of television lines

N_F = number of scanning lines per field

B_L = line blanking factor (active time/total times)

W = bandwidth of the video signal

F = television frame rate

and $F = \frac{2}{T_F}$ where T_F = times per field.

The left-hand member of the above equation is a dimentionless function of the aspect ratio horizontal resolution, scanning lines per field and blanking width. It is thus seen that a trade-off is possible between bandwidth and the frame rate. As an example of this, if the conventional television standards are considered for picture transmission with a difference of frames presentation time from 1/30th of a second to 10 seconds, then the bandwidth is reduced by a factor of 300. Thus about 300 simultaneous transmissions are possible in one television equivalent channel, neglecting frame identification information and the required audio bandwidth. Besides the narrow bandwidth required for slow-scan television, it has the advantage of increased resolution that can be realized from the vidicon tubes; this results because more time is available to discharge the screen as the scan time is increased.Increasing discharge time permits lower beam currents and, as a result, the scanning aperture (or beam size) can be reduced. The resolution of a vidicon tube is limited by the beam size; the resolution is increased as the beam size is decreased.

The slow-scan video information can be transmitted and received on a storage screen without the use of any memory unit. However, the viewer has to spend a certain amount of time prior to display of a complete picture. Even if the first picture is removed line by line as the next picture is laid, thus creating the effect of wipe moving, the above problem still exists. Other problems associated with this are: (i) while the audio is transmitted continuously the picture takes finite time to appear. Thus arrangements must be made to synchronize the audio with picture. One of the ways in which this can be done is to send the audio with synchronization information, ahead of the video information and then synchronize locally the audio and video. (ii) a set of new display apparatus with storage tube is required.

The above difficulties can be removed if the slow-scan information being transmitted through satellite is first stored at a central receiving point, converted into a standard television signal, and then retransmitted to user display receivers. A block diagram of a possible slow-scan transreceiver is shown in Figure 2.1. Scan converters are used at the transmitter and receiver to convert the standard signal format to slow-scan and slow-scan to standard format, respectively. The storage element is an important part of the scan converter system. Farr (13) discusses a slow-scan system for which phonograph records can be used as storage elements. Magnetic disc recorders have been suggested as another storage element (16).

Deutsch (17) has proposed another narrowband television transmission system. This is basically a slow-scan system but not a still-picture system. Bandwidth as low as 10 kHz has been reported in this case (18). Deutsch's system takes advantage of the low information content of the television picture, the tolerance of the human vision for motion

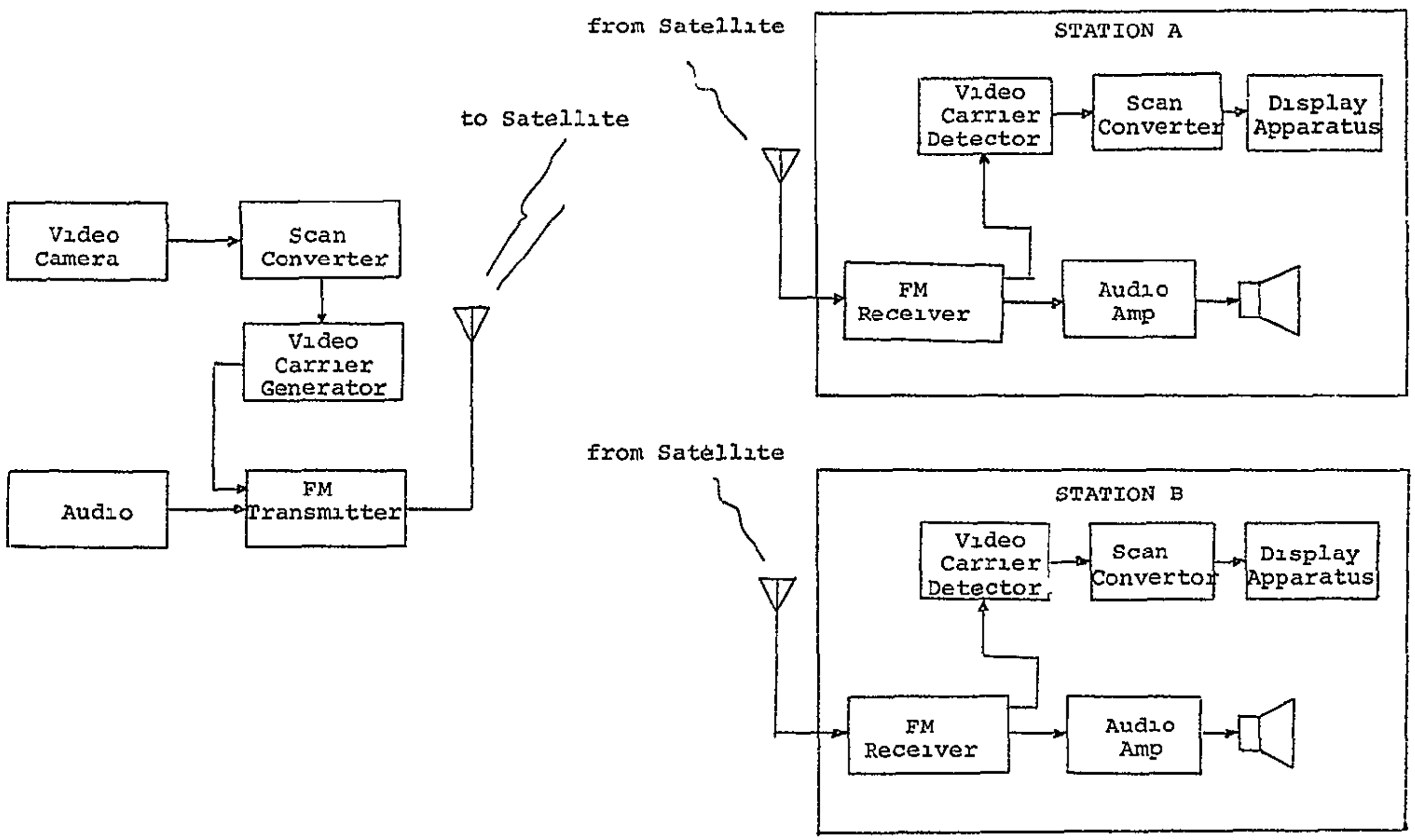


Figure 2 1 Block Diagram of a Possible Slow-Scan Transmission Scheme deterioration, and lower resolution than that used with conventional television. It has been stated (30) that the principal psychological requirements of human vision are satisfied by a video frame frequency of one or two frames per second. To avoid flicker and the illusion of drifting of lines (19), when line scanning is applied with such low frame rates, a pseudo-random dot scan is employed by Deutsch in conjunction with a long persistence phosphorous. Fifteen percent dot flicker has been shown to be tolerable. This system, though promising is not compatible with the conventional system, and needs new receiver structures.

2.1.2 Effect of Scanning Speed on the Signal to Noise Ratio of the Camera Tubes

The signal amplitude from a camera tube and its bandwidth vary directly with the scanning speed. Since the noise power is distributed over the whole frequency, the rms noise voltage must rise in proportion to the square root of bandwidth, and hence the square root of scanning velocity. Thus SNR is actually proportional to the square root of scanning speed. Thus for slow-scanning speed, the SNR can be expected to be smaller than at conventional speeds, but this statement can be modified by saying that the SNR of the camera can be made independent of scanning speed if the system parameters are optimized. The validity of the latter statement has been shown by Schreiber (20) by considering the three inherent sources of noise: (i) the signal shot noise; (ii) the thermal noise of the load resistor; (iii) and the amplifier noise.

The shot noise rms amplitude, inherent in a video tube with plate current I_s is $i_n = \sqrt{2eI_sF}$, where e is the electronic charge and F the video bandwidth. The SNR due to this is

$$\left(\frac{S}{N}\right)_{\text{shot noise}} = \frac{I_s}{\sqrt{2eI_sF}} = \frac{1}{\sqrt{2e}}\ \frac{I_s}{F} \qquad (1)$$

As the scanning speed changes I_s and F change accordingly, thus making the above SNR independent of scanning speed.

The SNR due to the load resistance is

$$\left(\frac{S}{N}\right)_{\text{Load resistor}} = \frac{I_sR}{\sqrt{4kTRF}} = \frac{1}{\sqrt{4k}} \cdot I_s\sqrt{\frac{R}{F}} \qquad (2)$$

where $\sqrt{4kTRF}$ is the rms thermal noise voltage generated due to the load resistor. The above expression can be made independent of the scanning speed if R is made inversely proportional to F, because the ratio $\frac{I_s}{F}$ is already independent of the scanning speed. Now, for a properly designed system, the noise generated within the preamplifier will be small compared to shot noise and the thermal noise, at least at low frequencies. So the amplifier bandwidth must be decreased in proportion to the scanning speed Thus to obtain this independence of scanning speed the load resistor and the amplifier bandwidth must be adjusted accordingly.

It can be easily seen from equation (2) that camera SNR does indeed vary as the square root of scanning speed if the load resistor is not optimized.

2.2 TIME-SHARED SPTV TRANSMISSION

In designing a multi-channel communication system, two parameters, time and frequency, can be utilized as a means of separating the sub-channels. A given amount of time-limited information can be transmitted in either domain with a constant time-bandwidth product; that is, if the bandwidth is reduced, the time duration is prolonged and vice versa. Although equal in their capabilities, the two methods differ in many other respects.

To create the illusion of motion in the standard television transmission format, a number of frames of slightly different spatial variation are transmitted. However, if motion is not a requirement, the same television frame has to be repeated, as long as it is being displayed on the receiver. Therefore, instead of sending the same television frame repeatedly for still-picture transmission, one frame can be transmitted for each picture, stored at the receiver, and displayed as long as desired by cyclically displaying the stored frame. Each succeeding frame can be sent when the preceding frame is no longer required for display. The time saved by this procedure can be used for sending other unrelated frames. Thus a time-sharing system can be used for sending the still-picture video information. The received video information may be stored at a receiving point and formed into a signal compatible with a conventional television receiver.

The audio information accompanying each still-frame can either be transmitted by frequency-sharing, placing the audio information band above the video frequency band, or by time sharing by expanding its frequency to video frequency level and transmitting along with the video information the corresponding audio.

2.2.1 Classification

Since the audio information of different sub-channels can be separated either in frequency or time domain, the transmission system can be classified into two categories: (i) time-shared-video, frequency-shared-audio, (ii) time-shared video, time-shared, time-compressed audio.

In the time-shared-video, frequency-shared-audio scheme, time division multiplexing (TDM) is used for the video information transmission and frequency division multiplexing (FDM) is used for audio information. On the other hand, only TDM is used in the time-shared-video, time-shared, time-compressed audio system. The detailed description and some of the technical problems associated with these systems are discussed in the subsequent sections.

2.2.2 Time-Shared-Video, Frequency-Shared-Audio, SPTV Transmission System

In this system of transmitting still-picture with continuous audio, time division multiplexing is used to transmit the different still video frames, while frequency division multiplexing is used for the continuous audio information accompanying each video slide. The use of several FDM audio channels requires a greater fraction of total bandwidth available compared to single audio in the standard television system. If only one television channel equivalent bandwidth is assigned for such a system, then the expanded audio bandwidth must be accommodated. This can be done either by decreasing the video frame rate, which in turn reduces the video bandwidth thus creating more bandwidth for audio, or by using some video bandwidth reduction technique (43,44,45).

Each sub-channel of video information consists of still pictures that are updated infrequently. These pictures are time-multiplexed into the transmitted video signal as an ordered sequence of individual frames. Thus, if there are N sub-channels of audio-visual information, frame 1, N+1, 2N+1, 3N+1,..., correspond to the sub-channel one, frame 2, N+2, 2N+2,..., correspond to sub-channel two, etc. [See Figure 2.2].

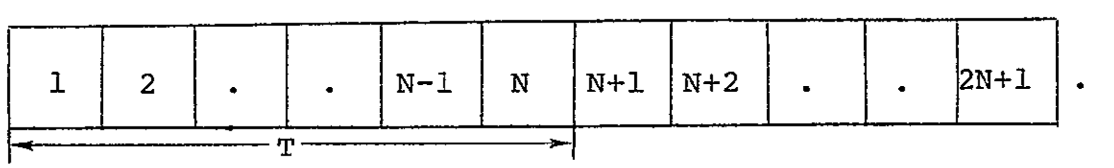


Figure 2.2

Timing Diagram for Time-Shared SPTV Video Information

The updating time for each video sub-channel, T, is N times the frame time of the transmitted video signal. For example, the updating time with 30 frames/sec is equal to $\frac{N}{30}$ sec. The audio is sent continuously by FDM. A frame synchronizing signal is required to identify the beginning of a frame sequence. The preprocessor then counts frames from the frame synchronizing signal to the frames corresponding to the desired sub-channel. This frame must be stored in a video frame buffer. Between updatings of the buffer, the stored video frame corresponding to a single still-picture is read periodically and combined with its companion audio signal to form an audio visual signal compatible with conventional television receivers used for display.

2.2.3 General System Considerations

The composite still-picture television signal consists of the time-division multiplexed video information along with the frequency division multiplexed audio information. A number of techniques are available to achieve this simultaneous transmission of video and audio information. Among these are. (i) separate RF carriers for time-shared video and each audio channel; (ii) separate RF carriers for time-shared video and multiplexed audio information, for example, if W_{c_v} and W_{c_a} are the RF carrier frequencies for time-shared video and multiplexed audio information, respectively. The frequency modulated signals are $x_1(t) = \sqrt{2}\ p \cos [W_{c_v} t + d_{f_1} \int^t a_1(u)du]$ and $x_2(t) = \sqrt{2}\ p\cos[W_{c_a} t + d_{f_2} \int^t a_2(u)du]$ for video and audio information respectively. The notations are:

$x_1(t)$ = video frequency modulated signal

d_{f_1} = deviation ratio for video

$a_1(t)$ = video signal

d_{f_2} = deviation ratio for multiplexed audio, and

$a_2(t)$ = multiplexed audio signal and is given by

$$a_2(t) = \sum_{i=1}^{N} A[1 + m_i b_i(t)] \cos W_i t$$

where m_i = modulation index

$b_i(t)$ = audio signal

W_i = audio sub-carrier frequency.

(iii) Multiple sound channels in the backporch (21) of the synchronizing pulse of video waveform. This method, however, gives one or two sound channels and receiver complexity is increased. Parameters such as satellite effective isotropically radiated power (EIRP), RF bandwidth, etc., are not affected. (iv) One RF carrier for both time-shared-video and multiplexed audio information. If frequency modulation is used for RF transmission, the tramsmitted signal can be written as

$$x_c(t) = \sqrt{2}\, p \cos [W_c t + d_f \int^t a(u)\, du]$$

where W_c = carrier frequency

d_f = deviation ratio for the combined video and multiplexed audio signal

$$a(t) = a_1(t) + \sum_{i=1}^{N} A(1+m_i b_i(t)] \cos W_i t$$

This system has been recommended by the International Radio Consultative Committee (CCIR) for terrestrial microwave systems (22). The ultimate choice of a method for this system depends on the following factors:

(a) Transmission base-bandwidth available.

