ARINC CHARACTERISTIC 741

AVIATION SATELLITE COMMUNICATIONS SYSTEM

PART 3

CIRCUIT MODE VOICE AND DATA SERVICES

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FOREWORD

Activities of AERONAUTICAL RADIO, INC. (ARINC)

and the

Purpose of ARINC Characteristics

Aeronautical Radio, Inc. is a corporation in which the United States scheduled airlines are the principal stockholders. Other stockholders include a variety of other air transport companies, aircraft manufacturers and non-U.S. airlines.

Activities of ARINC include the operation of an extensive system of domestic and overseas aeronautical land radio stations, the fulfillment of systems requirements to accomplish ground and airborne compatibility, the allocation and assignment of frequencies to meet those needs, the coordination incident to standard airborne communications and electronics systems and the exchange of technical information. ARINC sponsors the Airlines Electronic Engineering Committee (AEEC), composed of airline technical personnel. The AEEC formulates standards for electronic equipment and systems for the airlines. The establishment of Equipment Characteristics is a principal function of this Committee.

An ARINC Equipment Characteristic is finalized after investigation and coordination with the airlines who have a requirement or anticipate a requirement, with other aircraft operators, with the Military services having similar requirements, and with the equipment manufacturers. It is released as an ARINC Equipment Characteristic only when the interested airline companies are in general agreement. Such a release does not commit any airline or ARINC to purchase equipment so described nor does it establish or indicate recognition of the existence of an operational requirement for such equipment, nor does it constitute endorsement of any manufacturer's product designed or built to meet the Characteristic. An ARINC Characteristic has a twofold purpose, which is:

(1) To indicate to the prospective manufacturers of airline electronic equipment the considered opinion of the airline technical people, coordinated on an industry basis, concerning requisites of new equipment, and

(2) To channel new equipment designs in a direction which can result in the maximum possible standardization of those physical and electrical characteristics which influence interchangeability of equipment without seriously hampering engineering initiative.

ARINC CHARACTERISTIC 741
PART 3
TABLE OF CONTENTS

ARINC CHARACTERISTIC 741
PART 3
TABLE OF CONTENTS

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5.0 VOICE ENCODER STANDARDS

5.1 Introduction

This section describes, in a high-level language, the 9.6 kbit's multipulse-excited LPC voice coding algorithm recommended by the AEEC's satellite subcommittee for aeronautical mobile satellite communications. The voice coding algorithm is described in sufficient detail in this document, so that those skilled in the art can implement the algorithm precisely, in accordance with the standard.

The WE DSP32 (or the WE DSP32C) 32 bit floating point digital signal processor (DSP) is the preferred digital signal processing device for the implementation of the algorithm. DSP32 source code, corresponding to the algorithm description of this document, will be provided, royalty free, for aeronautical mobile satellite applications only, in accordance with the license statement submitted by British Telecom.

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Note: This voice encoding algorithm is subject to BT patent rights and copyrights. BT has agreed to grant royalty-free non-exclusive licenses under such patent rights and copyrights to all manufacturers of AES and GES equipment operating in the AMSS service. These manufacturers should enter into a royalty-free license agreement with BT Laboratories prior to incorporating the algorithm in equipments operating in the AMSS service.

5.2 General Introduction to Multipulse-Excited LPC

This section, starting from conventional LPC synthesis, provides a short introduction to multipulse-excited LPC speech coding.

5.2.1 LPC Synthesis

A simplified speech production model is presented in Figure 5-1. In this speech production model, the vocal-tract filter models the response of the vocal-tract, and the remaining elements model the excitation. The excitation model requires a decision as to whether the speech to be synthesized is voiced or unvoiced. If the speech is voiced, the voiced/unvoiced switch selects the upper excitation model, and unipolar, unity amplitude pitch pulses excite the vocal-tract filter. The pitch period of the voiced speech is controlled by the pitch period input to the pulse generator. For unvoiced speech, the voiced/unvoiced switch selects the lower excitation model, and the random noise generator excites the vocal-tract filter. For both voiced and unvoiced speech, the gain value is periodically updated to control the amplitude of the synthesized speech. For a LPC synthesizer, the vocal-tract filter model is an all-pole, time-varying digital filter. The filter coefficients are updated typically every 10-20ms; this is sufficient to model the slowly varying characteristics of the vocal-tract.

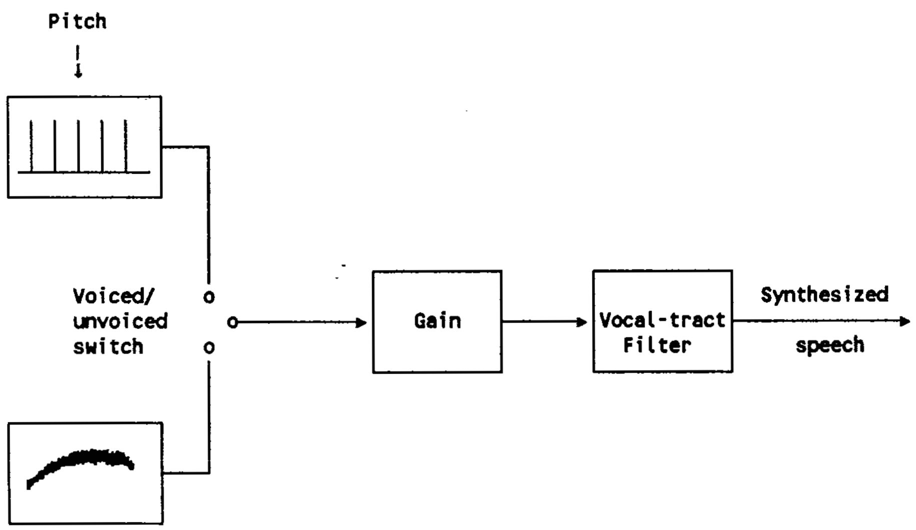


Figure 5-1 Simplified Speech Production Model

5.0 VOICE ENCODER STANDARDS (cont'd)

5.2.2 Multipulse-Excited LPC

A multipulse-excited LPC synthesizer is illustrated in Figure 5-2. The vocal-tract response is modelled by the same all-pole, time-varying digital filter as used in conventional LPC synthesis. The multipulse excitation model, however, does not distinguish between voiced and unvoiced speech; the excitation in both cases consists of a series of non-uniformly spaced pulses, having differing amplitudes. Multipulse-excited LPC thus avoids the need to make voiced/unvoiced decisions and pitch period estimations and replaces these by the determination of appropriate pulse positions and amplitudes. The pulse positions and amplitudes are determined for excitation frame durations of typically between 4 and 20 ms. The quality of the synthesized speech from a multipulse coder is determined by, among other things, the number of pulses per excitation frame - the more pulses per frame, the better the speech quality. All pulse positions and amplitudes, however, must be quantized and transmitted; therefore, the maximum number of pulses per frame is limited by the desired transmission bit-rate. A multipulse-excited LPC encoder is presented in the next section.

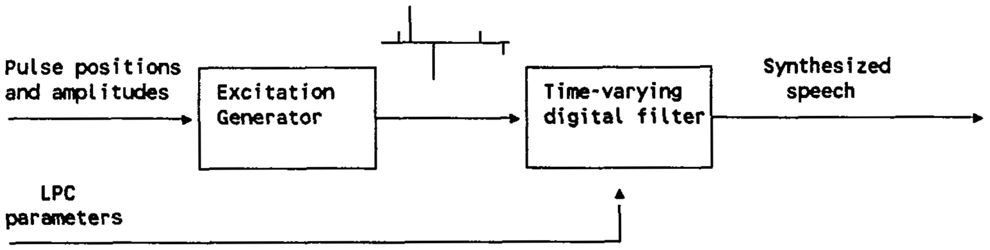


Figure 5-2 Multipulse-Excited LPC Synthesizer

5.2.3 Multipulse-Excited LPC Encoder

A multipulse-excited LPC encoder is illustrated in Figure 5-3. The encoding process involved splitting the incoming speech into frames of 10-20 ms duration to derive the LPC coefficients, followed by an analysis-by-synthesis process to obtain the pulse positions and amplitudes. The excitation analysis-by-synthesis procedure involves a search for the pulse positions and amplitudes which minimize the squared error between a block of the input and synthesized speech signals. The derivation of the excitation may be carried out for the entire LPC block length or the LPC frame may be split into several excitation frames and the excitation derived for each excitation frame separately. To find all the pulse positions and amplitudes simultaneously for a frame, requires the solution of non-linear equations and is thus very complex. Sub-optimal methods have therefore been developed, to derive the pulse positions and amplitudes sequentially.

The long-term predictions included in the encoding process takes advantage of the long-term correlations in speech, primarily as a result of pitch related correlations in voiced speech. With the inclusion of long-term prediction less pulses are required per pitch period to obtain the same speech quality.

5.2.4 Multipulse-Excited LPC Decoder

The multipulse-excited LPC decoder corresponding to the encoder of Figure 5-3 is presented in Figure 5-4. The decoding process is very straightforward requiring the formation of the excitation signal and the application of this excitation to the long-term and short-term predictors to synthesis the output signal.

5.3 Multipulse-Excited LPC Encoding

Extensive use of flow charts and a high-level language description are used in this section to define the encoding operations. A schematic representation of the overall processing algorithm is presented in Figure 5-11. However, before commencing the detailed description of the encoder, it is important to examine the framing structure which is employed by the encoder and outline the conventions used in the high-level language description which follows.

5.3.1 Introduction

The incoming speech signal, sampled at 8 kHz, is subject to the framing structure outlined below.

5.0 VOICE ENCODER STANDARDS (cont'd)

5.3.1.1 Encoder Framing Structure

The encoder uses three frame sizes, namely:

1. A 32 ms (256 sample) "window frame."
2. A 20 ms (160 sample) "speech frame."
3. A 4 ms (32 sample) "excitation frame."

The relationship of these three frames to one another is illustrated in Figure 5-5. As this figure illustrates the window frames are arranged such that the first and last 12 ms of each frame is overlapped with the previous and subsequent window frames. Each overlapped window frame is used in the calculation of the LPC coefficients. However, the LPC coefficients calculated from the window frame are only considered valid over the corresponding speech frame, i.e., the central 20 ms (160 samples) of the window frame.

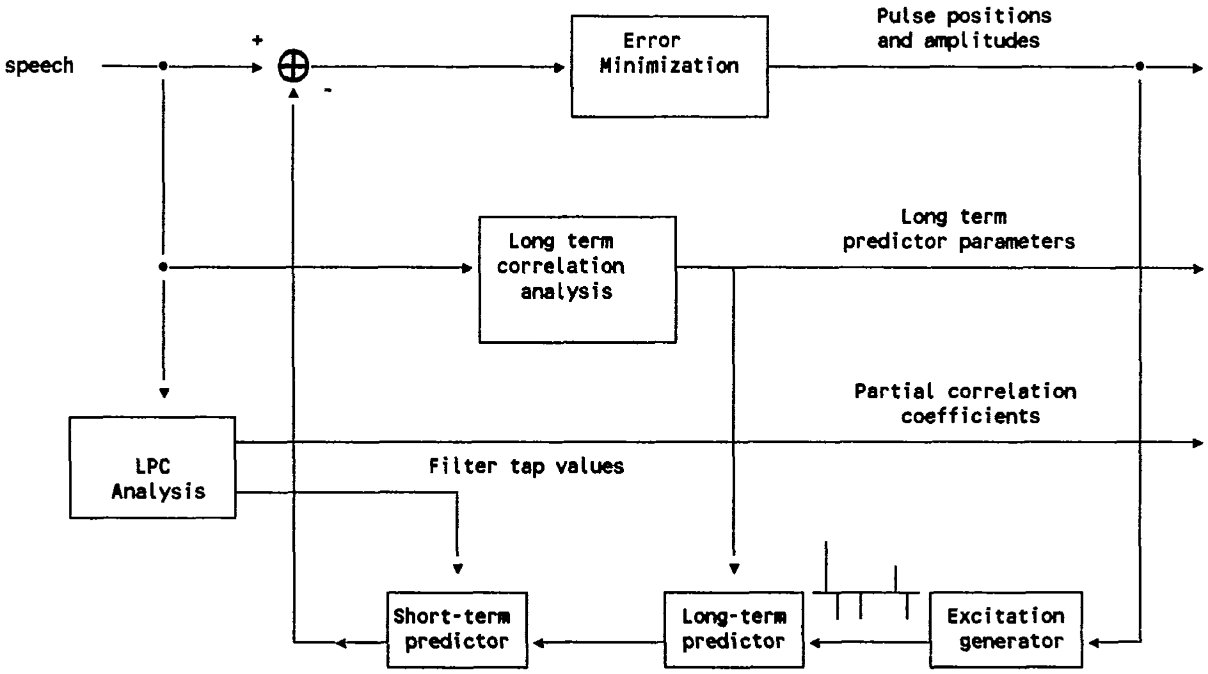


Figure 5-3 Multipulse-Excited LPC Encoder

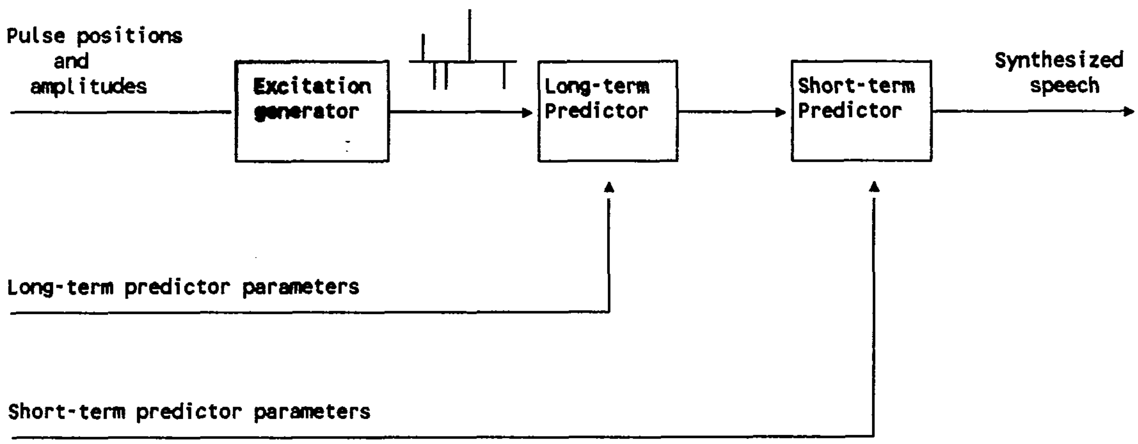


Figure 5-4 Multipulse-Excited LPC Decoder with Long-Term Prediction

5.0 VOICE ENCODER STANDARDS (cont'd)

5.3.1.1 Encoder Framing Structure (cont'd)

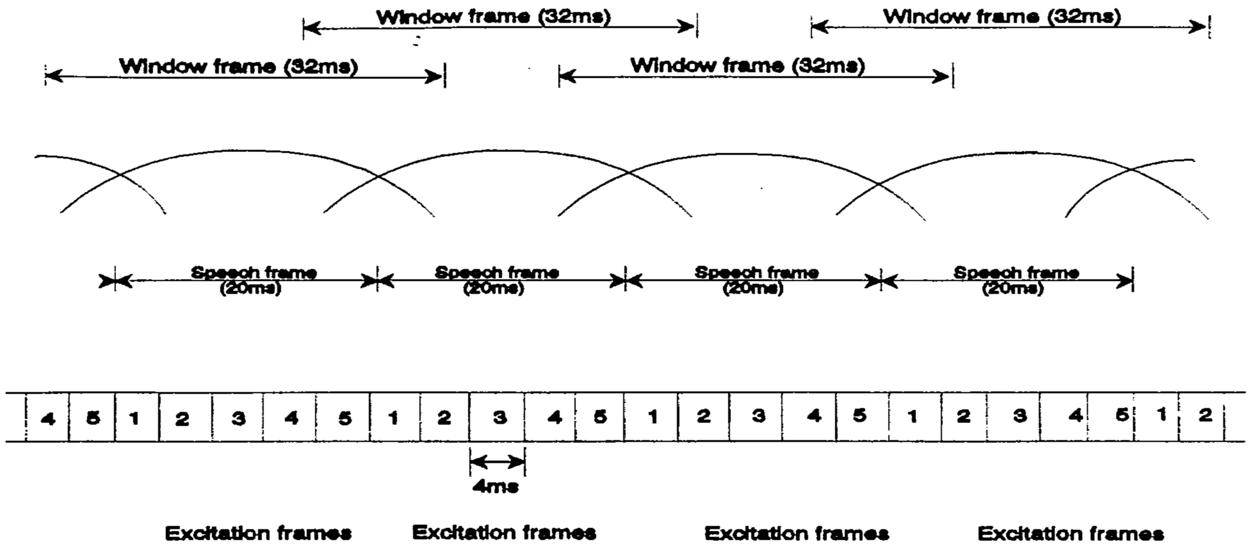


Figure 5-5 Relationship Between the Various Frames

The speech frame as well as being the frame size used in the updating of the LPC coefficients is also the frame size used in the updating of the long-term correlation parameters.

The excitation parameters, i.e., the pulse amplitudes and positions, are updated every excitation frame. The excitation calculation loop is thus executed 5 times for each execution of the LPC and long-term correlation calculation loops.

5.3.1.2 High-Level Language Conventions

The following conventions have been adopted for the high-level language algorithm descriptions of this and subsequent sections.

The high-level code description is based on the high-level language Pascal.

All high-level code is printed in *italics* and all high-level variable names are also printed in *italics* in the associated text, e.g., the scalar variable, 'scalar', would be printed in the text as, *scalar*.

Vectors or two dimensional arrays are distinguished from scalar variables in the text by appending '[]' to the variable name; e.g. the vector variable, 'vector', would be printed in the text as, *vector[]*.

All variables should be considered to be 32 bit floating point values unless otherwise stated. The exception to this rule is loop variables and indices which are always 16 bit variables.

5.3.1.3 Requirements for Numeric Calculations

Except where explicitly noted, and except for obvious integer variables such as vector indices and loop counters, all variables shall be maintained as floating-point values with a minimum mantissa precision of 24 bits inclusive of a sign bit. The range of the associated exponent shall be such that the magnitude of a positive or negative non-zero variable can vary over a minimum range of $5.87747*10^{39}$ to $3.40282*10^{38}$.

Note: Compliance with this minimum numeric range is essential if interoperability between different vocoder implementations is to be ensured.

5.0 VOICE ENCODER STANDARDS (cont'd)

5.3.1.4 Audio Input Characteristics

5.3.1.4.1 Audio Input Spectral Characteristics

Prior to sampling and quantization, speech audio which is presented to the encoding algorithm shall exhibit the following minimum spectral characteristics. These characteristics may be met by any combination of specific anti-aliasing filters and/or the natural spectral roll-off characteristics of transducers and audio channels:

Note: All specifications are referenced to 1020 Hz.

a) Over a passband of 300 to 3400 Hz, a gain variation of no more than ± 2 dB;
b) Over a transition band of 4000 Hz to 4600 Hz, a gain of -45 or lower; and
c) At 4600 Hz and above, a gain of -75 dB or lower.

Note 1: In addition to the listed spectral characteristics, additional filtering appropriate to the particular equipment installation should be provided in the low end response characteristics of the vocoder so as to mitigate the effects of power supply noise pickup. (e.g., 400 Hz for AES installations.)

Note 2: The input to the encoding algorithm is defined in Section 5.3.1.4.2 to be quantized audio samples with at least 12 bits of resolution. Consequently, all required spectral characteristics at the point of sampling are defined with this in mind. The use of u-law or A-law companded audio in conjunction with both reduced resolution (e.g., 8 bits) and appropriately relaxed transition-band and stop-band gain characteristics is permitted provided that the quantized samples are decompanded (linearized) prior to their presentation to the algorithm.

5.3.1.4.2 Sampling, Quantization, and Scaling

Audio which exhibits the spectral characteristics defined in Section 5.3.1.4.1 shall be sampled at a continuous 8 kHz rate, and the resultant series of samples shall be quantized to an equivalent time-ordered series of linear (i.e., non-companded) binary values. Each quantization operation shall be performed with a precision of at least 12 bits and a linearity of at least 10 bits. The maximum dynamic range of this series shall be scaled up or down in magnitude as appropriate so that it is tightly bounded by the floating-point range of 4095.0 to -4096.0 without exceeding it. The resultant time-ordered series of scaled values shall be presented to the segmentation and windowing logic defined in Section 5.3.3.1.

Note: The above bounding range of scaled values is indicative of a fixed-point signed value of 13 bits in length. Scaling of the time-ordered series of binary values to the indicated numeric range is required in order to take maximum advantage of the internal numeric range of the algorithm.

5.3.2 The Encoding Process

The encoder of Figure 5-3 is reproduced as Figure 5-6: the high-level variable names of key encoder parameters have been added to Figure 5-6 so that the reader may use this figure as a reference, while studying the detailed high-level description to follow. Figure 5-6 illustrates the main processes executed by the encoder, namely:

1. LPC analysis
2. Long term correlation analysis
3. Formation of a local decoder consisting of a short term and long term predictor
4. A closed loop error minimization process to calculate the excitation pulse positions and amplitudes.

These processes are described in detail in the following sub-sections.

