A Time Compression Multiplex System For Multiple Video and Data Distribution Using Existing Satellite Channels

Donald Kirk, Jr.

Digital Communications, Inc.

Abstract - A multiplex system is described in which sequential segments of two video signals are time compressed and interleaved in analog form for transmission over a satellite transponder. Channel information rate and S/N are considered. The Time Compression Multiplex approach is extended to data plus video and two way video cases. A signal enhancement system for S/N improvement is proposed.

Introduction Starting with WESTAR I in April of 1974, four domestic satellites have been placed in geostationary orbit to serve miscellaneous customers in the continental United States. They make available a total of 72 transponder channels, each of 36 megacycles bandwidth, for transmission of various kinds of signals. A number of services including multi-channel telephone and high rate digital data transmission have developed to use these channels. At present, one of the fastest growing users of satellite channels is the CATV industry with its need for color video transmission.

Because of the high cost of satellite transponder channels it is natural that we should question the efficiency with which they are used. For multi-channel audio or data the efficiency of time bandwidth utilization controls the selection of the multiplex method chosen to assemble the various signals for transmission through a common path. A number of such multiplex systems are described in the literature. (1)

The situation for video transmission is more complex. Several systems which can transmit two video signals over one transponder channel are suggested in the literature, $^{(2)}$ but none of them are in widespread use. It is the purpose of this paper to propose another system called Time Compression Multiplex which may overcome some of the problems encountered by the earlier systems.

The permissible cost of a multiplex system can be examined by considering the case of an operator who is currently delivering one channel of video via one satellite transponder to N earth stations. To deliver a second channel he may lease a second transponder and provide a second receiver at each earth station. The cost per earth station C is then:

 $C = R + \frac{T}{N}$, where R is the cost of a receiver and T is the present value of a transponder chan nel. A useful multiplex system should have a receiving terminal cost of less than C which is in the range of \$1.7 x 10⁴ to \$4.4 x 10⁴ as N varies from 300 to 100. (Assuming R = 4 x 10³ and T = 4 x 10⁶). If a value is assigned to spectrum conservation as well as to hardware the allowable terminal cost would be considerably larger.

Frequency division multiplex, or Frequency Division Multiple Access as it is called in Satellite circles, has been widely used for transmitting a large number of telephone signals over a single transponder channel. Figure 1 is a block diagram showing how this technique might be used to carry two video channels with one transponder. Unfortunately, when two independent signals are passed through the satellite channel the average power output must be reduced by something of the order of 6 dB to reduce the intermodulation products to acceptable values. In addition, the FM deviation must be reduced by about 8 dB because of the reduction in available bandwidth. For performance comparable to a single channel system with a 15 foot dish, a two channel system would require an antenna of over 50 ft. diameter.

Other approaches to Frequency Division Multiple Access experience the same signal to noise ratio problem. For example, let the proposal be to use two 6 MHz AM channels instead of the 18 MHz FM paths. The noise bandwidth of the receiver is now less, but there is no FM improvement, and the back-off in satellite power must be greater to prevent intermodulation problems.

The intermodulation problems of Frequency Division Multiple Access can be eliminated by going to some time division system in which the two information streams do not use the transmission path simultaneously. In order to examine the available Time Division Multiplex possibilities we need to first establish the data rates involved.

Channel Capacity 5 MHz video channel and use of an eight bit PCM code yields a data rate of 80 Megabits/sec. or 160 Megabits for two video channels.

Application of Shannon's Theorem for a band limited channel in the presence of white noise

(1) $R = B Log_2 (1 + S/N) bits/sec.$

to a satellite down-link of bandwidth

B = 36 MHz

and a carrier to noise ratio

S/N = 12 dB

yields a limit for the information rate of

R = 146.7 Megabits/sec.

Clearly, two signals requiring a total data rate of 160 Mb/s cannot pass through this channel.

Let us examine the video channel more closely to see whether the conventional PCM encoding approach was efficient. We know that a useable video signal can be passed through a circuit of about 4.2 MHz bandwidth with a delivered signal to noise ratio of about 46 dB P-P/rms. Reducing this to an rms/rms value of 37 dB and using (1) again we find that the greatest data rate which the video channel can have is 51.6 Megabits/sec.

Two different approaches have been reported in the search for a more efficient method for encoding a video channel. In one approach the video signal is stored and examined at the transmitting end of the system. To the greatest extent possible, redundancy is removed from the signal, and what is left is encoded and transmitted.