(b) Number of still-picture channels required.

(c) Receiving and transmitting station complexity considerations.

(d) Satellite EIRP considerations.

If a limited frequency spectrum, either in terms of RF bandwidth or base-bandwidth is available, and a substantial number of still-picture television channels are desired, then from minimum equipment complexity and satellite EIRP considerations, method 4 seems to be a suitable choice. A number of studies done on simultaneous transmission of video with multiple sound channels (23,24) for India, claim that this method is the least costly solution. This method has the advantage that a substantial number of still-picture television (SPTV) channels can be accommodated in a single satellite transponder with only a single RF carrier operation.

Once it is decided about the modulation format, the type of modulation for audio sub-carriers and their arrangement above video information has to be looked into so as to avoid the intermodulation products in the video band. The latter part of this problem depends on the first. For example, the frequency modulated sub-carriers have to be considered from a different point of view than the amplitude modulated ones, as the former contains many upper and lower side bands while the latter contains only one upper and one lower side band.

Practical and economical considerations (23,24) suggest that audio channel transmission in space broadcasting be done with sub-carriers modulated in accordance with the present standards for the audio carrier modulation. As previously stated, in nearly all television transmission systems, the sound carrier is frequency modulated with predetection bandwidth of 200 kHz. If a large number of SPTV channels are required, then the base-bandwidth requirement of this composite channel would be prohibitive For example, a base-bandwidth of at least 8.2 MHz would be required for a 50 channel SPTV system, with 4.2 MHz as video bandwidth. However, when sub-carrier arrangement for frequency modulated sub-carriers is considered, this base-bandwidth requirement greatly increases because the sub-carrier frequencies have to be chosen so that none of the sub-carrier bands overlap the third order products of the intermodulation between other sub-carriers and between any other sound subcarrier and the color sub-carrier (23,25). In addition the video,to audio carrier spacing of the given television standard is avoided (6), to simplify the filter requirements, which further increases the SPTV base-bandwidth.

The base-bandwidth requirements suggest that a modulation scheme with less complicated sub-carrier arrangement and less sub-carrier bandwidth is desired. This immediately suggests the idea of single-side band modulation scheme. But considerations of equipment complexity and oscillator stability seem to discourage this. However, amplitude modulation seems to be a reasonable choice. The problem of sub-carrier separation is automatically solved as amplitude modulation contains only the upper and lower side bands, and therefore a separation equal to or little more than twice the audio bandwidth will suffice.

The base-bandwidth for this system consists of the spectrum occupied by the video information along with all the modulated audio sub-carriers above this. Two cases. (i) with total base-bandwidth fixed and equivalent to one television channel, and (ii) with video base-bandwidth fixed and equal to standard video base-bandwidth can be considered. In either case the sub-carriers are placed above the video base-bandwidth. More details about these are given below.

2.2.3.1 Base Band-width Fixed and Equivalent to One Television Channel Bandwidth

If the video scanning rate is reduced by an appropriate amount, the frequency spectrum thus created can be used to accommodate the modulated audio sub-carriers. However, the number of total audio sub-carriers is limited by the maximum bandwidth which can be allocated for all the audio channels. A relation between the number of sub-channels that can be transmitted, the updating time of the picture frame, and the audio base-bandwidth can be derived as follows·

Let N = Number of channels to be transmitted

B = Total base-bandwidth

Ba = Audio sub-carrier bandwidth

K = constant depending on number of television lines, aspect ratio and horizontal resolution.

T = channel frame update time (See Figure 3.1).

Then the television frame rate is $\frac{N}{T}$. Since the television video frequency is directly proportional to the television frame frequency, we get the following relation between these parameters:

$$N = \frac{B - k\frac{N}{T}}{Ba}$$

and this can be written as

$$N = \frac{B}{Ba + \frac{k}{T}} \qquad (5)$$

where the video bandwidth is

$$f_v = k\frac{N}{T} \qquad (6)$$

From these relations, a curve relating the number of channels with frame update time has been prepared (Figure 2.3). It can be seen from this curve that for T=10 and Ba= 20 kHz, the number of channels, N, is 30; and to obtain this, the television frame frequency has to be reduced to 13 frames per second.

While this method of transmitting appears to be reasonable, the scan conversion problem seems to discourage it. If video base-bandwidth is kept fixed and the composite base-bandwidth is increased as the number of audio channels are increased, the scan conversion problem is alleviated. The RF bandwidth requirements for the latter case are not much different than for the case in which total base-bandwidth is kept fixed for the same transmission quality requirements. Therefore, it seems reasonable to keep the standard base-bandwidth of video with sub-carriers above this. Figure 2.4 shows the base-bandwidth spectrum of the composite signal with time-shared video and frequency-shared equally spaced, amplitude modulated audio information.

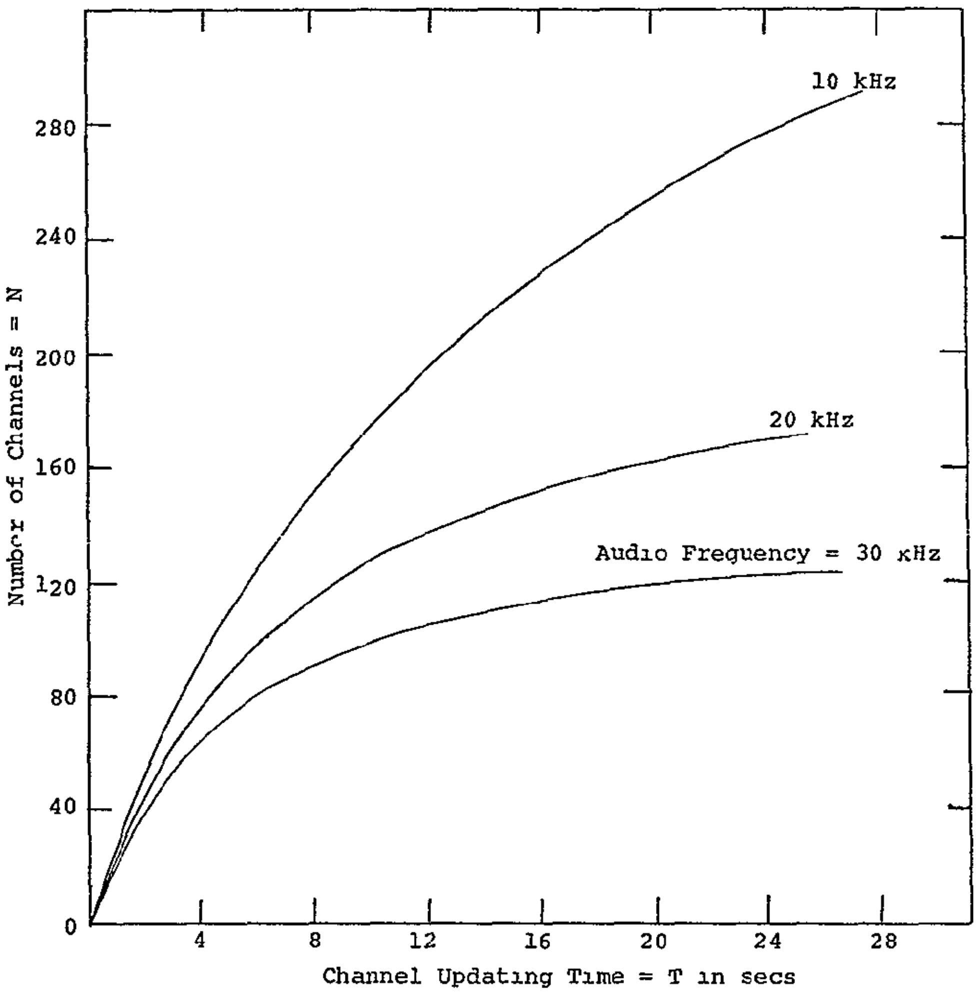


Figure 2.3

Relation between the Number of SPTV Channels and Channel Frame Updating Time for a Fixed Video Base-Bandwidth of 4 5 MHz

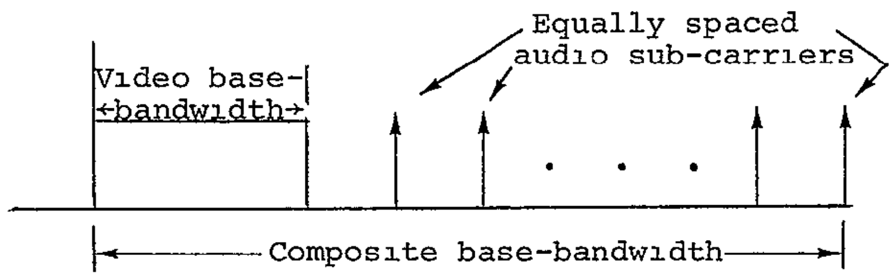


Figure 2.4
Base-bandwidth Configuration
of Composite Signal

For equally spaced carriers above video information, the number of channels and audio bandwidth and video bandwidth will determine the total base-bandwidth. The picture updating time is directly related to the number of channels in the sense that for 30 still-picture channels, the updating time will be 30 times the television frame time. Therefore, as the number of still-picture channels increses, the updating time increases in the same ratio.

2.2.4 Transmission and Reception

The general transmitter and receiver are shown in Figures 2.5 and 2.6, respectively. Typically, the video time division multiplexing can be obtained by using N+1 state counter, in which N states account for N channels and the remaining one state can be utilized for frame synchronization. Each video signal can be connected to a logic switch, which operates only when both the counter and the signal are present. These switches can be opened for one frame period so that one television frame of each video signal sent sequentially. The corresponding guard bands between the adjacent frames can be adjusted to match the

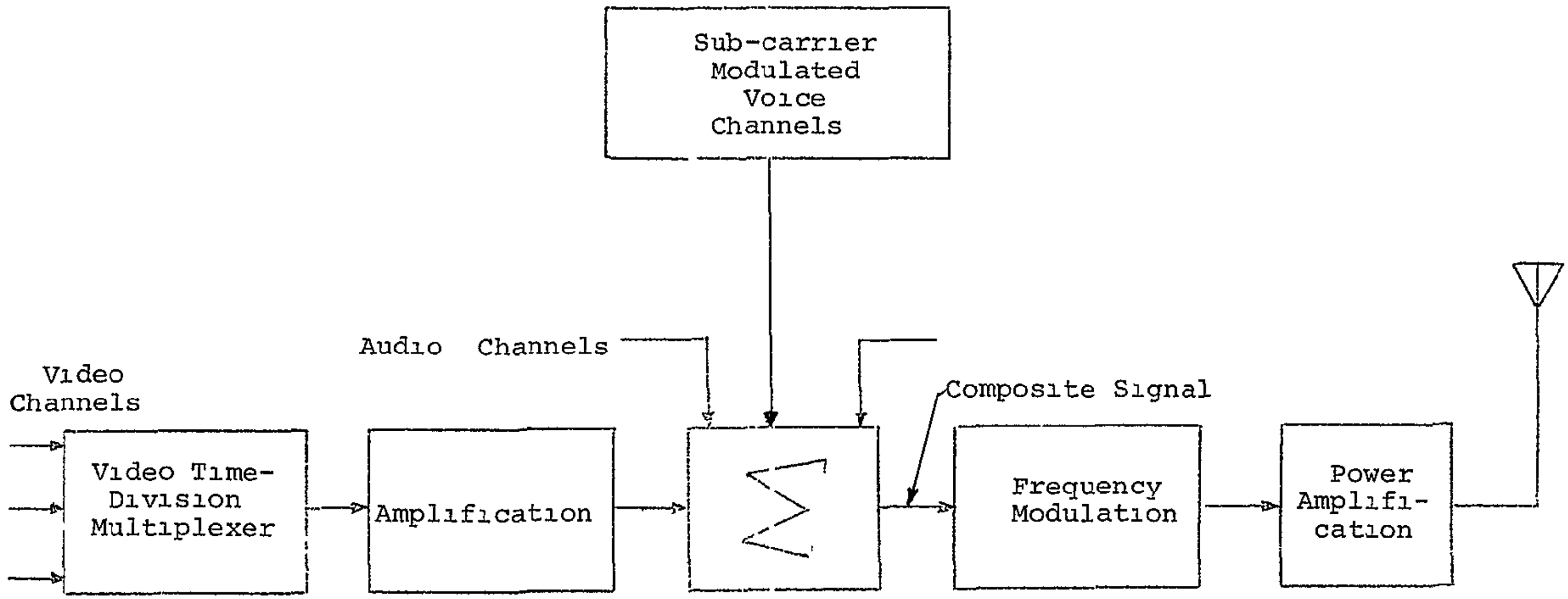


Figure 2.5

Block Diagram of a Time-Shared Video, Frequency-
Shared-Audio Transmitter

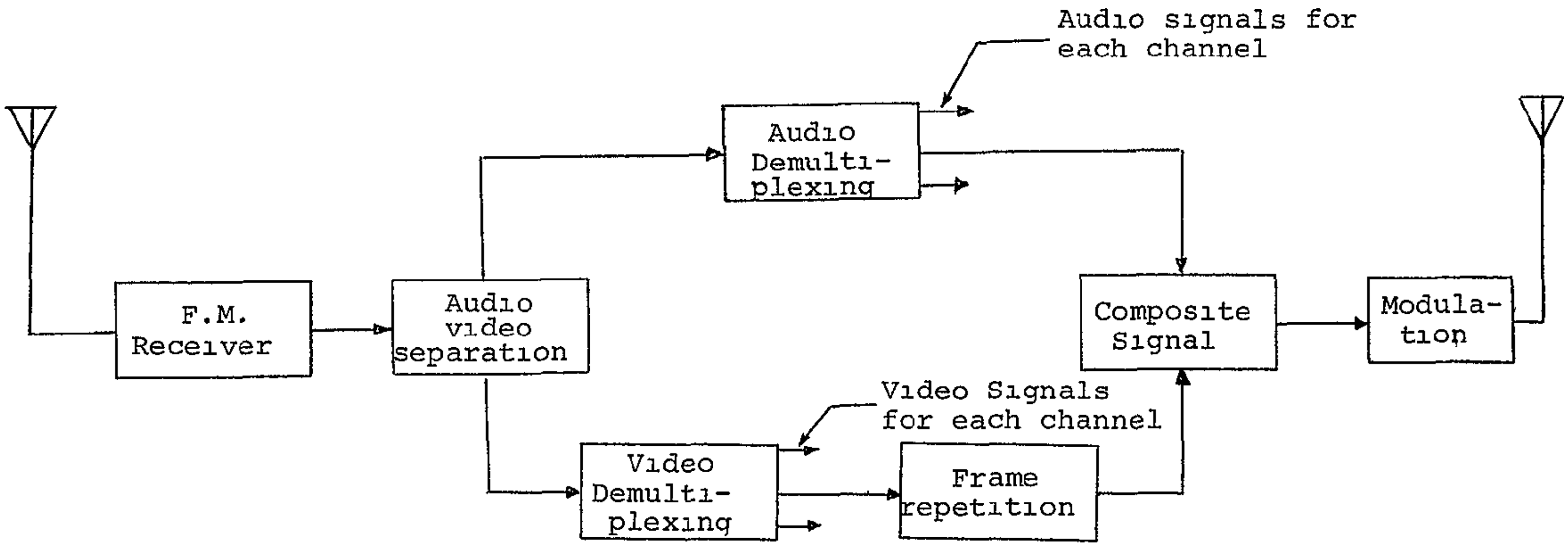


Figure 2.6

Block Diagram of a Central Receiving Station system requirements such as intermodulation effects and crosstalk level. The time division and multiplexed video signal is brought to a suitable power level to recombine this with the frequency division multiplexed audio signal, forming a composite audio-video signal. Finally, the composite signal is modulated and brought to the proper level for transmission to the satellite. The steps involved in transmission can be summarized as follows:

(1) Formulation of FDM video signal along with the frame sequence synchronizing signal.

