A TECHNICAL STUDY ON THE FEASIBILITY
AND CONCEPTUAL DESIGN OF A
TDM FORWARD LINK FOR MUSAT

MILLER
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A TECHNICAL STUDY ON THE FEASIBILITY AND CONCEPTUAL DESIGN OF A TDM FORWARD LINK FOR MUSAT

FINAL REPORT

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1.0 INTRODUCTION

In the present concept of the MUSAT system, it is proposed that a large, fixed SHF Central Control Station (CCS) establish strategic and tactical communications to several hundred small transportable and mobile UHF field terminals by means of a forward link operating in the frequency division multiplex (FDM) mode and with sufficient capacity to support 80 FDM channels at 25 kHz spacings. The forward link is to be established via the SHF-to-UHF transponder on board each MUSAT satellite.

As an alternative to the FDM mode of operation for the MUSAT forward link, however, time division multiplex (TDM) operation, which offers a number of technical and potential economic advantages, is also being considered.

It is thus the intention of this study to determine what advantages (or disadvantages) time division multiplex operation of the forward link might offer.

Specifically, the main objectives of this study are:

1. To establish conceptual designs for a TDM forward link which will satisfy both MUSAT's system performance specifications and the forward link traffic requirements.

2. To determine to what extent the inclusion or exclusion of a fighter aircraft service affects the conceptual TDM designs.

3. To determine what difficulties, if any, might be involved in the design and implementation of:
   a) a common TDM receiver for all UHF field terminals, excluding the fighter aircraft, and
   b) a TDM receiver specifically for the fighter aircraft.

4. To determine a method of reducing forward link capacity by 50% during eclipse operation and its effect on system performance specifications.

5. To determine the implementation risks, relative to an FDM approach, of any proposed conceptual TDM forward link designs.

2.0 FEASIBILITY ANALYSIS

2.1 Traffic Requirements

As described in Table 2.1, the MUSAT forward link is required to provide a mixture of demand assigned (D/A) voice and teletype channels, 2.4 kbps demand assigned and pre-assigned (P/A) channels, and optionally a 2.4 kbps common channel to fighter aircrafts (which have 10 dB less G/T than surface mobiles). A single SHF/UHF satellite transponder having an allocated bandwidth of 2.0 MHz and an EIRP to be specified (i.e. minimized) will support this traffic. This study considers the feasibility and possible advantages (e.g. reduced satellite EIRP requirement, simplified terminal equipment) of employing time division rather than frequency division to multiplex these digital signals.

SERVICE	BIT RATE	BER REQUIREMENT	TRAFFIC (ERLANGS)	NO. OF SIMULTANEOUS CHANNELS REQUIRED*
Clear Voice	16 kbps	10^{-2}	29	40
Secure Voice	16 kbps	10^{-3}	9	16
75 Baud TTY	75 bps	10^{-4}	5	10
300 Baud TTY	300 bps	10^{-4}	12	20
DATA (D/A)	2.4 kbps	10^{-5}	12	20
DATA (P/A)	2.4 kbps	10^{-5}	--	3
NFA (P/A)	2.4 kbps	10^{-5}	--	1

TABLE 2.1
TRAFFIC REQUIREMENT

* Based on grade of service due to demand assignment of channels with separate D/A pools for each type of service.

2.2 Evaluation of Theoretical Capacity

Detailed link calculations are avoided for the purposes of this study by describing the power limited channel in terms of single carrier capacity (C) versus bit error rate (p) capability. This representation is general with respect to the form of modulation, link margin and FEC coding employed, providing the channel possesses the necessary bandwidth to support the system.

The power-limited capacity of the MUSAT forward link for a specified bit error rate p is given by

$$C(p) = 10^{0.1[K+G/T-E_b/N_o]} \qquad (1)$$

where

$$K = e.i.r.p. - L - m - \Delta + 228.6 \qquad (2)$$

and

e.i.r.p.	=	satellite e.i.r.p. (dBW) at edge of coverage
L	=	free space path loss
m	=	downlink margin
Δ	=	reduction in overall C/N_o due to uplink noise
G/T	=	figure of merit of receiving terminal
E_b/N_o	=	energy per bit-to-noise density ratio at the specified probability of bit error, p.

For the MUSAT forward link, Table 2.2 lists the values of $\overline{C}(p)$ for various values of p assuming FFSK modulation (theoretical value of E_b/N_o + 1.5 dB implementation margin).

BER (p)	E_b/N_o (dB)	G/T (dB/K)	C(p)
10^{-2}	5.8	-17	1.000
10^{-3}	7.3	-17	0.708
10^{-4}	9.9	-17	0.389
10^{-5}	11.1	-17	0.295
10^{-5}	11.1	-27	0.0295

TABLE 2.2

Defining $\overline{C}(p) = C(p)/C(p_o)$ as the capacity of the link at a bit error rate p relative to that at a bit error rate p_o, equation (1) yields:

$$\overline{C}(p) = 10^{0.1[G/T-E_b/N_o]_p \; - \; 0.1[G/T-E_b/N_o]_{p_o}} \qquad (3)$$

where

K is assumed constant for the particular system configuration and

$[x]_p$ denotes the value of x at a bit error rate p.

For a power-limited channel which must support simultaneously various numbers of circuits n_i at data rates $R(p_i)$ corresponding to a specified set of error rates p_i, the following relation holds

$$\sum_i \frac{n_i R(p_i)}{C(p_i)} < 1 \qquad (4)$$

Using (4), the following expression is obtained for the minimum absolute capacity of the channel $C(p_o)$ at an error rate p_o which satisfies the specified traffic requirement

$$C(p_o) = \sum_i \frac{n_i R(p_i)}{\overline{C}(p_i)} \qquad (5)$$

Using Tables 2.1 and 2.2, and assuming the number of required transmission channels with DSI (digital speech interpolation) to be half that without DSI, equation (5) yields:

$$C(10^{-2}) = \begin{cases} 0.787 \text{ Mbps with DSI} \\ 1.287 \text{ Mbps without DSI*} \end{cases}$$

---

* The value of $C(p_o)$ given by (5) may be substituted into equation (1) to determine the minimum satellite e.i.r.p. (all other link parameters being specified) to support the desired traffic.

Using L = 174 dB, m = 10 dB and Δ=0.1 dB, the minimum satellite e.i.r.p. at edge of coverage is calculated as follows:

$$\text{e.i.r.p.} = L+m+\Delta-228.6 - [G/T-E_b/N_o]_{p_o} + 10 \log [C(p_o)] \text{ dBW}$$

$$= \begin{cases} 37.3 \text{ dBW with DSI} \\ 39.4 \text{ dBW without DSI} \end{cases}$$

These figures take no account of overhead required to synchronously multiplex the channels or possible penalty imposed by the need to vary the transmission rate within the TDM frame. These rates and corresponding duty factors with and without DSI are listed in Table 2.3.

SERVICE	NET THROUGHPUT (kbps)	DUTY FACTOR	TRANSMISSION RATE (kbps)
Clear Voice	320/640	.411/.494	787/1287
Secure Voice	128/256	.235/.285	545/899
75 baud TTY	.75	.0025/.0015	300/494
300 baud TTY	6.0	.020/.012	300/494
Data (D/A)	48	.211/.128	227/375
Data (P/A)	7.2	.032/.019	227/375
NFA (P/A)	2.4	.106/.064	22.7/37.5

TABLE 2.3

TDM CAPACITY ALLOCATION WITHOUT FEC CODING

If FEC coding were applied to all links requiring an error rate below a given value, say λ, the C(p) curve assumed in the previous analysis would be modified as follows:

$$C_{new}(p) = \begin{cases} C_{old}(p) & \text{for } p > \lambda \\ C_{old}(p) \cdot G(p) & \text{for } p \leq \lambda \end{cases}$$

where

$10 \log G(p)$ = real coding gain (in dB) at decoded error rate p.
$C_{old}(p)$ = C(p) channel characteristic without coding
$C_{new}(p)$ = C(p) channel characteristic with coding

yielding a revised (and improved) capacity profile curve as illustrated in Figure 2.1.

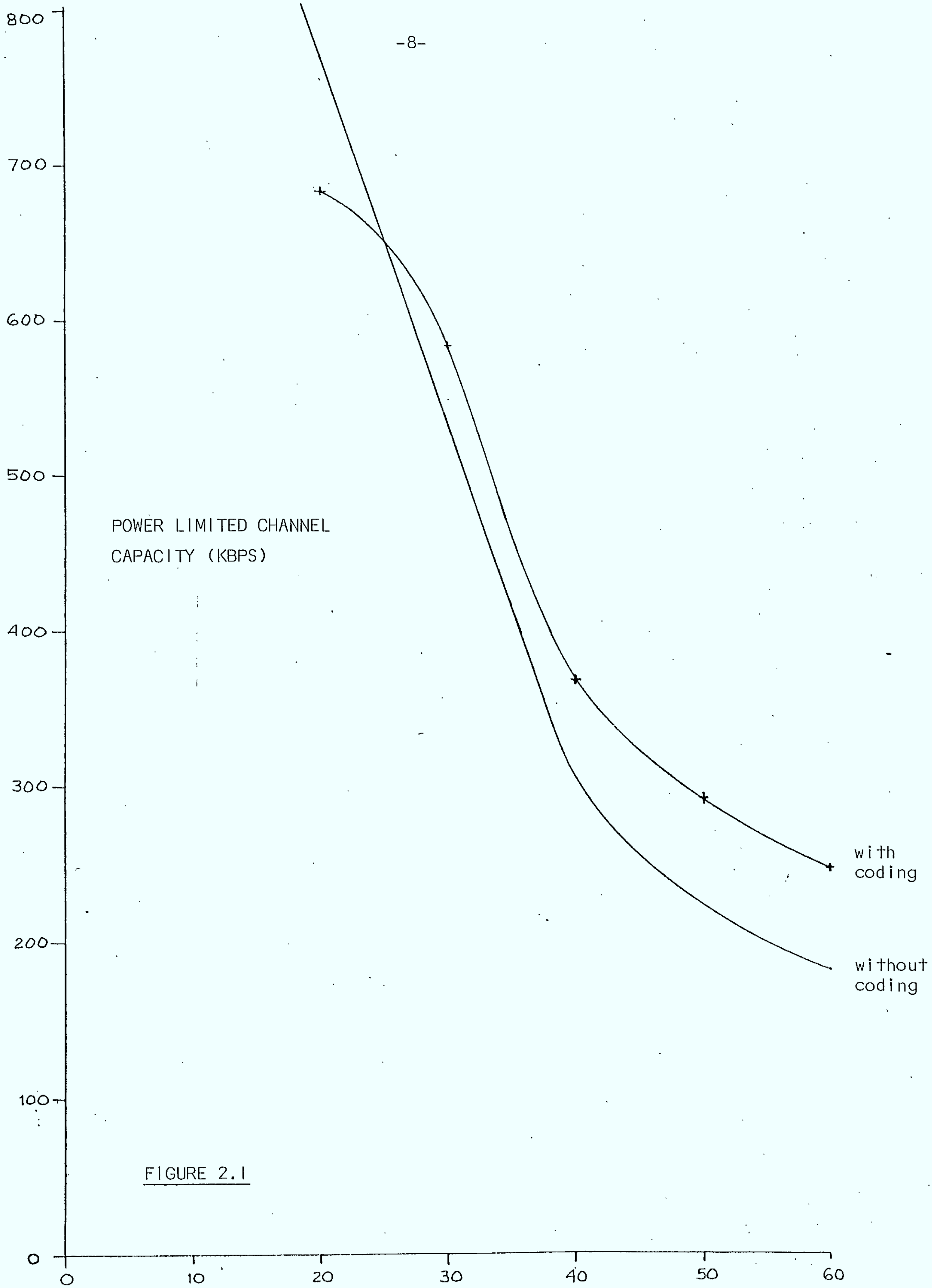
800
700
600
500
400
300
200
100
0
POWER LIMITED CHANNEL
CAPACITY (KBPS)
0
10
20
30
40
50
60
with
coding
without
coding

FIGURE 2.1

G(p) depends on the type of code employed, code rate (bandwidth expansion factor) and method of decoding (hard or soft). In all cases G(p) decreases with increasing p, becoming negative at some large value of p. This explains the cross-over of the coded and uncoded capacity curves in Figure 2.1, and why coding would first be considered on links requiring lower error rates.*

For illustrative purposes, the application of a simple rate 1/2 double error correcting convolutional code with threshold decoding is considered. Its theoretical (upper bound) performance in the presence of random errors is shown in Table 2.4.†

p	G
10^{-2}	-0.5 dB
10^{-3}	0.4 dB
10^{-4}	0.9 dB
10^{-5}	1.1 dB
10^{-6}	1.4 dB
10^{-7}	1.6 dB

TABLE 2.4

Rate 1/2 coding applied to the 2.4 kbps data links (.295 and .0295 entries in Table 2.1 are now multiplied by 1.288 corresponding to the 1.1 dB real coding gain) reduces the required forward link capacity to

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* In many practical situations, coding provides a further improvement by reducing the required modem margin (at increased bit error operating point); this impact of coding is not considered in the theoretical treatment presented here.