In the second approach alternate frames of each video signal are deleted at the transmitting terminal and the remaining frames are time interleaved for transmission on a common path. At the receiving end the incoming data is stored and made available to a computer which is programmed to predict what the missing frames probably were.

Using the first system, good picture quality has been reported with data rates of 33 to 43 Megabits per channel. Two channel operation over a single transponder with a 15 ft. receive dish has been demonstrated. Some picture degradation is reported with the second approach. With each of these systems some costly storage and computational hardware is required.

The TCM System If the second system is modified somewhat we can avoid the necessity for omitting part of the incoming data at the transmitting terminal and predicting its probable value at the receiver. Figure 2 shows an arrangement in which alternate blocks of data from a video input, A, are stored in two stores A-1T and A-2T. The same treatment is given a second signal, B. The stores are read sequentially into a common path in the order shown. The reading speed is twice the real time writing speed used to record the data in the store. Thus, in the common path, time is apparently compressed to half its real value, and spectral components of the incoming signals are doubled in frequency.

At the receiving terminal the process is reversed. The signals are read into the stores at high speed and read out into separate lines in real time. We will call this complete process Time Compression Multiplex (TCM).

To investigate the ability of a satellite channel to carry the TCM signal, recall that the signal to noise ratio of the FM channel is related to its carrier to noise ratio by

$$(2) \frac{S_{\rm rms}}{N_{\rm rms}} = \frac{C}{N} \left[-\frac{3}{4} \left(\frac{\Delta f}{f_{\rm m}} \right)^2 \frac{B}{f_{\rm m}} \right] \left[\frac{(2\pi f_{\rm m}T)^2}{3} \right]$$

where Δf is the peak deviation f_m is the highest modulating frequency B is the I F bandwidth and T is the de-emphasis time constant For our example let $f_mT = 1.25$; in which case the third factor, giving the effect of use of pre and de emphasis, is 13.1 dB.

Let the deviation and the highest base band frequency be related to the bandwidth by Carson's rule.

(3)
$$B = 2(\Delta f + f_m)$$

Using (1), (2), and (3) we can plot the information capacity of an FM channel as a function of highest modulating frequency. This has been done in Figure 3. For the single channel case where $f_m = 4.2$ MHz, we have an excess channel capacity of about 17 Megabits. For the dual channel case assuming f_m of 8.4 MHz the excess channel capacity is about zero. (In both cases we are assuming that the required data rate for the analog channel is 51.6 Mb/sec. as previously calculated.)

For S/N \gg 1, we may restate Shannon's Theorem, (1), as

$$R_1 = B \operatorname{Log}_2(S/N)_1 \quad \text{and} \\ R_2 = B \operatorname{Log}_2(S/N)_2$$

By subtraction

$$\underline{\Delta} \underline{R} = \text{Log}_2 \begin{bmatrix} S/N_1 \\ S/N_2 \end{bmatrix}, \text{ where } R_1 - R_2 = \Delta R.$$

Then

$$\frac{\Delta R}{B} = 10 \operatorname{Log}_{10}^2 = \Delta S/N \text{ in dB,}$$

is the relation between excess channel capacity expressed as bit rate and available noise margin expressed as S/N in dB.

It can be seen that unless the data rate of the two channel TCM system can be somehow reduced, the system will have no noise performance margin.

The previous computation which indicated that a video channel could represent a data rate of as much as 51.6 Mb/sec. assumed that all times were equally important in the video signal. This is not necessarily true. The active portion of the video scanning line occupies only 108 ths of the total time. The remaining 130

 $\frac{22}{130}$ ths of the time is used for horizontal

synchronizing signal which has very low information content. If we can save most of this time we can reduce the data stream by

$$2 \times 51.6 \times \frac{22}{130} = 17.5 \text{ Megabits/sec.}$$

Using this reduction in data rate for an 8.4 MHz channel in (4) indicates that we may be able to achieve a 6 dB improvement in Signal to Noise Ratio. All of this improvement will not be realized in the design of a system because some of the available data rate must be assigned to the audio channels and to synchronization.

Application of TCM to Two Video Channels

Before proceeding to a sample system design consider again the block diagram of the TCM system in Figure 2. The signals going into the storage units are analog in character. Those coming out into the common line are also analog. but they occupy half as much time, and their spectral components are doubled in frequency. The stores are not necessarily analog. They could be digital shift registers preceeded by an A/D converter and followed by a D/A converter. Recently there have become available some Charge Coupled Devices (CCD) sometimes called "Bucket Brigade Delay Lines" which can perform the storage function without the requirement for the A/D and D/A conversions. In these devices an amount of charge proportional to the analog input voltage is shifted through the cells of the device in response to a clock signal. When this charge eventually flows through an output resistor it reproduces the analog input voltage but delayed in time. By having available two clock frequencies one may place a line of video in store at one rate and read it out of store at a different rate.