(2) Formation of FDM audio signal.

(3) Formation of composite audio-still-video signal.

(4) Modulation of composite signal, and final power level adjustment for transmission to satellite.

The transmitted audio-still-video signal is received at a central receiving station, where it is formed into a compatible audio-still-video signal for the existing conventional receivers. The block diagram of the receiver shows the signal reception and formation of compatible signal at a central receiving point. Frame sequence synchronizing signals can be used to separate the video frames, while a number of band-pass filters can be used for the audio channel separation. The following steps are involved in the reception and demodulation of the above signal·

(i) Receiving and initial demodulation

(ii) Selection of TDM video and FDM audio

(iii) Video frame selection and frame repetition until the next frame.

(iv) Audio signal recovery

(v) Formation of audio-still-video signal by synchronizing the proper audio signal to the frame repeated video signal.

(vi) Placing the audio-still-video channel in proper frequency band and retransmitting them for contentional receivers.

2.2.5 Frame Sequence Synchronization

The synchronization of the proper video frame with the audio information is an important issue for the system described above. A composite frame consists of a number of television frames in time T. The period of time preceding transmission slots in the composite frame is designated as the synchronizing period. This assures the composite frame starting time for all frames, thus making the task of frame selection easy The characteristics of the synchronizing burst are·

(a) It must be a signal that is uniquely determined.

(b) It must establish a point in the time within a required tolerance.

A signal with these characteristics is sent from the transmitting station after each composite frame for a preassigned time period. This signal is then decoded and formed into a synchronizing pulse, which when applied to the N+1 counter at the receiver resets it to the original position. Thus, the cycle begins again.

A possible decoding arrangement is shown in Figure 2.7.

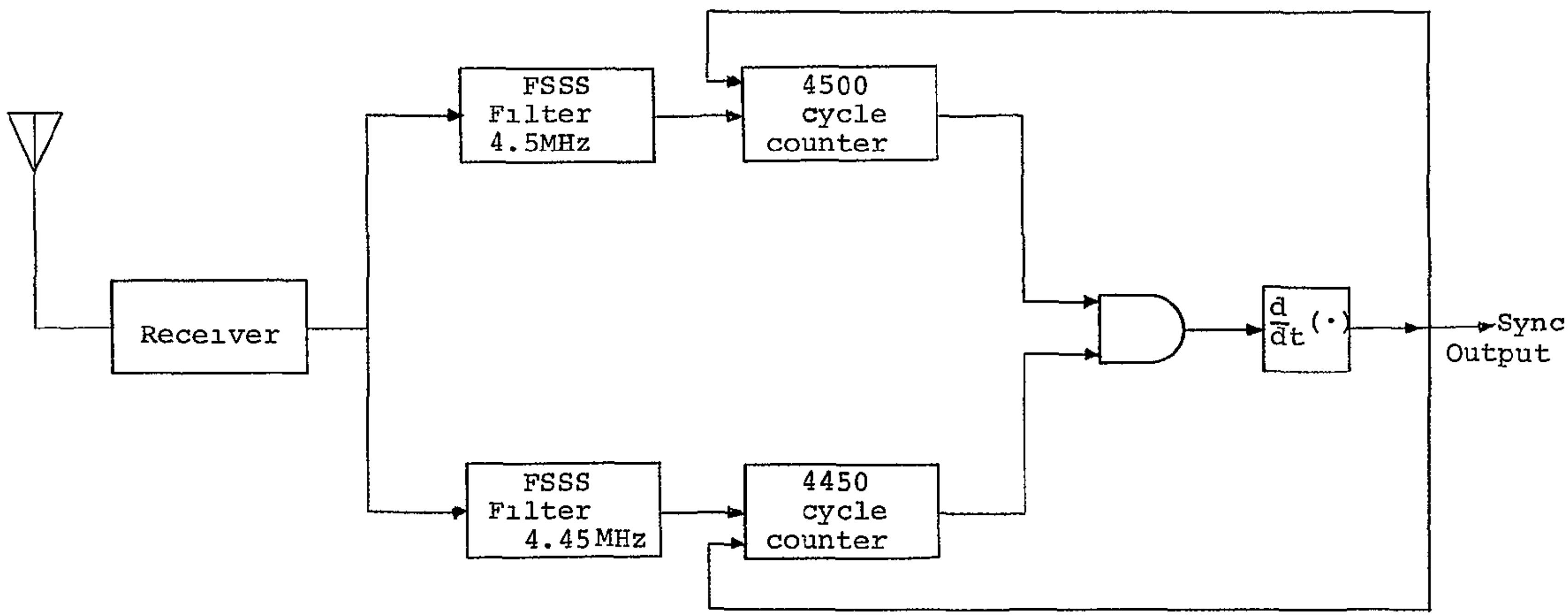


Figure 2.7

A possible decoder.

A synchronizing signal with frequency between the video frequency and FDM audio is chosen so that is satisfies the first first characteristic stated above. Two fixed frequencies say 4.5 MHz and 4.45 MHz with a difference of 50 kHz are transmitted simultaneously for one millisecond. The frame sequence synchronizing signal (FSSS) filter gives these outputs to the previously reset counters. A 4500 cycle counter is made to give output only on the 4500th cycle, and a 4450 cycle counter on the 4450th cycle.

These outputs go through the AND gate and the output [illegible] This pulse is also used to reset the counters to the "OFF" position. The synchronizing pulse so obtained can be fed to the synchronizer in the main receiving system, where it can be used to restart the operation.

2.2.6 Frame Repeating System

The basic frame repeating sequence for one of the video sub-channels is shown in Figure 2.8 Each video sub-channel signal is extracted from the TDM composite frame. Each television frame of the extracted signal of a channel is repeated until the next television frame of the same channel is received, and the same procedure is repeated for each successive frame. One of the possible arrangements for television frame repetition is shown in Figure 2.9.

With the switch S closed through A, the video input is coupled to the frame memory. With the switch S closed for transmission through B, the output of the memory is

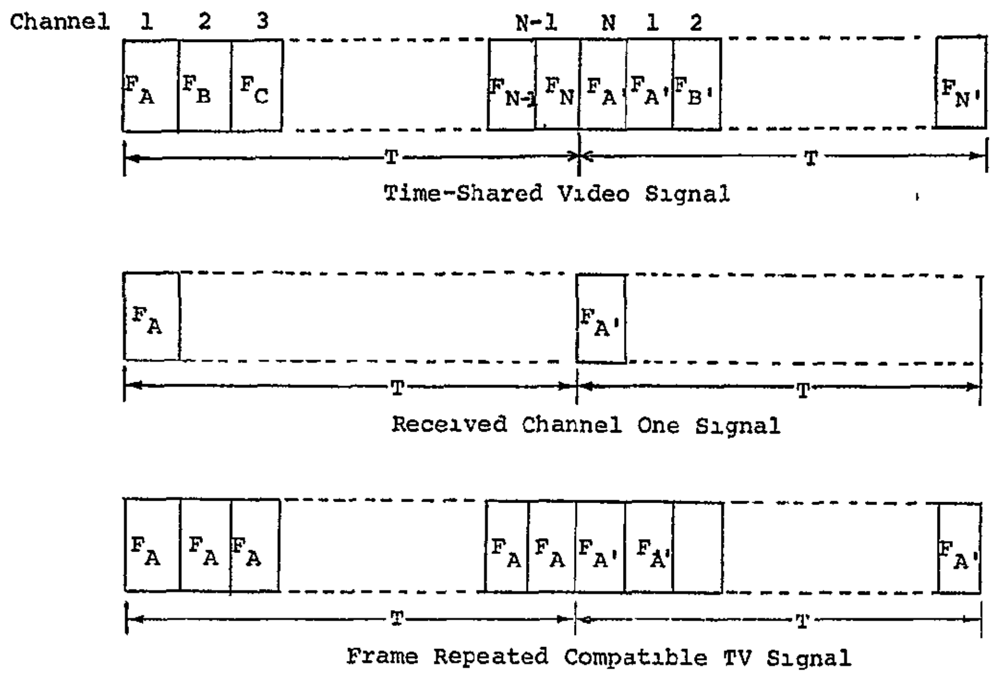


Figure 2 8

Basic Frame Repetition Sequence

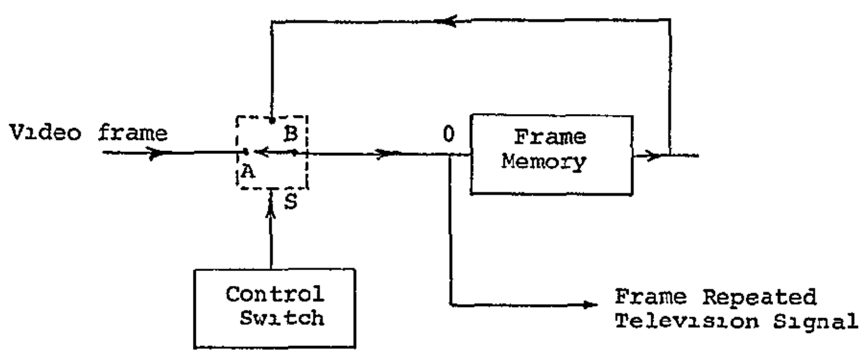


Figure 2.9

Frame Repeat System

coupled back to the input of the memory. If one frame is introduced in output and input of the frame memory, then the information previously stored in the memory is recirculated via path B. Thus a frame repeated signal can be taken out at the point 0.

Frame memory is an important component of this system. In some of the experiments done at Bell Laboratories (31,32, 33) concerning the frame repetition, a number of delay lines have been used for this purpose. For a low resolution, 160 line television frame storage system, high speed ultrasonic delay lines have been used (34) Each line has a delay of 4.2 milliseconds insertion loss of 34 dbs and a bandwidth of 3 MHz at a midband frequency of 5 MHz. A number of lines are used to give a total delay of one frame period. The other frame memory that can be suggested is a video magnetic recorder in which the writing and reading heads are arranged so that the readout is delayed by a frame period. The detailed technical considerations have to be investigated

Subjective measurements of the apparent increase in noise level due to frame repetition (32) indicate that this increase is small. It has been reported to be less than 3 dbs when a frame is repeated for less than 10 times. Further investigations about the subjective measurements of noise have to be done if a frame is repeated for a large time.

The following questions must be answered before such a modulation format can be put into practice: (i) what is the best and most economical frame memory?, (ii) how does the frame repetition affect the video signal quality?; (iii) how exact is the frame sequence synchronization. These questions are hard to answer analytically, but experimental tasks can possibly give reasonable answers.

3. TIME-SHARED-VIDEO, TIME-SHARED COMPRESSED-AUDIO, SPTV TRANSMISSION

In this method of SPTV transmission, time-sharing is used for both video and audio information with the audio information of each sub-channel time-compressed and sent with the corresponding video information. The audio time-compression is determined by the ratio of video bandwidth to audio bandwidth. The duration of the compressed audio information placed next to its video information is determined by the product of the composite frame duration and the audio compression ratio. The composite frame duration is defined as the time in which one video frame and its corresponding audio in compressed form is transmitted for each sub-channel. These things will be explained in detail later.

Like the time-shared video frequency shared audio, only one R F channel is required for this type of modulation. The sub-channels to be multiplexed are arranged sequentially in time, with the video information followed by the corresponding compressed audio information. Time compression is used to expand the bandwidth of audio information to the video information bandwidth. The audio information to be transmitted is stored during the entire composite frame interval. This information is recovered in a shorter duration and placed next to the sub-channel video frame. At the receiver: first, the video information with compressed audio information of different sub-channels separated; then the video and audio of each sub-channel is separated. The audio is expanded in time and combined with the corresponding video information to form a standard television signal.

This method offers some inherent advantages in terms of simplicity compared to one discussed previously. These will be clear in the subsequent sections and are mentioned here.

3.1 RELATIVE MERITS

1) Unlike the previously described system, where the total base-bandwidth increases as the number of sub-channels are increased, the same base-bandwidth can be used, irrespective of the number of channels. Of course, the picture updating time increases as the number of channels increases. Thus the problem of scan conversion considered, to keep the base-bandwidth fixed as the number of channels increases, can be avoided. This may not only offer more simplicity, but may be desirable on economic grounds.