5.3.3 LPC Analysis

The LPC analysis process can be broken into the following steps:

- Segmentation and windowing
- Autocorrelation
- Durbin's recursion
- Quantization of the partial correlation values

5.0 VOICE ENCODER STANDARDS (cont'd)

5.3.3 LPC Analysis (cont'd)

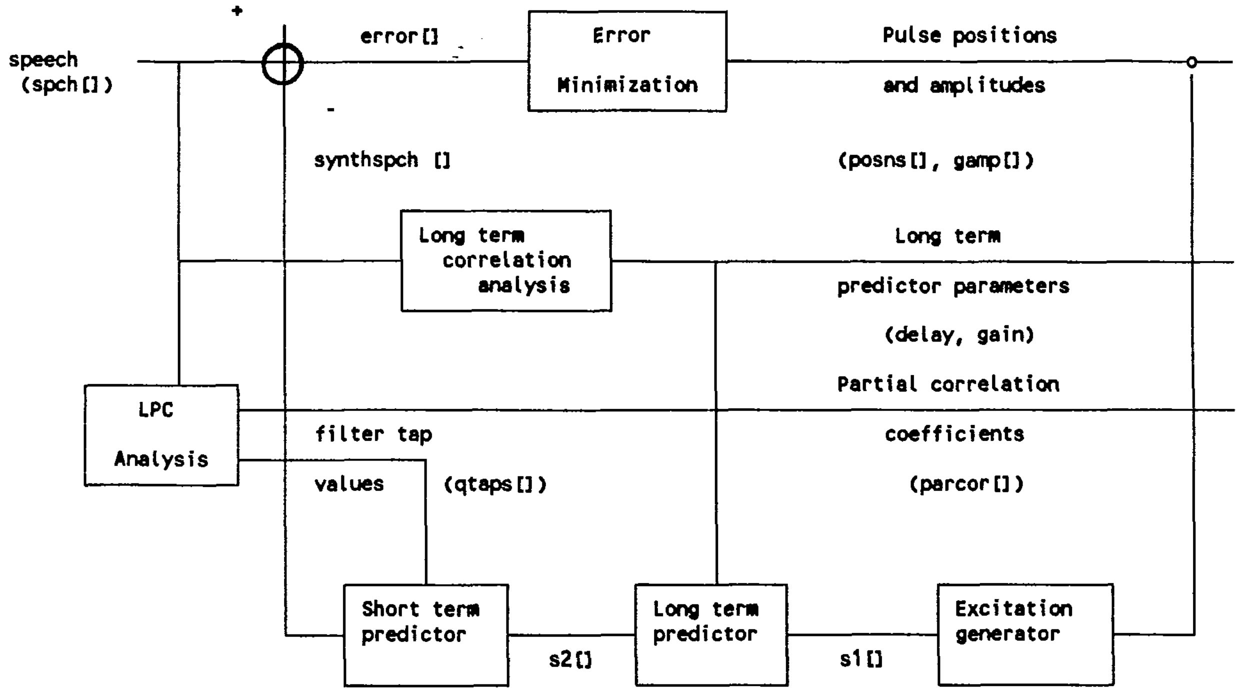


Figure 5-6 Multipulse-excited LPC encoder

- Step-up procedure

These steps are detailed below.

5.3.3.1 Segmentation and Windowing

The speech signal is first divided into 32 ms (256 samples) overlapping segments *segspch []*. The first and last 12 ms (96 samples) of each segment is overlapped with the previous and subsequent 32 ms segments (Figure 5-5). Windowing is performed on each 32 ms segment using a Hamming window: the Hamming window vector, *hamwindow[]*, is defined as:

*FOR n:=0 TO 255 DO hamwindow[n]:=0.54-0.46*COS((2*3.1415927*n/255);*

and the windowing operation is defined as:

*FOR n:=0 TO 255 DO windspch[n]:=segspch[n]*hamwindow[n];*

It is the 256 element *windspch[]*, vector which is used in the calculation of the LPC filter coefficients, however, the LPC coefficients thus calculated, are only considered valid over the central 20 ms (160 samples) of each window frame (see Figure 5-5); i.e., over the corresponding speech frame.

An additional time-ordered vector *spch[]* corresponding to the central 160 samples of each *segspch[]* vector shall be generated by the operation defined by:

FOR n:=0 TO 159 DO spch[]:=segspch[n+48];.

5.3.3.2 Autocorrelation

A 10th order LPC analysis requires the calculation of the 11 element autocorrelation vector, *corr[]*, as follows:

5.0 VOICE ENCODER STANDARDS (cont'd)

```
FOR i:=0 TO 10 DO
BEGIN
dot:=0;
    FOR n:=0 TO 255-i DO dot:=dot+windspch[n]*windspch[n+i];
corr[i]:=dot
END;
```

5.3.3.3 Durbin's Recursion

The partial correlation coefficients, *parcor[]*, are derived from the autocorrelation vector, *corr[]*, using Durbin's recursion given below:

```
IF corr[0]=0 THEN FOR i:=1 TO 10 DO parcor[i]:=0;
ELSE
    BEGIN
    parcor[1]:=corr[1]/corr[0];
    taps[1]:=parcor[1];
    error:=(1-sqr(parcor[1]))*corr[0];
    FOR i:=2 TO 10 DO
        BEGIN
        FOR j:=1 TO pred(i) DO alpha[j]:=taps[j];
        parcor[i]:=corr[i];
        FOR j:=1 TO pred(i) DO parcor[i]:=parcor[i]-alpha[j]*corr[i-j];
        parcor[i]:=parcor[i]/error;
        taps[i]:=parcor[i];
        FOR j:=1 TO pred(i) DO taps[j]:=alpha[j]-parcor[i]*alpha[i-j];
        error:=(1-sqr(parcor[i]))*error
        END
    END;
```

5.3.3.4 Quantization of the Partial Correlation Coefficients

The quantization of the partial correlation coefficients is achieved using a set of non-uniform quantizers. The decision levels for the partial correlation quantizers is represented in Table 5-1. The decision levels are stored in the two dimensional array *pardl[]*, where the first dimension of the array refers to the partial correlation coefficient number and the second dimension is the decision level index.

The quantization levels corresponding to the decision levels of Table 5-1 are derived as illustrated below. The two dimensional array, *parql[]*, contains 10 sets of quantization values corresponding to the 10 partial correlation coefficients. The appropriate value for each coefficient is selected and inserted in *qparcor[]*.

```
{Quantizer initialization procedure}

{No. of quantizer
levels for each
coefficient}

plevel[1]:=64;
plevel[2[:=64;
plevel[3]:=32;
plevel[4]:=32;
plevel[5]:=16;
plevel[6]:=16;
plevel[7]:=8;
plevel[8]:=8;
plevel[9]:=4;
plevel[10]:=4;

FOR i:=1 TO 10 DO
   FOR j:=0 TO pred(plevel[i]) DO
        parql[i j]:=pardl[i,pred(j)]+pardl[i,j]/2;
```

5.0 VOICE ENCODER STANDARDS (cont'd)

5.3.3.4 Quantization of the Partial Correlation Coefficients (cont'd)

pardl[1,-1], pardl[1,0], pardl[1,1], pardl[1,2]	-0.7512804,	-0.7269750,	-0.7017226,	-0.6755562
pardl[1,3], pardl[1,4], pardl[1,5], pardl[1,6]	-0.6485097,	-0.6206185,	-0.5919189,	-0.5624482
. . . .	-0.5322449,	-0.5013483,	-0.4697985	-0.4376369
. . . .	-0.4049051,	-0.3716459,	-0.3379026	-0.3037192
. . . .	-0.2691401,	-0.2342104,	-0.1989756,	-0.1634817
. . . .	-0.1277748,	-0.0919014,	-0.0559084,	-0.0198425
. . . .	0.0162492,	0.0523198,	0.0883222,	0.1241096
. . . .	0.1599351,	0.1954524,	0.2307150,	0.2656771
. . . .	0.3002931,	0.3345180,	0.3683071,	0.4016164
. . . .	0.4344026,	0.4666229,	0.4982354,	0.5291989
. . . .	0.5594731,	0.5890184,	0.6177965,	0.6457698
. . . .	0.6729020,	0.6991577,	0.7245026,	0.7489037
. . . .	0.7723293,	0.7947488,	0.8161331,	0.8364543
. . . .	0.8556859,	0.8738029,	0.8907816,	0.9066001
. . . .	0.9212375,	0.9346749,	0.9468949,	0.9578813
pardl[1,59], pardl[1,60], pardl[1,61], pardl[1,62]	0.9676200,	0.9760983,	0.9833051,	0.9892311
pardl[1,63], pardl[2,-1], pardl[2,0], pardl[2,1]	0.9938684,	-0.9687151,	-0.9600082,	-0.9502578
pardl[2,2], pardl[2,3], pardl[2,4], pardl[2,5]	-0.9394747,	-0.9276705,	-0.9148581,	-0.9010514
. . . .	-0.8862653,	-0.8705161,	-0.8538207,	-0.8361974
. . . .	-0.8176653,	-0.7982445,	-0.7779561,	-0.7568223
. . . .	-0.7348658,	-0.7121107,	-0.6885817,	-0.6643043
. . . .	-0.6393048,	-0.6136106,	-0.5872494,	-0.5602500
. . . .	-0.5326418,	-0.5044546,	-0.4757191,	-0.4464666
. . . .	-0.4167289,	-0.3865382,	-0.3559275,	-0.3249299
. . . .	-0.2935792,	-0.2619094,	-0.2299549,	-0.1977505
. . . .	-0.1653312,	-0.1327322,	-0.0999890,	-0.0671371
. . . .	-0.0342122,	-0.0012501,	0.0317133,	0.0646423
. . . .	0.0975010,	0.1302537,	0.1628649,	0.1952991
. . . .	0.2275210,	0.2594956,	0.2911882,	0.3225643
. . . .	0.3535898,	0.3842311,	0.4144548,	0.4442280
. . . .	0.4735184,	0.5022941,	0.5305240,	0.5581772
. . . .	0.5852238,	0.6116344,	0.6373802,	0.6624333
pardl[2,62], pardl[2,63], pardl[3,-1], pardl[3,0]	0.6867664,	0.7103531,	-0.6209860,	-0.5817980
pardl[3,1], pardl[3,2], pardl[3,3], pardl[3,4]	-0.5412097,	-0.4993190,	-0.4562265,	-0.4120361
. . . .	-0.3668540,	-0.3207891,	-0.2739522,	-0.2264559
. . . .	-0.1784146,	-0.1299440,	-0.0811606,	-0.0321819
. . . .	0.0168742,	0.0658897,	0.1147467,	0.1633275
. . . .	0.2115152,	0.2591939,	0.3062488,	0.3525667
. . . .	0.3980361,	0.4425475,	0.4859939,	0.5282706
. . . .	0.5692760,	0.6089114,	0.6470813,	0.6836939
pardl[3,29], pardl[3,30], pardl[3,31], pardl[4,-1]	0.7186612,	0.7518988,	0.7833269,	-0.8819578
pardl[4,0], pardl[4,1], pardl[4,2], pardl[4,3]	-0.8576206,	-0.8311930,	-0.8027395,	-0.7723294
. . . .	-0.7400368,	-0.7059404,	-0.6701235,	-0.6326731
. . . .	-0.5936806,	-0.5532411,	-0.5114532,	-0.4684186
. . . .	-0.4242423,	-0.3790320,	-0.3328978,	-0.2859522
. . . .	-0.2383097,	-0.1900863,	-0.1413995,	-0.0923682
. . . .	-0.0431116,	0.0062499,	0.0555963,	0.1048072
. . . .	0.1537625,	0.2023432,	0.2504306,	0.2979076
pardl[4,28], pardl[4,29], pardl[4,30], pardl[4,31]	0.3446585,	0.3905694,	0.4355282,	0.4794255
pardl[5,-1], pardl[5,0], pardl[5,1], pardl[5,2]	-0.5728675,	-0.5044545,	-0.4327131,	-0.3581167
. . . .	-0.2811575,	-0.2023432,	-0.1221939,	-0.0412383
. . . .	0.0399893,	0.1209531,	0.2011188,	0.2799576
. . . .	0.3569492,	0.4315858,	0.5033747,	0.5718424
pardl[5,15], pardl[6,-1], pardl[6,0], pardl[6,1]	0.6365371,	-0.7032794,	-0.6508853,	-0.5951883
. . . .	-0.5364712,	-0.4750316,	-0.4111817,	-0.3452452
. . . .	-0.2775567,	-0.2084599,	-0.1383052,	-0.0674487
. . . .	0.0037500,	0.0749297,	0.1457292,	0.2157892
pardl[6,14], pardl[6,15], pardl[7,-1], pardl[7,0]	0.2847542,	0.3522742,	-0.4617792,	-0.3299494
pardl[7,1], pardl[7,2], pardl[7,3], pardl[7,4]	-0.1913133,	-0.0487307,	0.0948572,	0.2364883
pardl[7,5], pardl[7,6], pardl[7,7], pardl[8,-1]	0.3732410,	0.5022942,	0.6209860,	-0.6287930
pardl[8,0], pardl[8,1], pardl[8,2], pardl[8,3]	-0.5098411,	-0.3801884,	-0.2425563,	-0.0998334
pardl[8,4], pardl[8,5], pardl[8,6], pardl[8,7]	0.0449848,	0.1888589,	0.3287691,	0.4617791
pardl[9,-1], pardl[9,0], pardl[9,1], pardl[9,2]	-0.4259395,	-0.1271548,	0.1839465,	0.4772301
pardl[9,3], pardl[10,-1], pardl[10,0], pardl[10,1]	0.7242872,	-0.4968801,	-0.2763557,	-0.0399893
pardl[10,2], pardl[10,3]	0.1986693,	0.4259395		

Table 5-1 Decision levels for the partial correlation coefficient quantizers

5.0 VOICE ENCODER STANDARDS (cont'd)

The partial correlation coefficient quantizer is defined as:

```
FOR i:=1 TO 10 DO
    BEGIN
    j:=0;
        WHILE (parcor[i]>pardl[i,j] and (j<pred(plevel[i])) DO j:=succ(j);
    qparcor[i]:=parql[i,j]
    END;
```

The quantized partial correlation coefficients are stored in vector *qparcor[]*. A line code corresponding to each quantized partial correlation coefficient contained in qparcor[] shall be generated by the table look-up operation defined by:

a) For *qparcor[1]* use Table 5-2a;
b) For *qparcor[2]* use Table 5-2b;
c) For *qparcor[3]* use Table 5-2c;
d) For *qparcor[4]* use Table 5-2d;
e) For *qparcor[5]* use Table 5-2e;
f) For *qparcor[6]* use Table 5-2f;
g) For *qparcor[7]* use Table 5-2g;
h) For *qparcor[8]* use Table 5-2h;
i) For *qparcor[9]* use Table 5-2i; and
j) For *qparcor[10]* use Table 5-2j.

5.0 VOICE ENCODER STANDARDS (cont'd)

5.3.3.4 Quantization of the Partial Correlation Coefficients (cont'd)

QUANTIZER	LINE CODE
-0.7391278	000000
-0.7143489	000001
-0.6886395	000010
-0.6620330	000011
-0.6345642	000100
-0.6062688	000101
-0.5771836	000110
-0.5473465	000111
-0.5167966	001000
-0.4855734	001001
-0.4537177	001010
-0.4212710	001011
-0.3882755	001100
-0.3547743	001101
-0.3208109	001110
-0.2864296	001111
-0.2516752	010000
-0.2165930	010001
-0.1812287	010010
-0.1456282	010011
-0.1098381	010100
-0.0739049	010101
-0.0378754	010110
-0.0017966	010111
0.0342845	011000
0.0703210	011001
0.1062659	011010
0.1420724	011011
0.1776938	011100
0.2130837	011101
0.2481961	011110
0.2829851	011111

QUANTIZER LEVEL	LINE CODE
0.3174056	100000
0.3514125	100001
0.3849618	100010
0.4180095	100011
0.4505128	100100
0.4824292	100101
0.5137172	100110
0.5443360	100111
0.5742458	101000
0.6034074	101001
0.6317832	101010
0.6593359	101011
0.6860299	101100
0.7118301	101101
0.7367031	101110
0.7606165	101111
0.7835391	110000
0.8054410	110001
0.8262937	110010
0.8460701	110011
0.8647444	110100
0.8822923	110101
0.8986909	110110
0.9139188	110111
0.9279562	111000
0.9407849	111001
0.9523881	111010
0.9627507	111011
0.9718592	111100
0.9797017	111101
0.9862680	111110
0.9915497	111111

Table 5-2a Line Codes for Partial Correlation Coefficient No. 1

5.0 VOICE ENCODER STANDARDS (cont'd)

QUANTIZER	LINE CODE
-0.9643617	000000
-0.9551330	000001
-0.9448663	000010
-0.9335726	000011
-0.9212644	000100
-0.9079548	000101
-0.8936584	000110
-0.8783907	000111
-0.8621684	001000
-0.8450091	001001
-0.8269314	001010
-0.8079550	001011
-0.7881004	001100
-0.7673892	001101
-0.7458441	001110
-0.7234883	001111
-0.7003462	010000
-0.6764430	010001
-0.6518046	010010
-0.6264577	010011
-0.6004300	010100
-0.5737497	010101
-0.5464459	010110
-0.5185482	010111
-0.4900868	011000
-0.4610929	011001
-0.4315977	011010
-0.4016336	011011
-0.3712329	011100
-0.3404287	011101
-0.3092546	011110
-0.2777443	011111

QUANTIZER LEVEL	LINE CODE
-0.2459321	100000
-0.2138527	100001
-0.1815409	100010
-0.1490317	100011
-0.1163606	100100
-0.0835630	100101
-0.0506746	100110
-0.0177312	100111
0.0152316	101000
0.0481778	101001
0.0810716	101010
0.1138773	101011
0.1465593	101100
0.1790820	101101
0.2114100	101110
0.2435083	101111
0.2753419	110000
0.3068762	110001
0.3380771	110010
0.3689105	110011
0.3993429	110100
0.4293413	110101
0.4588732	110110
0.4879062	110111
0.5164090	111000
0.5443506	111001
0.5717005	111010
0.5984291	111011
0.6245073	111100
0.6499068	111101
0.6745999	111110
0.6985598	111111

Table 5-2b Line Codes for Partial Correlation Coefficient No. 2

5.0 VOICE ENCODER STANDARDS (cont'd)

5.3.3.4 Quantization of the Partial Correlation Coefficients (cont'd)

QUANTIZER	LINE CODE
-0.6013920	00000
-0.5615038	00001
-0.5202643	00010
-0.4777727	00011
-0.4341313	00100
-0.3894451	00101
-0.3438216	00110
-0.2973706	00111
-0.2502041	01000
-0.2024353	01001
-0.1541793	01010
-0.1055523	01011
-0.0566713	01100
-0.0076539	01101
0.0413820	01110
0.0903182	01111

QUANTIZERLEVEL	LINE CODE
0.1390371	10000
0.1874214	10001
0.2353546	10010
0.2827214	10011
0.3294078	10100
0.3753014	10101
0.4202918	10110
0.4642707	10111
0.5071322	11000
0.5487733	11001
0.5890937	11010
0.6279963	11011
0.6653876	11100
0.7011775	11101
0.7352800	11110
0.7676129	11111

Table 5-2c Line Codes for Partial Correlation Coefficient No. 3

QUANTIZER	LINE CODE
-0.8697892	00000
-0.8444068	00001
-0.8169663	00010
-0.7875345	00011
-0.7561831	00100
-0.7229886	00101
-0.6880320	00110
-0.6513983	00111
-0.6131769	01000
-0.5734609	01001
-0.5323472	01010
-0.4899359	01011
-0.4463305	01100
-0.4016372	01101
-0.3559649	01110
-0.3094251	01111

QUANTIZER	LINE CODE
-0.2621310	10000
-0.2141980	10001
-0.1657429	10010
-0.1168839	10011
-0.0677399	10100
-0.0184309	10101
0.0309231	10110
0.0802017	10111
0.1292848	11000
0.1780529	11001
0.2263869	11010
0.2741691	11011
0.3212830	11100
0.3676139	11101
0.4130488	11110
0.4574769	11111

Table 5-2d Line Codes for Partial Correlation Coefficient No. 4

5.0 VOICE ENCODER STANDARDS (cont'd)

QUANTIZER	LINE CODE
-0.5386610	0000
-0.4685838	0001
-0.3954149	0010
-0.3196371	0011
-0.2417503	0100
-0.1622685	0101
-0.0817161	0110
-0.0006245	0111

QUANTIZER	LINE CODE
0.0804712	1000
0.1610360	1001
0.2405382	1010
0.3184534	1011
0.3942675	1100
0.4674802	1101
0.5376085	1110
0.6041898	1111

Table 5-2e Line Codes for Partial Correlation Coefficient No. 5

QUANTIZER	LINE CODE
-0.6770824	0000
-0.6230369	0001
-0.5658298	0010
-0.5057514	0011
-0.4431067	0100
-0.3782134	0101
-0.3114009	0110
-0.2430083	0111

QUANTIZER	LINE CODE
-0.1733826	1000
-0.1028770	1001
-0.0318494	1010
0.0393399	1011
0.1103295	1100
0.1807592	1101
0.2502717	1110
0.3185142	1111

Table 5-2f Line Codes for Partial Correlation Coefficient No. 6

QUANTIZER	*LINE CODE*
-0.3958643	*000*
-0.2606314	*001*
-0.1200220	*010*
0.0230632	*011*

QUANTIZER LEVEL	*LINE CODE*
0.1656727	*100*
0.3048646	*101*
0.4377676	*110*
0.5616401	*111*

Table 5-2g Line Codes for Partial Correlation Coefficient No. 7

QUANTIZER	LINE CODE
-0.5693170	000
-0.4450148	001
-0.3113724	010
-0.1711949	011

QUANTIZER	LINE CODE
-0.0274243	100
0.1169218	101
0.2588140	110
0.3952741	111

Table 5-2h Line Codes for Partial Correlation Coefficient No. 8

5.0 VOICE ENCODER STANDARDS (cont'd)

5.3.3.4 Quantization of the Partial Correlation Coefficients (cont'd)

QUANTIZER	LINE CODE
-0.2765472	00
0.0283959	01

QUANTIZER LEVEL	LINE CODE
0.3305883	10
0.6007586	11

Table 2i Line Codes for Partial Correlation Coefficient No. 9

QUANTIZER	LINE CODE
-0.3866179	00
-0.1581725	01

QUANTIZER LEVEL	LINE CODE
0.0793400	10
0.3123044	11

Table 2j Line Codes for Partial Correlation Coefficient No. 10

5.3.3.5 Step Up Procedure

The final step in the LPC analysis procedure is the step up process from the quantized partial correlation coefficients to the quantized LPC filter tap values, *qtaps[]*, used in the inverse LPC filter and LPC short term predictor.

```
qtaps:=qparcor; {all 10 elements of vector qtaps set equal to corresponding elements in qparcor}
FOR i:=2 TO 10 DO
BEGIN
   FOR j:=1 TO pred(i) DO alpha[j]:=qtaps[j];
         FOR j:=1 TO pred(i) DO qtaps[j}:=alpha[j]-qtaps[i]*alpha[i-j]
   END;
```

5.3.4 Long Term Correlation Analysis

The operations involved in the long term correlation analysis are summarized in Figure 5-7. All the operations involved in the LPC analysis block to calculate the quantized filter tap coefficients, *qtaps[]*, have been described above: it is these quantized filter tap values, *qtaps[]*, which are employed by the inverse LPC filter in the calculation of the LPC residual signal.