† Results are based on the assumption that the decoded error rate throughout a constraint length =.5 if there are more that 2 demodulated errors in this constraint length (14 bits).For threshold decodable code considered, p can then be written as a binomial series sum,

$p = \sum_{i=3}^{14} \binom{14}{i} p_{in}^{\,i} (1-p_{in})^{14-i}$ where p_{in} is the input error rate, easily evaluated by computer.

$$C(10^{-2}) = \begin{cases} .726 \text{ Mbps with DSI} \\ 1.227 \text{ Mbps without DSI} \end{cases}$$

This implies only a .25 dB (with DSI) or .15 dB (without DSI) reduction in required forward link satellite EIRP. Note that coding in this case decreases slightly the transmission bandwidth required (determined by the maximum rate traffic, clear voice) although it more than doubles the rate during the lower speed 2.4 kbps data portions of the frame. This can be observed by comparing Tables 2.3 and 2.5.

SERVICE	NET THROUGHPUT (kbps)	DUTY FACTOR	TRANSMISSION RATE (kbps)
Clear Voice	320/640	.441/.522	726/1227
Secure Voice	128/256	.249/.295	514/869
75 baud TTY	.75	.0027/.0016	282/477
300 baud TTY	6.0	.021/.013	282/477
Data (D/A)	48	.174/.103	552/932
Data (P/A)	7.2	.026/.015	552/932
NFA (P/A)	2.4	.087/.052	55.2/93.2

TABLE 2.5
TDM CAPACITY ALLOCATION WITH RATE 1/2
FEC CODING APPLIED TO 2.4 kbps DATA ONLY

While more powerful codes (which are more complex to implement) than the one considered here are available[1], it is clear that the predominantly high error rate traffic of the system limits the utility of coding to reduce the power requirements of the forward link. FEC coding must still be considered, however, as a means of achieving a constant modem rate or to protect critical TDM signalling information.

[1] R. Lyons and L. Beaudet, "Application of Forward Error Correction over Aeronautical Satellite Data Links", CRC Report No. 1314, Ottawa, April 1978.

2.3 Multi-Destination TDM Carrier Transmission Alternatives

Section 2.2 presented a method of evaluating the theoretical capacity of a time division multiplexed broadcast channel supporting transmission at different rates to different size receive terminals or at different error rates. The means by which the high efficiency implied in this analysis could be achieved by practical multiplexing and modulation methods was, however, not considered.

This section examines three approaches to time sharing the MUSAT forward link:

(i) non-synchronous variable rate
(ii) synchronous variable rate
(iii) FEC coded synchronous common rate

2.3.1 Non-Synchronous Variable Rate

The transmission of non-synchronously modulated bursts of phase continuous carrier obviates the need to synchronize the digital message signals at the CCS, but imposes the need for the receiving terminals to recover bit timing on a serial burst by burst basis (imposes requirement for burst preamble), or maintain several (asynchronous) clocks. In essence the data synchronization problem (which complicates the transmitting terminal) has been traded for a burst demodulator synchronization problem (which complicates the receivers).

This tradeoff is not beneficial in our case due to the large number of receive terminals present and the desire to keep them simple. Furthermore:

(i) If the large number of asynchronous bit streams at the CCS are not at least partially synchronized, the frame efficiency of a burst transmission system with serial timing recovery will be low, or the buffer sizes (frame length) high. A parallel timing recovery approach must be regarded as high risk and would greatly complicate the receive terminal.

(ii) Compression buffering of inputs at the CCS is still required for asynchronous TDM transmission. Given these buffers synchronising their output rates can be achieved at modest cost.

(iii) Burst transmission rates will necessarily be rational number multiples of incoming asynchronous rates. Clocks must be available for all the incoming data lines and since their stability is not directly under the control of the modem designer, the demodulator may be required to acquire over a wide range of uncertainty about each nominal data rate.

(iv) A variable rate, asynchronous burst modem is difficult to implement, and would probably be realized by a number of gated burst modems operated in parallel. Such is not the case for a synchronous variable rate modem.

(v) The multiplicity of asynchronous data rates in a multi-channel receive terminal complicates the design of the demultiplexer.

(vi) The single access mode of operation permits continuous, synchronously modulated variable rate carrier schemes to be considered as alternatives (see Section 2.3.2).

2.3.2 Synchronous Variable Rate

In this case the carrier is successively modulated at rates which are exact integer multiples of a fundamental rate. The resulting carrier has, in fact, received a special case of modulation at the highest rate, an n sub-multiple of this rate being obtained simply by repeating each symbol n times. This corresponds to n bit repetition on the I and Q channels of a quadriphase PSK or FFSK modem, i.e., these inputs are simply held high or low for the desired multiplied clock period.

The structure of a synchronous, variable rate TDM system can therefore be visualized as in Figure 2.2.

Three continuous synchronous data inputs a, b, and c are to be bursted over the channel at rates R, R/m and R/n respectively as shown in Figure 2.2.* The resulting multiplexed variable rate data pattern can be transmitted and received efficiently using a coherent modem designed for continuous operation at a fixed rate R. (For linear, i.e. coherent demodulation, it is unnecessary to reduce the IF input noise bandwidth during lower rate transmission periods; for optimum performance the enhanced performance can be obtained purely by adjusting the baseband filtering).

The demodulator bit timing recovery circuit will produce line components at all three clock rates R, R/n and R/m. We could tune to any one (or combination) of these synchronous rates to derive the desired reference for the receive clock unit. Since much of the traffic will, in fact, be at the highest rate (clear voice), the normal (for an R-rate modem) recovered clock rate R is most appropriate. At this point there is nothing to distinguish our modem, which must operate with any combination of R sub-multiple rates, from a constant R-rate modem.

The difference, in fact, appears in the detector, consisting of baseband filter, sampler and threshold detector. It is required to

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* Synchronous multiplexing and CW modulation requires that m and n be rational numbers, i.e., of form M/N where M and N are integers. In this discussion m and n are restricted to be integers, which results in some simplification of the terminal design.

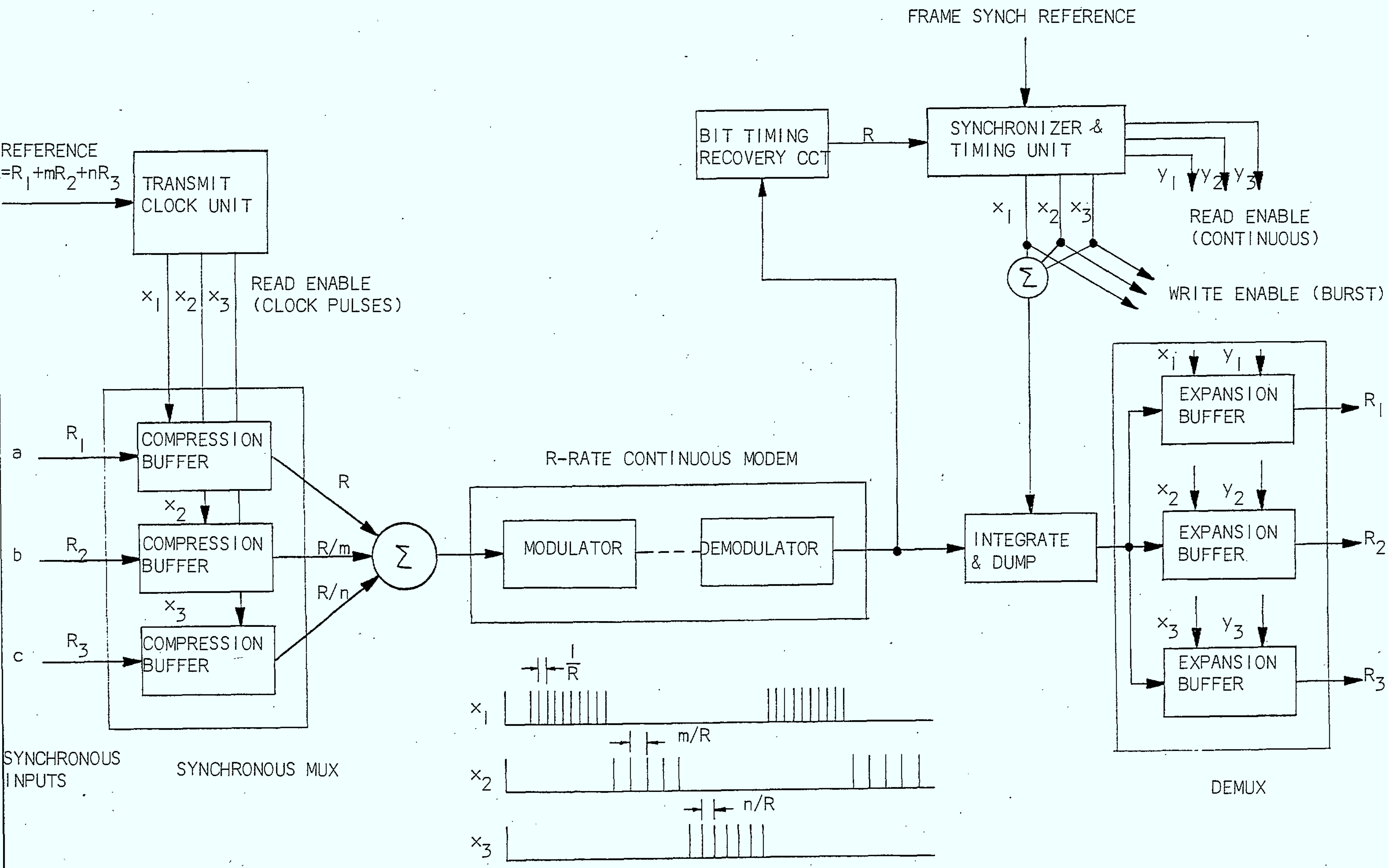

FIGURE 2.2

SYNCHRONOUS, VARIABLE RATE TDM SYSTEM

adjust the pre-detection filter bandwidth and detection interval whenever the effective incoming data rate changes. This is most conveniently obtained by employing an integrate and dump detector clocked appropriately. Integrate and dump detection implies the need for a transmission bandwidth about twice the Nyquist rate (i.e., 2 x symbol rate), which is acceptable providing a four level scheme (i.e., FFSK or QPSK) is employed. FFSK is an especially appropriate candidate because it is both bandwidth efficient and employs integrate and dump detection. Only the timing unit would require additional circuitry for the generation and appropriate selection of the necessary sub-harmonics (under instruction from the frame synchronizer) as shown in Figure 2.2.

The theoretical performance of a variable rate, synchronous, coherent TDM system is based on the following expression for bit error rate

$$P_e = \frac{1}{2}\,\mathrm{erfc}\left(\sqrt{\frac{C}{N_o}\,\frac{i}{R}}\right) \qquad \text{(without differential encoding*).}$$

where $\frac{R}{i}$ is the transmission rate (sub-multiple of highest rate R).

Table 2.5 presents a transmission plan based on i=2 for secure voice, i=3 for teletype, i=4 for 2.4 kbps data, and i=40 for 2.4 kbps data to the fighter aircraft. As before results are presented both with and without DSI.