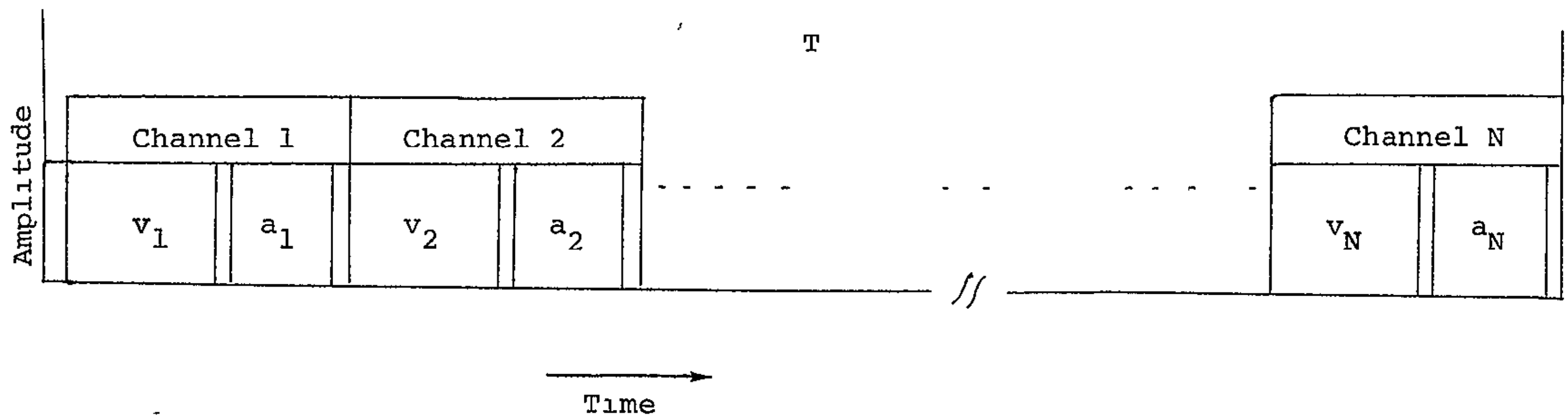
2) Each composite frame is complete in the sense that it contains all the audio-visual information required for that part and can be separated independently. Thus a channel selector, which selects the time-shared information along with a synchronizing unit (as discussed in section 3.3.1), a frame repeating system and audio storage system can possibly form as a front end receiver augmentation which can make the receivers direct from satellite receiving systems.

3) It offers the transmission system flexibility. Unlike the previous case, in which only one transmitting station can be used at a time due to technical requirements, a number of transmitting stations can transmit simultaneously because of the time-sharing techniques of both video and audio information. Of course, the synchronizing requirements become very important and have to be considered carefully.

3.2 COMPOSITE FRAME AND TIME ALLOCATIONS

Figure 3.1 shows the time allocations for N audio-still-video channels, allowing necessary time for guard bands and synchronizing bursts. The time axis of the diagram shows composite frame beginning with a synchronizing burst followed by transmission time for each channel. The transmission time for each channel includes the guard bands and the actual message time, which is video with compressed audio. A number of composite frames form the multiplexed system.

The duration of the composite frame is determined by the number of channels desired. Corresponding to each channel, one television frame time is assigned for the video information, while the audio information time is determined by the ratio of composite frame time and audio compression. A simple relationship between the number of channels, the composite frame time and the time required for the compressed audio information to be transmitted along with this can be derived as follows:



Composite Frame

$v_1, v_2, v_3, \ldots, v_N$ = still video frames transmitted during T.

$a_1, a_2, a_3, \ldots, a_N$ = time-compressed audio information corresponding to each video frame.

Figure 3.1

Channel Allocations for Time-Sharing of Video and Audio Information of Each Channel

If N = total number of channels desired,

T = composite frame time in seconds,

c_a = audio compression factor,

t_a = audio transmission time for each channel in one composite frame,

t_f = video transmission time for each channel in one composite frame, then since one television frame consists of an audio information equal to the length of the composite frame, and therefore

$$T = c_a \cdot t_a \qquad (1)$$

and also transmission time t, for each channel in one composite frame is

$$t = t_a + t_f, \qquad (2)$$

(assuming guard interval, $t_g << t_a$ or t_f)

Again, since N channels are transmitted in T seconds and therefore

$$N = \frac{T}{t_a + t_f} \qquad (3)$$

From Equation 1, we have

$$N = \frac{T}{\frac{T}{c_a} + t_f} \quad .$$

Now, if t_f is assumed as the time for one conventional television frame and c_a is found for certain audio frequency desired, a curve relating N and T can be drawn. Figure 3.2 is such a curve in which c_a is taken as 400. From this curve it can be seen that for a composite frame time of 10

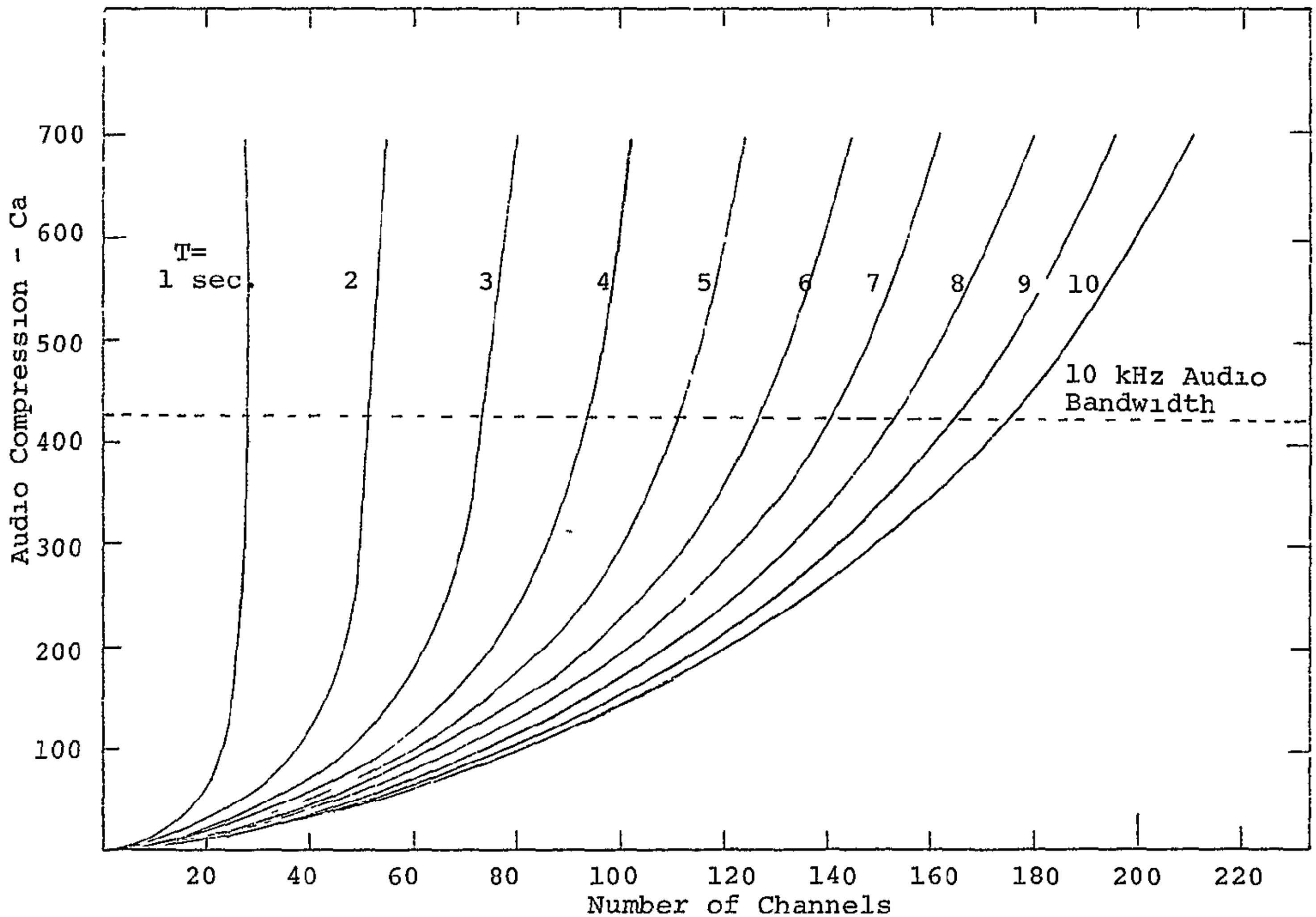


Figure 3.2

Number of Channels vs. Audio Compression Ratio for Different T seconds, as many as 170 channels can be transmitted simultaneously over one equivalent television channel. Since the audio information for each channel has to be stored for one composite frame time, the upper time limitation comes from this storage device.

Figure 3.3 shows the relation between the audio compression and the number of channels for different values of T. This gives the corresponding audio compression for certain values of T, and number of channels.

3.3 GENERAL TRANSMISSION AND RECEPTION

Figure 3.4 shows the timing diagram of the time-shared video and audio signals. Since the video and compressed audio information are sent sequentially, to synchronize the video information with incoming audio information at the receiver, video frame can be delayed by one frame period,. If this much delay, (one television frame = .33 ms.), is tolerable in audio information, it is not necessary. In the diagram video frame is shown delayed by one television frame. At a transmitter and receiver, the three main units, apart from the usual amplifying and modulating or demodulating units, are:

1) decoder and synchronizer unit
2) audio compression and expansion unit
3) video time sharing unit.

The block diagrams in Figures 3.5 and 3.6 show the transmitting and receiving scheme of this system. The decoding and synchronizing unit gives all the timing signals

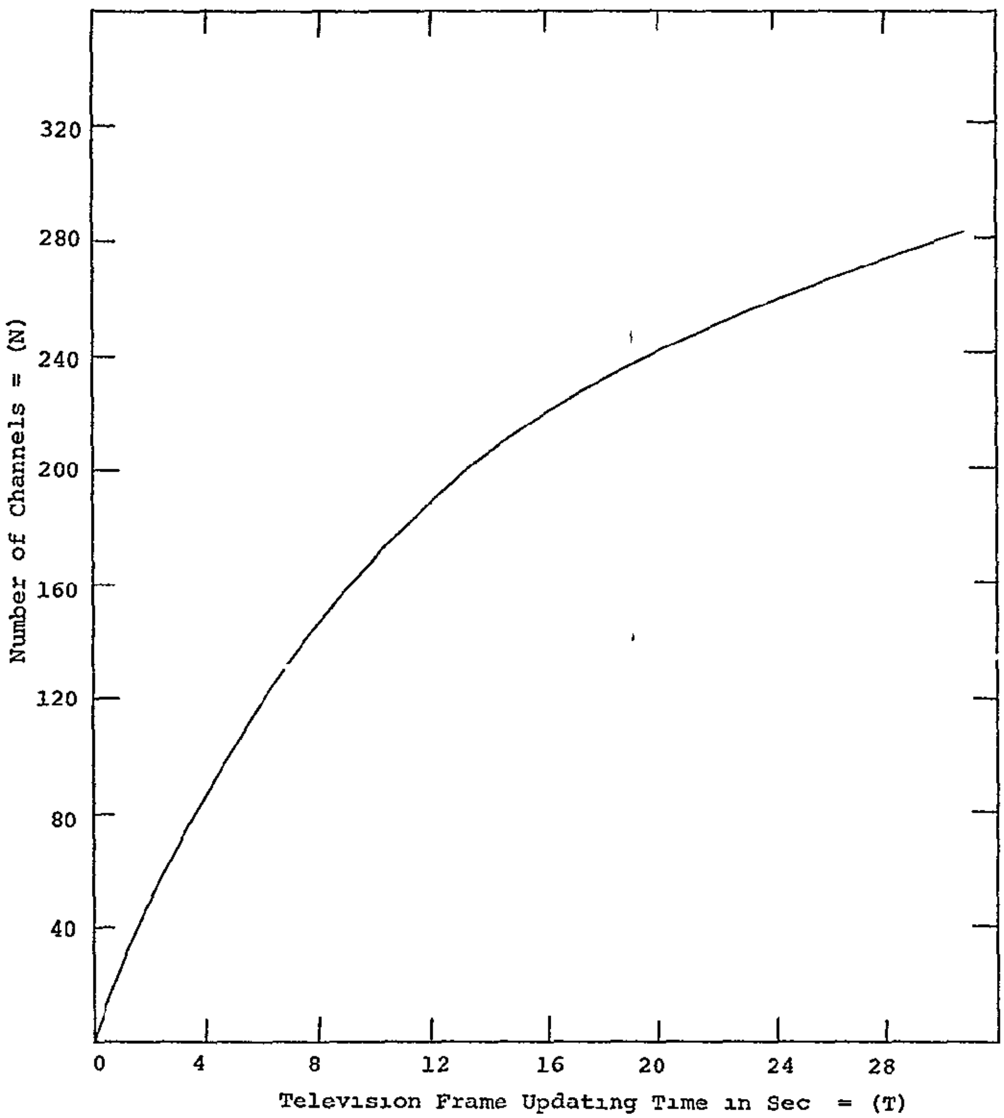


Figure 3 3

Number of Channels vs. Television Frame Updating Time for Audio Compression = 400, Television Frame Time = 1/30 sec [Sync and Guard Time neglected]