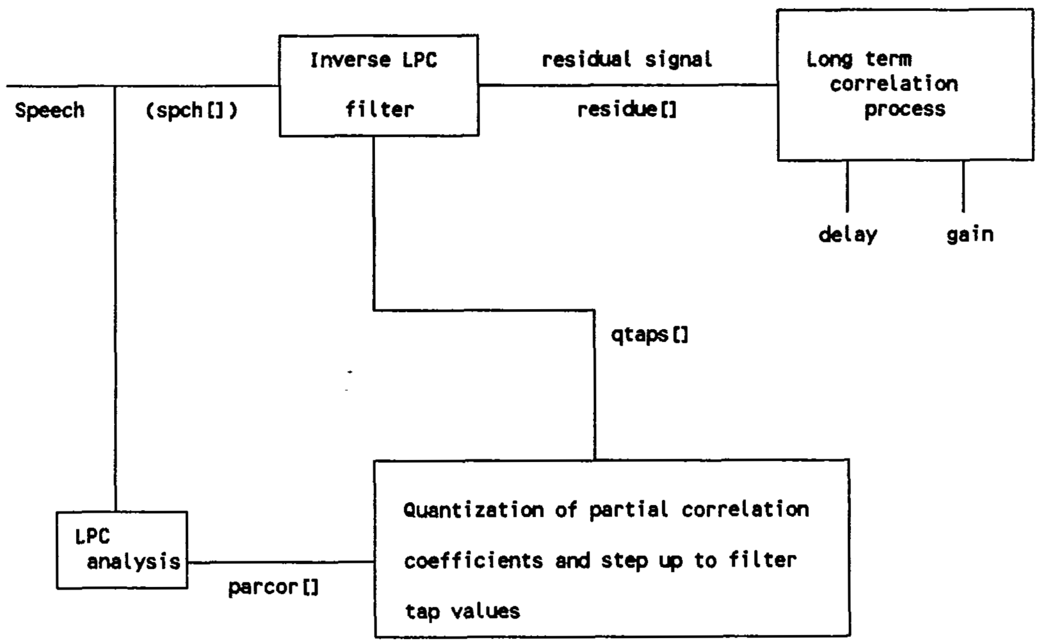


Figure 7 Long Term Correlation Analysis

5.0 VOICE ENCODER STANDARDS (cont'd)

5.3.4.1 Inverse Filtering

The LPC inverse filtering operation is applied to each speech frame , *spch []*, (corresponding to the central 160 samples of the current window frame). The output from the inverse filter is known as the LPC residual signal: the operations involved in forming the residual, *residue[]*, are defined as follows:

```
FOR n:=0 TO 159 DO
   BEGIN
   sum:=spch[n];
   FOR i:=1 TO 10 DO sum:=sum-qtaps[i]*spch[n-i];
   residue[n]:=sum
   END;
```

Note: The values near the start of each speech frame, for which the index of *spch[]* is less than zero, refers to speech samples at the end of the previous speech frame. For the first frame processed by the encoder, these values are all considered to be zero.

5.3.4.2 Long Term Correlation Process

The final output from the long term correlation analysis is the gain (*gain*) and delay (*delay*) parameters: these parameters are updated at the same rate as the LPC coefficients, i.e., every 20 ms (160 samples). The calculation of the delay and gain values from the residual signal is defined by the high-level code presented below. The range of values chosen for the delay parameter is 32 to 95 samples (inclusive). Note that the parameter, *delay*, is a 16 bit integer value.

```
max:=0
FOR lag:=33 TO 96 DO
   BEGIN
   sum:=0;
   FOR n:=0 TO 159 DO sum:-sum+residue[n]*residue[n-lag];
   IF sum>max THEN
         BEGIN
         max:=sum;
         delay:=lag
         END
   END;
sum:=0;
FOR n:=0 TO 159 DO sum:=sum+sqr(residue[n-delay]);
IF sum=0 THEN gain:=0 ELSE gain:=max/sum;
```

Note: Negative values of the indices, *n-lag*, and *n-delay*, point to LPC residual values calculated in the previous speech frame. For the first frame processed the residual samples of the previous frame are considered to be zero.

5.3.4.3 Quantization of Long Term Correlation Gain

The long term predictor gain value, *gain*, is quantized with a 2 bit non-uniform quantizer as illustrated below:

```
IF gain>0.75 THEN qgain:=0.9 ELSE
     IF gain>0.45 THEN qgain:=0.60 ELSE
         IF gain>0.2 THEN qgain:=0.325 ELSE qgain:=0.1
```

The line codes allocated to the quantized long term correlation gain parameter are presented in Table 5-3.

Long-term predictor gain parameter, gain	Line code	Decoded gain parameter, *qgain*
gain <=0.2	00	0.100
0.2 < *gain* <= 0.45	01	0.325
0.45 < *gain* <= 0.75	10	0.600
gain > 0.75	11	0.900

Table 5-3 Quantization and Encoding of the Long-Term Predictor *Gain* Parameter

5.0 VOICE ENCODER STANDARDS (cont'd)

5.3.4.4 Line Codes Allocated to the Long Term Correlation Delay Parameter

As the delay value can only have 64 distinct values (32-95), no quantization is required and the 6 bit line code presented in Table 5-4 below is allocated directly to it.

Delay value	Line code
32	000000
33	000001
34	000010
35	000011
,	,
,	,
,	,
63	011111
64	100000
65	100001
66	100010
,	,
,	,
,	,
92	111100
93	111101
94	111110
95	111111

Table 5-4 Line codes allocated to the long-term predictor delay parameter

5.3.5 The Local Decoder

The local decoder consists of the excitation generator, the long term predictor and the short term predictor.

Note: The local decoder logic is used to generate a locally-derived version of the encoded speech for feedback to the excitation analysis logic defined in Section 5.3.6. The local decoder logic is also used in the definition of the decoding process in Section 5.5.

5.3.5.1 The Excitation Generator

The excitation generator forms the excitation signal, for each excitation frame, from the pulse position and amplitude information derived in the encoder squared error minimization block. The excitation signal, *s1[]*, thus formed, consists of pulses located at the defined positions, *posns[]*, and having the corresponding quantized amplitudes, *qamp[]*, as illustrated below.

FOR n:=b TO b+31 DO s1[n]:=0;
FOR pulse:=1 TO 3 DO s1[psns[pulse]+b]:=qamp[pulse];

where *b* is the sample number in the current speech frame corresponding to the start of the current excitation frame and, *posns[]*, is the vector of pulse positions belonging to the current excitation frame.

Note: For the first, second, third, fourth and fifth excitation frames of each speech frame, the value of *b* equals 0, 32, 64, 96, and 128 respectively.

5.0 VOICE ENCODER STANDARDS (cont'd)

5.3.5.2 The Long Term Predictor

The long term predictor is defined by the high-level code description presented below. The input to this recursive filter is the excitation vector corresponding to the current excitation frame, *s1[]*, and the output, *s2[]*, in turn becomes the input to the short term predictor.

```
FOR n:=b TO b+31 DO s2[n]:=qgain*s2{[n-delay]+s1[n];
```

Note: Values of the index, *n-delay*, are always less than b (the beginning of the current excitation frame) and the index thus points to values in the long term predictor filter memory. For the first frame processed these values will be undefined and should be taken as zero.

5.3.5.3 The Short Term Predictor

The input to the short term predictor recursive filter is the output from the long term prediction filter, *s2[]*. The operations performed on the vector, *s2[]*, to calculate the output of the short term predictor, *synthspch[]*, are defined below.

```
FOR n:=b TO b+31 DO
BEGIN
sum:=0;
   FOR i:=1 TO 10 DO sum:=sum+qtaps[i]*synthspch[n-i];
   synthspch[n]:=s2[n]+sum
END;
```

Note: The index, *n-i*, points to values in the short term predictor memory. For the first speech frame processed, the contents of the short term predictor filter memory are set to zero.

5.3.6 Excitation Analysis

As Figure 5-6 illustrates the determination of the excitation involves a closed loop error minimization process. The error which is minimized is the square of the difference, on a sample-by-sample basis, between the input speech vector, *spch[]*, and the output from the local decoder, *synthspch[]*. The frame size over which this squared error minimization occurs is the excitation frame size of 4 ms (32 samples): three pulses are derived per excitation frame.

Note: The transmission frame that is generated by this encoding algorithm contains five excitation frames. Each group of five excitation frames is associated with the same speech frame (spch[]) from which the partial correlation coefficients of the current transmission frame were calculated.

5.3.6.1 Derivation of the Error Signal

As the short term and long term predictors, defined above, are recursive, the effect of the excitation signal in previous excitation frames must be accounted for, before the error minimization process commences for the current excitation frame. The output of the local decoder, due to the excitation signal in previous frames, is calculated by taking the current excitation as zero, and calculating the current excitation as zero, and calculating the output of the local decoder. (Note: It is the quantized filter taps, *qtaps[]*, and long term correlation parameters for the current speech frame which are used in this calculation.) The error signal, *error[]*, is then formed as the difference between the speech signal, *spch[]*, and the output of the local decoder, *synthspch[]*, as illustrated below.

```
local_synthspch:=synthspch; {create a local copy of the short term predictor filter memory}
local_s2:=s2; {create a local copy of the long term predictor filter memory}
FOR n:=b TO b+31 DO
   BEGIN
   sum:=0;
         FOR i:=1 TO 10 DO sum:=sum+qtaps[i]*local_synthspch[n-i];
   local_synthspch[n]:=qgain*local_s2[n-delay]+sum;
   error[n]:=spch[n]-local_synthspch[n]
   END;
```

Note: Changes to the short term and long term filter memories values are kept local to this routine.

The aim of the closed loop error minimization process is to select the three pulses for this current excitation frame which minimize the square of the error over the period of the excitation frame.

5.0 VOICE ENCODER STANDARDS (cont'd)

5.3.6.1 Derivation of the Error Signal (cont'd)

The selection of the three pulse positions and amplitudes to minimize the squared error is performed sequentially. Three independent error minimizations are thus required, i.e., the effect of the first pulse selected is taken account off in the search for the second pulse, and the combined effect of the first and second pulses is taken account of in the search for the third pulse. It can be shown that this squared error minimization is equivalent to maximizing the normalized cross-correlation function defined below.

5.3.6.2 Impulse Response Calculations

The first step in the error minimization process is the formation of the cross-correlation between the error vector, *error[]*, and the impulse response of the short term and long term predictor combination, for all possible (32) positions that a pulse may occupy within an excitation frame. As the excitation frame size is 32 samples long, the impulse response need only be calculated for 32 sample values, and only once per speech frame, as the short and long term predictor parameters do not change over that period. The impulse response, *iresp[]*, is calculated as shown below.

```
iresp[0]:=1;
ipwr[31]:=1;
FOR n:=1 TO 31 DO
   BEGIN
   sum:=0;
   FOR j:=1 TO 10 DO IF n>=j THEN sum:=sum+qtaps[j]*iresp[n-j];
   iresp[n]:=sum;
   ipwr[32-succ(n)]:=ipwr[32-n]+sqr(sum)
   END;
```

Note: As the minimum value that the delay parameter may not be less than 32, the long term predictor has not influence on the first 32 samples of the impulse response.

The calculation of the energy, *ipwr[]*, of that part of the impulse response which falls within the current excitation frame, for all possible positions of a pulse within a frame, is also illustrated above: these values are used later to normalize the cross-correlation function.

5.3.6.3 Cross-Correlation Calculation

The calculation of the cross-correlation, *xcorr[]*, of the impulse response with the error vector, *error[]*, is illustrated below.

```
FOR n:=0 TO 31 DO
   BEGIN
   dot:=0;
         FOR j:=n TO 31 DO dot:=dot+error[b+j]*iresp[j-n];
   xcorr[n]:=dot
   END;
```

where *b* is the sample number of the start of the current excitation frame within the current speech frame. For the first, second, third, fourth and fifth excitation frames of each speech frame, the value of b equals 0, 32, 64, 96, and 128 respectively.

5.3.6.4 Pulse Selection Procedure

Using the *xcorr[]* vector, the three pulse positions and amplitudes per excitation frame are calculated as shown below.

```
FOR pulse:=1 TO 3 DO
   BEGIN
   max:=0;
   FOR n:=0 TO 31 DO
         IF (sqr(xcorr[n])>=max*ipwr[n]) and not (n in posn_set) THEN BEGIN
         max:=sqr(xcorr[n])/ipwr[n];
         pos:=n
   END;
   posn_set:=posn_set+[pos];
   posns[pulse]:=pos;
   amp:= xcorr[pos]/ipwr[pos];
   qamp[pulse]:=quantamp(amp,pulse);
```

5.0 VOICE ENCODER STANDARDS (cont'd)

```
{Account for the effect of the quantized pulse just calculated on the error to be minimized}
        WHILE pulse<npulse DO
        BEGIN
                FOR n:=0 TO pred(pos) DO
                BEGIN
                dot:=0;
                        FOR j:=0 TO 31-pos DO dot:=dot+iresp[j]*iresp[j+(pos-n)];
                xcorr[n]:=xcorr[n]-qamp*dot
                END;

                FOR n:= pos TO 31 DO
                BEGIN
                dot:=0;
                        FOR j:=0 TO 31-n DO dot:=dot+iresp[j]*iresp[j+(n-pos)];
                xcorr[n]:=xcorr[n]-qamp*dot
                END
        END
END;
```

where *posn_set[]*, is a set used to hold the pulse positions for the current excitation frame, and *quantamp* is a function used to quantize the pulse amplitudes (see Section 5.3.6.5).

Note: The second and third pulses selected in each excitation frame are not allowed to occupy a position already occupied by a previously selected pulse in that excitation frame.

The pulse selection routine above, selects the best position for each pulse, as the position in the excitation frame (value of the index n) at which the normalized cross-correlation function:

sqr(xcorr[n])/ipwr[n]; n=0 to 31

is a maximum.

After the first pulse has been selected, and the pulse amplitude quantized, the effect of this pulse on the error signal must be accounted for, before the selection of the second and third pulses take place. It is in fact easier to account for the effects of this pulse directly, by modifying the cross-correlation vector, *xcorr[]*, as illustrated above.

5.3.6.5 Pulse Amplitude Quantization

The three pulse amplitudes selected per excitation frame are quantized using 4 bits each, for the first and second pulses in each excitation frame, and 3 bits for the third pulse in the frame. The quantization of each pulse amplitude involves two separate normalization stages: the first normalization process takes account of the average variation in the modules of the pulse amplitudes within an excitation frame and the second process takes account of the variation in speech energy from one excitation frame to the next. No explicit information is transmitted to the receiver regarding either of these normalizations.

After each pulse has been normalized it is quantized using a non-uniform quantizer: a 4-bit quantizer is employed for the first and second pulses in each excitation frame. The decision thresholds, decoded output levels and the line codes for this 4 bit quantizer are illustrated in Table 5-5 below.

Index	Decision Threshold	Line Code	Decoded Output Level
7	-2.4010	1111	-2.7330
6	-1.8440	1110	-2.0690
5	-1.4370	1101	-1.6180
4	-1.0990	1100	-1.2560
3	-0.7996	1011	-0.9424
2	-0.5224	1010	-0.6568
1	-0.2582	1001	-0.3881

5.0 VOICE ENCODER STANDARDS (cont'd)

0	-0.0000	1000	-0.1284
0	0.0000	0000	0.1284
1	0.2582	0001	0.3881
2	0.5224	0010	0.6568
3	0.7996	0011	0.9424
4	1.0990	0100	1.2560
5	1.4370	0101	1.6180
6	1.8440	0110	2.0690
7	2.4010	0111	2.7330

Table 5-5 Line Codes and Decoded Output Levels for the First and Second Pulse Amplitude Quantizers in Each Excitation Frame

For the third normalized pulse amplitude, a 3-bit non-uniform quantizer is employed. Table 5-6 defines the pulse amplitude quantizers for the third pulse.

Index	Decision Threshold	Line Code	Decoded Output Level
3	-1.7480	111	-2.1520
2	-1.0500	110	-1.3440
1	-0.5006	101	-0.7560
0	-0.0000	100	-0.2451
0	0.0000	000	0.2451
1	0.5006	001	0.7560
2	1.0500	010	1.3440
3	1.7480	011	2.1520

Table 5-6 Line Codes and Decoded Output Levels for the Third Pulse Amplitude Quantizer

The normalization of the pulse amplitudes before quantization is crucial to the successful operation of the pulse amplitude quantizer. The two normalization stages are explained in the next sub-section.

5.3.6.6 <u>Pulse Amplitude Normalization</u>

The first stage of the normalization process is an intraframe normalization and is achieved by multiplying each pulse in an excitation frame by a set of fixed scaling factors (see Table 5-7 below).

Pulse	Scaling factor
First pulse found via the sequential search procedure	1.0
Second pulse found via the sequential search procedure	1.6
Third pulse found via the sequential search procedure	1/0.375(~2.67)

Table 5-7 Scaling Factors Applied to the Pulses Within an Excitation Frame

5.0 VOICE ENCODER STANDARDS (cont'd)

These scaling factors adjust the pulse amplitudes within an excitation frame so that on average they are more nearly equal.

The pulse amplitude information is transmitted in the order in which the pulses are found. Thus the decoder can take account of the encoder scale factors by dividing the decoded pulse amplitudes by 1, 1.6 and 2.67 respectively, for the first, second and third pulses received for each excitation frame.

The second pulse amplitude normalization stage accounts for the variation in speech energy from one excitation frame to the next. The gain factor used for this normalization is adapted in a backward fashion according to the line codes generated by the pulse amplitude quantizers. After quantization of the normalized first pulse amplitude the gain factor is adjusted by the multipliers presented in Table 5-8 below.

4-Bit Quantizer Output Code	Gain Factor Multiplier
1000 or 0111	2.000
1001 or 0110	1.500
1010 or 0101	1.250
1011 or 0100	1.000
1100 or 0011	0.875
1101 or 0010	0.875
1110 or 0001	0.875
1111 or 0000	0.750

Table 5-8 Multipliers for Gain Factor Adjustment After the First and Second Pulse Amplitudes Have Been Quantized

Thus if the line code issued for the quantized first pulse amplitude is either 0111 or 1000, corresponding to the most positive and most negative quantizer levels respectively, the gain factor is increased by a factor of 2. If the line code issued, however, is either 0000 or 1111, corresponding to the smallest quantizer levels, then the gain factor is decreased by a factor of 0.75. The gain factor adaptations for all other line codes issued for the first and second pulses can be obtained by reference to Table 5-8.

The gain factor adaptations after quantization of the third pulse amplitude is defined by Table 5-9 below.

3-Bit Quantizer Output Code	Gain Factor Multiplier
100 or 011	1.500
101 or 010	1.000
110 or 001	0.875
111 or 000	0.875

Table 5-9 Multipliers for Gain Factor Adjustment After the Third Pulse Amplitude Has Been Quantized

After the gain factor is adjusted according to the line code just issued, the gain factor may be further increased by examining the line code issued for the previous pulse. If the line code just issued and the line code of the previous pulse, indicate that both pulses were either the most positive and negative quantizer values, then the gain factor is increased further. This further gain factor increased is by a multiplication factor corresponding to the most extreme positive and negative quantizer levels; i.e., for the gain adjustment after quantization of the first and second pulses, the gain factor may be increased by a further factor of 2 and following the quantization of the third pulse, the gain factor may be further increased by a factor of 1.5.

Note: This further adjustment of the gain factor, according to the line codes issued for the pulse just quantized and the previous pulse, is continuous across excitation and speech frame boundaries.