In comparing Tables 2.2 and 2.5, we note that the need to synchronize all transmission rates to sub-multiples of the clear voice transmission rate increases the channel power and bandwidth requirements by factors of 1.30 (1.13 dB) with DSI and 1.22 (0.85 dB) without DSI. This increase does, however, provide excess margin on the links as indicated in Table 2.5. Note also that convenient fundamental rates (1.0 and 1.544 Mbps) have been chosen resulting in duty factors which total to less than one. Some small part of the frame will, in fact, be required for synchronizing, multiplexing, and possibly signalling purposes; this is considered in Section 2.6.

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* Due to the variable rate (variable error probability) nature of the frame, differential encoding would impose a 1 symbol overhead per sub-frame, certainly for the NFA burst. On the other hand, depending on the phase stability of the recovered carrier and the frame length, the synch work could be used to resolve phase ambiguity once per frame and eliminate both the double error probability and overhead loss of differencial encoding.

SERVICE	NET THROUGHPUT (kbps)	DUTY FACTOR	SYNCHRONOUS TDM RATE (kbps)	EXCESS MARGIN (dB)
Clear Voice	320/640	.320/.415	1000/1544	0.00
Secure Voice	128/256	.256/.332	500/772	1.50
75 Baud TTY	.75	.00225/.00146	$333\frac{1}{3}/514\frac{2}{3}$	0.67
300 baud TTY	6.0	.018/.012	$333\frac{1}{3}/514\frac{2}{3}$	0.67
Data (D/A)	48	.192/.124	250/386	0.72
Data (P/A)	7.2	.0288/.0187	250/386	0.72
NFA (P/A)	2.4	.096/.062	25/38.6	0.72

$\Sigma = .913/.965$

TABLE 2.6
SYNCHRONOUS TDM CAPACITY ALLOCATION
WITHOUT FEC CODING

With little penalty in channel power and bandwidth, a synchronous programmable variable rate system offers the following advantages:

(i) A receive terminal requires only one demodulator/demultiplexer unit. This unit is common to all surface mobiles and fighter aircraft regardless of specific traffic requirements.

(ii) A very large dynamic range of transmission rates ($R_i = R/i$) can be accomodated. The effective "coding" gain at all transmission rates and demodulated error rates is 0 dB. No single easily implementable FEC coding scheme can boast such performance over such a range.

(iii) Changes in frame format or the introduction of new links (at new rate R_i) are easily accomodated

(iv) Similarly, overall transmission capacity can be traded in discrete steps up to maximum R for link margins by re-adjusting burst rates. This flexibility is useful in assuring links are neither over-designed nor under-designed for operational requirements. It can also be exploited to permit operation at half capacity during eclipse operation.*

2.3.3 FEC Coded Common Synchronous Rate

The synchronous variable rate scheme described in Section 2.3.2 could be regarded as a special case of coding to a common synchronous rate R. The codes employed are simple majority logic codes (bit repeated n times) with soft decision (coherent integration of rectangular pulses) majority logic decoding giving 0 dB real coding gain for all values of n and error rates.

The implementation implied here, however, consists of a modem (including detector) operating at a single rate, with FEC coding/decoding applied to channels in the multiplex requiring improved error performance. This results in the functional representation of the TDM system depicted in Figure 2.3.

Since a great variety of code rates m/n are possible, the "excess margin" loss (see Table 2.5) of the synchronous sub-multiple variable rate scheme described in Section 2.3.2 can in theory, be largely avoided. FEC coding may give real coding gains which are positive, further reducing channel power requirements. Also, the configuration in Figure 2.3 does not necessitate the use of integrate and dump detection or even coherent demodulation as did the one in

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* The normal rates R, R/2, R/3, R/4 and R/40 would be divided by 2 and the multiplex logic altered to accomodate half the number of channels in each category.

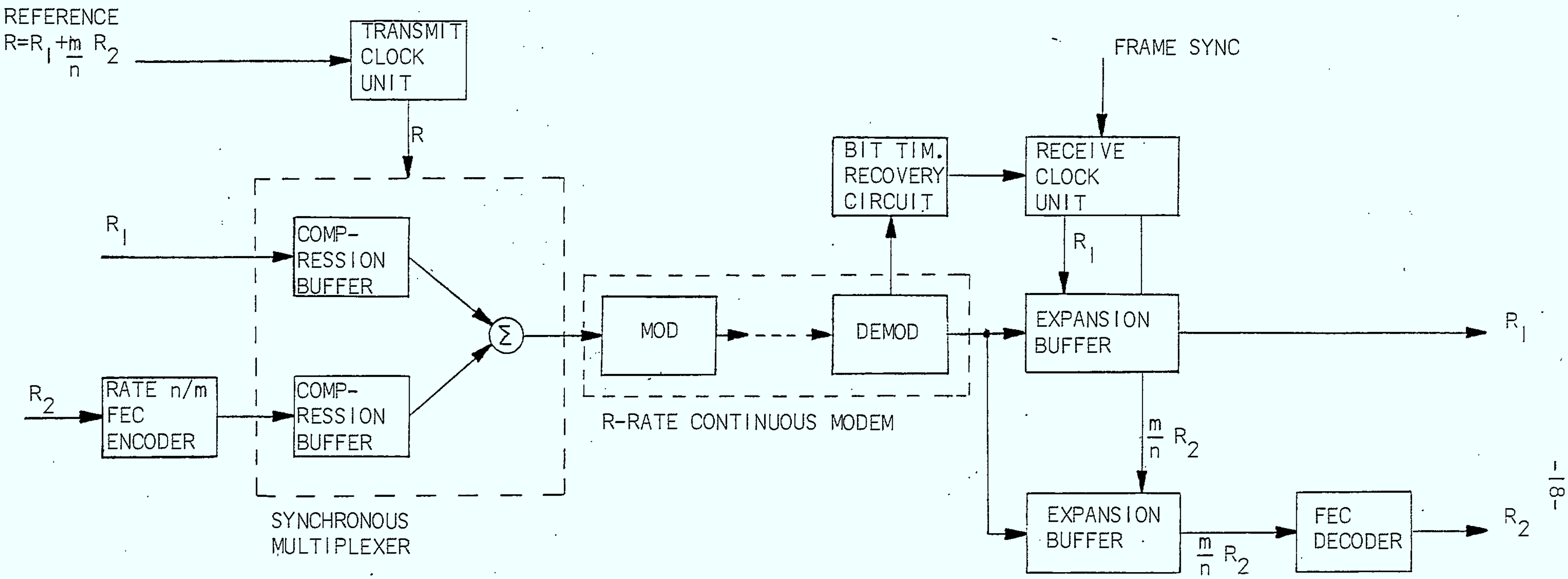

FIGURE 2.3

FEC CODED COMMON SYNCHRONOUS RATE TDM SYSTEM

Figure 2.2. However, there are a number of serious disadvantages to taking the FEC coding approach as outlined below:

(i) Cost and complexity of the receive terminal is increased significantly by the presence of FEC decoders (not simple devices) at various rates.

(ii) Little coding gain can be expected at the 10^{-3} (secure voice) and 10^{-4} (TTY) error rates.

(iii) An efficient very low rate code for the 2.4 kbps fighter aircraft data channel would prove difficult and costly to implement, and would require the demodulator to operate effectively at negative E_b/N_o's.

(iv) To trade capacity for link margin requires alteration of the code rate (i.e., selection of a new code) and redesign/refitting of the codec.

2.4 Two Level Multiplex Structure for Synchronous, Variable Rate TDM

In order to implement the preferred approach in Section 2.3, namely synchronous, variable rate TDM, a two level multiplex scheme[2] is required. As illustrated in Figure 2.4, the outer multiplex synchronizes (where necessary) and combines signals[3] to be transmitted at a particular synchronous rate. Table 2.5 indicates that a minimum of five outer multiplexers are required. A single inner multiplexer interfaces with the outer multiplexers and the continuous modem to generate and separate bursts of variable rate synchronous data. The inner multiplexer also has responsibility for ensuring overall frame synchronization can be acquired and maintained by all receive terminals, including the lower G/T fighter aircraft. Receivers will possess common inner multiplex equipment and, depending on particular traffic requirements, appropriate outer multiplex equipment. (The fighter aircraft will only be capable of receiving the low rate 2.4 kbps data stream).

To accommodate demand-assigned operation, the outer demultiplexers at the surface mobiles must be programmable (i.e., capable of extracting <u>any</u> m out of n channels in a sub-frame) and, of course, be capable of extracting the pre-assigned 2.4 kbps DAMA channel.

The frame synch word must be transmitted at the lowest bit rate (R/40) to ensure proper detection by the fighter aircraft. Its demodulator (see Figure 2.2) may* be unable to maintain bit timing synch throughout the frame implying the need to:

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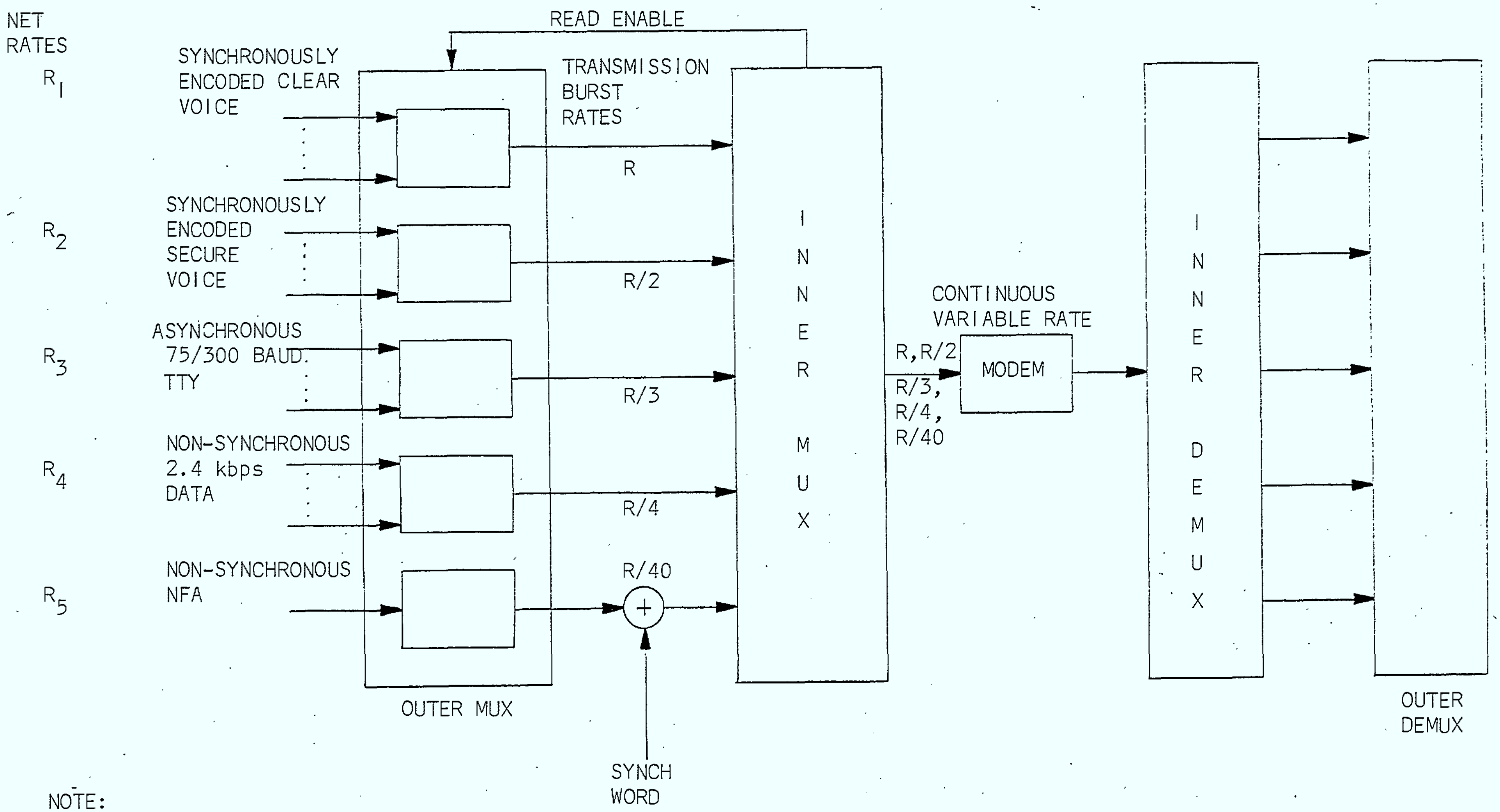
[2] P. Benowitz et al, "Digital Multiplexers", B.S.T.J. Vol. 54, No. 5, May-June 1975, pp.393-418.