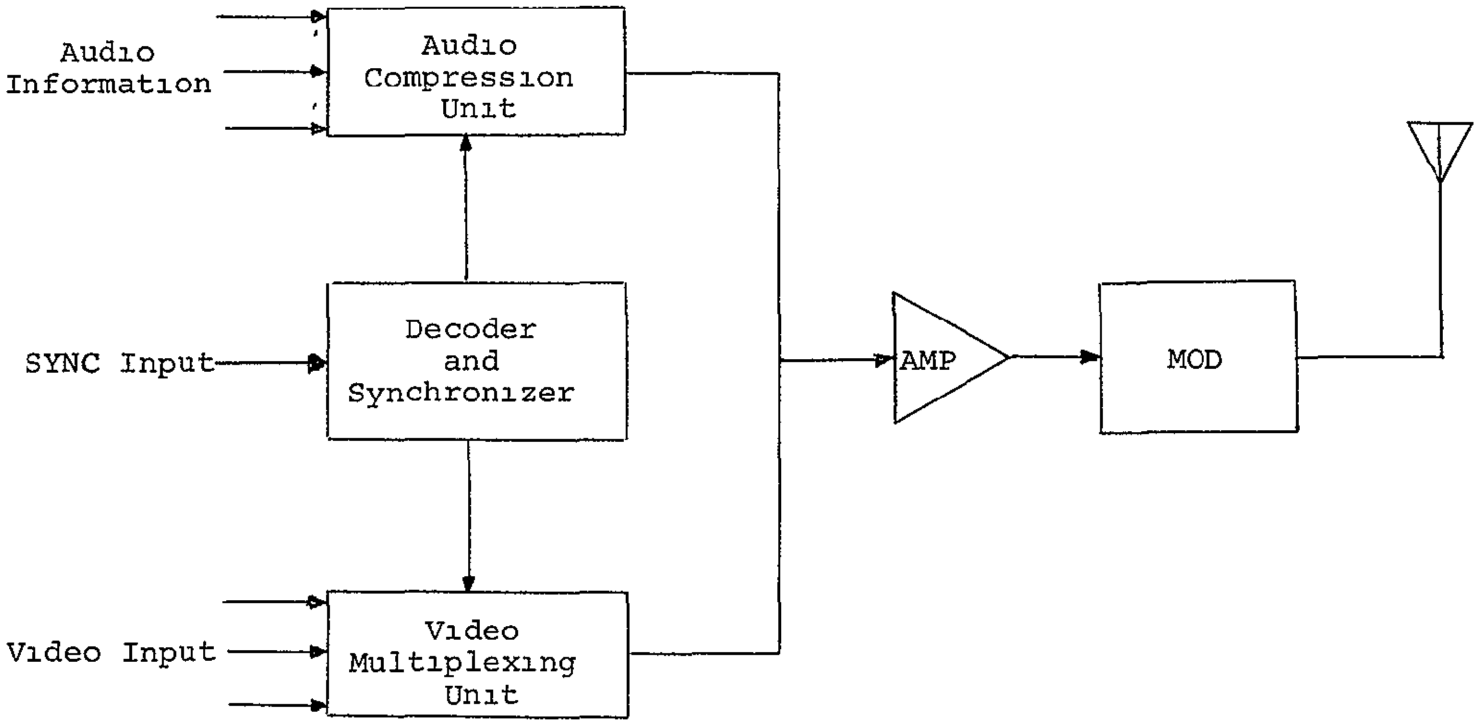


Figure 3.5

Transmitter

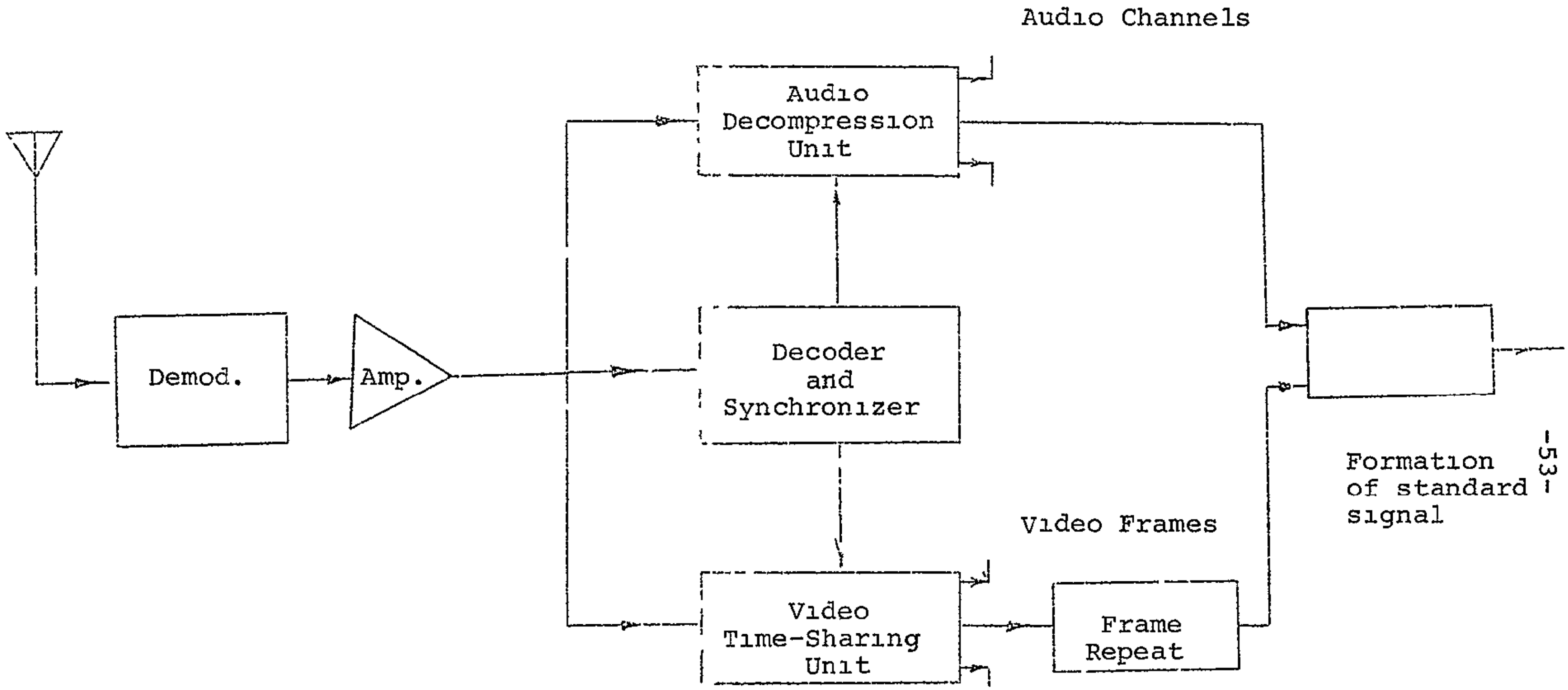


Figure 3.6

Receiver

required for video time-sharing and audio-video interleaving in synchronization with other such units. The decoder transforms the incoming synchronization burst to a pulse, which is used to synchronize the synchronizer with other synchronizers.

Audio compression and expansion consists of a number of stores. The audio information of each channel is written in the stores and read out at faster speeds. Thus an audio information of a longer duration is reduced to a short duration. An analog time-compression is employed for this method of transmission, in view of the following advantages:

1) maximum possible product of bandwidth and number of sound channels,

2) minimum storage requirements.

If this signal, for example, is first converted into digital form, then into a pulse modulated signal, and if this pulse signal would be transmitted in time-compression shape, then either the bandwidth or the number of sound channels capable of being transmitted would be substantially smaller and the storage requirements would be substantially higher.

The video time-sharing unit is a time-division multiplexing unit with an equivalent time slot equal to the time required for one television frame with its time-compressed audio information. A detailed discussion of these units is given in the subsequent sections.

3.3.1 Synchronization

Since there is an interleaving of time-shared video and corresponding time-compressed audio information, like other time-sharing systems, the information on absolute time is very important to ensure the relation between various channels. The synchronizing burst is one solution.

Synchronizing arrangements as suggested by Jacob and Mattern (36) seems to be suitable for this purpose. A synchronizing burst of certain duration is sent from a master controlled station. It is decoded and converted to a pulse at the receiving station. It is then fed into the local synchronizer unit which generates the required timing pulses.

A possible synchronizer for a composite frame length of 5 seconds is described here (Figure 3.7). This is based on reference 36. It employs a digital counter controlled by a master clock (crystal oscillator). The counter supplies the actual ON-OFF synchronizing signals that time the system operation. Figure 3.3 gives the number of channels for a frame time of 5 seconds with the desired compression ratio. For a compression ratio assumed as 420, the number of channels comes out to be 110. So 11,000 cycle counter is taken here to be controlled by a clock running at 2.2 MHz. The output of the clock is divided by 1000 providing 2.2 kHz pulses through gate G_2 to the 11,000 counter. As long as gate G_2 is enabled, the 11,000 cycle counter continues to count through 11,000 cycles, at which time it resets and counts again.

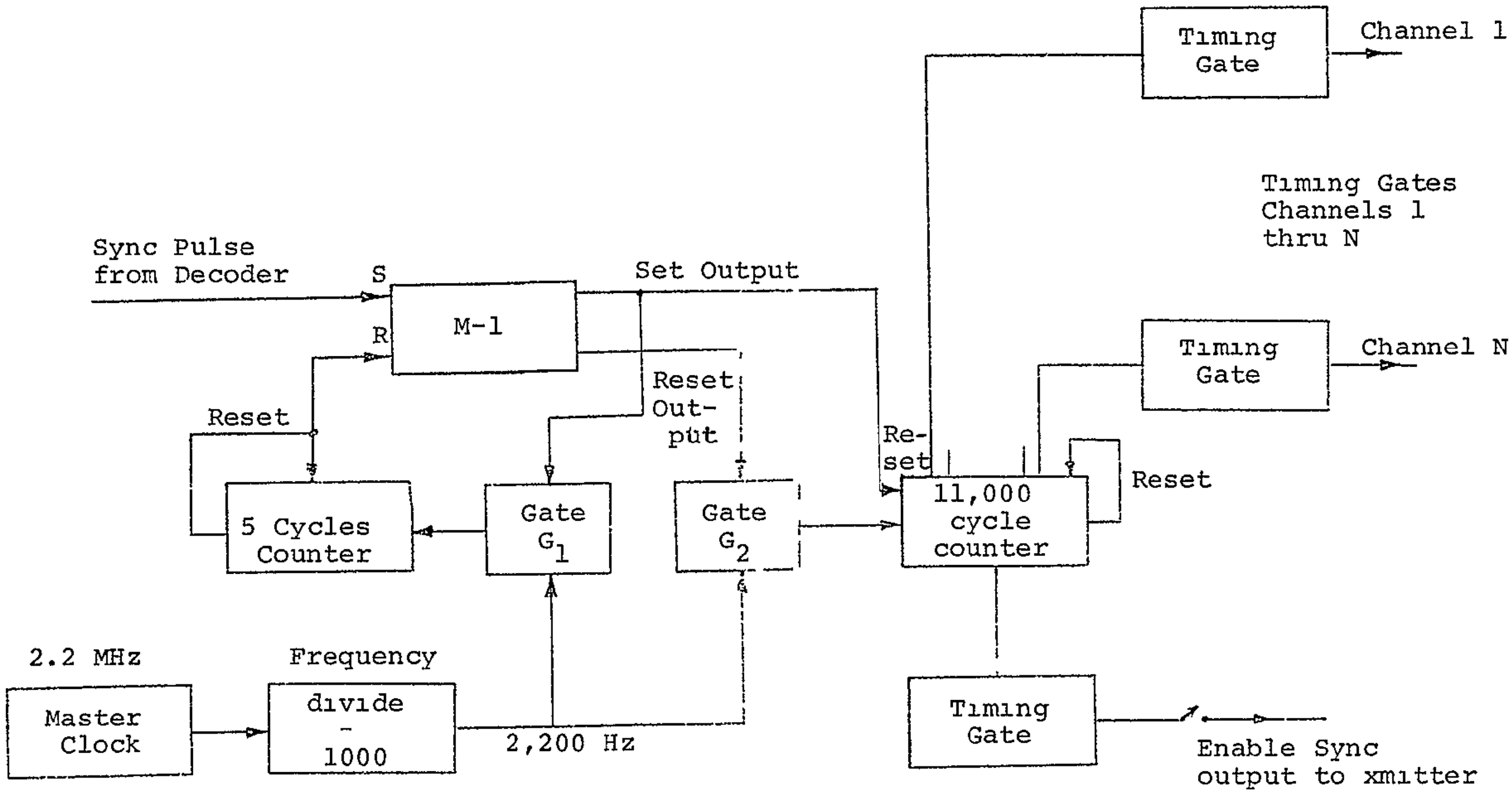


Figure 3.7

A Typical Synchronizer

If the synchronizing pulse is received from the decoder, (see Section 2.2.5), then a pulse is applied to the multi-vibrator M-1. In turn, the M-1 removes the enabling voltage from G_2, resets the counter, and enables the gate G_1 to apply the 2.2 kHz pulse to the 5 cycle counter. This counter resets after 5 seconds, resets M-1 to its original position, thus enabling the 11,000 cycle counter to begin the count again.

Thereafter, the synchronizing pulses come on cycle 10,995 of the 11,000 cycle counter. If the crystal oscillator does not maintain its frequency, the synchronizing pulses correct the 11,000 cycle counter by resetting the counter ahead or behind the 10,995 count as required.

During the absence of synchronizing pulses, the 11,000 cycle counter continues to free-run through 11,000 counts and provides proper synchronization as long as the master clock stays within the required tolerance.

The 11,000 cycle counter generates all the timing pulses required for the time-sharing of video and audio information. Timing gates (multiple input AND gates) for each channel develop a group of timing pulses. These pulses define the beginning and end of the frame voltages. These control voltages can be generated by multi-vibrators that are set and reset by appropriate timing pulses.

The synchronizer may act as a master control station by selecting a voltage from the 11,000 cycle counter at the proper cycle. This voltage would then be applied to the synchronization burst generator that provides the proper burst for transmission.

3.3.2 Audio Compression and Decompression Unit

This consists of a number of storage elements arranged in parallel (Figure 3.8). The audio information of each channel is written in these and read out at a faster speed in an appropriate time interval. A storage element with simultaneous read and write head is needed for each channel. However, the number can be reduced to only two for all the channels if Flood and Urquhart-Pullen's (35) approach is taken for audio time compression. This is explained as follows·

Figure 3.9 shows an audio time compression expansion unit with two storage elements at each station. At the sending terminal, signals of the N channels are sampled regularly by means of pulse trains, producing the amplitude modulated pulses on the common input lead 'a'. This lead is connected to two gates G_1 and G_2, which in turn are connected to the storage elements A and B. The output of these is connected to the gates G_3 and G_4, which operate on receiving the output from gates G_5 and G_7. The gates G_1, G_2, G_5, and G_6 are operated by the write and read waveforms A, B, and C shown in Figure 3.10. When the waveform A is on, storage element A writes through the gate G_1, and B reads through G_4 which is operated by the gate G_6, which in turn is operated by waveforms A and C. Gate G_6 operates only when A and C both are