After all adjustments of the gain factor, *gfact_leak*, it is hard limited, ensuring that it is bounded, such that:

5.0 VOICE ENCODER STANDARDS (cont'd)

5.3.6.6 Pulse Amplitude Normalization (cont'd)

$$0.5 =< gfact\_leak <= 512$$

During initialization both the encoder and decoder gain factors are set to unity. Provided there are no transmission error the decoder pulse amplitude gain factor will track the encoder pulse amplitude gain factor exactly. To minimize the effects of transmission errors a degree of leakage is included in the gain factor adjustment at both the encoder and decoder. The gain factor calculated from the adaptation rules, and including leakage, *gfact_leak*, is given by:

$$gfact\_leak := gfact\_leak^{0.98}$$

It is the leaked gain factor, *gfact_leak*, which is used to normalize each pulse amplitude, i.e., normalized pulse amplitude = pulse amplitude / *gfact_leak*. A flowchart summarizing the operation of the pulse amplitude quantizer is presented in Figure 5-8 and the high-level code description of the quantizer is presented below.

```
{Pulse amplitude quantizer initialization}

gfact_leak:=1; maxadapt:=false;
adjust[1]:=1; adjust[2]:=1.0/0.625; adjust[3]:=1.0/0.375;

{3 bit quantizer initialization}

adapt[3,0]:=0.875; adapt[3,1]:=0.875; adapt[3,2]:=1.0; adpat[3,3]:=1.5;
quant[3,0]:=0.5006; quant[3,1]:=1.050; quant[3,2]:=1.748;
inv_quant[3,0]:=0.2451; inv_quant[3,1]:=0.7560; inv_quant[3,2]:=1.344;
inv_quant[3,3]:=2.152;

{4 bit quantizer initialization}

adapt[4,0]:=0.75; adapt[4,1]:=0.875; adapt[4,2]:=0.875; adapt[4,3]:=0.875; adapt[4,4]:=1.0;
adapt[4,5]:=1.25; adapt[4,6]:=1.5; adapt[4,7]:=2.0;
quant[4,0]:=0.2582; quant[4,1]:=0.5224; quant[4,2]:=0.7996;
quant[4,3]:=1.099; quant[4,4]:=1.437; quant[4,5]:=1.844; quant[4,6]:=2.401;
inv_quant[4,0]:=0.1284; inv_quant[4,1]:=0.3881; inv_quant[4,2]:=0.6568;
inv_quant[4,3]:=0.9424; inv_quant[4,4]:=1.256; inv_quant[4,5]:=1.618; inv_quant[4,6]:=2.069;
inv_quant[4,7]:=2.733;

FUNCTION quantamp(lev:real, pulse:integer):real;

VAR     i,qbits      : integer
        sgn   : -1..1;
BEGIN
lev:=lev*adjust[pulse]
IF pulse<3 THEN BEGIN qbits:=4; qlevels:=7 END
   ELSE BEGIN qbits:=3; qlevels:=3 END;
IF lev>0 THEN sgn:=1 ELSE sgn:=-1;
i:=0;
WHILE (abs(lev)>quant[qbits,i]*gfact_leak) and (i<qlevels) DO i:=succ(i);
quantamp:=inv_quant[qbits,i]*gfact_leak*sgn/adjust[pulse];
gfact_leak:=EXP(0.98*LN(gfact_leak));

IF i=qlevels THEN
   BEGIN
   IF maxadapt THEN gfact_leak:=gfact_leak*adapt[qbits,qlevels]; maxadapt:=true
   END ELSE maxadapt:=false;
gfact_leak:=gfact_leak*adapt[qbits,i];
IF gfact_leak>512.0 THEN gfact_leak:=512.0 ELSE
   IF gfact_leak<0.5 THEN gfact_leak:=0.5
END { of quantamp. };
```

5.0 VOICE ENCODER STANDARDS (cont'd)

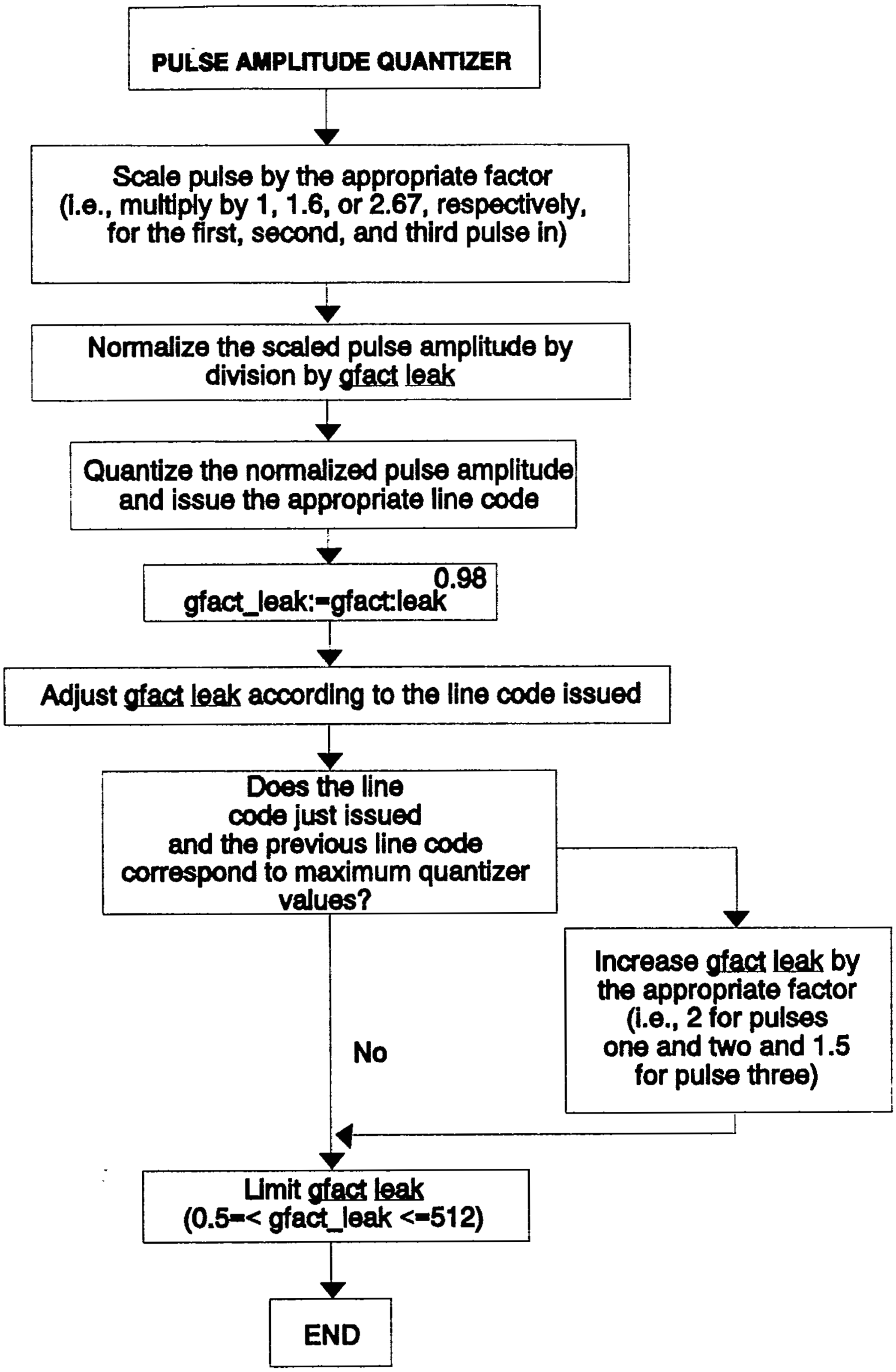


Figure 5-8 Pulse Amplitude Quantizer

5.0 VOICE ENCODER STANDARDS (cont'd)

5.3.6.6 Pulse Amplitude Normalization (cont'd)

Note: This operation is performed for each of the five excitation frames contained within a transmission frame.

Note: The line codes for the pulse amplitudes are available after execution of the line of code:

*WHILE (abs(lev) > quant[qbits,i]*gfact_leak) and (i < qlevels) DO i:=succ(i);*

The line code for pulses 1 and 2 are defined in Table 5-5 and the line code for pulse 3 is defined in Table 5-6. The value of the variable "i" at completion of the execution of the line of code above acts as the index to these tables. The value of the variable "sgn" at this same point in the function determines whether the msb of the line code is zero or one corresponding to a positive or negative pulse amplitude respectively (see Tables 5-5 and 5-6).

5.3.6.7 Pulse Position Encoding

Each pulse position is encoded with reference to the beginning of the excitation frame within which it falls. With excitation frame durations of 4 ms, and 8 kHz sampling, there are 32 sample positions per excitation frame. A 5-bit line code is thus required to encode each pulse position. Positions within each excitation frame are numbered from 0 to 31, with 0 corresponding to the first sample position within each frame and 31 corresponding to the last sample position. Table 5-10 below presents the line codes corresponding to various pulse positions.

Pulse Position	Line Code
0	00000
1	00001
2	00010
3	00011
4	00100
,	,
,	,
15	01111
16	10000
,	,
,	,
,	,
29	11101
30	11110
31	11111

Table 5-10 Pulse Positions and Corresponding Line Codes

5.4 Parameter Transmission

This section describes the transmission frame structure used by the encoder.

5.4.1 Introduction

A specification summary of the multipulse-excited LPC encoder is presented in Table 5-11. All the parameters presented in this Characteristic have been defined in Section 5.3, except the error protection codes which are associated with the partial correlation coefficients.

5.0 VOICE ENCODER STANDARDS (cont'd)

Parameter	Values
Input bandwidth	0.2 - 3.4 KHz
Sampling rate	8.0 KHz
Short-term Predictor	10th order
LPC analysis	Autocorrelation analysis with 32ms Hamming window and 20ms predictor coefficient update rate
Partial correlation coefficients	
1,2	6 bits
3,4	5 bits
5,6	4 bits
7,8	3 bits
9,10	2 bits
Error Protection	14 bits
Total	54 bits/20ms
Long-term Predictor	one-tap
Long-term correlation lag	6 bits
Gain parameter	2 bits
Total	8 bits/20ms
Excitation	Three pulses per 4 ms excitation frame (i.e., 5 excitation frames/LPC frame)
Pulse position 1	5 bits
Pulse position 2	5 bits
Pulse position 3	5 bits
Pulse amplitude 1	4 bits
Pulse amplitude 2	4 bits
Pulse amplitude 3	3 bits
Total	26 bits/4ms

Table 5-11 Specification Summary

5.4.2 Error Protection

Single bit error correction/double error detection is applied to 26 bits out of the 40 bits used to quantize the partial correlation coefficients. Figure 5-9 illustrates which of the partial correlation coefficient bits are protected. Burst error detection is also implemented and burst errors occurring within the same 26 partial correlation coefficient bits or the single bit error/double error detection bits (Figure 5-9) can be detected.

5.4.2.1 Single Bit Error Correction and Double Error Detection

Single bit error correction is provided for the protected partial correlation coefficient bits by a 5-bit Hamming Codeword. A parity bit (odd parity) is augmented to the Hamming codeword to provide double error detection. The Hamming function presented below defines the operations involved in finding a Hamming codeword.

5.0 VOICE ENCODER STANDARDS (cont'd)

5.4.2.1 Single Bit Error Correction and Double Error Detection (cont'd)

	parcor						
	1	2	3	4	5	6	7
bytes 1 - 4	* * * * * * 6 5 4 3 2 1 £ £ £ £ £ £	* * * * * 6 5 4 3 2 1 £ £ £ £ £	* * * 5 4 3 2 1 £ £ £	* * * 5 4 3 2 1 £ £ £	* * 4 3 2 1 £ £	* * 4 3 2 1 £ £	* * 3 2 £ £

	parcor				Hamming code	Burst error detection bits	gain	delay	Excitation
		8	9	10					posn. 1
bytes 5 - 8	* 1 £	* 3 2 1 £	* 2 1 £	2 1	6 5 4 3 2 1 £ £ £ £ £ £	8 7 6 5 4 3 2 1	2 1	6 5 4 3 2 1	5 4

	frame 1						Excitation	
bytes 9 - 12	3 2 1	amp 1 4 3 2 1	posn 2 5 4 3 2 1	amp 2 4 3 2 1	posn 3 5 4 3 2 1	amp 3 3 2 1	posn 1 5 4 3 2 1	amp 1 4 3 2

	frame 2					Excitation		
bytes 13 - 16	1	posn 2 5 4 3 2 1	amp 2 4 3 2 1	posn 3 5 4 3 2 1	amp 3 3 2 1	posn 1 5 4 3 2 1	amp 1 4 3 2 1	posn 2 5 4 3 2 1

	frame 3			Excitation frame 4				
bytes 17 - 20	amp 2 4 3 2 1	posn 3 5 4 3 2 1	amp 3 4 3 2 1	posn 1 5 4 3 2 1	amp 1 3 2 1	posn 2 5 4 3 2 1	amp 2 4 3 2 1	posn 3 5 4

			Excitation frame 5					
bytes 21 - 24	3 2 1	amp 3 3 2 1	posn 1 5 4 3 2 1	amp 1 4 3 2 1	posn 2 5 4 3 2 1	amp 2 4 3 2 1	posn 3 5 4 3 2 1	amp 3 3 2 1

* Bits covered by Hamming code   £ Bits covered by burst error detection

Figure 5-9 Transmission Frame Structure

Generation of Hamming Codewords

```
TYPE hamrng = 1.26;
     hamword = array [hamrng] of boolean;

FUNCTION hamming (word:hamword):integer;
VAR n,count,ham  :integer;
i               :hamrng;
BEGIN
n:=4; count:=3; ham:=0;
FOR i:=1 TO 26 DO
     BEGIN
     IF word[i] THEN ham:=EXOR(ham,count);
     count:=succ(count);
     IF count=n THEN BEGIN n:=n*2; count:=succ(count)END
     END

hamming:=ham
END{of hamming.};
```

Note: 'EXOR' function is bit-by-bit exclusive - OR operation on two integer values

The odd-parity bit is prepended to the 5-bit codeword as the most significant bit.

5.0 VOICE ENCODER STANDARDS (cont'd)

5.4.2.1 Single Bit Error Correction and Double Error Detection (cont'd)

Single error correction is provided with a (31,26) Hamming code which is further augmented with a 32-nd parity bit to provide double error detection. The Hamming code is defined by a parity check matrix *H*, where *H*-transpose is

$$H^T{}_{31,5} = \left[\frac{-P36,5}{I_{5,5}}\right]$$

and

G, the generator matrix, is

$$G_{26,31} = (I_{26,26} \backslash P_{5,26})$$

For binary elements recal $P = -P$

The row vector codeword c_{31}

$$= i_{26}\, G_{26,31}$$
$$= i_{26} \times (I_{26,26} \backslash P_{26,5})$$
$$= i_{26} \backslash (i_{26} \times P_{26,5})$$

where

I is the identity matrix of the appropriate dimension,
i is the 26-bit information-bit vector.
and
iP forms 5 parity check bits.

Note that only the 5 parity bits $p_5 = iP$ need be calculated. These are simply appended to the *i* vector to form $c = i \backslash p$.

On decoding, where the ' denotes bits possible received in error, the syndrome

$$s_5 = c'_{31}H^T$$
$$= (i' \backslash p'_5) \times \begin{pmatrix} P26,5 \\ I_5 \end{pmatrix}$$
$$= (i'P) + (p'I)$$

Thus the syndrome can be found by "reencoding" the received information bits to find a new set of "parity" bits, *i'P*, and then adding the new set to the received set, p'. The syndrome corresponds to the column of *H*, or row of H^T, where the single error occurred. This can be determined by table look-up. If not error occurred, $s = 0$. In both cases the same Pascal code for forming *iP (or i'P)* is the basis of their calculation.

Note that for both encoding, where *iP* is needed, and for decoding, where *i'P* is needed, the same program statements can be used.

The particular systematic Hamming code chosen uses

$$H^T{}_{31,5} \times (-P^T \backslash I) = (3,5,6,7,9,10,11,12,13,14,15,17,18,19,20,21,22,23,24,25,26,27,28,29,30,31$$
$$\backslash 1,2,4,8,16$$

where each digit represents the corresponding 5-digit column vector.

For example:

$$,5, = \begin{matrix} 1 \\ 0 \\ 1 \\ 0 \\ 0 \end{matrix}$$

Notice that the values of 2^n, n = 0,1,2,3,4, form the identity matrix at the end.

We need to form *iP* for encoding, or *iP* for decoding. The Pascal code utilizes the variable *i* to store the information bits. Rather than perform 5 sets of 26 multiply/accumulates to generate the 5 parity bits, the entire group of 5 is generated in the variable *'ham'* in one set of 26 exclusive-OR operations. Beginning with the first row of *P*, *'count'* = *3* = ... *00011*, *'ham'* is ex-OR'ed with *'count'* 26 times. A trap is set to skip the first integer power of 2 encountered, $n = 4$. When 4 is encountered, it is skipped, and the trap is reset for the next integer power of 2, and so on.

5.0 VOICE ENCODER STANDARDS (cont'd)

5.4.2.2 Burst Error Detection

To enable the decoder to detect burst errors occurring across the most critical partial correlation coefficients bits, a further 8 bits are augmented to the 26 most important partial correlation bits and their associated 6 bits of error protection. The 26 partial correlation coefficients bits, p, the 6 single bit error correction/double error detection bits, e, and the 8 burst error detection bits, b, are arranged in a 8 by 5 matrix structure as illustrated below.

```
pppppppp
pppppppp
pppppppp
ppeeeeee
bbbbbbbb
```

Even parity bit error protection is provided, on a column-by-column basis, to this matrix of bits, using the 8 burst error detection bits. The 8 burst error detection bits are exclusive-ORed with the mask 10101010 before transmission. At the decoder the received burst error detection bits are similarly exclusive-ORed with this mask, before the error detection process commences. Each column of the resulting matrix are then examined for a potential lack of even parity. The matrix and the associated column parity error information are then submitted to the error detection and correction process defined in Section 5.5.2. If that process indicates that the corrected error level of the received coefficient line codes is adequate, the coefficients are then decoded from the line codes. However, if the corrected error level is inadequate, the line codes are then discarded and the output of the decoder is muted as per Section 5.5.5.1.

Note: The purpose of the 8 Burst Error Detection bits is to guard against a pathological case where an extremely long burst error series might render '0' bits in all 40 positions of the Calculation Matrix. In that case, the exclusive-OR operation on the "b" bits of the matrix would cause odd parity to be made in four of the columns, thereby forcing the error detection and correction process to discard the transmission frame.

5.4.3 Transmission Frame Structure

For a transmission bit-rate of 9.6 kbit/s and a 20 ms speech frame size, 192 bits are available per speech frame for voice encoding and synchronization. Frame synchronization is accomplished using a 20 ms frame synchronization signal available from the modem. The order in which the multipulse parameters are transferred to the transmission controller hardware follows precisely the order in which they are calculated. The order of parameter derivation per 20ms speech frame is as follows:

1. Partial correlation coefficients
2. Long-term predictor parameters
3. Pulse positions and amplitudes for excitation frame number 1.
4. Pulse positions and amplitudes for excitation frame number 2.
5. Pulse positions and amplitudes for excitation frame number 3.
6. Pulse positions and amplitudes for excitation frame number 4.
7. Pulse positions and amplitudes for excitation frame number 5.

The ordering of the parameter line code bits prior to transmission is illustrated in Figure 5-9, where each line code bit is numbered: the larger the number the more significant the bit. Before the parameters are finally transmitted to the decoder the parameter line code bits are grouped into bytes (bytes are formed starting with partial correlation coefficient number one). The most significant bit of each byte is the left-most bit. Each byte is transmitted least significant bit first, starting with the left-most byte.

5.5 The Decoder

5.5.1 Introduction

The structure and operation of the decoder is identical to the structure and operation of the local decoder of Section 5.3.5. However, the parameters of the decoder, unlike those of the local decoder of Section 5.3.5, requires appropriate decoding of the transmitted bit stream.

5.5.2 Decoding of the Partial Correlation Coefficients

Before the partial correlation coefficients are decoded the partial correlation coefficients error protection codes are first examined. The burst error detection bits (after being exclusive-ORed with the 10101010 mask) and the Hamming code word protection are used to direct the error correction/detection process as illustrated by the flowchart of Figure 5-10.

The flowchart has the following outcomes:

5.0 VOICE ENCODER STANDARDS (cont'd)

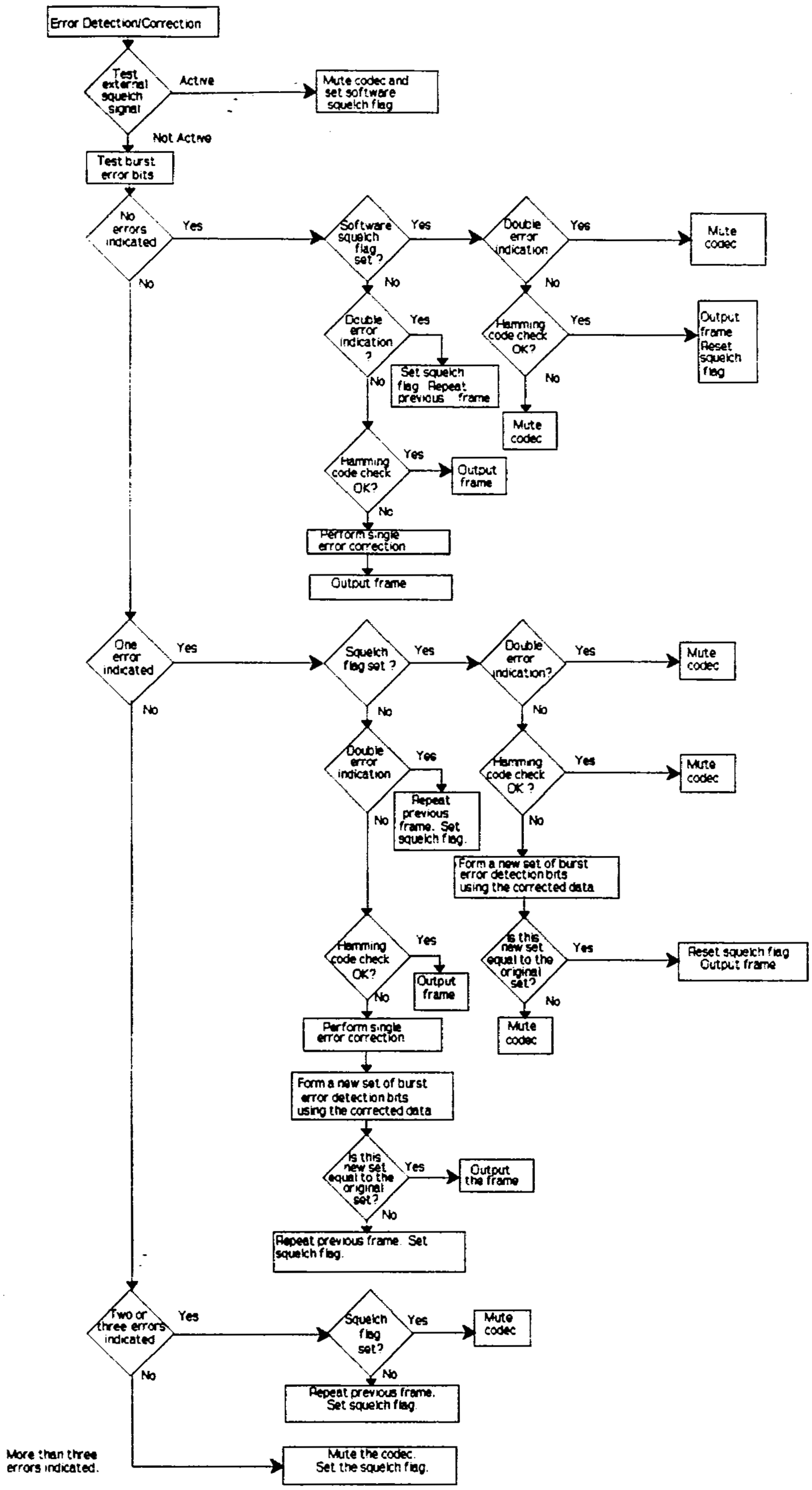


Figure 5-10 Error Correction/Detection Strategy

5.0 VOICE ENCODER STANDARDS (cont'd)

5.5.2 Decoding of the Partial Correlation Coefficients (cont'd)

1. External squelch signal is active. Mute the codec.

2. No errors indicated by burst or Hamming error codes. Decode the received line codes in accordance with the high-level code and line code Table 5-2, presented in Section 5.3.

3. Burst and Hamming error codes indicate that a single bit error has occurred in the critical partial correlation coefficient bits. Correct error and decode the corrected line codes.

4. Burst and Hamming error codes indicate that a double bit error has occurred in the critical partial correlation coefficient bits. If the partial correlation coefficients from the previous frame were received with no more than one bit in error, then use the partial correlation coefficients from the previous frame, otherwise mute the codec.

5. Burst error code indicates two or three bits in error. If the partial correlation coefficients from the previous frame were received with no more than one bit in error, then use the partial correlation coefficients from the previous frame, otherwise mute the codec.