[3] V. Johannes and R. McCullough, "Multiplexing of Asynchronous Digital Signals Using Pulse Staffing with Added-Bit Signalling", IEEE Trans. Communication Technology, Vol. COM-14, No. 5, Oct. 1966, pp.562-568.

*The fact that the demodulator operates at negative E_b/N_o over most of the frame does not imply that bit timing cannot be continuously tracked. This depends on the bandwidth of the timing recovery circuit, which in turn depends on the stability of the transmitter clock. Acquisition time does not constrain the design because the system operates continuously.



NOTE:

asynchronous = irregular rate of arrival of characters

nonsynchronous = constant bit rate clocked independently of local clock

FIGURE 2.4

SYNCHRONOUS, VARIABLE RATE TDM SYSTEM

- transmit the synch word just before data to the fighter aircraft;

- ensure the synch word is of sufficient length (and possesses sufficient number of transitions, to allow acquisition of bit timing as well as word detection;

- tune the clock recovery circuit of the fighter aircraft to R/40 rather than R and design it for fast acquisition.

It is presently unclear which of the following three approaches is most appropriate for the fighter aircraft demodulator:

(i) T-clock component tracked throughout frame,

(ii) 40 T-clock component tracked throughout frame;

(iii) 40 T-clock recovered once per frame.

The non-linear relationship between recovered clock error and input carrier-to-noise ratio favours option (ii) while (i) has the advantage of a higher average T-clock component in the carrier. The burst demodulation approach (iii) should be avoided if possible because it complicates the demodulator and reduces frame efficiency (the synchronization word must be appended by a 0101... timing recovery preamble and be immediately followed by the NFA burst). For purposes of describing a common receive terminal at both fighter aircraft and surface mobile it is assumed that one of the first two approaches can be taken.

2.5 Frame Format Tradeoffs

2.5.1 Inner Multiplex

From Figure 2.4, a complete frame consists of a sequence of bursts of clear voice, secure voice, teletype, 2.4 kbps data and synch word/NFA traffic transmitted at rates R, R/2, R/3, R/4 and R/40 respectively. The transmission plan is simply to repeatedly transmit frames of this type and control integrate and dump and demultiplexing operations at the receiver (see Figure 2.2) by counting down from a synchronization marker appearing at least once per frame (probably once per NFA burst - see Section 2.4). Selection of the length and sequencing of the sub-frames (i.e., the frame format) influences both the buffer sizes and logic complexity of the multiplexer. It has no effect on frame efficiency since, for continuous fixed format transmission, it is not necessary to annotate the position of any sub-frame in the frame.

There is clearly an inverse relationship between the rate at which multiplex switching occurs and the length of sub-frames. The period between successive sub-frames of a given type (assumed fixed) multiplied by the net source rate (denoted R_i in Figure 2.4) determines the compression/expansion buffer sizes required for inner multiplexing/demultiplexing. This implies that to minimize overall buffer requirements, the sub-frame repetition rate for each type of traffic should be directly proportional to the source rate, and the sub-frame lengths (in bits) should be identical. Since sub-frame repetition rates must also be a factor of 2 related for simple multiplexing and demultiplexing based on sequential logic rather than look-up table, a compromise must be made as illustrated in the following table of quasi-optimum relative sub-frame repetition rates and lengths with and without DSI.

TRAFFIC	RELATIVE SUB-FRAME REPETITION RATE	MESSAGE LENGTH (# UNITS)	RELATIVE TRANSMISSION PERIOD	TRANSMIT DOUBLE BUFFER SIZE (# UNITS)
Clear Voice	16	N_a = 4/8	4/8	8/16
Secure Voice	8	N_b = 4/8	8/16	8/16
Combined 2.4 kbps	4	N_c = 3	12	6
Combined TTY	2	N_d = 1	3	2
Synch Plus NFA	1	N_e = 1	40	2
			TOTAL	26/42

TABLE 2.7

RELATIVE SUB-FRAME LENGTH ASSIGNMENT

This yields a 32 element frame format described by:

	a	b	a	c	a	b	a	d	a	b	a	c	a	b	a	e	a	b	a	c	a	b	a	d	a	b	a	c	a	b	a	x
a	×		×		×		×		×		×		×		×		×		×		×		×		×		×		×		×	
b		×				×				×				×				×				×				×				×		
c				×								×								×								×				
d								×																×								
e																×																

FIGURE 2.5

A lower bound on the total buffer requirement for an optimized frame format with minimum sub-frame length of one unit is 12 units, indicating no great penalty with the scheme described in Table 2.7. However, this scheme permits simple 5 bit counter-based multiplex-demultiplex switching logic as illustrated in Figure 2.5. Since the sub-frames are not of equal length, the counter is clocked from a programmable source which sends a pulse only at the end of each sub-frame.

The number of basic clock pulses which this source must itself count is determined by the current switch position, which identifies the rate and the number of bits which must be read from the compression buffer or written into the expansion buffer. Figure 2.6 presents a functional block diagram of the multiplex logic required to implement the frame format of Figure 2.5.

Note in Figure 2.5 that the overhead loss due to the presence of the 32'nd slot in the frame could be eliminated without added complication by having the counter instantaneously "skip" its all '0' state. Alternatively, this logic state could initiate switching to a synchronization or signalling input to the multiplex.

The foregoing discussion has considered tradeoffs associated with fixed format frames. Fixed format implies a fixed number of demand assignable channels in each traffic category - clear voice, secure voice, teletype, and data - and hence the possibility of blocking in one category with unused capacity in another. This suggests consideration of adaptive multiplexers capable of handling various combinations of data inputs. Low speed data multiplexers of this type are commercially available.

Since a user can always access a 16 kbps channel in the secure voice sub-frame if the clear voice sub-frame is filled, virtually no improvement in grade of service for voice traffic is obtainable with

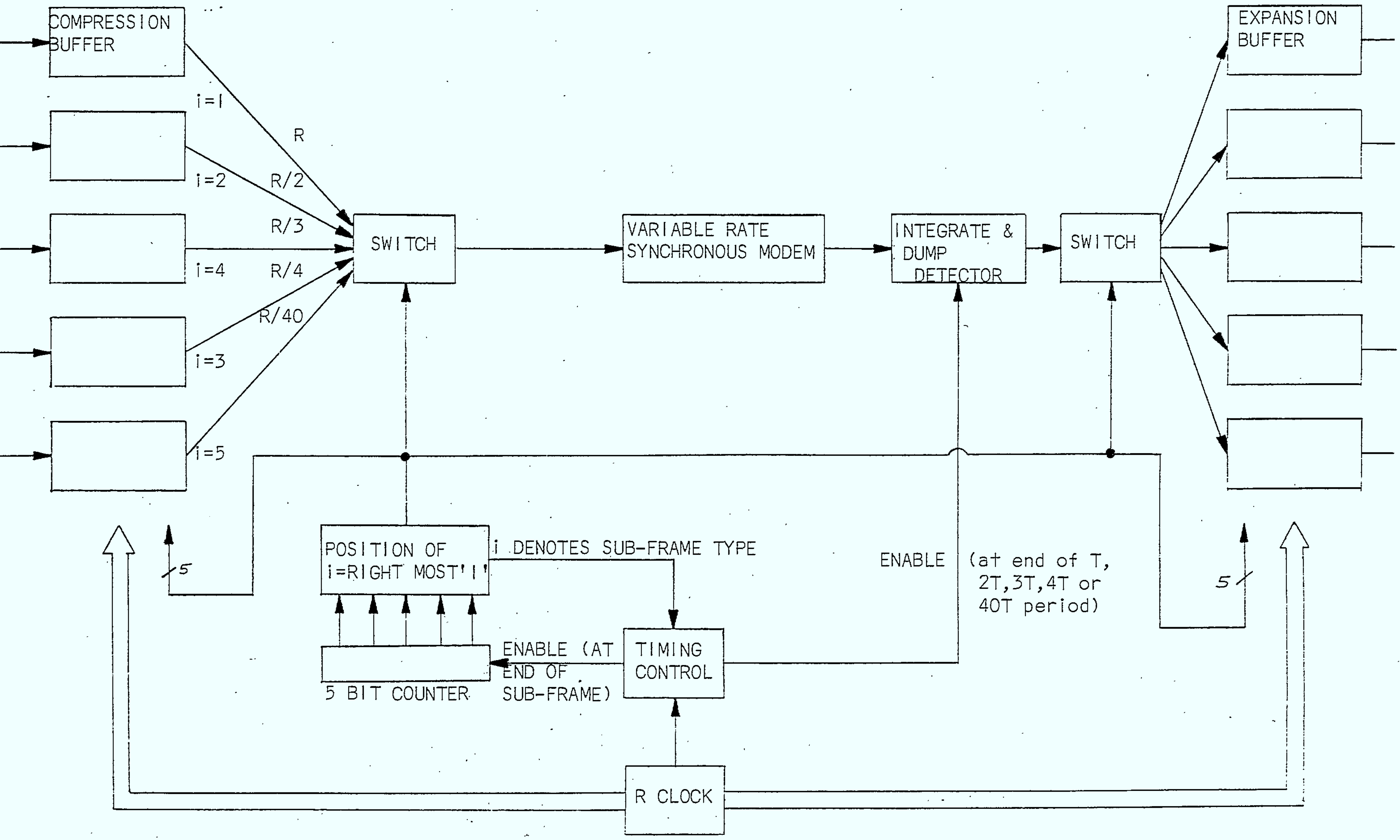


FIGURE 2.6

INNER MULTIPLEX/DEMULTIPLEX LOGIC FOR SYNCHRONOUS VARIABLE RATE TRANSMISSION adaptive formatting. Some improvement in 2.4 kbps data grade of service is possible by allowing extra channels to be made available as required from unused portions of the voice sub-frames. However, the resulting improvement in overall transmission efficiency (satellite power and bandwidth required) is small because the voice traffic is so dominant. An improved grade of service to the data users would probably be more economically met by adding channels to the multiplex and increasing transmission capacity slightly rather than going to an adaptive scheme, which significantly complicates the demultiplex equipment at the receive terminals.

2.5.2 Outer Multiplex

It is therefore envisaged that the inner multiplex arrangement will be fixed format with inner multiplex/demodulator equipment common to all receive terminals. Only those expansion buffers in Figure 2.6 necessary to support a required type of traffic will, however, actually be fitted into a terminal (e.g. the fighter aircraft will only expand the 2.4 kbps NFA bursts).

Following the regeneration of the desired combination of clear voice, secure voice, TTY, 2.4 kbps, and NFA data into continuous digital streams, it will be necessary (in the first four cases) to recover those individual channels actually required at a terminal. This is the role of the outer demultiplexers. Since many of the channels are demand assignable, the outer demultiplexers must be programmable (i.e., capable of presenting any n of the m channels at its n output ports). Control addresses will be distributed to the demultiplexers by the demand assignment channel decoder. The capacity (n) of a demultiplexer will depend on the traffic requirements of the individual terminal served.

If expansion buffering is performed prior to demultiplexing as indicated in Figure 2.6, then the inner and outer multiplex frame formats can be designed independently. On the other hand, this arrangement requires the storage of data only a small part of which will ultimately be presented to the data interface. An alternative approach is to remove the expansion buffers from the inner demultiplex and extract only those individual voice, TTY and data channels required from the output bursts prior to regenerating continuous data streams. This implies that the inner demultiplexer operates as a memoryless (except for counters) routing switch, and the terminal expansion buffer requirements are determined by the number of channels terminated rather than the total capacity in a particular traffic category.

However, in order to employ simple counter-based logic to demultiplex (on a programmable basis) channels on a burst by burst basis, it is necessary that the burst length (in bits) be an integer multiple of the number of channels multiplexed. Referring to Figure 2.5, this requires that a, b, c, and d be exact multiples of 40, 16, 50 and 23 bits respectively. These constraints trade off with inner frame format tradeoffs discussed in 2.5.1.