For either the digital or the analog the data is acquired by a sampling process, and the time between samples can be no greater than the Nyquist interval or $T = 1/2 f_m$ sec.

There is no requirement that samples of the input signal be taken all the time. During the sync interval the value of the signal is known, and no samples need to be taken. This reduces the total number of samples to be transmitted and thus reduces the maximum frequency in the common line.

There is no requirement that the two clock rates be related by a 2:1 factor. The clock which reads information out of store and into the common line is chosen to be fast enough to move the required number of samples in the available time leaving enough time to insert a small data stream containing the two sound channels, system sync, etc.

The common line of Figure 2 corresponds to the Earth Station Transmitter - Satellite Transponder - Earth Station Receiver of a satellite relay system. In this line, the highest frequency necessarily present would be

$$f_m = \frac{1}{2}$$
 Total number of Samples from
two video lines + DIGITAL
Time of one horizontal line

The digital group representing sync and sound channels would be treated as a frequency shift keyed signal of either Binary or M'ary type.

In the common line we need to be concerned with the time of transition from one of the analog signals to the other. In general, it is limitations arising in this transition time which have restricted the usefulness of Pulse Amplitude Modulation Systems to which the TCM system may be compared. Limitations in the phase response of the common line can cause a sudden amplitude transition to be accompanied by a precursor or tail (ringing or overshoot). By this mechanism, cross talk between channels is created.

We can avoid half of the problems of this cross talk by making the transition from one analog signal to the other during the sync interval of one of the signals. At the receive end of the system, we will generate and reinsert clean sync in this channel. Thus the effects of the cross talk will be removed.

In the general case the two video signals will have no fixed relation between their sync times. Thus, while the above fix works for one master channel (to which system sync is keyed), it is of no help for the slave channel. For the slave channel the problem may be avoided by transmission of a small amount of additional information.

In Figure 2, let the samples stored in Store A-1T be taken from Channel A video starting and ending during A blanking. Then add a few samples of B video taken from the B line just preceeding the samples used to fill store B-1T from the B line (with a pause in loading during B sync time). Load a few extra samples of B video after the complete line is loaded. (These samples will be duplicates of the first few samples in B-2T).

When read into the common line there will be no channel transition at the time when loading of B-1R or B-2R starts or stops because the common line is carrying B information on both sides of these points. Let us make a trial system design based on a video channel of about 4.2 MHz bandwidth and a store which can hold 455 samples. For NTSC video all of the frequencies and intervals are related to a master frequency, $F_{\rm M}$ = 14.31818 MHz. The time building block is:

$$\Gamma_{\rm A} = \frac{7}{F_{\rm M}}$$

The horizontal line time is 130 T_A , and this becomes the time in which we must read out both video stores plus the digital store. In loading the store let this horizontal line time be equal to 532 sample times, (T_{SI}) . Then, maximum base band frequency we can handle is

$$f_{\rm m} = \frac{1}{2 T_{\rm SI}} = \frac{532}{2 \times 130 T_{\rm A}} = \frac{532 F_{\rm M}}{2 \times 7 \times 130} = 4.19 \text{ MHz}$$

The active portion of the video line is $108\,T_A$ long which corresponds to approximately 442 T_{SI} . A typical load for an A store would then be:

4	samples	of A	Blanking level
442	samples	of A	active video
4	samples	of A	Blanking video
5	samples	of \mathbf{B}	video for cross talk
			prevention

455 Total samples

For the B stores the load would be the same except that the 8 samples of blanking would come somewhere in the middle of the store, and there would be a pause during loading in response to B sync.