6. Burst error codes indicate that more than three errors have occurred. Mute the codec.

After error correction/detection has been applied, the decoder's set of filter tap values, *qtaps[]*, are derived from the partial correlation coefficients, using the step up procedure (see Section 5.3.3.5).

5.5.3 Long Term Predictor Parameter Decoding

The gain and delay long term predictor parameters are decoded in accordance with the line code Tables 5-3 and 5-4 respectively.

5.5.4 Excitation Parameter Decoding

5.5.4.1 Pulse Positions

The pulse positions for each excitation frame are decoded in accordance with line code Table 5-10. Each position thus decoded, is referenced to the beginning of the current excitation frame.

5.5.4.2 Pulse Amplitudes

The pulse amplitudes in each excitation frame are received in the order in which they were transmitted. Thus after decoding the first pulse amplitude, the amplitude normalizations described in Section 5.3.6.6 must be taken account of. For the first pulse in each excitation frame the decoded pulse amplitude is multiplied by the decoder value of *gfact_leak*, to give the final pulse amplitude. The decoder, *gfact_leak*, parameter is then updated in the same way as the encoder, *gfact_leak*, value.

The decoded second pulse amplitude is multiplied by the updated *gfact_leak* parameter and the second stage, intraframe, normalization is applied. For the second pulse this means dividing the result of the above multiplication by 1.6 (see Section 5.3.6.6) to give the final pulse amplitude value. The *gfact_leak* parameter is updated.

The third pulse amplitude is obtained by multiplying the decoded value by *gfact_leak* and dividing the result by the third pulse intraframe normalization constant 2.67 (see Section 5.3.6.6). The *gfact_leak* parameter is again updated.

The amplitude compensation process described above is defined by the following process:

```
{Initialization}

gfact_leak:=1; maxadapt:=false;
adjust[1]:=1; adjust[2]:=1.0/0.625; adjust[3]:=1.0/0.375;

{3 bit initialization}

adapt[3,0]:=0.875; adapt[3,1]:=0.875; adapt[3,2]:=1.0; adapt[3,3]:=1.5;
inv_quant[3,0]:=0.2451; inv_quant[3,1]:=0.7560; inv_quant[3,2]:=1.344; inv_quant[3,3]:=2.152;

{4 bit initialization}

adapt[4,0]:=0.75; adapt[4,1]:=0.875; adapt[4,2]:=0.875; adapt[4,3]:=0.875; adapt[4,4]:=1.0;
adapt[4,5]:=1.25; adapt[4,6]:=1.5; adapt[4,7]:=2.0;
inv_quant[4,0]:=0.1284; inv_quant[4,1]:=0.3881; inv_quant[4,2]:=0.6568;
inv_quant[4,3]:=0.9424; inv_quant[4,4]:=1.256; inv_quant[4,5]:=1.618; inv_quant[4,6]:=2.069;
inv_quant[4,7]:=2.733;

VARi,qbits: integer
sgn: -1..1;
```

5.0 VOICE ENCODER STANDARDS (cont'd)

5.5.4.2 Pulse Amplitudes (cont'd)

```
FOR pulse:= 1 TO 3 DO
BEGIN
IF pulse<3 THEN BEGIN qbits:=4; qlevels:=7 END
ELSE BEGIN qbits:=3; qlevels:=3 END;
i:=0;
WHILE (abs(qamp[pulse])>inv_quant[qbits,i]) and (i<qlevels) DO i:=succ(i);
qamp[pulse]:=qamp[pulse]*gfact_leak/adjust[pulse];
gfact_leak:=EXP(0.98*LN(gfact_leak));

IF i=qlevels THEN
BEGIN
IF maxadapt THEN gfact_leak:=gfact_leak*adapt[qbits,qlevels]; maxadapt:=true
END ELSE maxadapt:=false;
gfact_leak:=gfact_leak*adapt[qbits,i];
IF gfact_leak>512.0 THEN gfact_leak:=512.0 ELSE
IF gfact_leak<0.5 THEN gfact_leak:=0.5
END;
```

If muting of the decoder output is indicated (Section 5.5.5.1), the contents of qamp[] shall be set to zero <u>after</u> the completion of the above operation.

5.5.5 Operation of the Decoder

Once all the parameters have been appropriately decoded, the decoder operates in an identical fashion to the local decoder of Section 5.3.5. The operation of the excitation generator, short term and long term predictors, which constitute the decoder, have been defined in Sections 5.3.5.1, 5.3.5.2 and 5.3.5.3 respectively.

5.5.5.1 The Muting Process

The muting process is invoked when there is an indication that a severe error condition exists. To mute the codec:

- the pulse amplitude received line codes for the first and second pulses are set to binary '0100' and the received line code for the third pulse is set to binary '010'. (By forcing the received line codes to these values the gain factor used for pulse amplitude normalization, *gfact_leak*, is adjusted to a mid-range value in readiness for the start of normal decoding.)

- set the decoded pulse amplitudes to zero and operate the decoder as outlined in Section 5.3.5.

5.5.5.2 Speech Output Transcoding

Each speech sample derived from the vector *synthspch[]* will exhibit the internal floating point numeric range as defined in Section 5.3.1.3. The numeric range of the speech sample stream should be appropriately scaled prior to any subsequent digital transcoding or analog speech reconstruction.

5.0 VOICE ENCODER STANDARDS (cont'd)

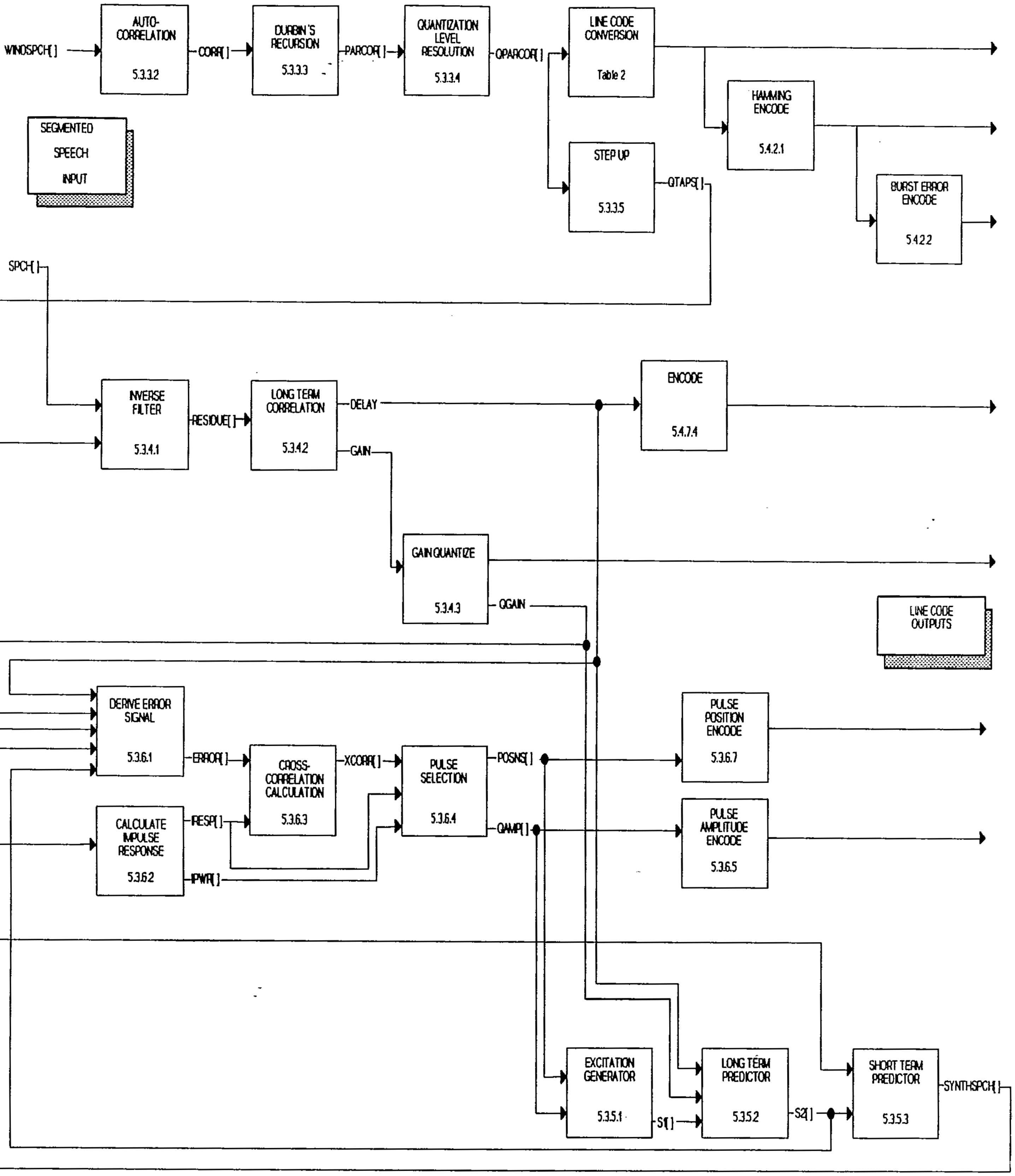


Figure 5-11 Overall Voice Encoding Algorithmy

6.0 DATA INTERFACE UNIT FUNCTIONAL REQUIREMENTS

6.1 General

The data interface unit (DIU) shall interwork with a user's CCITT V series modem and with the DIU at the distant end through a C-channel pair in accordance with requirements in the following sections.

6.2 Plesiochronous Interconnection Requirements

The clock rate accuracy of the received data from the terrestrial network is, in general, worse than the required clock rate stability of the satellite channel. It is therefore necessary to implement a plesiochronous interconnection arrangement in both is not required in the return direction, if the voice-band modems associated with both the AES and GES are synchronized to the satellite C-channel clock.

The rate adaption scheme described in Section 6.3 provides the means to transfer a variable number of octets in each C-channel frame. This capability, referred to as "stuff control," is used to adjust the plesiochronous buffer contents when a buffer overflow or underflow occurs. As a result, integrity of the user data stream is preserved by avoiding the occurrence of slips during buffer adjustment.

6.3 Rate Adaption Scheme

This rate adaption scheme applies to both the forward and return directions. The plesiochronous buffer and the associated buffer adjustment mechanism would not be used in the return direction, if the voice-band modems associated with both the AES and GES are synchronized to the satellite channel clock.

The received data is arbitrarily divided into 20 ms lengths which are referred to as described in Table 2A.1. The available transmission capacity on the C-channel is also framed into 20 ms primary fields as shown in Attachment 2X, Figure 3A. The basic transmission scheme is to divide each primary field into a number of subfields, each of length 48 bits as shown in Figure 3G. Provided the user data rate is less than the available channel capacity, the last subfield has a special format consisting of three copies of a 2-octet field made up as follows:

1. One octet being either all zeros (the normal case) or a 1-octet adjustment made to the plesiochronous buffer at the transmit end (PBA).

2. One octet packing code (PC) which indicates one of three conditions; i.e., overflow, no change, or underflow of the plesiochronous buffer.

The coding rules for the PC field and the corresponding contents of the PBA field and the reiterated data subfields for each code value are given in Table 2A.2.

The subfields from the start of the primary field carry 48-bit data fields as received from the incoming side reiterated to the extent possible in the capacity available. The possible C-channel primary field formats for the various channel rates and the different user data rates are shown in Tables 2B.1, 2B.2, and 2B.3.

6.0 DATA INTERFACE UNIT FUNCTIONAL REQUIREMENTS (cont'd)

User Data bit/s	First 48 bits	Second 48 bits	Third 48 bits	Fourth 48 bits
300	d1**	-	-	-
1200	d1*	-	-	-
2400	d1	-	-	-
4800	d1	d2	-	-
7200	d1	d2	d3	-
9600	d1	d2	d3	d4

where,
* two copies of 24 bits (bit interleaved)

** eight copies of 6 bits (bit interleaved)

Table 2A.1 Circuit-Mode User Data Segmentation

PC Value	Meaning	PBA Contents	dn Contents
1000 1111	Underflow	0000 0000	40 data + 8 zero bits
1111 0001	No change	0000 0000	48 data bits
0111 1110	Overflow	8 add. data bits	48 data bits

where,

dn   d1, d2 or d3, being the one present with the highest suffix value
PC   Packing Code
PBA   Plesiochronous Buffer Adjustment

Table 2A.2 PC/PBA/dn Contents

6.0 DATA INTERFACE UNIT FUNCTIONAL REQUIREMENTS (cont'd)

CIRCUIT-MODE DATA CHANNEL RATE ADAPTION

Subfield/User Bit Rate	1
300	d1
1200	d1
2400	d1

Table 2B.1 C-Channel Primary Band Rate = 2400 bps

Subfield/User Bit Rate	1	2
300	d1	PBA/PC
1200	d1	PBA/PC
2400	d1	PBA/PC
4800	d1	d2

Table 2B.2 C-Channel Primary Band Rate = 4800 bps

Subfield/User Bit Rate	1	2	3	4
300	d1	d1a	d1b	PBA/PC
1200	d1	d1a	d1b	PBA/PC
2400	d1	d1a	d1b	PBA/PC
4800	d1	d2	d1a	PBA/PC
7200	d1	d2	d3	PBA/PC
9600	d1	d2	d3	d4

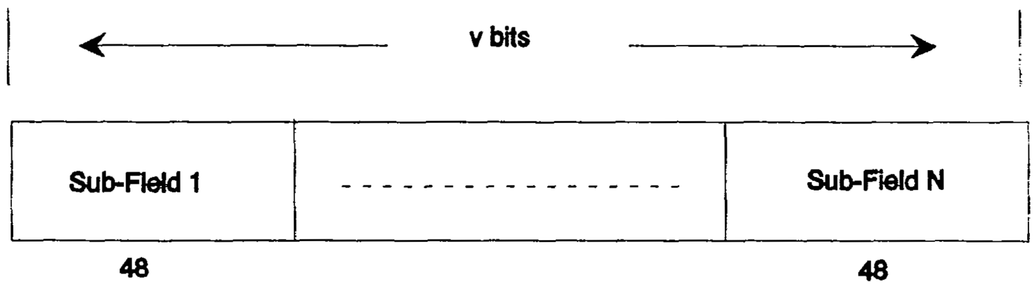
where,

a, b denote copies of the suffixed bit sequence;

PC/PBA denotes 3 copies of a 16 bit sequence composed of:
- 8 bit Plesiochronous Buffer Adjustment field
- 8 bit Packing Code field

Table 2B.3 C-Channel Primary Band Rate = 9600 bps

6.0 DATA INTERFACE UNIT FUNCTIONAL REQUIREMENTS (cont'd)



for 2400 bps C-channel primary rate, N = 1; see Table 2B.1 for sub-field contents
4800 bps C-channel primary rate, N = 2; see Table 2B.2 for sub-field contents
9600 bps C-channel primary rate, N = 4; see Table 2B.3 for sub-field contents

Figure 3G C-Channel Primary Field Content Circuit Mode Data Service

7.0 TERMINAL INTERFACE FUNCTIONAL UNIT (TIFU)

7.1 Introduction

For information on the Terminal Interface Function Unit (TIFU) refer to Attachment 3X. The material in Attachment 3X has been reprinted with the permission of Inmarsat with all rights reserved. Change Notice 64 is incorporated.

ATTACHMENT 3X

6 THE TERMINAL INTERFACE FUNCTION (TIF)

6.1 General

The TIF is a semi-transparent function using in-band signalling in the primary-band of the Aeronautical System C-channel and may be implemented entirely within the same hardware platform as the voice codec software.

6.1.1 The TIF uses a blocked frame structure, in-band signalling, and is autonomous in operation for user data rates up to and including 4.8 kbit/s, requiring no interaction with the access and control functions of the AES or GES at these rates.

6.1.2 Higher user data rates, i.e. 7.2 and 9.6 kbit/s, may require a higher quality channel, or may be accommodated using other throughput enhancing techniques, such as real-time image/data compression.

6.1.3 At higher user data-rates, i.e. 7.2 and 9.6 kbit/s, signalling is required between the TIF and the host equipment to change the Power Control Decision Table in the GES to enable a higher quality transmission path through increased forward and return link EIRPs. This signalling shall be in-band.

6.1.4 A packet-oriented structure is used to accommodate the user data at the 9.6 kbit/s data-rate with in-band signalling.

6.1.5 **In order to ensure at least a minimum level of interoperability, all TIF implementations must support the 2400/4800 bit/s facsimile function (using CCITT V.21, $V.27_{ter}$), and the 1200/2400 bit/s data function (using CCITT $V.22_{bis}$). The implementation of all other modem types and data rates is optional.**

6.2 TIF Performance Requirements

6.2.1 User Data Error Rate Performance

6.2.1.1 The TIF user-data error protection/correction algorithms are designed to ensure that the user data is adequately protected against the information channel errors likely to be encountered in the Aeronautical System which are bursty in nature following the Viterbi decoder.

6.2.1.2 The error protection/correction scheme selected for 2.4 and 4.8 kbit/s facsimile ensures that the picture quality is high at the nominal voice-mode operating BER.

6.2.1.3 Error protection/correction schemes for higher user data rates are [TBD].

6.2.1.4 The effect of the coding schemes adopted for 2.4 kbit/s $V.27_{ter}/V.22_{bis}$ data and 4.8 kbit/s $V.27_{ter}$ data is such as to produce the following average Line Errors Per Page (LEPPs) and/or information rate BERs at the nominal Aeronautical System channel BER of 1.0E-3.

TABLE 12: Performance of the TIF Error Protection/Correction Schemes for User Data

User Data Rate	Coding Method	LEPPs	Information Rate BER
2.4 kbit/s	Majority-Voting	1.52	0.7E-4
4.8 kbit/s	Cyclic Code	1.53	4.4E-4

ATTACHMENT 3X (cont'd)

6.2.2 **Facsimile Picture Data**

6.2.2.1 The TIF operates with the same quality channel as the normal voice channel for the picture element data rate of 2.4/4.8 kbit/s and provides high-quality facsimile transmissions at the voice-mode C-channel BER of 1.0E-3 or better, consistent with the SDM specification for the normal link operating BER.

6.2.2.2 A high-quality facsimile is assumed to be one in which standard graphics or text test charts, e.g. the CCITT Test Chart No. 2, etc., show no visible, subjective evidence of any errored lines in any single sheet and are delivered with no satellite network-related transmission-related error reports from the airborne or ground-based facsimile terminal equipment.

6.2.3 **Voice-Mode Integrity**

As the TIF is implemented entirely within the voice-codec function, the integrity of the voice function is preserved first and foremost and ensures that transitions from voice-to-data and vice-versa are functionally separable and are handled in a reliable fashion. Switching between voice and data modes, or just to the data mode, is performed automatically by the function as described in detail in Section 8.

The default mode of the TIF decoder is to process voice frames; a voice-to-data transition by the function's decoder is only permitted when the start of a data frame has been detected with a high degree of confidence.

6.2.3.1 The decoder decision logic can deal with severely corrupted voice or data frames by the number of parity errors detected in an 8 bit parity check word, the number of bits successfully matched in a 31 bit signalling control codeword and consideration of past decoded frames in each 24-byte, 20 ms, frame. Further details are given in Section 8.2.4.

6.2.3.2 Tests using 5,600 data-mode frames have shown that the TIF decoder's in-band control signalling scheme always remains in data mode, never switching incorrectly to voice mode, at an average channel BER of 6.4E-3. The decision logic only begins to exhibit false switches between data and voice mode at a BER of 0.9E-2.

6.2.3.3 The TIF processes all 'unrecognised' speech-band data signals as speech.

6.2.4 **AES Demodulator Preamble-Less Reacquisition**

6.2.4.1 Because the circuit-mode data signals may be continuous, for example facsimile data during Phase C, burst-mode carrier-activation based on an 'active' state signalled by a voice detected/present signal from the voice codec, may not take place in the forward direction during a data transmission once it has commenced.

The transmission of forward link burst preambles will therefore not recur once facsimile picture data has commenced until the phase has completed. Short interruptions to the forward signal longer than a few tens of milliseconds, caused by deep fading for instance, will not be followed by the commencement of a new burst with a preamble at some point when voice-activation takes place to enable the demodulator to reacquire synchronization.

6.2.4.2 It is therefore a requirement that the AES demodulator shall be capable of preamble-less reacquisition following a break in signal.

7 TIF FUNCTIONAL REQUIREMENTS

Airborne and ground-based calls supported by the Inmarsat Aeronautical System can be enhanced using the Terminal Interface Function (TIF) to include analogue-interconnect 1988 (Blue Book) CCITT Recommendations T.4 and T.30 Group-3 facsimile and V series voice-band data services in manual, automatic, private circuit or PSTN modes. Such circuit-mode data calls can be made through analogue 2/4-wire or digital telephone circuits (e.g. PCM encoded 64 kbit/s) which are carried by a pair of 21 kbit/s C-channels across the satellite subnetwork in the Inmarsat Aeronautical System.

ATTACHMENT 3 X (cont'd)

The TIF will support the following functions; this list is not necessarily complete, other modem types may also be supported by the TIF if required.

The use of square brackets indicates that the requirements contained in this Section of Module 5 are not supported by tested solutions.

7.1 Facsimile

Group-3 facsimile, supporting analogue-interconnect 1988 (Blue Book) CCITT Recommendations T.4 and T.30 only, with the following combination of data rates and V series modem types:

7.1.1 300 bit/s 300 baud half-duplex operation for binary-coded signalling sequences using CCITT Recommendation V.21 Channel 2.

7.1.2 2.4/4.8 kbit/s 1200/1600 baud half-duplex operation using CCITT Recommendation V.27$_{ter}$.

7.1.3 [7.2/9.6 kbit/s 2400 baud half-duplex operation using CCITT Recommendation V.29].

7.2 Voice-Band Data

The following CCITT V series voice-band data modems:

7.2.1 [300 bit/s 300 baud duplex operation using CCITT Recommendation V.21].

7.2.2 [600 bit/s/1.2 kbit/s 600 baud duplex operation using CCITT Recommendation V.22].

7.2.3 1.2/2.4 kbit/s 600 baud duplex operation using CCITT Recommendation V.22$_{bis}$.

7.2.4 [2.4 kbit/s 1200 baud duplex operation using CCITT Recommendation V.26$_{ter}$].

7.2.5 [4.8/7.2/9.6 kbit/s 2400 baud duplex operation using CCITT Recommendation V.32].

7.3 TIF Transparency

7.3.1 The TIF shall be totally transparent to the resolution, 1-D picture element coding and options such as 2-D picture element coding, error correction and limiting modes described in CCITT Recommendation T.4. The TIF shall not interpret or modify the content or format of any of the data, codewords or message formats used by any of these standard or optional facilities.