Given the low cost, weight and power consumption of commercially available buffer memories, it is not presently clear which of the above approaches is optimum. If only low terminal capacities are required a channel select/expand approach is favoured, however, at relatively little cost, an expand/channel select demultiplexer capable of presenting a large number of channels to the data interface can be implemented and used as a standard throughout the system. Individual terminals are fitted with the interface equipment (delta decoders, etc.) required for their own purposes. This approach takes maximum advantage of the flexibility offered by a TDM approach.

2.6 Coding for Frame Synchronization

Frame synchronization is required to ensure:

- a receiver, when turned on, can establish the necessary frame reference for its demultiplexer and integrate and dump detector

- a continuously operating receiver will recover synch in the event of demodulator clock slips.

These two requirements, denoted the acquisition requirement and the steady state tracking requirement, impose somewhat different constraints on the synchronization word and the rate at which it is repeated. Without going into details, the following points are pertinent:

(i) The frame synchronization scheme must be designed to meet minimum requirements on the "weak" CCS to fighter aircraft link.

(ii) This implies that the synch word must be sent at the lowest demodulator rate 1/40 and possibly appended to the NFA message (see discussion in Section 2.4).

(iii) The synch word length N must be much larger than the number of correctable bit slips M occuring between synch words (or per frame).

(iv) Tracking Performance:

Presuming an ideal synch word autocorrelation function (of type achievable with a pn sequence of length $N=2^n-1$)

$$R(i) = \begin{cases} N \text{ for } i=0 \\ 0 \text{ otherwise,} \end{cases}$$

and a detector threshold of $\frac{N}{2}$, the probabilities of missed-detection and false alarm in a detection window of length 2M+1, M<<N, are

$$P_{MD} < P\{>N/4 \text{ errors in N positions}\} \simeq \binom{N}{N/4} P_e^{\frac{N}{4}}$$

and

$$P_{FA} \simeq (2M+1)P_{MD}$$

respectively, where

P_e = bit error rate into binary correlator

Assuming synchronization is deemed lost (and receiver returns to acquisition mode) whenever a synch word is not detected in K consecutive windows:

$$P_{loss} = P\ \{\text{return to acquisition mode}\}$$

$$\simeq P_{MD}^{K} + P_{FA}\ P_{MD} P\ \{>1 \text{ bit slip per frame}\} +$$

$$P\{> \text{M bit slips per frame}\}$$

$$= P_{MD}^{K} + P_{FA}\ P_{MD} Lp + L(L-1)\ldots(L-M)p^{M+1}$$

Here

L = frame length (in equivalent number of bits at rate R/40)

p = bit integrity

The three terms in P_{loss} respectively correspond to K successive mis-detections, incorrect translation of the detection window (to decentre the synch word) immediately followed by a bit slip which removes the synch word from the window, and a single long slip which moves the synch word out of the window.

(v) Acquisition Performance

In the acquisition mode, the window is removed and the receiver continuously searches for the synch word. To reduce the false alarm rate, the threshold is set at N-I where I=allowable number of errors in synch word. Now,

$$P_{MD} \simeq P\{>I \text{ errors in } N \text{ positions}\} \simeq \binom{N}{I} P_e^{\,I+1}$$

and

P_{FA} = probability of occurence of synch word with up to I arbitrary errors in one frame of demodulated random data

$$\simeq L\binom{N}{I} 2^{-(N-I)}$$

Assuming acquisition is deemed to have occured when K consecutive detections L bits apart occur,

$$P_{FD} = P\text{ (incorrect synchronization)}$$

$$= P_{FA}\left[\binom{N}{I} 2^{-(N-I)}\right]^{K-1}$$

and

$$T_{acq} = \text{mean time to acquire synchronization}$$

$$\simeq K(1+2P_{MD}(K)) \text{ frames}$$

where

$$P_{MD}(K) = (1-P_{MD})^K \ll 1$$

Consider for example:

L = frame length = 2048

N = synch word (pn sequence) length
= 15

P_e = bit error rate
$= 10^{-2}$

P = bit integrity
= 1 part in 10^9

$M = 2$

$I = 1$

$K = \begin{cases} 4 \text{ for acquisition} \\ 3 \text{ for tracking} \end{cases}$

Then substituting in the above equations:

T_{acq} = mean time to acquire synchronization
= 4.5 frames

P_{FD} = probability of incorrectly acquiring synch
$\simeq 2 \times 10^{-9}$

P_{loss} = probability of losing synch
$\simeq 2 \times 10^{-15}$

P_{FRAME} = frame integrity
= percentage of time frame is demultiplexed incorrectly
$\simeq 10^{-6}$

Reasonable theoretical synchronization performance with a frame efficiency of better than 99% is thus obtainable under rather degraded performance ($P_e = 10^{-2}$, $p = 10^{-9}$) using binary correlation of demodulated data. Because the E_b/N_o following correct demodulation and integrate and dump detection at a rate R/40 is high (to yield $P_e = 10^{-4}$ at the fighter aircraft and essentially error free reception at a surface mobile), synchronization based on matched filter (analog) correlation is unnecessary and should not be considered. On the other hand, transmission of the synch word at the R/40 rate introduces the ambiguity problem discussed in (i) below, suggesting the following approach should be considered as an alternative:

a. Transmit synch word possessing desired autocorrelation properties at rate R. Integrate and dump would then be performed at this rate, and the bit correlator could acquire synch to the nearest $\pm T/2$ interval. This circumvents the need (see (i)) to independently recover T/40 clock at the receive terminal, and removes any constraints this imposes on positioning the synch word in the frame.

b. While this simplifies the surface mobiles (and would certainly be employed if the NFA were not included in the system), it requires analog correlation (employing a SAW device) for acquisition and tracking at the NFA implying the need for two types of receive terminals.

The variable rate nature of the frame imposes the following constraints on synchronization using a synch word possessing bits of length 40T:

(i) In the acquisition mode, both the surface mobiles and the fighter aircraft must perform continuous integrate and dump detection at the R/40 rate. To avoid the requirement to search over 40 start of integration positions, an R/40 clock reference <u>synchronous in phase</u> with the synch word clock must be provided to the integrate and dump detector. Such a source cannot be obtained by dividing the R clock component present in the signal. Providing the separation between the synch word

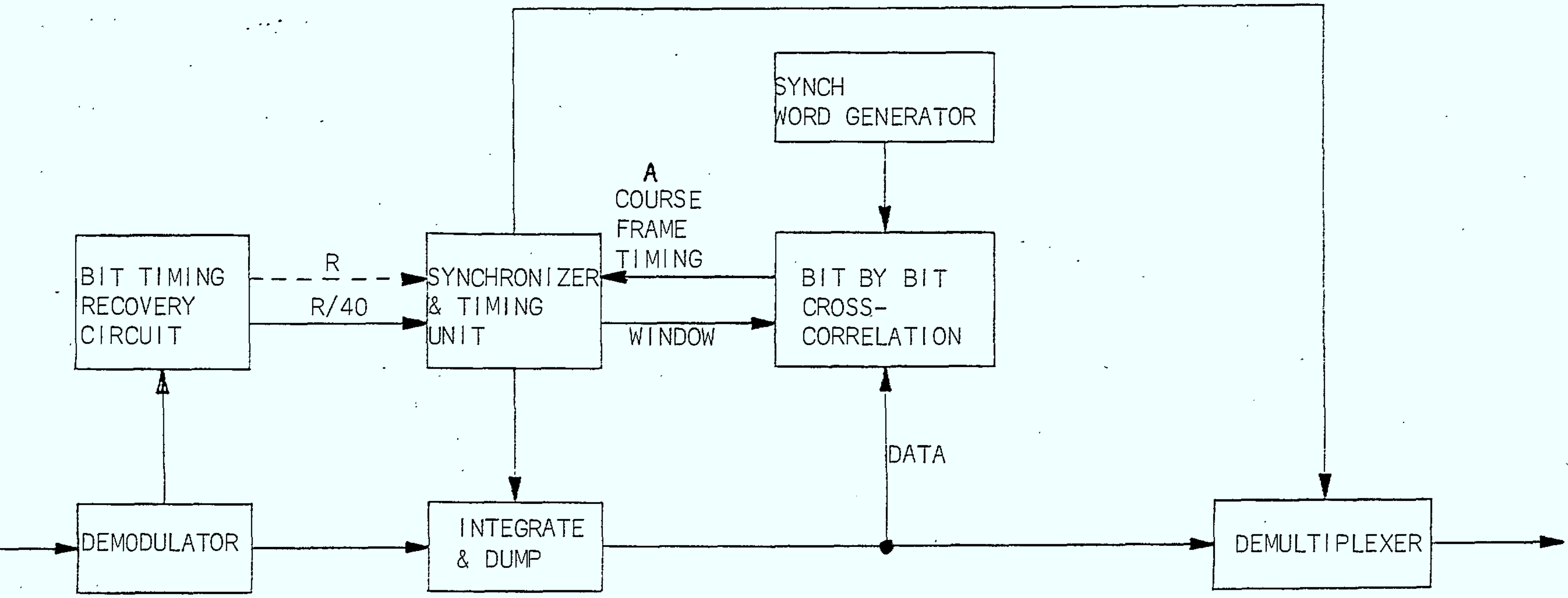


FIGURE 2.7

TDM RECEIVER TIMING RECOVERY

and NFA burst is an integer multiple of 40T a coherent clock component at R/40 will be present in the TDM carrier about 10% of the time. Figure 2.7 indicates that the bit timing recovery circuit must recover this component and provide the desired reference to the synchronization and timing unit.†

(ii) In the tracking mode, the fighter aircraft may continue to detect at the R/40 rate since separation between synch word and NFA burst is a multiple of 40T throughout the frame, hence R clock from the timing recovery circuit is not required in this case. A surface mobile must be capable of effective integrate and dump detection at various rates (R, R/2, R/4, R/40) throughout the frame, and hence needs a continuous accurate reference. This is best achieved by recovering the dominant R clock component in the signal, as shown in Figure 2.7, supplemented if necessary during the NFA burst by the R/40 clock.

(iii) Proper acquisition of frame synchronization for demultiplexing purposes at a surface mobile (i.e. rate R) requires a factor of 40 more accurate detection of the synch word correlation peak than that provided by the bit correlator (see Figure 2.7). This additional resolution, i.e., 1 in 40 decision, is made by the synchronizer and timing unit on the basis of R clock input. Hence R clock as well as R/40 clock is required to complete the acquisition process at a surface mobile.

None of the above constraints appears to seriously complicate the design of the system or jeopardize its performance.

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† The reference is actually required only during the period the synch word can be expected: e.g., if the NFA burst immediately precedes the synch word, clock need only be held for a relatively short period. The rise and fall of the R/40 clock component in the received spectrum could then be used to advantage and limit the Search interval further improving acquisition performance.

2.7 Digital Speech Interpolation

As indicated in Table 2.5, the key to the efficiency of a TDM system is the use of digital speech interpolation (DSI) to reduce by a factor of 2 required clear and secure voice transmission capacity. This results in an overall reduction in required power (i.e. satellite EIRP) and bandwidth of 1000/1.544 = .648 or 1.9 dB. In an FDM system, this power (but not bandwidth) advantage is realized through voice activation of the carriers. Unfortunately, DSI in a TDM system is more complex as it involves the continual reapportionment of available channels in the multiplex to active talkers, and the need to contend with overload (when the number of talkers exceeds the number of transmission channels).

Two DSI schemes, time-assigned speech interpolation (TASI) and speech predictive encoded communications (SPEC) have been analysed and tested.[4] In the absence of overload, these systems are equivalent. A 2N bit sub-frame preamble containing $M < N$ '1''s indicates which talkers are active and identifies those delta decoders which, in order, are to be replenished by the contents of the first M data fields of the sub-frame. The others will be squelched or artificially fed with an alternating '0101' pattern corresponding to low level noise. The sub-frame format is shown in Figure 2.8. The DSI frame efficiency is given by

$$E = \frac{L}{2N+L}$$

where L is the total number of bits per channel in a DSI frame. Very high efficiencies are achievable without voice clipping.