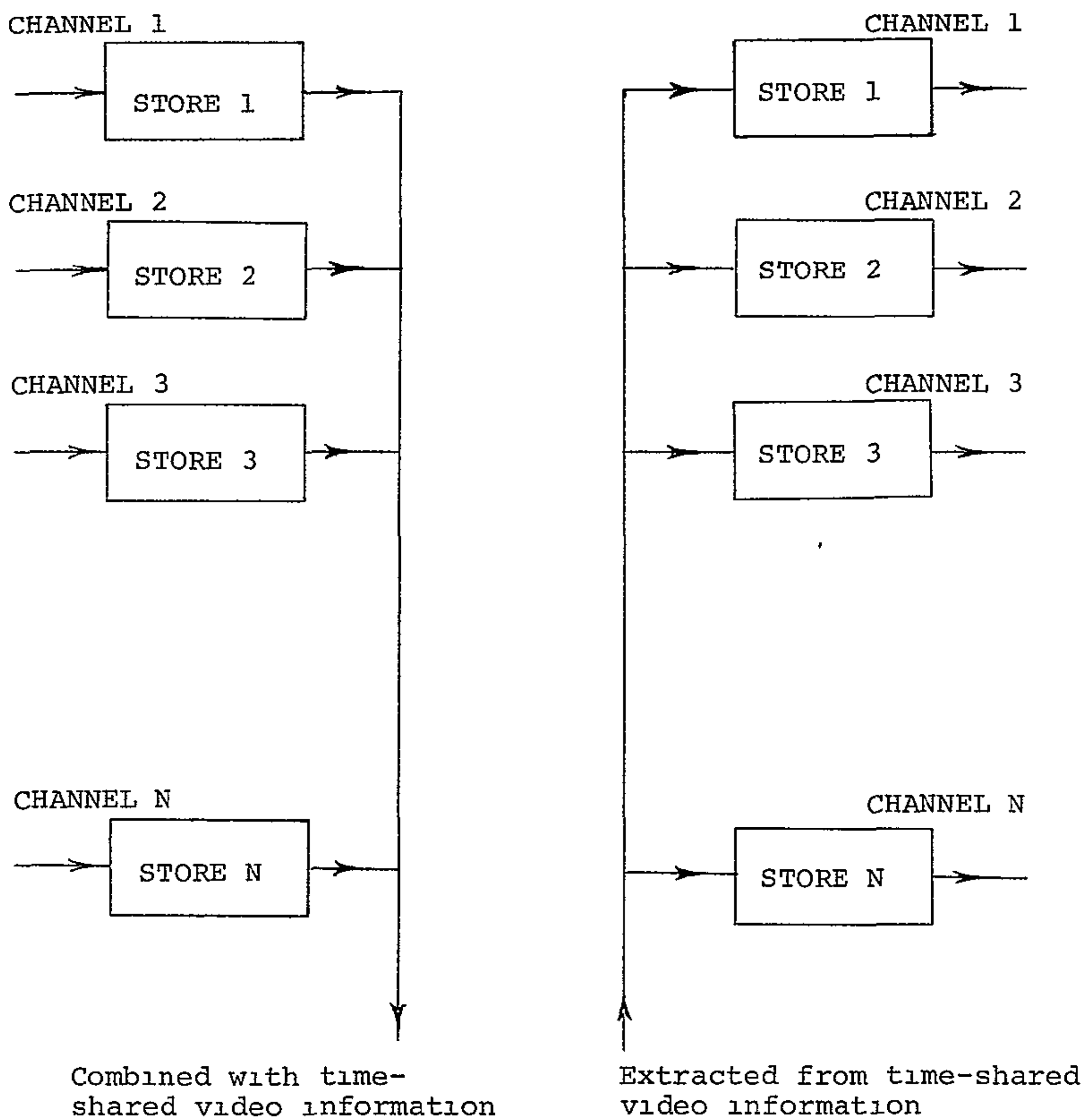


Figure 3.8

Elementary Audio Compression Decompression Unit

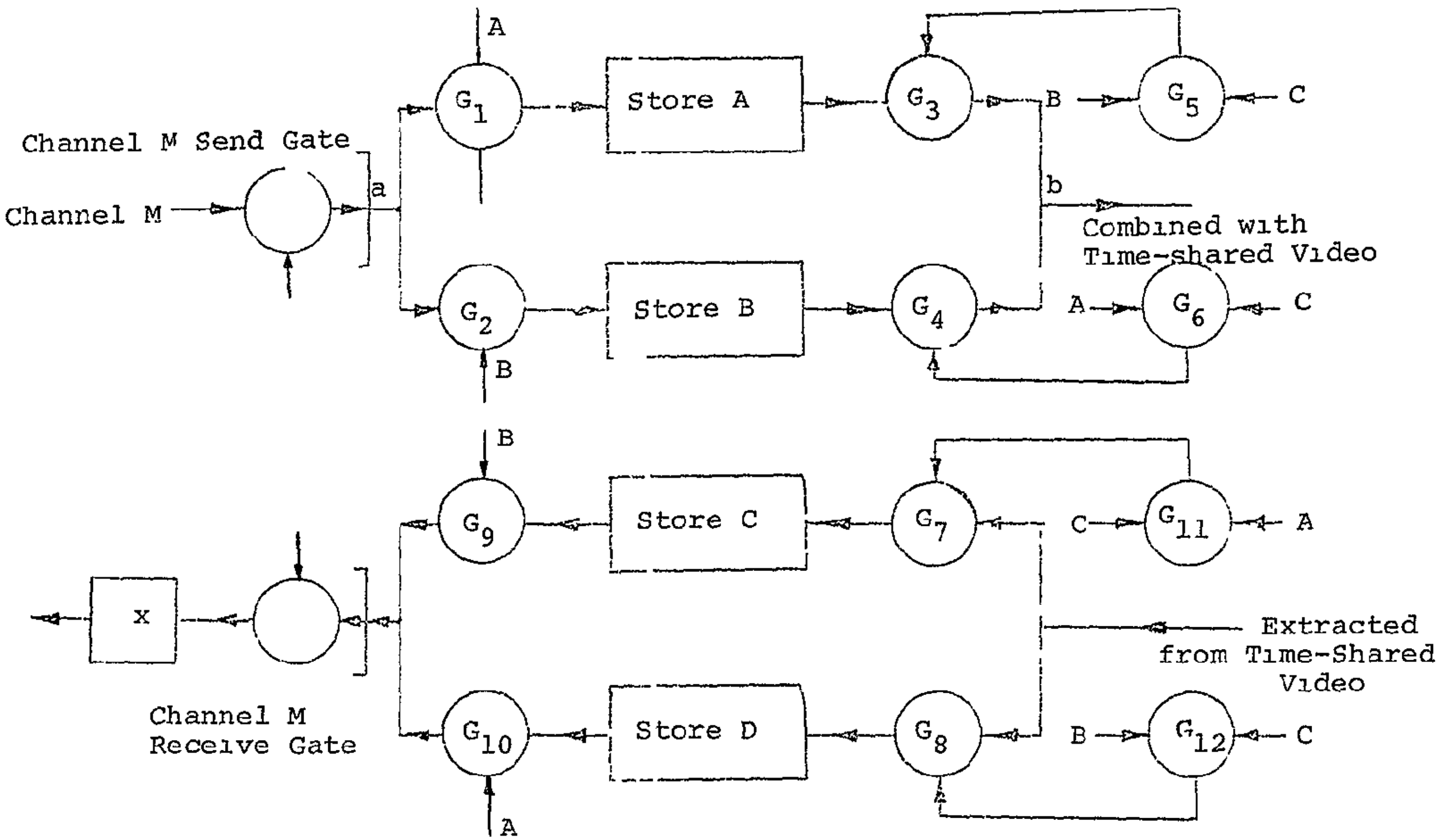


Figure 3.9

Transmission Reception with Common Storage Elements

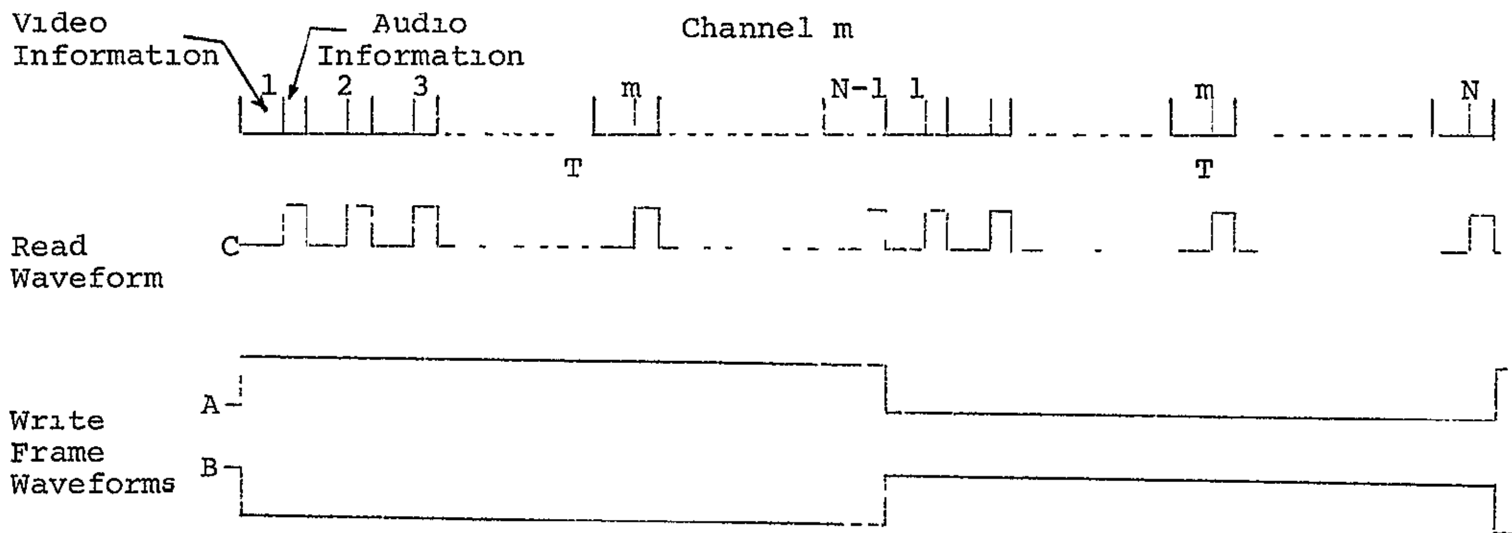
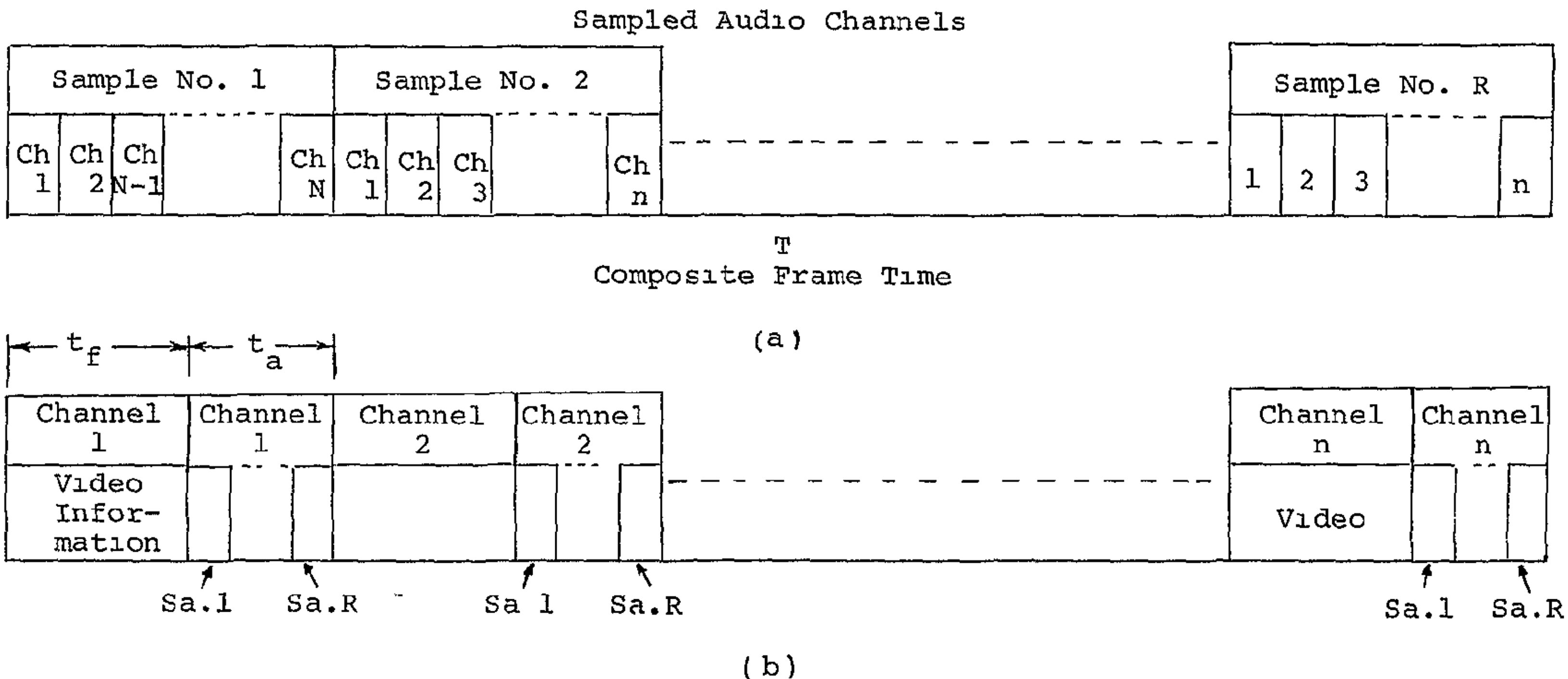


Figure 3.10

Timing Diagram for Common Storage Elements Arrangement



Sa = Sample
Ch = Channel

Figure 3.11

Rearrangement of Samples After Store

present and therefore it operates intermittently as shown in the timing diagram of read waveform C. Similarly, when storage element B writes, A reads intermittently, providing proper time intervals for video information. The information read from the storage element is then transmitted sequentially with video information.

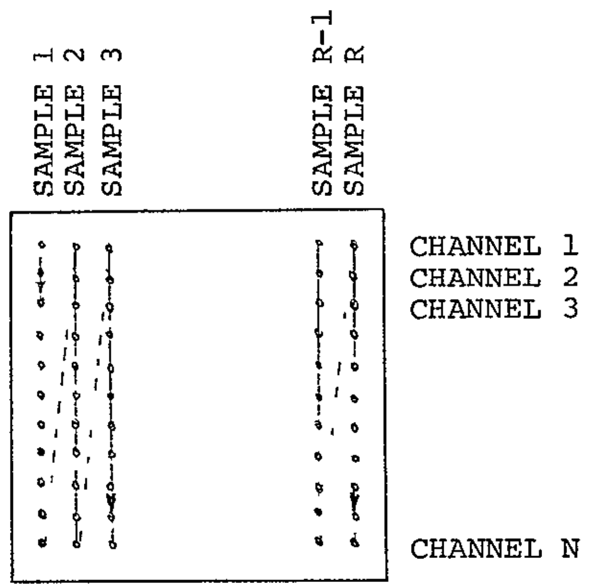
At the receiving terminal audio signals received from the common transmission path are extracted from the video waveform and just the inverse of the described modulation format is performed. The incoming intermittent time-compressed signal is written in the storage elements C and D through the gates G_7 and G_8 and is read at the previously written speed. The original audio signal is recovered from received amplitude modulated pulses by demodulating it by a low pass filter.

The channel send gates produce samples of each channel with successive samples of the same channel occurring every nth pulse. The samples written in the storage element are shown in Figure 3.11a. The function of the storage elements at the sending and receiving end is to change the order in which samples of the channel occur with faster speed. The faster speed thus creates the time needed for the video information. The sample read out from the storage elements occurs in the order shown in Figure 3.11b. The R successive samples of the one channel are followed by R samples of the next channel with a video gape in between them. The reverse process takes place at the receiving end.