When the samples are read out into the common line, let us use the following sample assignment:

455 samples from A-1T
455 samples from B-1T
32 sample times for Master sync.
10 sample times for A/B Sync difference
16 sample times for A audio PCM
16 sample times for B audio PCM
16 sample times - Spare at this time

1,000 Total sample times

The sample time into the output common line is 130 $T_{\rm A}$

$$T_{SO} = 1000$$

We can now calculate the maximum base band video frequency which must be transmitted over the satellite as

$$f_{\rm m} = \frac{1}{2 T_{\rm SO}} = \frac{1000}{2 x 130 T_{\rm A}} = \frac{1000 \ {\rm F}_{\rm M}}{2 x 7 x 130} = 7.87 \ {\rm MHz}$$

One may well object to a system design which calls for sampling at the Nyquist interval. Note, however, that this is an internal problem which may be resolved by using a larger number of cells in the store. The ratio of the top frequency in the video input to the highest frequency presented to the satellite transmitter base band is:

$$\frac{f_{\rm m} \text{ video}}{f_{\rm m} \text{ B-B}} = \frac{532}{1000} = \frac{4.19 \text{ MHz}}{7.87 \text{ MHz}}$$

If a store with 910 rather than 455 cells had been used this ratio would have stayed the same. Sampling rates within the multiplex equipment would have doubled and the digital samples would have been twice as long in terms of sample times but the same length in real time.

The numbers chosen for the example come from an attempt to design around an existing CCD storage device of 455 cells which may be driven at sampling rates as high as 16 Megasamples/sec.

Let us compare the noise performance of this two channel TCM system with that of a conventional single video channel over a satellite transponder. The path of interest is the downlink and we will use a carrier to noise ratio of 12 dB for the TCM system. For the single channel system the C/N will be increased to 12.8 dB to take into account the fact that a receiver of 30 rather than 36 MHz bandwidth is often used in current practice.

A relationship which is frequently used is

(5)
$$\frac{S_{p-p}}{N_{rms}} = \frac{C}{N} + 20 \log \Delta f + 10 \log B - 30 \log f_m + 10 \log 6 + EW$$

Where the emphasis weighting factor is 13.1dB

For the single channel case where

 Δ f = 10.5 MHz peak deviation and

 $f_{\rm m}$ = 4.2 MHz highest base band frequency the result is S/N = 50.2 dB

For the two channel TCM system the applicable factors are:

and the result is S/N = 41.7 dB

The single channel system would normally transmit video with sync having a maximum amplitude of 160 IRE units (including maximum color signal). In the TCM system the sync signal would be regenerated and added at the receiving end. Thus, the maximum P-P value for its video would be 140 IRE units. A 1.2 dB improvement in S/N can be realized by taking advantage of this. In addition, a portion of the transmitter deviation in the single channel case was attributable to the sound subcarrier which is not needed in the TCM case. If a 1.5 dB adjustment is made for this the result is:

S/N Enhancement

There is an additional step which we can take to improve

the signal to noise ratio of the TCM system. Note that before each line of video is read into the common output line, the entire line is in storage. The maximum peak to peak amplitude of the line could have been determined as it was written into the store. This information can be used to increase the gain of the transmitter modulator and reduce the gain of the receiver video amplifier by equal amounts for any video line having less than the maximum allowable amplitude. The result will be no change in signal at the receiver, but the received noise will be attenuated.

This possibility is shown in block diagram form in Figure 4. Here the S/N enhancement circuitry has been shown separated from the TCM circuit. In this arrangement the incoming video signal is alternately stored in one of two stores.

As a particular line of video is written into storage, its peak to peak amplitude is monitored. As that line of video is read out of storage the results of the monitoring process is used to set a digital attenuator such that the amplitude of the line will be increased to the maximum which can be accepted by the transmitter. A code word to tell the receiving end of the system how to set its attenuator is inserted in the digital portion of the TCM format.

At the receiving end of the system, the sync

timing and the instruction code words are recovered directly from the common line. An instruction generated from the code word is used to set a digital attenuator to remove the results of the extra deviation inserted at the transmitter. In doing this, it brings the signal back to the proper level and reduces the noise power introduced by the satellite link.

This enhancement process could, of course, be used in other applications such as line of sight microwave or video tape recording.

Figure 5 shows one attempt to divide video waveforms into categories which could be designated with a 4 bit digital word. The approach used here was to divide the problem into two parts. First, an off-set voltage was picked as the average amplitude of the signal measured from black level. Two bits were used to describe this off-set. Then, the peak to peak variation around this off-set was given a two bit designation. To use this with the signal enhancement circuit described the instruction generators at the transmitter and receiver ends of the system would control both a four level current generator and a four level digital attenuator.

The amount of signal to noise improvement which can be achieved with this circuit cannot be calculated. It must be determined through subjective tests. We have no reason to feel that a given amount of noise power will have an equally degrading effect in a high contrast portion of the picture and a portion of little contrast.

The circuit can, of course, remove much more noise power from a low contrast picture than it can from one with large changes of light level. It is interesting to note that for a video stream in which the wave forms shown in Figure 5 were equiprobable the measured reduction in noise provided by the enhancement circuit would be 5 dB.