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[4] S. J. Campanella, "Digital Speech Interpolation", Comsat Technical Review, Spring 1976, pp. 127-158.

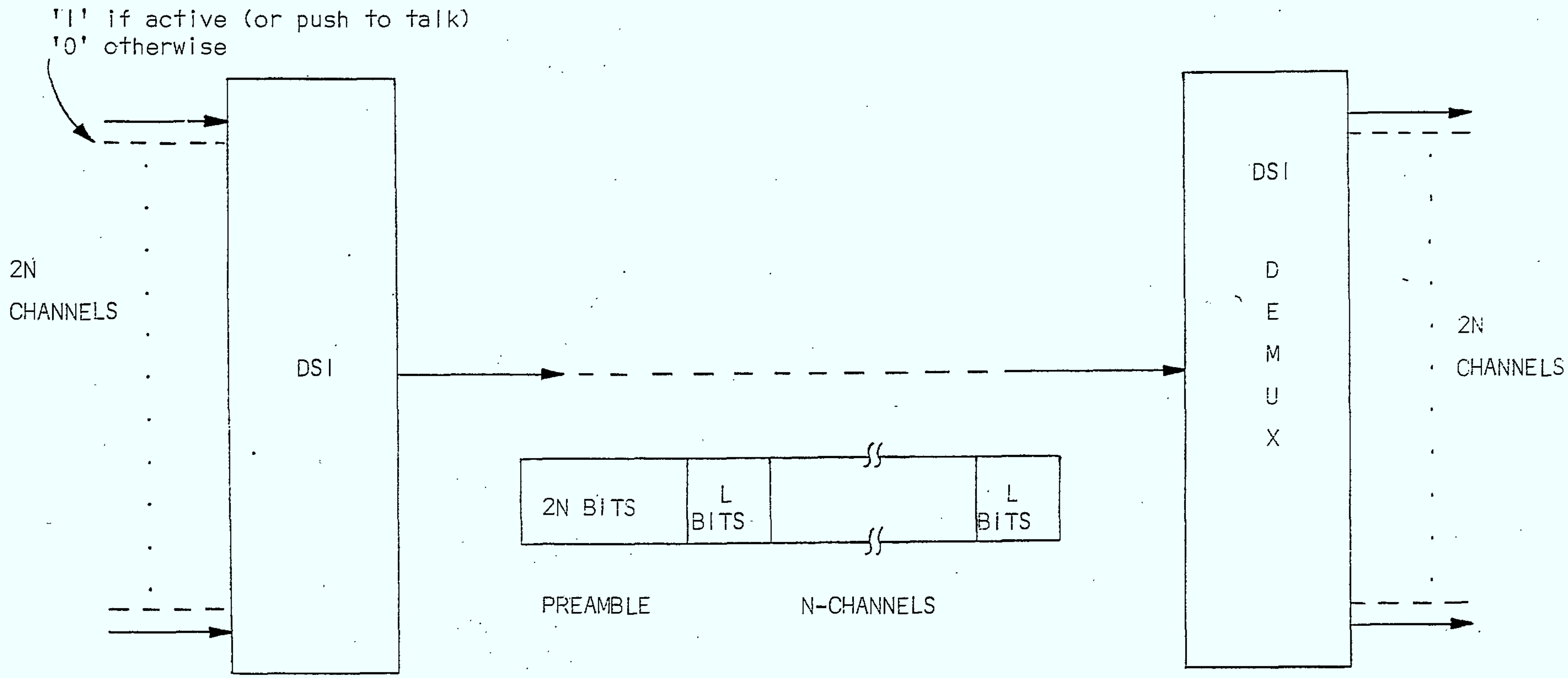

FIGURE 2.8

DSI TRANSMISSION FORMAT FOR SECURE/CLEAR VOICE CHANNELS

DSI operation in the absence of overload is rather simple, as shown in Figure 2.8. Most of the complexity of DSI multiplexers is contained in the logic required to handle the (infrequent) overload condition. TASI and SPEC handle overload differently. With TASI, the codec rate is reduced under overload conditions to accommodate the additional channels. This imposes the need for the multiplexer/demultiplexer to adapt the frame format. SPEC, a more easily implemented and higher performance scheme, replenishes active channels in decreasing order of differential (sample to sample) changes in amplitude level. All unreplenished active decoders automatically repeat the last word transmitted to them. A fixed multiplex format can be retained at the expense of one sample memory at the multiplexer and demultiplexer.

Both of these schemes are designed for PCM channels and are not suitable for delta channels. DSI multiplexing of delta channels does not appear to have been investigated. Limited examination by the author has demonstrated that considerable difficulty is posed by the companded, differential nature of delta. A variable codec rate (i.e, TASI) approach offers the greatest likelihood of success.

A DSI scheme which gracefully handles overload may not in fact be essential in this application. For many years analog TASI which briefly freezes out (in logical order) talkers has operated successfully over some submarine cable circuits. It is especially favoured here because demultiplexer/decoder logic at the mobiles would be greatly simplified for the following reason. When the number of active talkers exceeds N, the central station multiplexer simply selects (on a programmed basis to ensure the sharing of and hence minimum disruptions due to overload) N which will be serviced. The DSI frame preamble will never contain more than N '1' 's, and the DSI demultiplexer logic need not accommodate the overload condition.

The listeners, however, will be exposed to freeze out[5]. While statistical analysis could show the rate and duration of freeze outs to be small, small disruptions can be significant and subjective testing would be required to ensure acceptable performance. In particular, the combined effect of satellite delay and freeze out (which occurs in the middle of speech and is therefore not the same as clipping) on a telephone circuit has not been evaluated.

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[5] C. Weinstein, "Fractional Speech Loss and Talker Activity Model for TASI and for Packet-Switched Speech", IEEE Trans. Communications, August 1978, pp. 1253-1257.

2.8 Effect of Vehicle Dynamics

A continuous carrier, synchronous TDM link to the fighter aircraft is much less sensitive to doppler frequency modulation due to vehicle dynamics than is an FDM link for the following reasons:

(i) The doppler frequency shifts applied by the channel would be the same whether the carrier were TDM or FDM.

(ii) However, the faster signalling rate (25 kbps vs 2.4 kbps typically) and higher power of the TDM carrier allows about an order of magnitude larger carrier phase tracking bandwidth for the same degradation in demodulator performance.

This means the TDM link can tolerate doppler frequency offsets and rates of change of doppler about one order of magnitude larger than its FDM equivalent.

A further discussion and analysis of doppler, differential doppler and multipath effects on UHF transmission via geostationary satellite to an aircraft is given in [6].

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[6] L. Beaudet, "A Study of Binary DECPSK and DPSK for Aerosat Data Channels", to be released as CRC report (ref. Allan Sewards).

3.0 CONCEPTUAL DESIGN OF A TWO-LEVEL, SYNCHRONOUS, VARIABLE RATE TDM SYSTEM

The conceptual design of a two-level, synchronous, variable rate TDM system is presented in this section. Block diagrams of the TDM multiplexer and demultiplexer together with the details of one possible frame format are presented. The impact of the inclusion/exclusion of a fighter aircraft service on the multiplexer/demultiplexer and frame format is briefly discussed. Storage requirements for the compression buffers for the proposed frame format are also given.

In order to satisfy the forward link traffic requirements, the application of DSI (digital speech interpolation) is assumed for both secure and clear voice Δ channels.

With the exceptions of the secure and clear voice circuits (assumed synchronously encoded), all traffic is assumed to be asynchronous with respect to the CCS station clock.

3.1 TDM Multiplexer

The TDM multiplexer is required to multiplex the following traffic:

1. 1 channel of 2.4 kbps asynchronous data for the new fighter aircraft (NFA) - optional
2. 3 channels of 2.4 kbps asynchronous data (pre-assigned)
3. 20 channels of 2.4 kbps asynchronous data (demand-assigned)
4. 10 channels of 75 baud teletype (TTY) (demand-assigned)
5. 20 channels of 300 baud teletype (TTY) (demand-assigned)
6. 16 channels of synchronously encoded 16 kbps Δ secure voice (demand-assigned)
7. 40 channels of synchronously encoded 16 kbps Δ clear voice (demand-assigned).

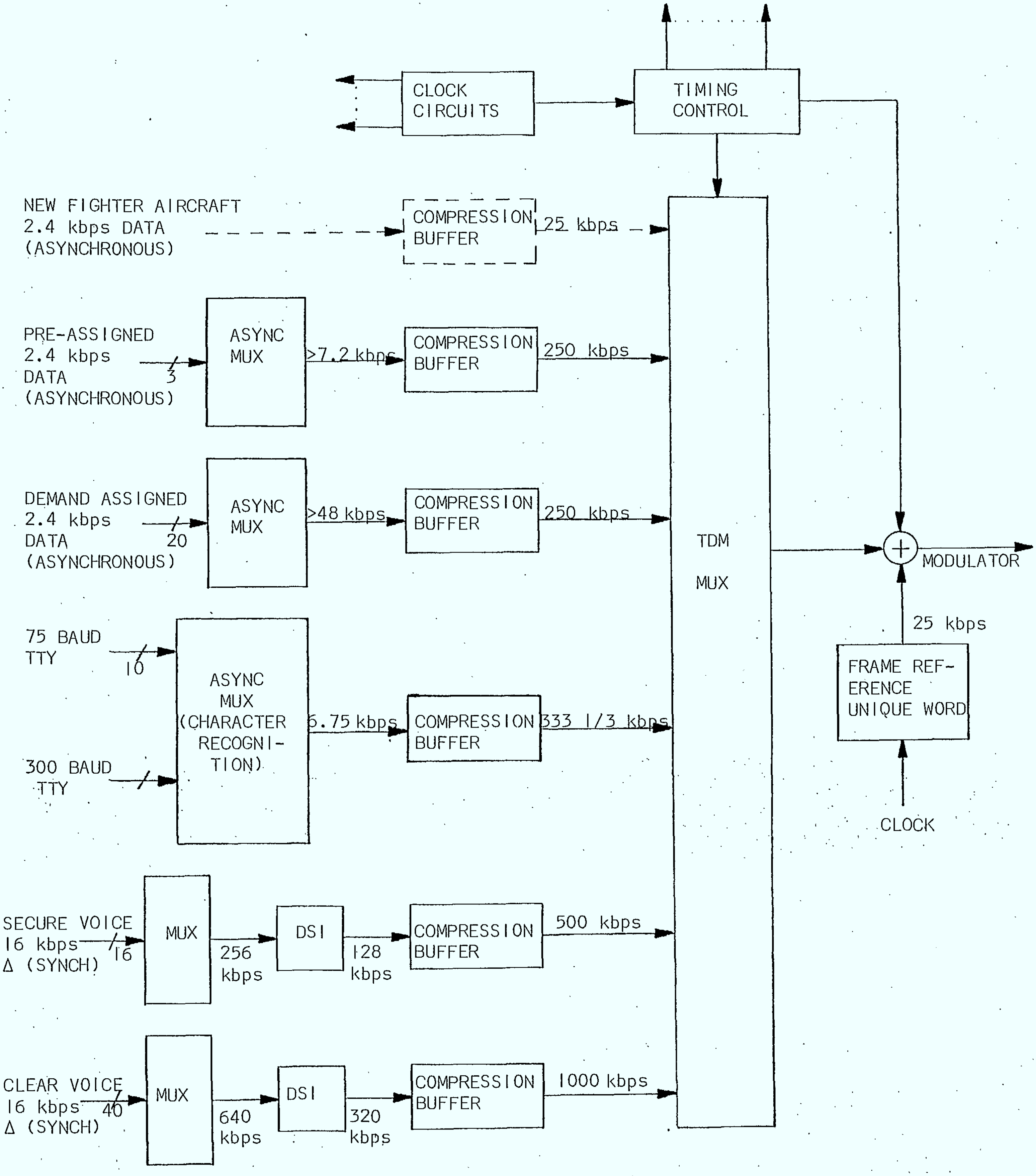
Figure 3.1 illustrates the block diagram of the conceptual TDM multiplexer.