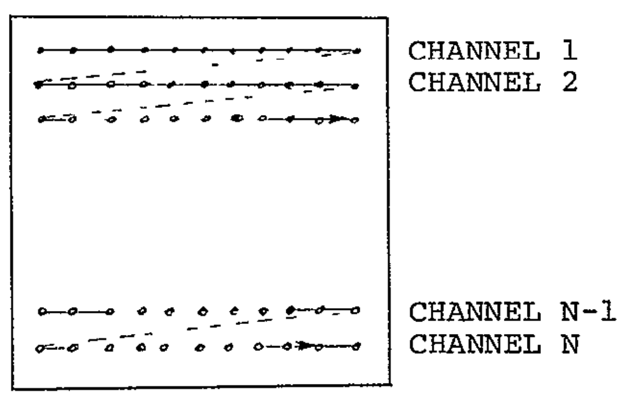
If the storage elements consist of rectangular matrix of storage devices as shown in Figure 3.12, input samples can be inserted in a vertical column, so that the successive samples of a channel occupy positions in the same horizontal row, or vice versa. The stored samples are read from each row with higher speed, thus providing the transmitting signal with video information space as shown in Figure 3.11b. Hence, it can be viewed as a reconstitution of the audio time division multiplexing into the speeded audio time-compression multiplexing.

It has been said that two storage elements are required both at the transmitter and receiver. The transmission paths through the storage elements may have slightly different gains, which in turn results in overall gain of every channel varying periodically and thus producing a distorted output. A detailed analysis of this is given in section 3.4.3 and it is found that for the distortion level to be 40 dbs below the signal level, the store gains must be within 1.4%. Therefore, to avoid this difficulty, a method that requires only one store seems more suitable. Flood and Urquhart-Pullen (35) have described a method in which one store is used and reading is interwoven with writing by using the vertical columns and horizontal rows of the store alternately.

As regards the storage devices which are required at the transmitter as well as the receiver, either cathode ray storage tubes or semiconductor storage devices seem to be



Scan for Writing into Sending Store
(or Reading from Receiving Store)



Scan for Reading from Sending Store
(or Writing into Receiving Store)

Figure 3.12

Scanning Patterns for Storage Matrix

adequate. The first practical use of cathode ray storage tubes appear to have been in the system of Jacob and Mattern (36). Cathode ray storage tubes have a high resolution and are potentially able to provide required storage (for example, Hughes' H-1213 resolution 1600 TV lines per diameter).

The use of semiconductor storage devices has been reported by various authors (37,38,39). Analog memory systems reported by Garsmann (38) and Flood and Urquhart-Pullen can be realized with integrated circuit techniques and may eventually be prefered to the storage tubes on economic grounds.

3.3.3 Video Multiplexing Assembly

Figure 3.13 shows the block diagram of a video multiplexing assembly. Each channel output is connected to a gate which is operated by the timing signals from the synchronizing unit. The time duration for which each gate remains on is also controlled by the synchronizing unit. The time-shared video obtained here is combined with the speeded time-compressed audio to produce the composite signal.

3.4 GENERAL SYSTEM CONSIDERATIONS

This is basically a time-shared system with video and analog time-compressed pulse amplitude modulated audio information sent sequentially. The composite signal is band-limited to video bandwidth. Since the same transmission path is subjected to both analog video signal and pulse modulated audio signal, the pulse response of the

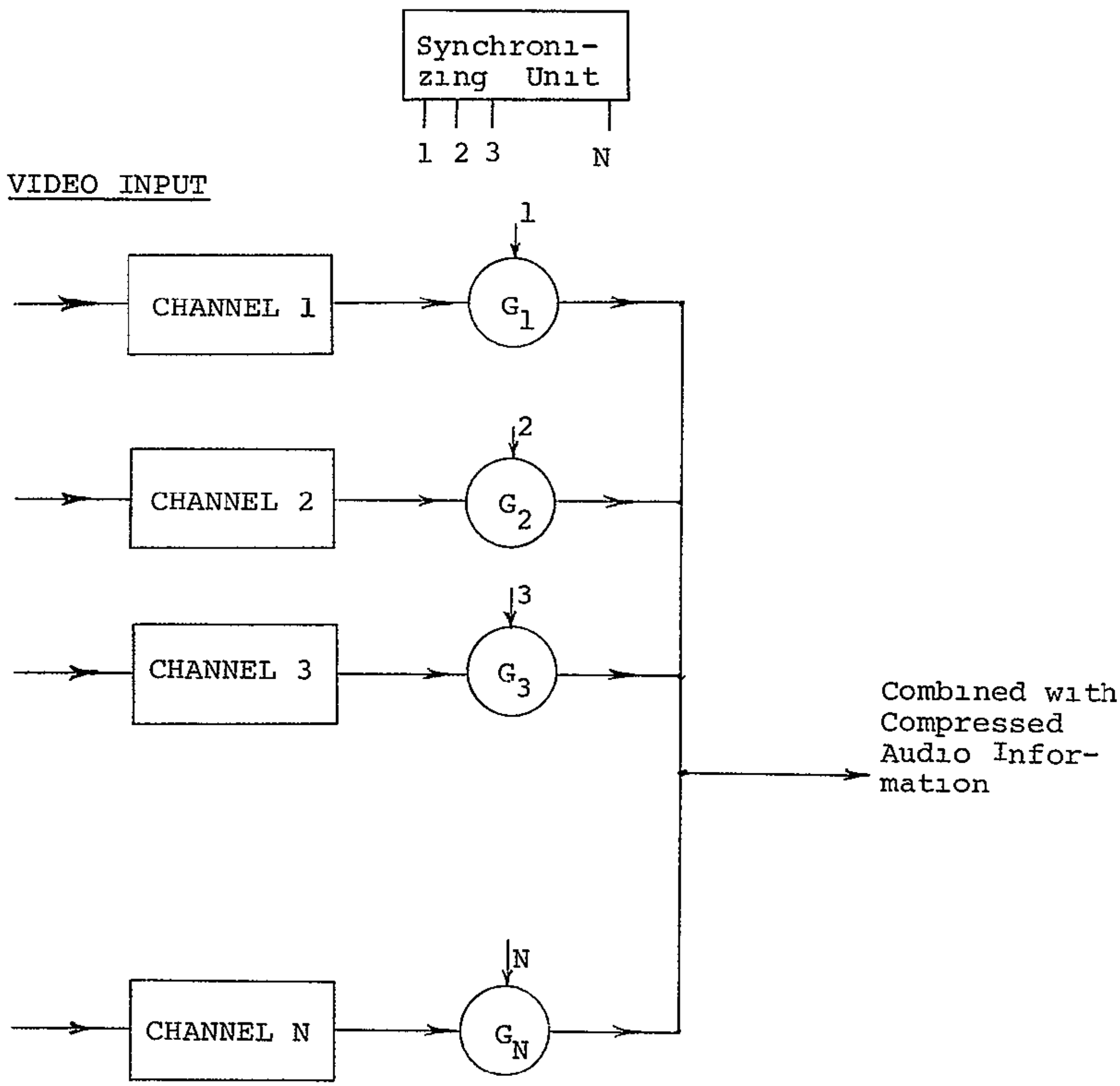


Figure 3.13

Video Multiplexing Assembly

transmission path has to be considered.

The basic characteristic of an idealized linear system may be expressed in the form

$$H(\omega) = G(\omega)e^{-j\phi(\omega)}$$

where $H(\omega)$ is transfer function of the system,

$G(\omega)$ is amplitude response of the system,

$\phi(\omega)$ is phase/frequency characteristics,

ω is impressed frequency in radians/second.

Again, since the system is band-limited, a band pass filter would describe the system characteristics. A transmission path with linear phase/frequency characteristic and Gaussian amplitude response seems to be a reasonable choice. In that case amplitude response is given by

$$G(\omega) = \exp\left[-\frac{(\omega\tau)^2}{2}\right] \qquad 3.4.1$$

The response of the transmission path to short pulse is approximate to its impulse response, which is given by

$$h(t) = h_0 \exp\left[-\frac{(t/\tau)^2}{2}\right] \qquad 3.4.2$$

where $h_0 = \dfrac{1}{\tau\sqrt{2\pi}}$

3.4.1 Audio-Video Crosstalk

In the audio time-compression scheme considered here the adjacent transmitted samples are associated with the same channel. The samples of the adjacent audio channel are separated from the previous one by the analog video information. When insufficient bandwidth and nonlinearities of the transfer characteristic of the transmission path cause the samples to spread in time, crosstalk is possible, unless a reasonable guard band is alloted between them. The calculation of the crosstalk and guard time can be carried out in several ways as described in reference (41). Staube (42) has given a method of evaluating crosstalk by considering separately two cases of insufficient high and low bandwidths. We will calculate crosstalk by using the model which we have postulated for the transmission path (35).

The crosstalk ratio, C_t, is defined as follows

$$C_T = \frac{\text{Magnitude of the signal of disturbed channel}}{\text{Magnitude of the signal of disturbing channel}}\ .$$

If Te = interval between the epochs of pulse amplitude modulated (PAM) audio pulse,

Tg = guard interval between the PAM audio and video (Figure 3.1),

and $p = \frac{Tg}{Te}$, then the crosstalk, considering the Gaussian amplitude response, is approximated from equation 3.4.2 as follows

$$C_T = \frac{h_0}{h_0 \exp\left[-\frac{1}{2}\ \frac{Te+Tg}{Y}\right]^2}$$

$$= \exp\left[+\frac{1}{2}\ \frac{(1+p)Te}{Y}\right]^2 \qquad 3.4.3$$

For a system of fixed bandwidth, there is a lower limit on the duration of pulses at the output. This minimum output pulse duration is related to the system bandwidth by (26)

$$Te \geq \frac{1}{2B_{eq}} \qquad 3.4.4$$

where B_{eq} is the system bandwidth and is approximated as follows (40) for Gaussian filter.

$$B_{eq} = \frac{1}{2\pi}\,\frac{1}{2G(0)}\int_{-\alpha}^{\alpha} G(\omega)\,d\omega$$

$$= \frac{1}{2\pi}\,\frac{1}{2}\int_{-\alpha}^{\alpha} \bar{e}\;\frac{\omega^2\tau^2}{2}\,d\omega$$

$$= \frac{1}{2\sqrt{2\pi}\,\tau}\quad \text{Hz} \qquad 3.4.5$$

and attenuation at this frequency is

$$\eta = 8.686 \cdot \frac{\left(2\pi B'_{eq}\tau\right)^2}{2}$$

$$= 8686 \cdot \frac{\pi}{2} \cdot \frac{1}{\tau^2} \cdot \tau^2 = 6.72 \text{ db.}$$

Therefore, from equations 3.4.3, 3.4.4, and 3.4.5, we get

$$C_T = \exp\ (1+p)\sqrt{2\pi} \qquad 3.4.6$$

Therefore, the crosstalk factor, C_T, in dbs.

$$C_{T_{db}} = 8.686 \cdot [(1+p)]\sqrt{2\pi} \qquad 3.4.7$$

If, for example, a crosstalk attenuation of 100 dbs is assumed, p=3.6, i.e., Tg=3.6Te. Thus, for a required crosstalk attenuation, the value of Tg can be found in terms of Te.

3.4.2 Relation Between Audio Bandwidth and Number of Channels

If the transmission path bandwidth is B_{eq} the minimum separation between their epochs is given by equation 3.4.6. For compressed audio transmission, the signal of each channel is stored at the sending terminal for a period T (see Figure 3.11a), the number of samples stored is waveform is a square wave with amplitude equal to the gain difference and period equal to 2T, where T is the composite frame period during which samples are stored in either of the stores. The received waveform is thus distorted as shown in Figure 3.14. The resulting signal to noise ratio can be calculated by considering the distortion waveform as a periodic square wave (35), with period equal to twice the storage time, i.e., 2T, which is amplitude-modulated by the signal.

Let us first consider the Fourier cosine series for an unmodulated voltage waveform with period 2T.

$$v(t) = bo + 2\sum_{n=1}^{\infty} bn \cdot \cos\omega nt$$

where $\omega = \frac{2\pi}{2T}$ and the coefficients bn are given by

$$bn = \frac{1}{2T}\int_0^{2T} v(t)\ \cos n\omega t dt$$

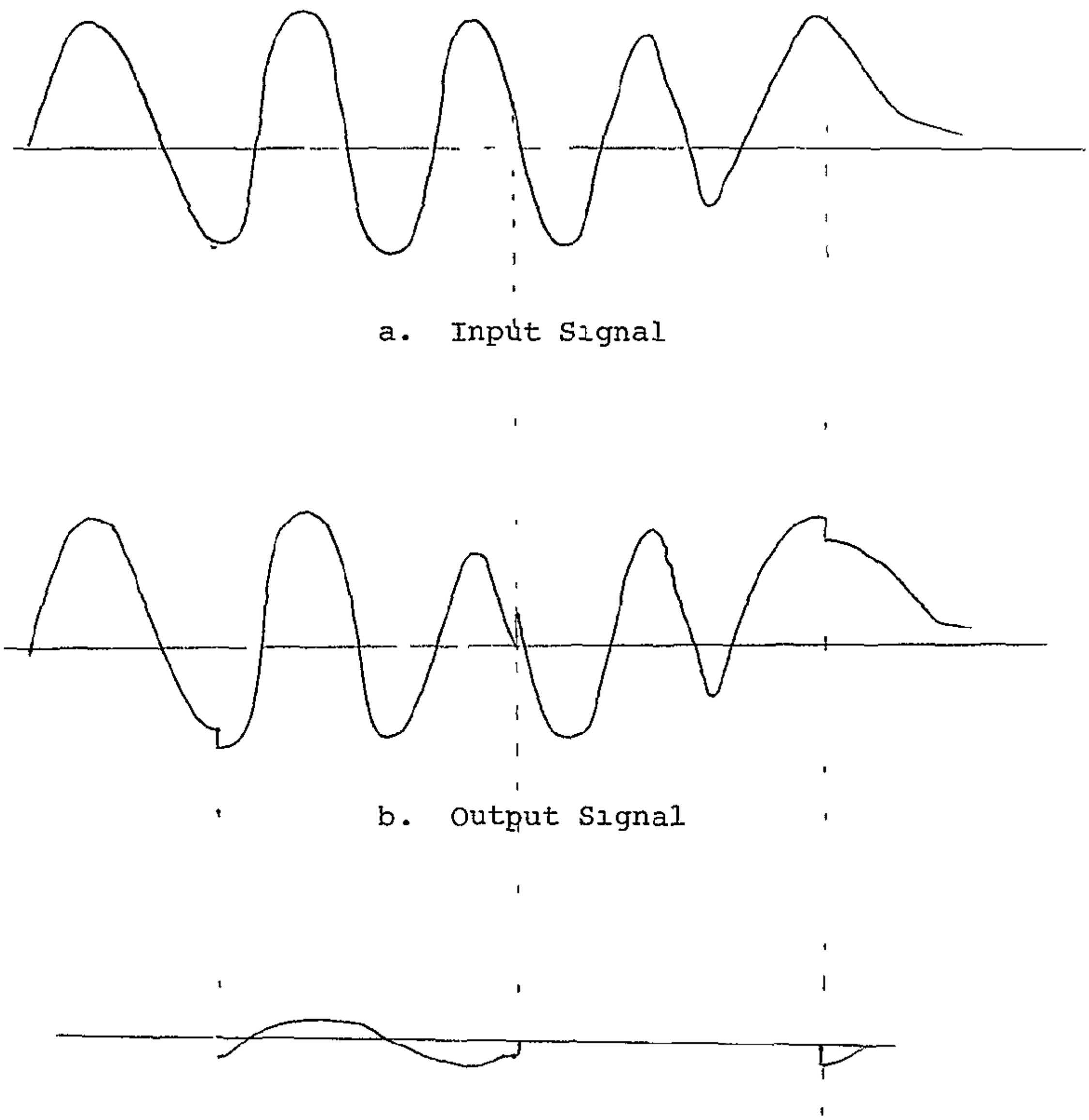
The power contained in the waveform is the total power contained in the d.c. component and all the harmonics. From Parseval's theorem, the power produced by v(t) in load of unit resistance is

$$\bar{v}^2 = bo^2 + \sum_{n=1}^{\alpha} bn^2$$

Therefore

$$\sum_{n=1}^{\alpha} bn^2 = \frac{1}{2T}\int_0^{2T} v^2(t)dt - bo^2 \qquad 3.4.9$$

The instantaneous amplitude of the above waveform is determined by the signal transmitted. For purposes of this analysis, this may be assumed to be



a. Input Signal

b. Output Signal

c. Distortion, i.e., difference between curve a and b.