A Video and Data Application

The usefulness of the TCM

approach is not limited to the transmission of two video programs over one transponder channel. In the example discussed, a digital stream representing two audio channels was interspersed with the two video signals. The different signal sources accommodated by a Time Compression Multiplex system may have greatly differing information rates. The time alloted to each source will be approximately proportional to its information rate. The system is flexible in that it can use different modulation processes to achieve different error performances for the various subchannels.

For example, a low data rate channel requiring excellent error performance might be handled as Binary Frequency Shift Keying. At some increase in error rate and a worthwhile improvement in signalling speed a channel could use Quaternary or M'ary FSK. For high values of M the channel error performance would be limited by phase equalization requirements unless a high correlation exists between adjacent symbols thus limiting the ringing or overshoot introduced by large transitions in a channel of poor equalization. It is the high intersymbol correlation of a video signal which allows us to increase M without bound and transmit video in its analog form on a TCM system.

To investigate the versatility of the TCM approach let us try to design a system which can intermix a standard video channel, its companion sound channel, and a low data rate digital channel (of the order of 10^5 bits/sec.)

Make it a further requirement that we be able to receive the digital channel with an inexpensive Receive Only Earth Station. The cost of an earth station for receiving satellite signals is determined by its G/T figure of merit. This term, usually expressed in $dB/^{O}K$ is simply

 $G/T = 10 \text{ Log } G_{ANT} - 10 \text{ Log } T_S$, where

 $G_{\rm ANT}$ is the power gain of the receiving antenna and $T_{\rm S}$ is the noise temperature of the system determined primarily by the input low noise amplifier used (LNA).

For a typical receive only video terminal using a 15 ft. dish and a 150° K LNA the G/T rating would be 21.5 dB/ $^{\circ}$ K

To achieve a low cost receive terminal for our low data rate channel we would like to use a 4 ft. dish and perhaps a 750-1000^oK LNA. For this combination the figure of merit would be around zero $dB/^{o}K$.

Let us assume some system parameters and see how they affect the performance which can be expected. First assign to the video channel 3/4of the available time and allow the remaining 1/4for the data channel. This means that in the video receiver the top base band frequency will no longer be 7.87 MHz as it was in the example using two video channels. It will be reduced by a factor of 1.5/1 to 5.25 MHz. The noise performance of the video channel will be improved by 30 Log 1.5 for reduced base band frequency and by 20 Log $\frac{12.75}{10.13}$ for increased possible

deviation. These combine to predict a S/N of 51.6 dB for the single channel with data compared to the 44.4 computed for two video channels.

Now let us select Quaternary Phase Shift Keying as a modulation method for the data channel. Allow a receive bandwidth of $B_{receive}$ where $B_{rec} = 0.75 f_{data}$ and f_{data} is the data rate transmitted. Since data is only transmitted during 1/4 of the time

 $f_{data} = 4 \times 10^5$ Bits/sec. for our assumed input data rate of 10^5 bits/sec.

The receiver bandwidth is

$$B = 0.75 \times 4 \times 10^{5} = 3 \times 10^{5} Hz$$

Assume that the Quadraphase modulation is differentially coherent and that a carrier to noise ratio of 14 dB will give adequate error performance. We can now compute the required figure of merit for the data service ground station.

Recall that the carrier to noise ratio of a receiving earth station is given by

(6)
$$\frac{C}{N} = \frac{G}{T} - L_D - K + EIRP_{SAT} - -10 \text{ Log B, where}$$

G/T is the receiving figure of merit L_D is the total loss in the down link

K is a constant -228.6 $dBW/^{O}K$

 $EIRP_{SAT}$ is radiated satellite power, and B is the receiver bandwidth.

For the video link we have been considering with B = 36 MHz, we assumed a C/N of 12 dB when G/T was 21.5 dB/ O K.

For the same satellite, we can rearrange (6) to give the required figure of merit for our data channel as

(7)
$$G/T = C/N - 66 + 10 \text{ Log B}$$
.

From this we have

$$G/T = 14 - 66 + 54.77 = 2.8 \text{ dB}/^{\circ}\text{K}.$$

To use a 4 ft. receiving dish, we would need a 750° K LNA.

Figure 6 shows the block diagram of a combination video and data system using TCM. Since different types of modulation are proposed for the two different kinds of input, the two streams are essentially separate until they feed a final transmitting amplifier. Both of the streams use angle modulation.