Basically two stages of multiplexing are employed. At the first stage, the outgoing data channels are multiplexed into a single synchronous bit stream which is then written into a two-stage compression buffer. This occurs for each type of traffic except for the 16 kbps Δ voice channels (clear and secure) for which the composite bit streams are first processed in the DSI equipment before being written into their respective compression buffers.

To facilitate frame formatting, the pre-assigned and demand-assigned 2.4 kbps data channels are not combined together into a single

FIGURE 3.1

TDM MULTIPLEXER BLOCK DIAGRAM



data stream at the first stage of multiplexing although their TDM burst rates are identical.

At the second stage of multiplexing, the compression buffers are periodically dumped at their respective synchronous TDM transmit rates under the control of the timing control unit to form a composite multi-rate, synchronous TDM bit stream at the output of the TDM multiplexer. After appending the frame reference unique word (FRUW) required for frame synchronization at the receive mobile terminals, the TDM bit stream is routed to the modulator for transmission.

A brief description of the multiplexing subsystems follows:

NFA 2.4 kbps Asynchronous Data

To indicate its optional inclusion in the system, the NFA data path is depicted by dotted lines. It is clear from Figure 3.1 that the inclusion or exclusion of this service has minimal hardware impact on the TDM system.

The inclusion/exclusion of the NFA service does, however, impact on the TDM transmit rate for the FRUW since the lowest G/T terminal (i.e. the fighter aircraft) in the system must be capable of correctly detecting the FRUW. Inclusion of the NFA service requires the FRUW to be transmitted at the same TDM rate as the former, i.e., 25 kbps (assuming bit by bit correlation at NFA).

Consequently, the TDM demultiplexers in land mobiles will be required to recover an R/40 clock component (in addition to an R-clock component) in order to also correctly retrieve the FRUW (see section 2.6).

Exclusion of this service would clearly nullify the requirement for the FRUW to be transmitted at an R/40 rate, and hence the need to recover an R/40 clock component in land mobiles, in addition to freeing approximately 10% of the TDM frame capacity.

The FRUW must either be inserted immediately after all NFA data bursts or be located at an integral number of 40T time intervals apart (see section 2.6). The latter approach has been adopted in the conceptual design.

Pre-assigned 2.4 kbps Asynchronous Data

Since the incoming data channels are assumed to be asynchronous with respect to the CCS station clock, asynchronous multiplexing (pulse stuffing) of the three pre-assigned data channels (demand-assignment control channel plus two fleet broadcast channels) will be required.

The stuff control and framing bits to enable proper de-stuffing at the receiving terminals are multiplexed along with the information and stuffed bits to produce a composite synchronous bit stream at a net throughput rate slightly greater than three times the incoming data rate of 2.4 kbps i.e. 7.2 kbps. The bit stuff rate will be determined by the maximum expected uncertainty in the incoming 2.4 kbps data clocks.

The output synchronous bit stream is next written into a two-stage compression buffer. Data is periodically read out of the buffer at a TDM burst rate of 250 kbps.

Demand-assigned 2.4 kbps Asynchronous Data

Asynchronous multiplexing of the 20 demand-assigned 2.4 kbps data channels is accomplished in the same manner as in the previous case i.e. through bit stuffing synchronization. The net throughput rate in this case, however, will be slightly more than 20 X 2.4 kbps or 48 kbps. The TDM transmit rate (buffer read rate) for this service is 250 kbps.

75 Baud and 300 Baud TTY

Since TTY transmissions occur on a character by character basis at some average character rate, they are asynchronous in nature. Consequently an asynchronous data multiplexer capable of simultaneously receiving at either baud rate and of outputting a composite synchronous bit stream at a net throughput rate of 6.75 kbps to the compression buffer will be required.

Pulse stuffing synchronization will not be required; instead some form of teletype character recognition technique together with a suitable synchronous encoding technique might be employed to ensure that the transmission path appears transparent to both the transmitting and receiving TTY terminals.

The TDM transmit rate for this service will be 333 1/3 kbps.

16 kbps Δ Secure Voice (Synchronously Encoded)

The incoming 16 kbps Δ (synchronously encoded) channels are synchronously multiplexed into a composite bit stream with a net throughput rate of 256 kbps and then routed to the DSI equipment. At the output of the DSI equipment is a 128 kbps data stream consisting of 8 data channels preceded by a coded DSI speech assignment word (SAW) to enable proper allocation of the transmitted data bits at the receiving terminals.

Since the 16 bit (uncoded) DSI speech assignment word must be received at a bit error rate (BER) of typically 1 in 10^7 or better, coding of the SAW will be required since the bit error rate of the transmission path for the secure voice transmission will only be 1 in 10^3. For coherent PSK transmission, a coding gain of approximately 4.5 dB will be required to ensure detection of the SAW at a BER of 1 in 10^7. For a 16 bit word, this will require a coded word of length 16 x $\text{antilog}_{10}(0.45)$ or 45 bits. For ease of implementation, a 48 bit coded SAW word will be broken into four rate 1/3 identically coded 12 bit blocks which are easily decoded and each undergo the necessary 0 dB real coding gain. (Majority logic coding could lead to even simpler decoding, but would require a longer SAW word to compensate the negative real coding gain).

The 128 kbps output of the DSI equipment is next routed to the two-stage compression buffer which is periodically depleted at a TDM burst rate of 500 kbps.

16 kbps Δ Clear Voice (Synchronously Encoded)

Similarly to the secure voice circuits, the 40 incoming clear voice channels are synchronously multiplexed into a composite bit stream at a net throughput rate of 640 kbps which is then routed to the DSI equipment. The latter outputs a data stream at a rate of 320 kbps consisting of 20 data channels preceded by a 200 bit coded (40 bits uncoded) DSI SAW word. Similarly to the secure voice transmission, for detection of the SAW word at a BER of 1 in 10^7 over a transmission path with a BER of 1 in 10^2 (in this case), a coding gain of approximately 7 dB will be required. For a 40 bit SAW word, this will require a coded word of length 40 x $\mathrm{anilog}_{10}(0.7)$ or 200 bits. Again this coding gain might be realized with some simple coding scheme.

The output data stream of DSI equipment is written into and read out of the compression buffer at 320 kbps and 1000 kbps respectively.

Frame Reference Unique Word (FRUW)

To facilitate TDM frame synchronization, a 20 bit FRUW (reference Section 2.6) is inserted at the start of each TDM frame. If the NFA service is included, the FRUW TDM burst rate will be 25 kbps; if not, the FRUW may be transmitted at the 1 Mbps rate of the clear voice service.

3.2 TDM Frame Format

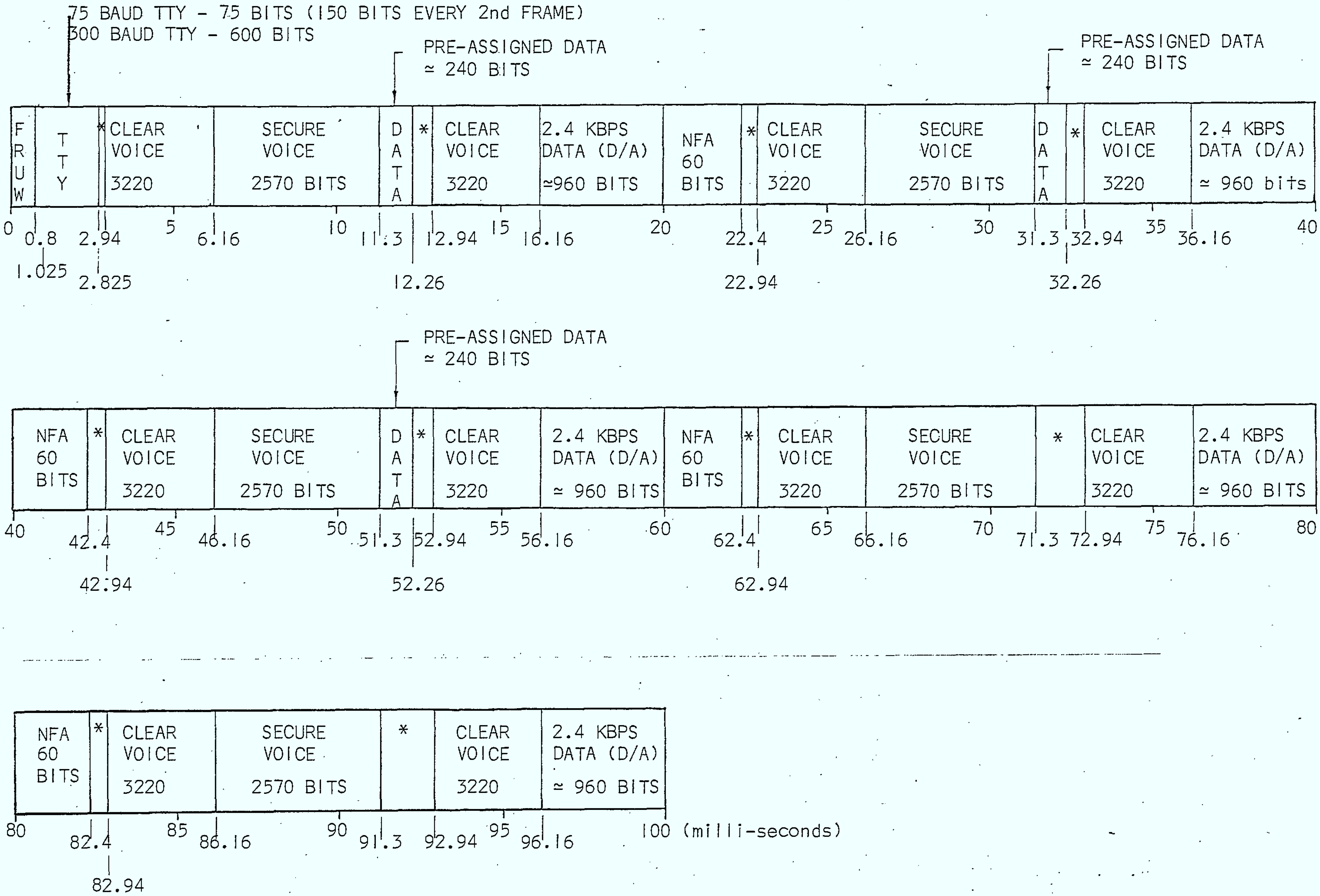
For TDM operation, the frame rate is typically determined by the character rate of the slowest service multiplexed i.e., 75 baud TTY in this case. Assuming an average character bit rate of 10 bits/character/second, a TDM frame rate of 10 frames/second is selected for the conceptual TDM system. This effectively sets the TDM frame length at 100 milliseconds.

The sub-frame repitition rate within a TDM frame, where applicable, has been selected in such a manner as to allow the realization of the compression buffers with standard capacity (e.g. 256 bit, 1024 bits, etc.) static RAM's (random access memories) not exceeding 4 kilo bits (4,096 bits) of storage. A proposed TDM frame format is illustrated in Figure 3.2 and the distribution of the bits per sub-frame for the various services is given in Table 3.1. Note that for the pre-assigned and demand-assigned 2.4 kbps data, the sub-frame time intervals are calculated for net throughput rates of 7.2 and 48 kbps respectively. These, therefore, do not take into account the overhead transmission time required to accomodate the necessary additional stuffed pulses and stuff control signalling bits as indicated earlier.

A summation of the sub-frame time (Table 3.1) for each of the services yields a total burst time of 92.405 ms. Thus, for a 100 ms frame length, there is about 7.595 ms of frame time remaining for such overhead functions as switching guard times, pulse stuffing control and frame bits for the 2.4 kbps data service (pre-assigned and demand-assigned), signalling, etc.

A guard time of one overhead symbol between sub-frames is required to accommodate finite linear mux/demux switching times and ensure transients are not applied to the modulation or outer demultiplexer.

The distribution of empty data slots as shown in Figure 3.2 therefore depicts a somewhat idealised situation. Depending on the frame overhead, some of the spare time slots might be used to accomodate a limited expansion of 2.4 kbps data or TTY services.