7.3.2 The TIF shall be semi-transparent to the binary coded control signalling employed in the above CCITT-recommended services and transmission standards.

However, all pre-message, in-message and post-message command and response signals shall be conveyed through the TIF without modification of their contents. (It is possible that the TIF may optionally inspect the NSF frame and modify its contents to disable non-standard transmission modes, as further described in Section 7.4.)

7.3.3 Similarly, the TIF shall not in any way modify or disable the fall-back of circuit-mode data equipment from a higher-speed to a lower one if the transmission path is not of an adequate quality to support the former. The transparent nature of the TIF naturally supports the fall-back feature.

7.4 Non-Standard Facilities

7.4.1 The TIF supports neither transmissions from Group 1/2 facsimile equipment nor Binary Code Procedural Signals that are transmitted at the optional 2.4 kbit/s rate.

7.4.2 The design of the TIF does not guarantee that transmissions to or from any Group-3 facsimile equipment which uses non-standard and/or proprietary procedural protocols and/or transmission modes will be supported other than the 1988 (Blue Book) CCITT Recommendations T.4/T.30.

ATTACHMENT 3X (cont'd)

[Some facsimile equipment is capable of operating in non-standard, proprietary modes, often when two machines of the same model number and type produced by the same manufacturer are connected. These procedures are usually bypassed when dissimilar machines, or machines from different manufacturers, interwork.]

7.4.3 It is highly desirable that the airborne facsimile machine shall be capable of being commanded, either by a manual or stored input, to ignore non-standard facilities when connected to the TIFU. Many, but not all, facsimile machines have the capability to disable these non-standard facilities by keypad entries which modify the internal memory in the machine and hence its configuration. Such a setting of the airborne machine automatically cancels non-standard modes of operation in both the air-to-ground and ground-to-air directions.

7.4.4 The TIF may incorporate a non-transparent facility to inspect and decode the FIF (Facsimile Information Field) to determine whether the non-standard capabilities field is transmitted by the called FTE. If so, the TIF may modify the NSF (Non-Standard Facilities) frame so as to disable the capabilities. The methods of modifying the NSF frame are [TBD].

7.5 Functional Blocks

The TIFU is comprised of the following functional blocks:

The numbers preceding each function indicate that the particular function is:

1) required for facsimile functions only; and/or
2) required for voice-band data functions only; and/or
3) required for speech functions only.

1,2,3 - 2/4-wire analogue/[CEPT E1 B-channel digital] I/O interfaces.
1,2,3 - I/O buffering.
1,2 - Threshold detection.
1,2 - Data modem preamble/carrier/training sequence detection.
1,2 - FTE/DTE-side speech/data switching.
1,2 - Line-side speech/data signalling decoding.
1,2 - Error protection for user data and speech/data mode signalling.
1,2 - Error detection and correction for user data and signalling decoder.
3 - Inmarsat Aeronautical System MELPC speech encoder and decoder.
1 - CCITT Recommendation V.21 (half-duplex, channel 2) FSK voice-band demodulator and remodulator.
1 - CCITT Recommendation $V.27_{ter}$ [half-duplex] demodulator and remodulator.
2 - CCITT Recommendation $V.22_{bis}$ [duplex] demodulator and remodulator.
1 - [CCITT Recommendation V.29 [half-duplex] voice-band data demodulator and remodulator].
1,2 - Plesiochronous buffering.
1,2 - [AES/GES control signalling].
1,2,3 - Encoder line interfacing.
1,2,3 - Decoder line interfacing.

8. TIF OPERATION

The circuit-mode data functionality can be either integrated with the speech codec or implemented in an independent platform.

The operation of the TIF when processing circuit-mode data signals shall be as follows and assumes an integrated design:

8.1 Circuit-Mode Data Call Control

No C-channel sub-band signal units are generated or required for the operation of the TIF other than those utilised in normal circuit-mode telephony calls.

8.1.1 Call Set-Up Procedure

ATTACHMENT 3 X (cont'd)

8.1.1.1 The call set-up sequence for a circuit-mode data call shall be identical to a standard telephone call set-up sequence, as defined in Sections 6.5 and 6.6 of Module 1 of the Inmarsat Aeronautical System SDM for all forms of analogue-interconnect circuit mode data services provided by the TIF.

8.1.1.2 The same AES and GES power control tables and algorithms shall also be used for user data rates of 4.8 kbit/s and less, as those used for the telephony service. The TIF error protection and correction algorithms have been designed to ensure that the data is adequately protected against the Aeronautical System channel error characteristics.

8.1.1.3 There is no requirement to identify, either to the GES or the AES, whether the TIF is present or not within the speech codecs for user data rates of 4.8 kbit/s or less.

8.1.1.4 No signalling shall be required between the TIF and the GES control processor to implement a higher quality channel at user data-rates of 4.8 kbit/s or less.

8.1.1.5 If the TIF supports higher user data rates for facsimile and voice-band data modems, e.g. 7.2 or 9.6 kbit/s V.29 facsimile data modems, a higher-quality channel with a BER of 1.0E-5 shall be established, either at the onset of transmission or at some stage during the initial procedural facsimile call set-up before the training sequence (in both facsimile and voice-band data modem cases). This is because the facsimile training check sequence, TCF, and all facsimile in-message signals are transmitted at the data-rate of the high-speed message channel. This is discussed further in Section 10.

8.1.2 Circuit-Mode Data Activation and Deactivation Procedures

8.1.2.1 All circuit-mode data activation and deactivation procedures relating to circuit-mode data operation shall be automatically performed within the TIF using only the primary band of the assigned C-channels. This shall apply to all implementation levels of the TIF and for all user data rates up to and including 9.6 kbit/s and is a fundamental TIF design principle.

8.1.2.2 No sub-band C-channel signal units shall be generated or required to indicate TIF data-mode operation other than those utilised in normal circuit-mode telephony calls.

8.1.2.3 All high-speed data, i.e. non-FSK, TIF modem types shall be recognised by performing a spectral analysis of a portion of the training/retraining sequences employed either during the initial stages of interworking or following loss of equalization. This spectral analysis will identify the characteristics of the signal in terms of carrier frequency and baud rate to identify the modulation type. Alternative methods other than spectral analysis may be used to identify the modem type, and hence modulation in use, if the TIF encoder/decoder delays, signal onset and termination timings given in Section 11 are not exceeded.

8.1.2.4 Certain V series voice-band data modem signals, e.g. V.22_{bis}, require additional in-band signalling to transfer information between interworking AES and GES TIFs. This is necessary because only the TIF connected to the called DTE may be able to detect the modulation type used in the synchronization stages of a transmission. The modulation detection process relies upon a spectral analysis of the signal which requires a portion of unscrambled repetitive data to produce the required information about baud-rate and carrier frequency. For example, in the V.22_{bis} calling modem case the handshaking signals start with scrambled 1's at 1.2 and 2.4 kbit/s, which yields no spectral information to enable the TIF to determine baud rate and carrier frequency which could identify the modem type.

However, the called V.22_{bis} modem transmits unscrambled binary 0's and 1's at 1.2 kbit/s and is therefore readily identified by spectral analysis.

This information shall therefore be communicated by additional in-band signalling to the remote TIF connected to the calling modem to enable it to select the appropriate user data rate and hence V.22/V.22_{bis} modulation processing modes (see Sub-Section 8.2.3.3).

ATTACHMENT 3X (cont'd)

8.1.3 Call Termination Procedures

User-initiated call termination may occur at any time during a circuit-mode data call using the procedures defined in Sections 6.5.5 and 6.6.4 of Module 1 of the Inmarsat Aeronautical System SDM.

8.2 Encoding

8.2.1 Input Sampling and Echo Control

8.2.1.1 Referring to the simplified TIF encoder functional block diagram shown in Figure 11 and the flow chart in Figure 13, the incoming signals, assumed to be analogue, are first converted to 8 bit PCM μ-law samples every 125μs in an A-D circuit using an 8 kHz clock. [If the incoming signals are from a CEPT E1 B-channel source, then the signals will be 8 bit PCM A-law.]

Note: Figure 13 only shows switches for the 2.4 kbit/s $V.27_{ter}$ version of the TIF.

8.2.1.2 The 8 bit μ-law binary samples are then converted to linear 32 bit floating-point samples and formed into two 20 ms long input buffers each holding 160 values; the current and previous input buffers. (The input signal is, however, first divided into overlapping segments 32 ms long, each containing 256 samples identical to the process in the speech codec (see Section 3.1.1). The windowing to reduce leakage by a Hamming window is not however performed for data signals.)

8.2.1.3 The relationship between the various frames in the TIF, number of samples, number of overlapped samples, etc., shall be identical to that used in the speech codec (see Section 3.1.1, etc.).

8.2.1.4 The circuit-mode data modem call establishment procedure tones, CED and CNG, shall pass through the speech codec transparently in speech mode. No action shall be taken or is required by the TIF encoder to detect or interpret these tones.

8.2.1.5 Echoes will be generated in a circuit-mode data call across the Inmarsat Aeronautical System in both directions from 2/4-wire terminations in the aircraft and in the terrestrial network. (There will be no local echo at most GESs as these are generally 4-wire PCM-connected to their respective international exchanges/PSTNs.)

In an analogue interconnect version of the TIF, good quality audio hybrids shall be used between the airborne FTE/DTE and the TIF to maintain a high degree of isolation between TIF modulator output and demodulator input. A minimum hybrid leakage figure of better than 20 dB is recommended.

8.2.1.6 Careful setting of the TIF output power and TIF threshold value (see Section 8.2.2.2) is also necessary to maximise the signal-to-echo ratio at the input to the TIF demodulator.

8.2.1.7 A minimum signal-to-echo ratio of around 25 dB is considered to be adequate for the TIF software to be unaffected by delayed or local echo.

[In the case of a full-duplex modem type which use the same bandwidth for both transmission directions, e.g. $V.26_{ter}$, echo cancellation will be required to provide the necessary minimum signal-to-echo ratio of around 20 dB. This in turn will require the demodulator to have access to the modulator output waveform, i.e. there shall be a direct connection between the TIF encoder and decoder. This is an implementation issue that can be resolved if a single DSP, e.g. a DSP32C, performs both functions and therefore provides the necessary internal communication path.]

8.2.2 FSK BCS Processing in the TIF Encoder for Facsimile Signalling

8.2.2.1 The TIF shall receive and demodulate all V.21 FSK BCS signals defined in CCITT T.30 which appear on the incoming voice-band circuit. The signals shall be remodulated at the distant TIF onto the outgoing voice-band circuit. Both demodulation and remodulation processes shall conform to the half-duplex, channel 2 definition in CCITT Recommendation V.21.

ATTACHMENT 3 X (cont'd)

8.2.2.2 A threshold level detector shall be used to calculate the level of the signal in each buffer.

If the input power level is less than -43 dBm, the windowed samples shall always be processed by the voice encoding algorithm as all signals below this level shall be assumed to be due to local, delayed echo or noise.

Higher level signals shall be passed to a voice/data switch algorithm to determine the characteristics of the signals and shall then be routed to the voice encoder/data demodulators as appropriate.

8.2.2.3 All voice frames shall be examined for the commencement of FSK-modulated BCS signals, using an unsynchronised FSK demodulation technique, typically by using a pair of near-optimum digital filters representing the "0" and "1" frequencies and measuring the energy in each. FSK demodulation shall be unsynchronised at this stage to provide rapid FSK data "recognition" times.

8.2.2.4 The TIF shall search the input data for the presence of an 8 bit HDLC flag sequence, which indicates the start of an FSK facsimile BCS data frame. **This is a non-transparent feature of the TIF design.**

The provisionally-demodulated FSK data is then tested for the occurrence of the sequence 0X11111X0*, the HDLC flag sequence contained in the preeamble portion of all T.30 binary coded signal (BCS) facsimile control sequences and the power contained in the "1's" and "0's" detected in each frame. (* X represents "don't care" bit.)

8.2.2.5 The CCITT T.30 recommendation specifies that the BCS frame commences with approximately one second's worth of HDLC flags (typically 38). The TIF shall acquire the start of an FSK BCS frame within two flags, i.e. within a time period of less than 54 ms.

8.2.2.6 The first full flag detected in this process may be discarded by the TIF, shortening the frame by 8 bits. If the first flag is discarded, the final delay between the incoming and the outgoing, remodulated BCS sequences at the TIF decoder shall not be altered.

8.2.2.7 Once an FSK HDLC flag byte has been detected by the unsynchronised demodulator, the rest of the BCS control sequence bits shall be detected by a "full" synchronised FSK detector from which the demodulated bits are encoded and transmitted. All the bits shall be remodulated and outputed by the TIF decoder with the possible exception of the first HDLC flag byte.

8.2.2.8 Normally there may be 6 FSK data bits detected per 20 ms block of input samples or subframe. Timing differences between the FTE/DTE and the TIF encoder may occasionally cause this to become 5 or 7 data bits.

8.2.2.9 Two buffer control bits shall therefore be allocated to each set of 5, 6 or 7 data bits in the final output buffer to indicate to the TIF decoder whether there is an underflow, overflow or correct number of bits in each 24-byte output frame.

8.2.2.10 A majority voting error protection and correction scheme shall be implemented on the user-data in the TIF up to 2.4 kbit/s. [Majority voting has been chosen for all user data rates from 300 bit/s to 2.4 kbit/s because of the small memory and computational overheads involved and the adequate level of error protection provided by such a scheme.]

8.2.2.11 The TIF output buffer, which is 24-bytes long in total, shall therefore accommodate 17 copies of 5, 6 or 7 FSK data bits, together with 17 copies of the two plesiochronous buffer control bits, a 31 bit signalling control code and 8 parity bits.

8.2.2.12 For the frame in which the BCS flag sequence is detected and if the TIF had been in the voice mode prior to the detection of data, the V.21 31 bit FSK-start, i.e. "VDS" (Voice-to-Data Switch), signalling control code shall be inserted in the appropriate locations in the 24-byte output buffer.

[See Table 21 for the allocation of the V.21 31 bit signalling control codes and Table 24 for its location.]

ATTACHMENT 3X (cont'd)

8.2.2.13 If the TIF had already been in the V.21 FSK data mode prior to the detection of FSK, the V.21 31 bit "DATA" signalling control code shall be placed in the appropriate locations in the 24-byte output buffer.

[See Table 21 for the allocation of the V.21 31 bit signalling control codes and Table 24 for its location.]

8.2.2.14 FSK demodulation continues until an energy detector indicates that FSK-modulated data has ceased.

The energy detector shall measure the energy of each symbol, and shall signal the end of FSK data when this values falls to 10% of the previously established AGC-value. The FSK turn-OFF time shall be less than 4 ms.

The TIF shall then return to processing input data as if it could be either voice, FSK or high-speed data.

8.2.2.15 If a partially-filled buffer is encountered, the last FSK data bit is located in the buffer and a single "1" shall be placed at the end of the FSK data bits to indicate to the decoder the position of the last data bit. The remaining bits in the output buffer, if any, shall be filled with "0's."

This additional 1 bit must be added to unambiguously indicate the last bit of the FSK data. This shall necessitate the transmission of a further frame if the final data bit occupies the final bit position in the frame.

The V.21 31 bit "END" control signal shall be placed in the appropriate locations in the 24-byte output buffer of the final FSK data frame. This code word indicates that the frame is the last frame of data, **and may be only partially filled by FSK data.**

[See Table 21 for the allocation of the V.21 31 bit signalling control codes and Table 24 for its location.]

8.2.2.16 After the data has been placed in the 24-byte output buffer, it is available to be output at 9.6 kbit/s and multiplexed with the sub-band signalling data and framing bits to the final channel rate of 21 kbit/s and prior to channel error control coding, interleaving and modulation. Note that the data is implicitly interleaved in the output buffer because of the replication of 17 sets of the data in multiples of 5, 6 or 7 bits.

8.2.2.17 It is essential to carefully terminate the end of each remodulated FSK BCS frame output by the decoder with a minimum of one complete flag byte as required by CCITT Recommendation T.30 to ensure correct processing of the CRC information.

8.2.2.18 Differential delay of approximately 90ms at the output of the TIF remodulator may be encountered following the transmission of the DCS sequence and when a V.29 TCF sequence is processed through the voice codec.

[This depends on whether the calling and called facsimile machines are both initially operating with V.29 modulation. The TCF sequences in this case will be unrecognised by the TIF, unlike $V.27_{ter}$ modulation, and hence will be processed by the voice codec].

To accommodate this differential delay, a [100]ms delay shall be implemented at the end of each FSK sequence fully demodulated by the TIF to ensure that the two sequences will always be separated sufficiently in time to guarantee a response from the called machine unless a valid $V.27_{ter}$ TCF sequence is received and recognised in which case the delay shall not be required.

NOTE: Section 8.2.2.18 added by Corrigenda Version 1.0 to INMARSAT SDM MODULE 5/VER 3.0 [Incorporating CN64]

8.2.3 High-Speed Data Processing in the TIF Encoder

ATTACHMENT 3 X (cont'd)

8.2.3.1 **General**

The TIF shall receive and demodulate all high-speed facsimile message and data signals modulated at [600 bit/s][1], 1.2, 2.4, 4.8, [7.2 and 9.6][1] kbit/s as appropriate and as defined in CCITT Recommendations [V.22][1], $V.22_{bis}$, $V.27_{ter}$ [and V.29][1] which appear on the incoming voice-band circuit. Each transmission is preceded by a modem synchronising sequence as specified in the CCITT Recommendation appropriate to the modulation system used.

Within the complex demodulation processes carried out in the TIF encoder, carrier phase and symbol timing is derived and applied to the detection, filtering and adaptive equalization processes, which shall be symbol-synchronised. The TIF encoder shall perform these processes in accordance with the relevant CCITT V Series Recommendations.

The high-speed facsimile and data signals shall be remodulated by the distant TIF decoder and output onto the voice-band circuit using the modulation system defined in the appropriate CCITT Recommendation.

A threshold level detector shall be used to determine the level of the signal in each input buffer.

If the input power level is less than -43 dBm the windowed samples shall always be processed by the voice encoding algorithm as all signals below this level shall be assumed to be due to local, delayed echo or noise.

Higher level signals shall be passed to a voice/data switch algorithm to determine the characteristics of the signals and be routed to the appropriate voice encoder/data demodulators.

8.2.3.2 **$V.27_{ter}$ Signalling and Training**

8.2.3.2.1 The detection of the HDLC V.21 preamble in the TIF encoder shall cause an "FSK block counter" variable to be initialised to 1000. This counter is decremented for subsequent frames for which a $V.27_{ter}$ training sequence detector is called and is used to protect against false detection of tones triggering the TIF data encoder.

8.2.3.2.2 This detector shall be called during data demodulation and therefore the "FSK block counter" is not decremented for high speed data or FSK transmissions. After 1000 non-data frames, the counter will be 0, which will cause the decrementing to halt.

8.2.3.2.3 Whilst the "FSK block counter" is greater than 0, the TIF encoder shall test frames for the 2.4/4.8 kbit/s $V.27_{ter}$ unmodulated 1800 Hz synchronising signal carrier segment (Segment 1) using spectral analysis, e.g. using Fourier transform techniques.

8.2.3.2.4 The detection of $V.27_{ter}$ 1800 Hz carrier in the synchronising signal shall place the TIF encoder into $V.27_{ter}$ mode, and a $V.27_{ter}$ "VDS" signalling control code shall be transmitted for that frame.

[The $V.27_{ter}$ training sequence detector may, typically, be called 50 times a second, establishing a 20 second window within which the 1800 Hz carrier may be detected.]

8.2.3.2.5 Having detected a portion of the $V.27_{ter}$ synchronising signal unmodulated carrier segment in the "current" 20 ms frame, the signal power within any previously input samples remaining in the previous frame buffer shall be inspected to locate the actual starting point of the 1800 Hz carrier segment data.

8.2.3.2.6 The location of the start of the unmodulated carrier segment of the synchronising signal shall then be indicated by a transition in the TIF encoder output data from continuous "0" to continuous "1" bits.

Subsequent frames have corresponding 2.4 kbit/s $V.27_{ter}$ "DATA" signalling control codes and continuous "1" bits shall be transmitted in the TIF encoder output data until the $V.27_{ter}$ silence segment, Segment 2, is detected.

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[1] Optional TIF capabilities which require further study and definition.

NOTE: The information in Attachment 3X is equivalent to Annex 1 in the INMARSAT SDM MODULE 5/VER 3.0 [Incorporating CN64]

ATTACHMENT 3X (cont'd)

The transition may be detected by power inspection and shall be indicated by a transition from continuous "1" to continuous "0" bits in the TIF encoder output data.

8.2.3.3.7 After this transition, subsequent frames shall be tested for 2.4 and 4.8 kbit/s V.27$_{ter}$ phase reversal training signals, i.e. Segment 3.

It is recommended that spectral analysis techniques are again used.

8.2.3.2.8 When detected, the start of the phase reversals shall be indicated by a transition from continuous "0" to continuous "1" bits in the TIF encoder output data.

If the phase reversals are subsequently identified as corresponding to the V.27$_{ter}$ 2.4 kbit/s fallback rate, 2.4 kbits "DATA" signalling control codes shall be added to that frame and all subsequent ones, otherwise the V.27$_{ter}$ 4.8 kbit/s "DATA" signalling control code shall be used.

If valid phase reversals are not detected, then the TIF encoder shall return to voice mode.

8.2.3.2.9 Within the encoder demodulator, when the phase reversal training sequence has been detected, training of the actual TIF encoder V.27$_{ter}$ demodulator shall commence. Whilst the continuous "1's" are being transmitted, the demodulated data bits shall be examined within the TIF encoder for a recognisable pseudo-random sequence, Segment 5, indicating the end of the training segments and the beginning of user data.

8.2.3.2.10 On detection of this pseudo-random sequence, given in the CCITT Recommendation V.27$_{ter}$ the data "1's" outputed from the TIF encoder shall be replaced by the appropriate pseudo-random sequence, followed by the demodulated user data.

8.2.3.2.11 If the quality of the data after training sequence demodulation becomes unacceptably low for three consecutive frames, the TIF encoder shall return to voice mode.