Figure 3 14

Distortion in time-compressed Audio Information with Common Storage Elements

$$v_m(t) = A \cos(\omega_m t + \phi) \qquad 3.4.10$$

The output distortion waveform is thus given by

$$v_o(t) = v_m(t) \cdot v(t)$$

or

$$v_o(t) = Abo\cos(\omega_m t + \phi) + A\sum_{n=1}^{\alpha} b_n\{\cos[(\omega_m + \omega n)t + \phi] + \cos[(\omega_m - \omega n)t + \phi]\} \qquad 3.4.11$$

The first term merely represents a change in amplitude of the signal, but the other terms are unwanted products. Since $\omega_m > \omega$, the distortion products consists of upper and lower sidebands about the signal frequency, each sideband being a series of harmonics of frequency n . If the signal were a complex wave instead of a sinusoid, then such sidebands would be produced about each component of the signal.

The total distortion power $\bar{v}_n{}^2$ is the total power in the sidebands and thus

$$v_n{}^2 = A^2 \sum_{n=1}^{\alpha} bn^2 \quad .$$

Substituting from equation 3.4.9

$$v_n{}^2 = A^2\left[\frac{1}{2T}\int_0^{2T} v^2(t)dt - bo^2\right] \qquad 3.4.12$$

The output signal from equation 3.4.10 and 3.4.11 is

$$v_s(t) = A(1-bo)\cos\omega_m t + \phi$$

and the output power is

$$\bar{v}_s{}^2 = \frac{1}{2} A^2(1-bo)^2 \qquad 3.4.13$$

The output signal to noise ratio of audio is thus

$$\frac{S}{N} = \frac{\bar{v}_s^{\,2}}{\bar{v}_n^{\,2}}$$

where $\bar{v}_s^{\,2}$ and $\bar{v}_n^{\,2}$ are given by equations 3.4.12 and 3.4.13.

Now, if the unmodulated amplitude of the waveform, when common storage elements are used, is α, then $bo = \frac{1}{2}\alpha$ and

$$\frac{1}{2T}\int_0^{2T} v^2(t)\,dt = \frac{1}{2}\alpha^2$$

and therefore

$$v_n^{\,2} = A^2\left[\frac{1}{2}\,\alpha^2 - \left(\frac{1}{2}\alpha\right)^2\right] = \frac{1}{4}\,\alpha^2$$

and

$$v_s^{\,2} = \frac{1}{2}A^2\left(1 - \frac{1}{2}\alpha\right)^2$$

therefore

$$\frac{S}{N} = \frac{2\left(1 - \frac{1}{2}\alpha\right)^2}{\alpha^2} \qquad 3.4.14$$

From equation 3.4.14 it can be seen that for distortion level to be 40 db below audio signal level, the store gains must be equal to within 1.4%.

4. SUMMARY AND CONCLUSIONS

In the study reported here two main modulation formats, time-shared video, time-shared time-compressed audio have been proposed and investigated for still-picture television transmission. The issues explored are: (i) the number of still-picture television channels that can be realized in a limited video bandwidth, (ii) interrelation of various parameters to system constraints, such as maximum display time of still-picture to number of channels and consequently to the available bandwidth, (iii) a possible transmission and reception scheme for each of them, (iv) general system considerations for each system, for example, the trade-off between the picture quality and bandwidth, intermodulation and crosstalk considerations, etc.

From this research it is concluded that although all three formats discussed here have the basic characteristics for a real time still-picture transmission, the time-shared video, time-shared time-compressed audio seems to be a promising one. Independence of the audio-visual information of still-picture channel in a composite frame may be listed as one of the reasons. This gives a possibility of multiple station transmission on the lines suggested by Jacob and Mattern in their TICOSS system (36). Eventual preference of any system will have to be decided on experimental and cost analysis.

As far as the hardware is concerned, much of the technology needed for slow-scan is available in commercial market, while the other two formats need experimental exploration to evaluate the system performance for the desired purpose. Subsystems like the frame repeat system, synchronization system need experimental evaluation. The frozen noise problem in the frame repeat system is another issue on which very little has been done and needs to be pursued to evaluate more careful power consideration for various picture qualities. Since some of the hardware needed for the other two formats is either not developed or a very little documentation is available about them and so it is hard to evaluate their performance in terms of cost. Therefore, it seems hard to compare the actual system performance in terms of cost at this time. All that can be said now is that the formats described here are equally capable of still-picture television transmission, each with its own technical problems, and various solutions.

The still-picture transmission formats considered here are for real time transmission. A non real time format that can be suggested for still-picture transmission can be termed program multiplexed still-picture transmission. The basic principle of this is as follows. The still-pictures comprising a program are sent sequentially over a conventional video channel. The audio accompanying this video can be compressed and sent after the video information. The video and compressed audio information is stored at a receiving station. The video and audio information can be then converted to the required format at this receiving station and can be retransmitted for user display purposes. Again the actual performance can best be predicted by experimental evaluation along with a detailed technical analysis.

The still-picture television transmission schemes proposed in this report are idealized. Before they can be put into practice various technical and economic questions must be answered; some of these are:

(i) How does the audio time-compression affect the quality of audio signal?

(ii) What are the timing accuracies required for time-shared-video, time-shared audio?

(iii) What are the transmitter power trade offs for audio compression?

(iv) What guard-bands and other compensation must be incorporated to accommodate oscillator instability?

(v) What can be the possible cost of such a system?

(vi) How does the timing error affect the number of channels?

suitable answers to these and other questions can be obtained by further investigation of the proposed scheme.

5. ACKNOWLEDGMENTS

This work was supported by the National Aeronautics and Space Administration, Grant Number Y/NGL-26-008-054, Office of University Affairs, which the author gratefully acknowledges.

6. BIBLIOGRAPHY

(1) Godwin C. Chu and Wilber Schram, "Learning from Television, What Research Says," Stanford University, California, Institute for Communication Research, December, 1967, ERIC No. ED014,900.

(2) Rudy Bretz, Communication Media: "Properties and Uses," Rand Memo, RM 6070, NLM/PR, September, 1969.

(3) Draft Report E4t(X1) Space Services for Broadcasting Terminology, Conclusions of the Intrium Meeting of Study Group XL (Television) Palma-de-Mallora, 1968

(4) CCIR Document TV/176 (Rev. 1)-E, 1966-1969.

(5) Broadcast Satellite Service: Technical Considerations, Working Paper Submitted to Working Group on Direct Broadcast Satellites by the U.S. Delegation, United Nations General Assembly, Committee on the Peaceful Uses of Outer Space, 19 February 1969.

(6) CCIR Recommendation 417-1 Documents of XIth Plenary Assembly, Oslo, 1966.

(7) Television Broadcast Satellite Study, J. Jansen, P.L. Jordan et al., TRW Systems Group Prepared for NASA, Contract NAS3-9707.

(8) "Engineering Aspects of Television Allocation," Report of the Television Allocations Study Organization to FCC, March 1959.

(9) L.D. Collins, Signal-to-Noise Ratios for Television Transmission, Technical Note, Lincoln Laboratory, M I.T., 14 March 1969, ESD-TR-69-50

(10) Barstow and Christopher, "The Measurement of Random Video Interference to Monochrome and Color Television Picture," AIEE Transactions, November, 1962.

(11) C.E. Haehnle, "The Optimum Value of the Aural-to-Visual Power Ratio for Television Stations in the United States of America," IEEE Transactions on Broadcasting, July, 1965.

(12) H.E Ennes, "Slow Sweep TV for Closed Circuit Use," Electronics, 29, No. 11, p. 140, (November, 1956).

(13) S.K. Alles and H.E. Reed, "Slow-Scan Adapter for Conventional TV Signals," Electronics, 30, No. 6, p. 153, (June, 1957).

(14) K.E Farr, "Phonovid - A System for Recording Television Pictures on Phonograph Records," Journal of Audio Engineering Society, Vol. 16, No 2, April, 1968

(15) Lee Stratton, "Reviewing Slow-Motion Disc Principles," Broadcast Engineering, Vol 11, No. 2, pp. 14-18, February, 1969.

(16) J M. Kuljian, "A Random-Access Audio-Picture Retrieval System, Journal of SMPTE, Vol. 78, October, 1969.

(17) Sid Deutch, "Pseudo-Random Dot Scan Television System," IEEE Transactions on Broadcasting, Vol. BC11, No. 1, pp. 11-21, July, 1965.

(18) Sid Deutch, "Narrow-band Television Uses Pseudo-Random Scan," Electronics, Vol. 35, pp. 49-51, April, 1962.

(19) "Possibilities of Reduced Television Bandwidth," IRE Tranactions on Broadcasting and Television Receivers, Vol BTR2, 69-82, October, 1956.

(20) W.F. Schreiber, "Effect of Scanning Speed on the Signal to Noise Ratio of Camera Tubes," Proceedings of the IEEE, p. 217, 1964.

(21) T. Miyoshi and K. Matsumoto, "Simultaneous Transmission of Color Television and Sound Programs," Japan Telecommunication Review, Vol. 11, No. 3, 1969.

(22) CCIR Recommendation, Vol. 4, Pt. 1, p. 133, Oslo, 1966.

(23) D.K. Sachder, "Multiple Sound Channel Trade-offs for Satellite TV Broadcast Systems," IEEE Transactions on Broadcasting, Vol. BC16, No. 2, June, 1970.

(24) "An Indian Domestic Satellite System," Hughes-INCOSPAR, Report SSD 90205R.

(25) J. Jansen, P.L. Jordan, et al, "Television Broadcast Satellite Study," Final Report NASA CR72510, October 24, 1969.

(26) A.B. Carlson, Communication Systems, McGraw-Hill Book Company, 1968.

(27) C.C.I.T.T. IIIrd Plenary Assembly, Geneva, 1964, Redbook, Vol V.

(28) C.C.I.T.T. IIIrd Plenary Assembly, Geneva, 1964, Bluebook, Vol. III.

(29) CCIR "Document of the XIth Plenary Assembly," Oslo, 1966, Vol. IV, Part 1.

(30) D.H. Fender, "Control Mechanisms of the Eye," Scientific American, Vol. 211, pp. 26-33, July, 1964.

(31) F.W. Mounts, "Low-Resolution TV, An Experimental Digital System for Evaluating Bandwidth Reduction Techniques," BSTJ, pp. 167-198, January, 1967.

(32) F.W. Mounts and D.E. Pearson, "Apparent Increase in Noise Level When Television Pictures are Frame Repeated, BSTJ, pp. 527-539, March, 1969.

(33) F.W. Mounts, R.C. Brainard, and B. Prasada, "Low-Resolution TV: Subjective Effects of Frame Repetition and Picture Replenishment," BSTJ, pp. 261-271, January, 1967

(34) A.H. Meitzler, "Ultrasonic Delay Lines for Digital Data Storage," Transactions of the IRE, Professional Group on Ultrasonic Engineering, UE-9, pp. 1-8, 1962.

(35) J.E. Flood and D.I. Urquhart-Pullen, "Time-Compression-Multiplex Transmission," Proceedings of the IEE, Vol. 111, No. 4, pp. 647-668, April, 1966.

(36) M.L. Jacob and J. Mattern, "Time-Compressed Single-Sideband System (TICOSS)," IRE Transactions on Communication Systems, pp. 2-8, June, 1958.

(37) J.D. Heightley and A.J. Perneski, "An 80 Nanosecond Plated-Wire Store for a Time-Compression Multiplex Transmission System," IEEE Journal of Solid-State Circuits, Vol. Sc3, No. 3, pp. 276-279, September, 1968.

(38) G.G. Garsmann, "Twelve Sound Channels During the Vertical Sync Interval of the Television Signal, IEEE Transactions on Broadcast and Television Receivers (BTR), pp. 318-324, November, 1970.

(39) J.E. Flood and D.I. Urquart-Pullen, "Gated Capacitor Store for t.c.m. Transmission," Proceedings of the IEE (England), Vol. 111, No. 4, April, 1964.

(40) A.B. Carlson, Communication Systems: An Introduction to Signals and Noise in Electrical Communication, McGraw-Hill Book Company, 1968.

(41) F.P. Panter, Modulation Noise and Spectral Analysis, McGraw-Hill Book Company, 1965.

(42) H M. Straube, "Dependence of Crosstalk on Upper and Lower Cut-off Frequencies in PAM Time Multiplexed Transmission Paths," IRE Transactions on Communication Systems, September, 1962.

(43) W.T. Bisignam, G.P. Richards, and J.W. Whelan, "The Improved Gray Scale and Coarse-Fine PCM Systems, Two New Digital TV Bandwidth Reduction Techniques," Proceedings of the IEEE, Vol. 54, pp. 376-390, March, 1966.

(44) C. Cherry et al., "An Experimental Study of Possible Bandwidth Compression of Visual Images," Proceedings of the IEEE, Vol. 51, pp. 1507-1515, November, 1963.

(45) B. Prasada, "Some Possibilities of Picture Signal Bandwidth Compression," IEEE Transactions on Communications Systems, Vol. CS11, pp. 315-328, September, 1963.