* SPARE TIME SLOTS

FIGURE 3.2

TDM FRAME FORMAT

SERVICE	BITS PER 100 ms FRAME	TDM RATE	NO. OF SUB-FRAMES	NO. OF BITS PER SUB-FRAME	SUB-FRAME TIME	TOTAL SUB-FRAME TIME PER FRAME
2.4 kbps DATA (NFA)	240	25 kbps	4	60	2.4 ms	9.6 ms
2.4 kbps DATA (P/A)	$\simeq$ 720	250 kbps	3	$\simeq$ 240	$\simeq$ 1.92 ms	$\simeq$ 2.88 ms
2.4 kbps DATA (D/A)	$\simeq$ 4,800	250 kbps	5	$\simeq$ 960	$\simeq$ 3.84 ms	$\simeq$ 19.2 ms
75 Baud TTY	75 (150 bits every 2nd frame)	333 1/3 kbps	1	75	0.225 ms	0.225 ms
300 Baud TTY	600	333 1/3 kbps	1	600	1.8 ms	1.8 ms
16 kbps Δ Secure Voice	12,850	500 kbps	5	2,570	5.14 ms	25.7 ms
16 kbps Δ Clear Voice	32,200	1,000 kbps	10	3,220	3.22 ms	0.8 ms
						TOTAL $\simeq$ 92.405 ms

TABLE 3.1

3.3 TDM Demultiplexer

At the mobile receive terminals (including the fighter aircraft), the demodulator outputs a replica of the original transmitted multi-rate TDM bit stream. The R and R/40 (if required) clock components are derived in the clock recovery unit and are transmitted to the receiving timing control unit (see Figure 3.3). Frame synchronization is established by verifying acquisition of the FRUW over several TDM frames (frame sync acquisition and tracking are described in Section 2.6). Once frame sync is established, this condition is signalled to the timing control unit and the demultiplexer is then ready to retrieve the multiplexed data services from the multi-rate TDM data stream.

As in the TDM multiplexer, there are two stages of demultiplexing. The first stage involves the recovery of the composite data stream from the multi-rate data stream and is accomplished by varying the integrating interval (T,2T, ..., 40T) of the integrate and dump unit at the appropriate instant and enabling the appropriate demultiplexer downchain. This process is accomplished by the synchronizer and timing control unit.

Individual data channels are demultiplexed in the reverse order of the multiplexing process. The service channel(s) that are received at any one mobile terminal will be determined by the in-station DAMA unit in response to control information previously transmitted over the 2.4 kbps pre-assigned DACC channel.

For the case of the pre-assigned and demand-assigned 2.4 kbps data service which were asynchronously multiplexed, phase-locked loops are utilized in the second stage of demultiplexing to smooth the output data after deletion of the stuffed bits.

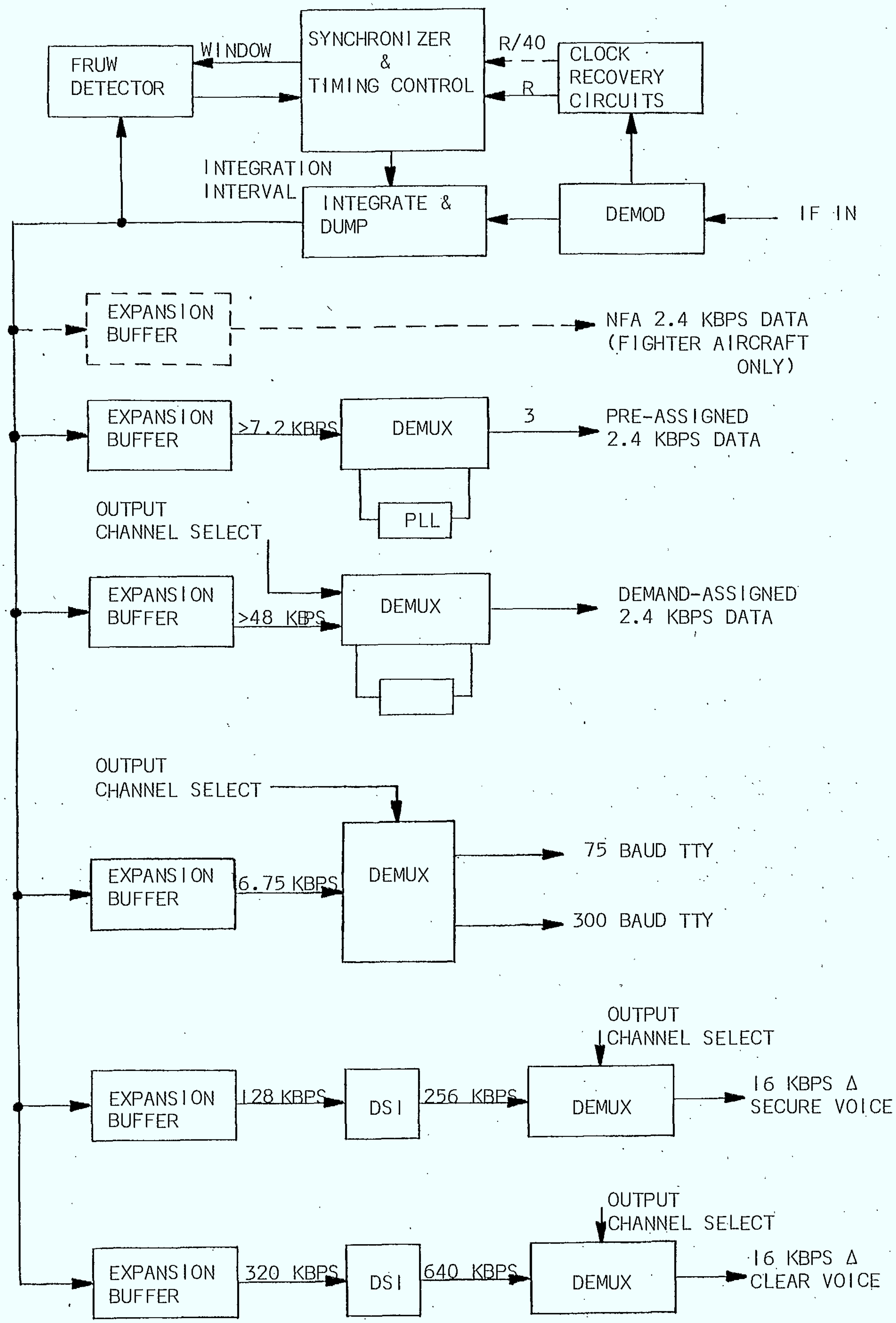


FIGURE 3.3

TDM DEMULTI-
PLEXER BLOCK
DIAGRAM

3.4 Compression/Expansion Buffer Requirements

With the use of two-stage compression buffers and the frame format indicated in Figure 3.2, the storage capacity of the compression and expansion buffers for the CCS and mobile terminals respectively are indicated in Table 3.2. Note that mobile terminals will not be required to receive the NFA data channel; the NFA will be required to receive only the pre-assigned NFA 2.4 kbps data channel.

As noted in Section 3.2, frame formatting has been selected in a manner which allows the realization of the compression buffers using standard capacity static RAM's.

SERVICE	COMPRESSION/EXPANSION BUFFER REQUIREMENTS
NFA 2.4 kbps Data	2 X 256 bit RAM *
2.4 kbps P/A Data	2 X 1 kbit "
2.4 kbps D/A Data	2 X 1 kbit "
75 Baud TTY 300 Baud TTY	2 X 1 kbit "
16 kbps Δ Secure Voice	2 X 4 kbit "
16 kbps Δ Clear Voice	2 X 4 kbit "

Total No. of 256 bit RAM = 2
Total No. of 1 kbit RAM = 6
Total No. of 4 kbit RAM = 4

TABLE 3.2

* Random Access Memory

4.0 COMPARISON OF TDM VS FDM APPROACH

The following summarizes apparent advantages and disadvantages of TDM over FDM:

(i) By permitting the transponder and earth station high power amplifier (HPA) to be operated close to saturation without intermodulation, a bandwidth and power limited TDM system typically requires about 4 dB less satellite EIRP and possibly a smaller HPA at the CCS than its FDM equivalent[7]. Operation of the satellite at saturation also reduces sensitivity to up-link fading and transmit power variations from the CCS[8]. To support TDM transmission, the transponder must possess a flat amplitude and group delay response.

(ii) A TDM system without DSI loses 1.9 dB of this advantage as compared to an FDM system which employs voice activated carriers. DSI schemes with graceful overload characteristics would significantly complicate the field terminal. Time assignment speech interpolation (TASI) that permits freeze out can be simply implemented; however, degraded circuit performance as compared to voice activated FDM operation will result. Unlike voice activated FDM operation, DSI in a TDM system saves bandwidth as well as power.

(iii) The proposed TDM system employs the same non-agile receiver (demodulator and demultiplexer) at all field terminals, including the fighter aircraft. A receive programmable synthesizer is unnecessary, lower downconverter short term and long term stabilities are tolerable, and separate channel equipment is not required for reception of each voice and data channel, including the demand assignment channel.

(iv) Acquisition of the demand assignment channel following equipment turn-on will be greater with TDM than FDM because of the need to first acquire frame synchronization. Once acquired, however, the TDM receiver can switch between channels in the

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[7] J.G. Puente, W.G. Schmidt, A.M. Wertin, "Multiple Access for Commercial Satellites, Proc. IEEE, Feb. 1971.

[8] R.G. Lyons, "The Evaluation of Satellite Link Availability", IEEE Transactions, Vol. Com-26, June 1978, pp. 847-853.

multiplex much more quickly than a frequency selective FDM receiver.

(v) While a TDM approach results in some simplification of the field terminal downconverter, IF common and demodulator equipment, it introduces the requirement for considerable equipment between the demodulator and data interface, viz., a two level TDM demultiplex with DSI, synchronization unit (to operate and switch as necessary between acquisition and steady state tracking modes), and variable rate synchronous clock source (controls both integrate and dump and demultiplex operations). Neglecting development costs (SCPC equipment is well developed), the cost of a TDM receiver is expected to exceed that of an FDM receiver by an amount which is very small compared to the overall cost of the terminal.

(vi) Since its operation depends almost completely on logical processing (i.e. digital and microprocessor circuits), and in no way stresses the transmission system, the risk associated with the TDM approach is minimal.

(vii) The main advantage of a TDM approach, apart from a reduction in satellite EIRP, is the resulting flexibility in selection of transmission rates to match operating conditions without any need to alter hardware (i.e. accommodated by software and programmable multiplexer). This flexibility permits the easy addition of a new class of field terminals to the network; the trading of margin for total systems capacity, possibly dynamically (i.e. instead of blocking under overload conditions rates are adjusted to increase capacity at reduced margin); and reduced capacity operation during eclipse.

(viii) A TDM system, because of its higher carrier power and bandwidth, is less sensitive to narrow band jamming than an FDM system. It is also more secure, and could be made very secure (without encryption of each individual channel) by periodically reprogramming the frame format via a single secure control channel.

(ix) A TDM link to the fighter aircraft is an order of magnitude more tolerant of all vehicle dynamics than is an FDM link.

(x) In terms of size, weight and power, a TDM receive terminal is at some disadvantage primarily because of the higher speed demodulator and buffer memories required. This discrepancy diminishes as the terminal capacity requirements increase (with an FDM system programmable synthesizers and demodulators with independent timing circuitry, etc. must be replicated). With the ready availability of low cost, reliable single chip RAM memories operating at the required speed and consuming only a few milliwatts, the TDM equipment should have a little incremental impact on the overall size and weight of the mobile terminal, and power consumption will continue to be dominated by the transmit portion of the terminal.

5.0 CONCLUSIONS

This study has revealed that a variable rate, synchronous TDM method of implementing the MUSAT forward link is a low risk technical solution which offers advantages of flexibility and may be economically attractive. This depends on the reduction in space segment cost due to reduced EIRP requirements versus a possible increase in ground segment cost (primarily at the CCS) due to buffering and synchronization requirements. This economic tradeoff was not quantified in the study, but should be prior to making a choice. Further technical evaluation of the FDM and TDM alternatives is not required to make the choice; in the view of the authors, the overall economica will probably favour a TDM solution.

A TECHNICAL STUDY ON THE FEASIBILITY AND CONCEPTUAL DESIGN OF A TDM FORWARD LINK FOR MUSAT: FINAL REPORT