8.2.3.3 V.22/V.22$_{bis}$ Signalling and Training

8.2.3.3.1 The 2.1 kHz V.22/V.22$_{bis}$ answering modem call establishment procedure tones shall pass through the speech codec transparently in speech mode. **No action shall be taken or is required by the TIF encoder to detect or interpret these tones.**

8.2.3.3.2 TIF encoders supporting V.22/V.22$_{bis}$ shall continuously test input frames for the presence of a 4-PSK unscrambled "1's" signal modulated onto a 2.4 kHz (high-channel) carrier.

This signal is transmitted by a V.22/V.22$_{bis}$ answering modem and may be detected using spectral estimation techniques.

8.2.3.3.3 TIF encoder frames shall then be tested for the presence of the first calling modem sequence which follows shortly afterwards in the opposite direction of transmission.

In the case of a V.22$_{bis}$ calling modem, this starts with a short, about 100 ms long, periodic signal (Signal S1), consisting of unscrambled "0011" dibits, which may also be detected by spectral estimation techniques.

V.22 calling modem sequences start with a scrambled "1's" sequence, Sequence S1, which in the absence of a recognisable signature in the frequency domain, may be recognised by energy threshold detection.

[Typically, to recognise a frame by energy threshold detection, the energy of each incoming frame shall be compared with a suitable threshold value and the signal shall be deemed to be present when the total energy of two successive frames exceeds this threshold.

The threshold level may be established by measuring the encoder input noise level at a time when silence is expected and setting the threshold 10 dB above this measured value; or, when the noise is found to be essentially zero, setting the threshold 10 dB above the idling noise level.]

ATTACHMENT 3 X (cont'd)

8.2.3.3.4 Threshold detection shall only be used when a V.22/V.22$_{bis}$ calling sequence is expected, i.e. when an answering signal is in progress, as the technique does not adequately discriminate between different signal types and the threshold detector could be falsely triggered by voice-modulated or other waveforms.

8.2.3.3.5 When the TIF encoder has identified the first frame to be processed for either the calling or answering direction, the signal shall be inspected to locate the position of the first symbol within the frame.

The location of the start of the V.22/V.22$_{bis}$ calling or answering segments shall be indicated by a transition in the data transmitted to the remote decoder from continuous '0' bits to a repeated 4 bit pattern in the TIF encoder output data.

The specific 4 bit pattern transmitted by the TIF encoder shall uniquely identify to the remote TIF decoder the type of segment in the handshake sequence that has been identified and processed in the TIF encoder.

The operating speed of the remote TIF is also set by these in-band quadbit signalling codes. The in-band codes shall be as specified in the following table.

Segment type	4 bit pattern
Unscrambled binary "1" at 1.2 kbit/s	1111
Scrambled binary "1" at 1.2 kbit/s	0101
Scrambled binary "0" at 1.2 kbit/s	0001
Scrambled binary "1" at 2.4 kbit/s	0111
Double dibit "0011" at 1.2 kbit/s	0011

There are eight possible complete training sequences which remain distinguishable when repeated, each sequence consisting of a combination of the above types of segment, as defined in CCITT Recommendations V.22 and V.22$_{bis}$.

8.2.3.3.6 The TIF encoder shall locate the start of user data by inspecting the demodulated bit pattern to detect a change from continuous scrambled "1's" within the permitted time window and at the end of the appropriate segment.

When the location of the start of the user data has been found, a 16 bit unique word, value "0110101011100000" (69E0H), is placed in the output bit stream immediately before the first data bit, replacing four of the 4 bit patterns originally placed there to indicate the appropriate signalling segment.

[Since the above unique word may become corrupted by transmission errors, a match shall be deemed to have been found if at least 13 of the 16 bits are found to match the unique word bit pattern. See Sub-Section 8.2.5.25.]

8.2.3.3.7 If an acceptable match is not found for the unique word at its anticipated location, or within two symbols thereafter, then a transition shall be forced to user data transmission.

8.2.3.3.8 The TIF encoder shall unscramble the bit stream and shall transmit the unscrambled data with the appropriate error-protection coding (see Sub-Section 8.2.3.4.7).

8.2.3.3.9 If the TIF encoder encounters more than 64 consecutive unscrambled "1's" during normal data demodulation, then this is interpreted as the start of a V.22/V.22$_{bis}$ test sequence.

The TIF encoder shall then transmit the V.22_test_start signalling control code, "VTS," with unscrambled data in this and subsequent frames.

ATTACHMENT 3X (cont'd)

8.2.3.3.10 The TIF encoder shall detect the end of testing as a break in transmission.

The TIF encoder shall then transmit the V.22_test_end signalling control code, "VTE," and repeated "0000" quadbits shall "mark" the duration of the break.

[See Table 21 for a description of these signalling control codes.]

8.2.3.3.11 Resumption of user-data transmission shall be signalled by a transition from repeated "0000" bits to normal, scrambled, data.

8.2.3.3.12 If the quality of user-data demodulation becomes too low in the TIF encoder for ten consecutive frames due to poor incoming signal quality, either during training or data transmission, the TIF encoder shall revert to voice mode.

8.2.3.3.13 A $V.22_{bis}$ retraining procedure shall be implemented to enable retraining to be initiated at any time when the modems are transmitting data and either a modem retraining procedure is initiated or the channel is momentarily interrupted at either calling or answering end.

The onset of a retraining procedure without a channel break shall be signalled by the $V.22_{bis}$ in-band signalling code "RETRAIN," to enable the modulator to be informed that it is to expect control patterns, rather than data, and respond accordingly.

If the calling or answering channels are interrupted whilst the modems are transmitting data then they will begin to transmit retraining sequences, but the TIF will return to voice mode. When this happens the TIF will detect a $V.22_{bis}$ answering signal and the modems will subsequently be reconnected once the channel has been reestablished. This may cause a problem with the TIF not recognising the data rate of the transmission because the calling demodulator may be turned on at an arbitrary point in the retraining sequence, missing the critical S1 segment. The TIFU implementation shall retain information concerning the previous $V.22_{bis}$ transmission so that it will know which type of signal to look for.

A 10-second countdown time shall be implemented which increments on each 20ms TIF frame, counting the number of frames since the cessation of the last successful $V.22_{bis}$ transmission.

When $V.22_{bis}$ signals are again detected by either answering or calling mode-connected TIF demodulator the counter can be inspected and if it has not expired then it can be assumed that a warm start, ie the stored information about the data rate etc, can be implemented and the modems reconnected.

NOTE: Section 8.2.3.3.13 added by Corrigenda Version 1.0 to INMARSAT SDM MODULE 5/VER 3.0 [Incorporating CN64]

8.2.3.4 High-Speed Data Processing

8.2.3.4.1 Once the samples are confirmed to be part of a high-speed data sequence in the 'current' 20 ms frame, any samples remaining in the previous frame input buffer are tested for the presence of high-speed data.

8.2.3.4.2 If the previous and current frames are detected as high-speed data and the TIF encoder had been in voice mode prior to the detection of high-speed data, the appropriate 31 bit high-speed data "VDS" signalling control code shall be added to the 24-byte TIF encoder output buffer.

[See Table 21 for the allocation of the high-speed data "VDS" 31 bit signalling control codes and Table 25-29 for their location.]

8.2.3.4.3 If the TIF encoder had already been in data mode prior to the detection of high-speed data, the appropriate 31 bit high-speed "DATA" signalling control code shall be added to the TIF encoder output buffer.

[See Table 21 for the allocation of the high-speed data "DATA" 31 bit signalling control codes and Table 25-29 for their location.]

ATTACHMENT 3 X (cont'd)

8.2.3.4.4 If the current frame is detected as voice data, the appropriate 31 bit high-speed "END" signalling control code shall be added to the TIF encoder output buffer, indicating that the frame is the last frame of data, and may only be partially-filled by valid high-speed data.

[See Table 21 for the allocation of the high-speed data "END" 31 bit signalling control codes and Table 25-29 for their location.]

8.2.3.4.5 The TIF encoder shall acquire the timing instant of the onset of high-speed data within one frame, i.e. the high-speed data turn-ON time shall be 20 ms.

8.2.3.4.6 Normally there shall be 24 bits detected per 20 ms block of input samples at a user data rate of 1.2 kbit/s, 48 bits at 2.4 kbit/s, 96 bits at 4.8 kbit/s, etc.

Timing differences between the terminal equipment and the TIF can cause this to become 23 or 25 bits for 1.2 kbit/s data, 47 or 49 bits for 2.4 kbit/s data, 95 or 97 bits for 4.8 kbit/s data, etc., on each occasion.

A pair of buffer control bits, PC1 and PC2, shall be allocated to each set of 23/24/25, 47/48/49 or 95/96/97 user data bits for 1.2/2.4/4.8 kbit/s data respectively in the final output buffer to indicate to the TIF decoder whether there is an underflow, overflow or correct number of bits in each 24-byte frame. Table 22 shows the coding scheme for PC1 and PC2 for each user data rate.

[In the case of 2.4 kbit/s data, there is exactly the right number of bits in the final, 24-byte long, output buffer to accumulate three copies of 47, 48 or 49 bits, together with three copies of the two plesiochronous buffer control bits, a 31 bit signalling control code and 8 parity bits. There may be some spare bits with other modulation types.]

8.2.3.4.7 A majority voting error protection/correction scheme is implemented for V.22/V.22$_{bis}$/V.27$_{ter}$ 1.2 and 2.4 kbit/s user data. A cyclic-code scheme with optimum decoding is used for V.27$_{ter}$ 4.8 kbit/s user data.

8.2.3.4.8 After the data has been placed in the 24-byte output buffer it is available to line at 9.6 kbit/s as required by the ACSE. In the case of user data rates less than 2.4 kbit/s, the data is implicitly interleaved in the output buffer because of the replication of a number of sets of the user data. This is useful additional protection against burst errors.

8.2.3.4.9 Demodulation continues until the energy detector indicates that high-speed modulated data has ceased and the TIF encoder shall return to processing input data as voice or data.

[It is recommended that the energy detector measures the energy of each symbol, and signals the end of high-speed data when this value falls to 10% of the previously established AGC-value over three successive symbols.]

8.2.3.4.10 The high-speed data turn-OFF time shall be less than 3 ms at 2.4 kbit/s, 2 ms at 4.8 kbit/s and 1.5 ms at 9.6 kbit/s.

8.2.3.4.11 If a partially-filled output buffer is encountered, the last high-speed data bit to be output is located in the buffer and a single "1" bit shall be placed at the end of the high-speed data to unambiguously indicate the position of the last data bit to the TIF decoder.

The remaining locations in the output buffer, if any, shall be filled with "0's." This shall necessitate the transmission of a further frame if the final user data bit occupies the final location in the frame.

8.2.4 TIF Decoder

Referring to the TIF decoder detailed functional block diagram, Figure 12, and the flow chart shown in Figure 14, the 9.6 kbit/s line signal is first buffered in a 24-byte digital input buffer.

Once the parity bits have been checked, the 31 bit signalling control code is examined to determine what deinterleaving, error correction and modulation process will be required for the data in the frame and whether the current frame is a start-of-data frame, a continuation data frame, or that the code is signalling the end of the current period of data transmission, i.e. an end-of-data frame.

ATTACHMENT 3X (cont'd)

Once this has been determined, the data is deinterleaved, buffer-control examined and the appropriate number of bits accumulated. The error correction logic is invoked, error-correction completed and then the appropriate data remodulator is activated to output the modulated analogue samples to the FTE/DTE.

8.2.4.1 The 8 parity check, or "burst," bits are inspected to determine the number of errors present in the parity word.

[The same parity check process is used as in the voice function (see Section 4.2.2). In the TIF voice-band data case it is expected that the parity bits will have nearly all bits in error as the parity bits have been OR'ed with a sequence which converts the parity from even to odd, i.e. the opposite of the case with speech frames.

Referring again to the detailed TIF decoder flowcharts shown in Figures 10 and 14, the voice decoder and data decoder decision logic respectively, the presence of >= six errors is a positive indication that the current frame is probably a data frame. No information can be output from the TIF decoder until all the buffer contents have been read, unlike the voice decoder which can start after 88 bits have been checked and decoded.

8.2.4.2 If four or more burst errors are detected then the voice decoder is muted, and if two or three burst errors are detected, then the previous LPC frame is repeated or the decoder is again muted.

8.2.4.3 Additional decision logic to support the control signalling scheme for the data functions is shown in Figure 14.

Note: The term "data-flag" is used in the following decoder decision logic explanation to represent a generic data operation and can be associated with as many V series modem functions as are present in the TIF.

8.2.4.4 When six, seven or eight burst errors are detected, then the input frame could either be a control frame, corrupted by channel errors, or, possibly, a badly corrupted voice frame. The additional logic shown in Figure 14 is designed to initiate appropriate action in the decoder.

8.2.4.5 For six or more burst errors, the first process checks the 31 bit signalling code field using a bit comparison with the stored codes.

8.2.4.6 If greater than 26 of the received bits match one of the stored codes than a positive identification of that code is noted.

[In Figure 15, variable IDm represents the number of bits matched with code m, where m=1 corresponds to the VDS code, m=2 the DATA code, and m=3 the END code. These code words are different for each V series modem function present in the TIF, see Table 21.]

8.2.4.7 If the VDS code is detected, then the signalling and data fields of the control frame are decoded, a flag set (the "data-flag") and the variable BFNUM set to zero.

8.2.4.8 A 0-to-1 transition in the "data-flag" will initiate a switch in the decoder from the speech algorithm to the appropriate data demodulator, and the transfer of the decoded signal and data information to the modulator.

8.2.4.9 Variable BFNUM is a bad control frame counter used to count the number of successive bad frames and is set to zero because IDm>26.

8.2.4.10 If the DATA code is detected, then the "data-flag" is checked. If the flag is set, then the modulator is deemed to be operating in a satisfactory manner and the data and signal information decoded. Variable BFNUM is set to 0.

ATTACHMENT 3 X (cont'd)

8.2.4.11 With the "data-flag" remaining set, the decoder will transfer the data and the data size information to the modulator and modulation will continue. If however the "data-flag" is found not to be set, than an abnormal situation has arisen as the "data-flag" should always be set for a DATA frame. In this case, if the previous frames were not of a DATA type, then it is decided that the DATA frame has been detected through the severe corruption of a frame from the voice encoder, and therefore the voice decoder is muted, the squelch flag set, and a DATA frame counter incremented by 1.

8.2.4.12 The identities of the previous frames are checked by inspection of a data frame counter, variable DFNUM. If DFNUM<2 then the previous frames were not of type DATA.

8.2.4.13 If DFNUM>1 then there have been three consecutive DATA frames and it is decided that the DATA identification is valid and therefore the "data-flag" is set, the data decoded and BFNUM set to 0. This branch of the decision logic allows the decoder to switch to the data mode after corruption has caused either the VDS code to be missed or the data mode to be wrongly terminated.

8.2.4.14 If the END code is detected and the "data-flag" is set, then the "data-flag" is reset, the data decoded and DFNUM is set to 0. The 1-to-0 transition of the "data-flag" will indicate the end of modulation to the decoder and the modulator will remain "ON" for the next frame in order to allow all the buffered data to be output.

8.2.4.15 For the frame following an END the modulation will be continued. If the "data-flag" is found not to be set, then the voice codec is muted, the squelch flag set and DFNUM set to 0.

8.2.4.16 If six or more burst errors have been detected but none of the IDm variables are greater than 26, then the "data-flag" is checked.

8.2.4.17 If the flag is set and the previous two frames were not both bad, then the frame is considered to be a corrupted DATA frame and modulation is continued. The data is decoded and the "data-flag" left set, but the occurrence of a bad frame is recorded by incrementing the bad frame counter, BFNUM, by 1. If, however, the two previous frames were bad (BFNUM>1), then the frame is considered to be a severely corrupted voice frame, and the "data-flag" is reset, the voice codec muted, the squelch flag set and DFNUM set to 0.

8.2.4.18 If six or more burst errors have been detected and the "data-flag" is found to be reset, then the frame is again considered to be a severely corrupted voice frame. The voice decoder is therefore muted and the squelch flag set.

8.2.4.19 The decision logic described so far has dealt with severely corrupted speech frames and DATA frames with little or no corruption indicated, with six or more burst errors detected. It cannot be expected that control information will be successfully communicated when severely corrupted, but the effects of such corruption can be reduced by consideration of past decoded frames. An essential requirement of the decision logic is to allow a switch from modulator to speech decoder when non-control frames are detected to reduce the effects of an END control code being missed at the decoder. However, such a capability will also allow severe corruption to initiate a modulator-to-speech decoder switch during a transmission of demodulated data. With the subsequent modem retraining such an action would be likely to cause significant corruption of a facsimile transmission. A decision logic branch has therefore been included for when less than six burst errors are detected.

8.2.4.20 If less than six burst errors are detected and the "data-flag" is not set, then the frame is treated as a speech frame, with DFNUM set to 0, and the speech decoder logic shown in Figure 10 then used. If the "data-flag" is set, then the bad frame counter (BFNUM) is checked. If the previous two frames are shown to be bad, then the "data-flag" is reset, the speech decoder muted, the squelch flag set and DFNUM set to 0. If the previous frames were both not bad, then the data is decoded, modulation continued and BFNUM incremented by 1.

8.2.4.21 At the decoder the $V.27_{ter}$ carrier, silence and training sequence transitions and durations shall be generated and controlled in the remodulator according to the "0-to-1" transitions placed in the data bits in the encoding process (see Sub-Section 8.2.3.2).

NOTE: The information in Attachment 3X is equivalent to Annex 1 in the INMARSAT SDM MODULE 5/VER 3.0 [Incorporating CN64]

ATTACHMENT 3X (cont'd)

8.2.4.22 In the case of $V.27_{ter}$ operation, if the location of the appropriate pseudo-random sequence is not identified within $\pm$ 5 symbols of its expected location in the incoming bit stream, then the incoming data shall be assumed to start at the expected location (which shall be based upon a symbol count from the start of the phase reversals).

8.2.4.23 The transitions between the segments in the $V.22/V.22_{bis}$ handshake sequences and the durations of the segments shall control the data generated by the remodulator in accordance with the changes detected in the incoming bit patterns (see the table in Sub-Section 8.2.3.3.6).

8.2.4.24 The $V.22/V.22_{bis}/V.27_{ter}$ segment identification sequences shall be replaced by the corresponding data patterns which are then modulated to regenerate an ideal version of the appropriate training waveform.

8.2.4.25 For $V.22_{bis}$ transmission, the TIF decoder shall locate the start of the user data by detecting the 16 bit unique word "0110101011100000" (69E0H) in the incoming bit stream.

Since the unique word may become corrupted by transmission errors, a match shall be deemed to be acceptable if at least 13 out of the 16 bits are found to match. If an acceptable match for the unique word is not found at its anticipated location, or within two symbols thereafter, the TIF decoder shall decode the following symbols as user-data.

8.2.4.26 The TIF decoder shall rescramble the user data bit stream in accordance with the scrambling defined in CCITT Recommendation $V.22_{bis}$.

8.2.4.27 When the TIF decoder detects the "VTS" signalling control code it shall remodulate the received symbols in that frame and subsequent frames in unscrambled form.

8.2.4.28 Following detection of the "VTE" signalling control code the TIF decoder shall turn off data remodulation and transmission for the whole time period that the "0000" sequence is being received.

8.2.4.29 Resumption of normal data scrambling, remodulation and transmission shall be detected by the TIF decoder when the data changes from the repeated "0000" pattern to normal data and when the first non-zero quadbit is received.

8.2.5 TIF Facsimile Function: Propagation Delay Compensation (optional)

The TIF shall optionally be capable of generating a data sequence which consists of repeated Flag patterns of HDLC frame and which is modulated in CCITT V.21 form, and transmitting it to the FTE over the outgoing analogue interconnect circuit. This sequence is provided in order to ensure that the response timeout limits set out in CCITT Recommendation T.30 are not violated at the FTE.

Generation of the sequence of Flags shall occur if reception of a valid V.21 data stream from the other TIF is not declared within 1400 ms (for GES TIF) or 2400 ms (for AES TIF) of the reception (from the FTE) of the end Flag of the FSK BCS signal requiring the response.

The binary coded procedural signals which will be received from the FTE on the incoming analogue interconnect circuit and which require a response from the distant TIF are as follows:

- DIS: Digital identification signal;
- DTC: Digital transmit command;
- TCF: Training check (preceded by DCS: Digital command signal);
- EOM: End-of-message;
- MPS: Multipage signal;
- EOP: End-of-procedure;
- CTC: Mode setting command;
- PPS.NULL: Partial page signal/partial page boundary;
- PPS.EOM: Partial page signal/EOM;
- PPS.MPS: Partial page signal/MPS;
- PPS.EOP: Partial page signal/EOP;
- EOR.NULL: End-of-retransmission/partial page boundary;
- EOR.EOM: End-of-retransmission/EOM;

ATTACHMENT 3 X (cont'd)

- EOR.EOM: End-of-retransmission/EOM;
- EOR.MPS: End-of-retransmission/MPS;
- EOR.EOP: End-of-retransmission/EOP; and
- RR: Receive ready.

If the TIF receives V.21 data stream from the other TIF within 6 seconds of the commencement of a transmission of the sequence of Flags, transmission of the sequence of Flags shall continue until an integral Flag (one-octet) has been completed. This shall be followed immediately by the signal modulated by the data stream received from the other TIF without an interruption of signal energy on the outgoing analogue interconnect circuit.

When either the sequence of Flags has lasted for the period of 6 seconds or another FSK BCS signal has been received on the incoming analogue interconnect circuit from the FTE while transmitting the sequence of Flags, the receiving TIF shall immediately terminate the transmission of the sequence.

9 TIF INTERFACE REQUIREMENTS

9.1 TIF Functional Interface

The TIF functional interface can be identical to the voice codec functional interface recommended in Module 1, Appendix 4, of the SDM for V.21/V.22$_{bis}$/V.27$_{ter}$ analogue interconnect services operating at user data rates below 7.2 kbit/s.

9.1.1 For voice-band modem types which operate with a common carrier frequency for both directions of transmission, e.g. V.26$_{ter}$, a direct connection between the TIF encoder and decoder will be required to enable echo control software to be installed. This can most conveniently catered for by implementing the two functions in a single DSP circuit and is the recommended architectural approach.

9.1.2 For higher user data rates a software interface shall be necessary from the TIF to the AES or GES control logic to signal the requirement for a high-quality channel.