There must be one connection between the two streams having to do with timing. It would probably be easiest to take system timing from the video side. This might cause some buffering problems on the data side which would require pulse stuffing to adjust the data rate to go with a store switching time set by the video.

At one receiving site a small dish and a narrow band receiver would receive the data signal. A number of possibilities exist for recovering timing. Recall that the beginning of the data stream of interest corresponds to switching off a FM signal which was shifted to a frequency representing black level of the video and switching on a DQPSK signal centered in the receiver pass band.

This should afford a start toward achieving a sync signal.

There is no particular requirement that the data subchannel be located in the center of the transponder passband. The location of a data subchannel might well be chosen to avoid interference from a similar channel on the same transponder channel of an adjacent satellite.

At a second location a larger dish and a wide band receiver are used to recover the video traffic. Analog stores spread this time compressed signal and deliver it at a real time rate to the output line.

Two Way ApplicationIn the previous example,
the actual interleaving of
the signals took place when they were in radio
frequency form on their way to a transmitter.The signals were actually separated in the
radiation downlink from the satellite. By an
extension of this idea we can use TCM to achieve
two way video communication via one satellite
channel.

There are two sets of timing requirements. The first stems from the fact that the two signals are completely independent in timing. The other from the fact that the transmission paths via the satellite are not fixed in length. To solve both of these, we start by nominating one video source as master and the other as slave. As shown in Figure 7, we locate a receiving ground station at the slave (B) location. At the master station, we transmit time compressed bursts of A video starting and stopping during blanking interval. Station A's transmission must have with it a digital group for audio and sync purposes.

At station B, we receive A's transmission and compute when B's transmitted burst would have to occur to fit on an interleaved basis. A time guard band must be provided to accommodate motion of the satellite relative to the two ground stations. We have two options to make provisions for B's commutating switch. A time base corrector can be used to GenLock B video to A video. It does this by storing a video frame and adding or deleting a blank line during the vertical interval as necessary to accommodate absolute differences in sync frequency. This is an expensive device, and it may be preferred to simply increase slightly the time guard band allowed and transmit extra B samples to avoid crosstalk from the commutating switch. The total guard band time allowed must take into account the fact that the time delay introduced in the transmissions from both stations is not constant. The satellite may be moving toward station A and away from station B. In this case the transmission from A will appear to arrive too early and that from B, too late. The time guard band must be large enough to prevent the last part of a B transmission from overlapping the start of an A transmission. This requirement is relatively easy to meet. A more severe requirement is that between one transmission burst and the next, the distance should not change enough from either station to alter the phase of the recovered color subcarrier more than some acceptable amount.

The distance from an earth station to a satellite of relative longitude L and latitude H is about

 $r= 26500 (1-0.295 \text{ Cos H Cos L})^{1/2}$.

If the satellite is at $H = 30^{\circ}$, $L = 50^{\circ}$, and it changes station by 0.1°, its distance will change by about 7.5 miles. For a satellite which keeps station to $\pm 0.1^{\circ}$, the worst case change of range will be 15 miles and the change in difference of distance from two earth stations will be about 5 miles. Thus, the worst case change in loop delay is 160 microseconds, and for difference in delay, the change is 27 microseconds. If the satellite drifts through its extreme positions in a one hour period, the rate of change of loop delay is 4.4×10^{-8} sec/sec. and the rate of change of delay difference is 7.5×10^{-9} sec/sec.

The rate of change of loop delay controls our ability to achieve proper color synchronization by sending a burst of color carrier once per field. With the rate of change of delay computed above, the color carrier would shift in phase by about 1° in 1/60 sec.

The rate of change in delay difference controls how much the time guard band between two interleaved transmissions can change in the time required to send a signal up to the satellite and get a reply back. This time is about 0.28 seconds, and in that time the delay difference could have changed by about two nanoseconds for the example given.

<u>Conclusions</u> The Time Compression Multiplex System described here is in a developmental status. A number of possible configurations must be built and tested before its full potential will be known. Subjective tests of picture quality in the laboratory and transmission tests over satellite paths must both be made before system parameters can be finalized.

V. References

- "Digital Communications by Satellite" J.J. Spilker, Jr. PhD. 1977, Prentice-Hall, Inc. Englewood Cliffs, New Jersey
- "Communication Satellite Systems: An Overview of the Technology" Edited by: R.G. Gould and Y.F. Lum 1975, IEEE Press New York, New York







Fig. 2 Time Compression Multiplex System



Fig.4. S/N Enhancement Circuit











Fig. 7 Two Way Video Transmission Using TCM

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