9.2 Analogue/Digital AES/GES Interfaces

Although airborne or ground-based circuit-mode voice-band data services users may interconnect by analogue means, it should be borne in mind that the interconnection to the TIF in either or both the AES and GES may be analogue or digital. Digital interconnection to the TIF is likely to be needed when either the GES, or aircraft cabin communications system(s), interfaces to the TIF using a digital PCM connection.

Note: Additional clock slip precautions must be taken if PCM digital interconnections are required between the TIF and the cabin or ground networks. One recommended course of action is to use an accurate clock in the AES itself and this solution is referenced in Module 2, Section 4.3.3, *C-Channel Transmitter Clock Accuracy*.

A more acceptable long-term solution will be to include a plesiochronous sliding interpolation process in the TIF decoder. The details of this process are TBD.

The TIF currently incorporates plesiochronous buffering, to be described in the next section, which eliminates completely any data slippage between analogue-connected terminal equipment, the satellite network and the PSTN, provided the links are analogue from the GES to the PSTN, which is not always the case.

Voice-band services shall use a common circuit-mode data channel format as described in Section 9.4.

ATTACHMENT 3X (cont'd)

9.3 Plesiochronous Interconnection Requirements

9.3.1 The clock rate accuracy of the voice-band data received from the FTE by the TIF is, in general, much worse than the SDM-specified clock rate accuracy of the satellite system C-channel equipment. It is therefore necessary to implement a plesiochronous buffer arrangement in both the AES and GES TIFs in the direction of transmission.

9.3.2 The rate adaption scheme described in Section 9.4.1 provides the means to transfer a variable number of data bits in each primary data field of 192 bits.

9.3.3 The rate adaption scheme shall be used to adjust the data field contents when an overflow or underflow occurs. Integrity of the user data is therefore assured, avoiding the occurrence of data slips for all combinations of FTE and satellite system data clock accuracies.

9.4 Channel Format Interfacing

9.4.1 Each circuit-mode data call shall utilise a pair of C-channels for the duration of the call.

9.4.2 The digital information bit stream carried in the primary-band of the C-channel shall be directly employed to transport the data traffic.

9.4.3 Each C-channel primary field of 192 bits shall contain only data signals or encoded voice signals, depending upon the current TIF mode.

9.4.4 The voice encoding/decoding algorithms normally associated with C-channel telephony signals shall not be used during the data transfers, although the TIF can switch between data and voice processing and transmission automatically throughout the call.

9.4.5 Rate adaption, error protection, plesiochronous buffering and bit-interleaving shall be used in the primary field to match the user data rate to the bit-rate of the primary band of the assigned C-channels and protect the user-data from slips and transmission errors.

9.4.6 Rate Adaption Scheme

The following rate adaption scheme for circuit-mode data shall apply to both the forward and return directions and only to the C-channel primary-band rate of 9.6 kbit/s.

9.4.6.1 The available C-channel transmission capacity is subdivided into primary data fields of 20 ms duration, i.e. 192 bits, at the primary-band data rate of 9.6 kbit/s.

9.4.6.2 The TIF shall allocate the user data bits in a primary data field of 192 bits for each of the V series data rates from 300 bit/s to 9.6 kbit/s, as shown in Tables 24 to 32.

ATTACHMENT 3 X (cont'd)

TABLE 13: Rate Adaption Summary

User Data Rate bit/s	No. of User Data Bits per Field (max)	No. of Copies per Field
300	7	17
600*	13	9
1200*	25	5
2400	49	3
4800	97	1
7200*	145	1
9600*	192	1

* This is only an indication of a possible rate adaption scheme for these particular user data rates.

TABLE 14: 300 bit/s User Data Rate (V.21)

Function	Code	No. of bits
Signalling Control Code	SC	31
Burst Error Detection Code	BD	8
Data	A1-A7	7
Plesiochronous Buffer Code	PC	2
Error Protected Bits (including Data & PBCs)	16x(PC1,2+A1-A7)	144
Total		192

TABLE 15: 600 bit/s User Data Rate (V.22)

Function	Code	No. of bits
Signalling Control Code	SC	31
Burst Error Detection Code	BD	8
Data	A1-A13	13
Plesiochronous Buffer Code	PC	2
Error Protected Bits (including Data & PBCs)	8x(PC1,2+A1-A13)	120
Unused	SPARE	18
Total		192

NOTE: The information in Attachment 3X is equivalent to Annex 1 in the INMARSAT SDM MODULE 5/VER 3.0 [Incorporating CN64]

ATTACHMENT 3X (cont'd)

TABLE 16: 1.2 kbit/s User Data Rate (V.22/V.22[illegible])

Function	Code	No. of bits
Signalling Control Code	SC	31
Burst Error Detection Code	BD	8
Data	A1-A25	25
Plesiochronous Buffer Code	PC	2
Error Protected Bits (including Data & PBCs)	4x(PC1,2+A1-A25)	108
Unused	SPARE	18
Total		192

TABLE 17: 2.4 kbit/s User Data Rate (V.22[illegible]/V.27[illegible])

Function	Code	No. of bits
Signalling Control Code	SCC	31
Burst Error Detection Code	BD	8
Data	D1	49
Plesiochronous Buffer Code	PC	2
Error Protected Bits	2x(PC1,2+A1-A49)	102
Total		192

TABLE 18: 4.8 kbit/s User Data Rate (V.27[illegible])

Function	Code	No. of bits
Signalling Control Code	SC	31
Burst Error Detection Code	BD	8
Data	D1	97
Plesiochronous Buffer Code	PC	2
Error Protection Bits	EP	54
Total		192

ATTACHMENT 3 X (cont'd)

TABLE 19: 7.2 kbit/s User Data Rate (V.29)

Function	Code	No. of bits
Signalling Control Code	SC	31
Burst Error Detection Code	BD	8
Data	D1	145
Plesiochronous Buffer Code	PC	2
Unused	SPARE	6
Total		192

TABLE 20: 9.6 kbit/s User Data Rate* (V.29)

Function	Code	No. of bits
[Signalling Control Code	SC	0]
[Burst Error Detection Code	BD	0]
Data	D1	192
[Plesiochronous Buffer Code	PC	0]
[Error Protection	EP	0]
Total		192

* The 9.6 kbit/s user rate data frames shall contain data only; the start and finish of a period of continuous 9.6 kbit/s data transmission shall be indicated by special signalling control frames.

9.4.7 Signalling Control Codes

9.4.7.1 The signalling control codes given in Table 21 shall communicate essential information to the TIF decoder to allow the correct mode of decoder operation to be determined for each received frame. Such information shall include:

- a voice-to-data switch (VDS) command;
- an 'in-data-mode' (DATA) command;
- a data-to-voice (END) command;
- a V.22$_{bis}$ test start and end (VTS/VTE) command and
- a V.22$_{bis}$ retrain (RETRAIN) command.

NOTE: Section 9.4.7.1 modified by Corrigenda Version 1.0 to INMARSAT SDM MODULE 5/VER 3.0 [Incorporating CN64]

9.4.7.2 A unique signalling code shall be transmitted for one of these commands in each frame whilst the decoder is operating in data mode, apart from the 9.6 kbit/s user rate data frame, which contains only data.

ATTACHMENT 3X (cont'd)

9.4.7.3 The different signalling control codes, which occupy bits 1 to 31 of each TIF frame, shall be allocated the appropriate 31 bit codeword listed in Table 21 to discriminate between the different control signals, as follows:

TABLE 21: 31 bit Signalling Control Codes

User Data Rate	Code	31 bit Code [S-1 to S31]
V.21 300 bit/s	VDS	1100110011001100110011001100110
	DATA	0011001100110011001100110011001
	END	0110011001100110011001100110011
$V.22/V.22_{bis}$	VDS	1010010110100101101001011010010
	DATA	1100001111000011110000111100001
V.22_Test_Start	VTS	0110100101101001011010010110100
V.22_Test_End	VTE	0110011010011001011001101001100
	END	0000111100001111000011110000111
V.22_Test_Retrain	RETRAIN	0101101001011010010110100101101
$V.27_{ter}$ 2.4 Fax	VDS	1111111111111111111111111111111
	DATA	1010101010101010101010101010101
	END	0000000000000000000000000000000
$V.27_{ter}$ 4.8 Fax	VDS	1111111111111111111111111111111
	DATA	1001100110011001100110011001100
	END	1111000011110000111100001111000

NOTE: Table 21 modified by Corrigenda Version 1.0 to INMARSAT SDM MODULE 5/VER 3.0 [Incorporating CN64]

9.4.7.4 Other codes can be introduced, if required, for other V series modems and other functions, such as signalling for V.26, V.29 and V.32.

9.4.8 Plesiochronous Buffer Control Codes (PCs)

9.4.8.1 The coding rules for the PC field, and the corresponding contents of the data and error protection fields are given in Table 22 below.

TABLE 22: Plesiochronous Buffer Codes

Bit/s	300	[600]	[1200]	2400	4800	[7200]	[9600]
	PC2, PC1 (Number of bits)						
Data bits	A1-7	A1-13	A1-25	A1-49	A1-97	A1-145	A1-192
Underflow	01(5)	11(11)	11(23)	01(47)	11(95)	11(143)	N/A
Normal	10(6)	00(12)	00(24)	10(48)	00(96)	00(144)	N/A
Overflow	11(7)	01(13)	01(25)	11(49)	01(97)	01(145)	N/A

NOTE: Table 22 modified by Corrigenda Version 1.0 to INMARSAT SDM MODULE 5/VER 3.0 [Incorporating CN64]

ATTACHMENT 3 X (cont'd)

9.4.8.2 The 2 bit PC field must be protected with the data bits, so that in the case of a user data-rate of 2.4 kbit/s, where the data sub-fields use a majority-voting rate = 1/3 code for transmission error protection, a total of 153 bits are protected.

9.4.9 Interleaving and User Data Error Protection Scheme

9.4.9.1 Post-Viterbi decoding error bursts in the line data sent to the TIF decoder are minimised by interleaving of the user data, including the error protection bits and plesiochronous buffer codes in the TIF encoder.

9.4.9.2 The 2.4 kbit/s user data and buffering bits are combined to give a 51 bit word which, because of the rate = 1/3 majority-voting error protection scheme employed for 2.4 kbit/s user data, is then repeated three times within the frame. This effectively spaces out adjacent bits in the user data by 51 bits, giving protection against error bursts of up to 51 bits in length.

9.4.9.3 The 4.8 kbit/s user data and buffering bits are combined to give a 99 bit word which is protected by a single burst error-correcting (153,99,24) optimised cyclic code based on the polynomial:

$$g(x) = x^{54} + x^{48} + x^{45} + x^{42} + x^{27} + x^{12} + x^{9} + x^{6} + 1$$

An optimal decoding method shall be employed which can correct an error burst up to 47 bits long.

9.4.10 Burst Error Detection Bits

9.4.10.1 As described in Section 8, it is essential that the signalling control codes are communicated reliably between the encoder and decoder incorporating the TIF and the voice codec function (see also Section 4.2.2). The operation of the encoder in a different mode to the decoder must be avoided and if the integrity of operation is threatened by a severe burst of errors then corrective action must be taken.

9.4.10.2 The basis for the mode protection scheme is the 2-D scheme used for the LPC bits in the voice codec function. The voice encoder transmits 40 LPC bits for each frame processed. These bits are the first bits transmitted in each 192 bit frame.

9.4.10.3 The voice codec transmission frame structure is shown in Figure 9, with K1[] to K10[] signifying the position of the ten predictor coefficients, quantised in PARCOR form to varying numbers of bits. 26 of the most important bits, listed in Table 23, are protected by a 6 bit extended Hamming code, (bits H-6 to H-1 in Table 23), and the resulting bits are transmitted immediately after the LPC bits. The 26 protected LPC bits and the 6 Hamming bits are further protected by 8 burst error detection bits, BD1-BD8, with 1 burst bit allocated to 4 protected bits. In the voice codec the burst bits are chosen to give 'vertical' even parity as shown in the following table:

TABLE 23: Burst Error Detection Bits

K1[6]	K1[5]	K1[4]	K1[3]	K1[2]	K1[1]	K2[6]	K2[5]
K2[4]	K2[3]	K2[2]	K3[5]	K3[4]	K3[3]	K4[5]	K4[4]
K4[3]	K5[4]	K5[3]	K6[4]	K6[3]	K7[3]	K7[2]	K7[1]
K8[3]	K9[2]	H-6	H-5	H-4	H-3	H-2	H-1
BD8	**BD7**	**BD6**	**BD5**	**BD4**	**BD3**	**BD2**	**BD1**

Note: The burst error detection bits are chosen to give odd parity of the columns in Table 23 for all non-voice encoded frames.

9.4.10.4 An odd-parity 'burst error detection' scheme, identical to the even parity scheme used in the voice codec function, shall be used in the TIF.

ATTACHMENT 3X (cont'd)

9.4.10.5 The use of odd parity burst 'protection' instead of even produces codes that will never be used by the voice encoder and would only ever be received at the voice decoder as voice frames after very severe corruption in transmission.

9.4.10.6 The burst error detection bits are the first to be examined in each frame when decoding after being exclusive-ORed with the 10101010 mask whether the frame is voice or data. The parity check is evaluated, and if less than or equal to three errors are detected the frame is assumed to be a voice frame and processed accordingly. If six, seven or eight errors are detected the signalling control code bits shall be decoded in accordance with Table 21 to determine the frame type.

9.4.11 Transmission Frame Structure

9.4.11.1 Detailed bit allocation tables for the blocked-frame structure for 300 bit/s, 600 bit/s, 1.2, 2.4, 4.8 and 7.2 kbit/s TIF user data rates are defined in Tables 24 to 29. The three special frames (start, data and finish) associated with the packetised structure for the 9.6 kbit/s user data rate are defined in Tables 30, 31 and 32 respectively.

9.4.11.2 The Signalling Control Code field shall always occupy the first 31 bits of the frame and the Burst Error Detection bits shall always occupy bits 47 to 54, apart from the packetised structure-based 9.6 kbit/s user data frame, which contains data only.

9.4.12 Output Interfacing - Octet Reversal

The TIF transmission frame formats shown in Tables 24 to 32 list the octets in MSB->LSB order reading from left to right. On output to line the octets are reversed, i.e. are read out of memory in the sequence 8 7 6 5 4 3 2 1, where bit 8 is the MSB, bit 1 the LSB.

10 POWER CONTROL FOR HIGHER-QUALITY CHANNELS

10.1 User data rates higher than 4.8 kbit/s require higher-quality both-way primary C-channels between AES and GES TIFs than the nominal voice-mode link BER provides.

10.2 It shall be necessary to indicate the commencement of circuit-mode data transmissions using the TIF to both the AES and GES to allow the GES Power Control Decision Table to be switched from that controlling AES and GES power levels for the circuit-mode voice service, providing a nominal channel BER of 1.0E-3 to 1.0E-4, to a new table, and hence new AES and GES power level settings, to provide a nominal channel BER of 1.0E-5 for circuit-mode data calls at user data rates higher than 4.8 kbit/s.

10.3 The new AES and GES levels shall be established in time for the receipt of the relevant modem training sequence, transmitted at the high-speed user data-rate.

10.4 Higher EIRPs for circuit-mode data calls at user data rates higher than 4.8 kbit/s can be provided by two alternative modes of operation as follows:

10.4.1 At the commencement of circuit-mode data call both AES and GES can easily determine that circuit-mode data operation is occurring from the voice/data detector. Therefore a 'data-mode' GES Power Control Decision Table can be implemented immediately irrespective of whether the data rate of the call requires a higher-quality channel or not by signalling from the GES TIF to the ACSE control software, using a simple single-line interface to the ACSE.

The higher-quality GES Power Control Decision Table shall be switched in within [TBD] ms of the commencement of a TIF-based circuit-mode data call.

If it is subsequently established by the GES-based TIF that the call is proceeding with a user data rate less than 7.2 kbit/s, by decoding the DIS/DCS BCS fields for instance, the voice-mode GES Power Control Decision Table can be switched back and link EIRP reduced to the nominal voice-mode level to maintain a BER better than 1.0E-3.

ATTACHMENT 3 X (cont'd)

10.4.2 The voice-mode GES Power Control Decision Table is used until the GES has determined that the requirements of either the calling/called FTE/DTE equipment (for instance by decoding the DIS/DCS BCS fields), and/or the AES and its own TIF capabilities, demand a higher quality channel in which case the data-mode GES Power Control Decision Table can be enabled and the power levels adjusted to provide an increase in channel quality.

The higher-quality GES Power Control Decision Table shall be switched in within [TBD] ms of commencement of a TIF-based circuit-mode data call.

10.5 Two C-channel Power Control Decision Tables therefore need to be available in the GES, one for circuit-mode voice and one for circuit-mode data services. The former would normally be active, except during circuit-mode data transmissions when the latter will govern the AES and GES C-channel signal power levels, enabling both EIRPs to be adjusted to a value capable of providing a higher quality, data-mode, BER performance around 1.0E-5 for user data rates greater than 4.8 kbit/s.

10.6 The additional GES Power Control Decision Table software flags, indicating that a switch from one table to another is required, can be processed in the GES control software for both air-to-ground and ground-to-air calling directions.

The flags can be included in a special in-band signalling field prefixed to the first modem procedural signal received at the GES in the case of ground-to-air calls, and by in-band signalling from the AES to the GES at the commencement of the call since the AES voice codec has knowledge of both its capability and mode of operation [TBD].

10.7 If a lower speed user data-rate only is supported in the TIF, i.e. 4.8 kbit/s or less, no in-band signalling, and hence no change in Power Control Decision table and AES/GES EIRPs will be requested.

11 SUMMARY OF VOICE AND TIF ENCODER/DECODER TRANSMISSION DELAYS AND TIMINGS

11.1 The maximum voice encoding delay shall be 46 ms.

The voice encoding delay measurement shall be made with a pair of voice codecs connected back-to-back.

The encoding delay time shall be measured from the instant the encoder receives the first input PCM sample to the instant the first channel bit is transmitted to line.

11.2 The maximum voice decoding delay shall be 25 ms.

The voice decoding delay measurement shall be made with a pair of voice codecs connected back-to-back.

The decoding delay time shall then be measured from the instant the decoder has received the first channel bit to the instant the first decoded PCM sample is output.

11.3 The maximum end-of-sequence delay in the TIF encoder for all types of circuit-mode data signals shall be constant at 2 sub-frames, i.e. 40 ms, plus 6 ms due to buffering in the input routine, making a total encoder delay of 46 ms.

11.4 The maximum delay in the TIF decoder shall be three 20 ms sub-frames only, i.e. 60 ms.

11.5 The maximum overall end-to-end delay between a pair of TIFs shall therefore be approximately 106 ms.

This overall delay figure refers to the delay to be measured between the end of the sequence input to the TIF demodulator and the end of the sequence emerging from the TIF remodulator. In the case of FSK BCS signals, the demodulating TIF may shorten the BCS sequence by up to 17 bits at 300 bit/s, i.e. 57 ms, making the end-to-end 'start' delay 163 ms. This is not significant because of the number of flag bytes transmitted at the start of each FSK BCS frame can vary within ±150 ms.

NOTE: The information in Attachment 3X is equivalent to Annex 1 in the INMARSAT SDM MODULE 5/VER 3.0 [Incorporating CN64]

ATTACHMENT 3X (cont'd)

The end-to-end 'finishing' delay may also be extended by a further 26 ms because the TIF will add an extra flag byte to an FSK-modulated BCS frame if only one flag byte is detected in the postamble, as permitted by CCITT Recommendation T.30.

Similarly, the end-to-end 'start' delay for high-speed data sequences may be increased by up to 20 ms, to 110-130 ms, because the TIF will truncate up to 20 ms of the carrier tone, again within the limits permitted by CCITT Recommendation T.30.

11.6 The maximum response times for 300 bit/s BCS FSK signal detection and termination shall be:

OFF to ON: less than 55 ms
ON to OFF: less than 5 ms

11.7 The maximum response times for high-speed data signal detection and termination shall be:

OFF to ON: less than 20 ms for all modulation types
ON to OFF: three symbols, i.e. less than 3 ms for 2.4 kbit/s PSK data

12 APPLICABLE DOCUMENTS

[1] *The Inmarsat Aeronautical System Definition Manual*, The International Maritime Satellite Organization, London.

[2] *Standardized test charts for document facsimile transmissions*, CCITT Recommendation T.21, 1988.

[3] *Standardization of Group 3 facsimile apparatus for document transmission*, CCITT Recommendation T.4, 1988.

[4] *Procedures for document facsimile transmission in the general switched telephone network*, CCITT Recommendation T.30, 1988.

[5] *Data Communications over the telephone network, Recommendations of the V Series*, CCITT Recommendations $V.27_{ter}$ and $V.22_{bis}$.

ATTACHMENT 3 X (cont'd)

13 GLOSSARY OF TERMS AND ACRONYMS

1-D	One-dimensional
2-D	Two-dimensional
2-FSK	2-Frequency Frequency Shift Keyed
3-D	Three-dimensional
4-DEPSK	4-Phase Differentially-Encoded Phase Shift Keyed
16-QAM	16-State Quadrature Amplitude Modulation
BTL	BT Laboratories
CCITT	Comité Consultatif International de Téléphonie et de Télégraphie
TIF	Terminal Interface Function
TIFU	Terminal Interface Function Unit
FTE	Facsimile Terminal Equipment
AES	Aeronautical Earth Station
GES	Ground Earth Station
ACSE	Access Control and Signalling Equipment
A/C	Aircraft
AGC	Automatic Gain Control
SDM	System Definition Manual
BCS	Binary Coded Signal
BER	Bit-Error Rate
C/N_o	Carrier-to-Noise Power Spectral Density
CED	Called Station Identification
DFT	Discrete Fourier Transform
DMA	Direct Memory Access
DSP	Digital Signal Processor
DIS	Digital Identification Signal
CSI	Called Subscriber Identification
CCS	Cabin Communications System
CTU	Cabin Telecommunications Unit
CDS	Cabin Distribution System
NSF	Non-Standard Facilities
ECM	Error Correction Mode
E_b/N_o	Bit-Energy-to-Noise Power Spectral Density Ratio
FEC	Forward Error Correction
FIFO	First-in-First-out (buffer)
DIU	Data Interface Unit
DMA	Direct memory access
HDLC	High-Level Data Link Control
I/O	Input/Output
NSF	Non-standard facilities
PCM	Pulse Code Modulation
PRBS	Pseudo Random Binary Sequence
PSK	Phase Shift Keyed
DPSK	Differential Ph
PEL	Facsimile Picture Element

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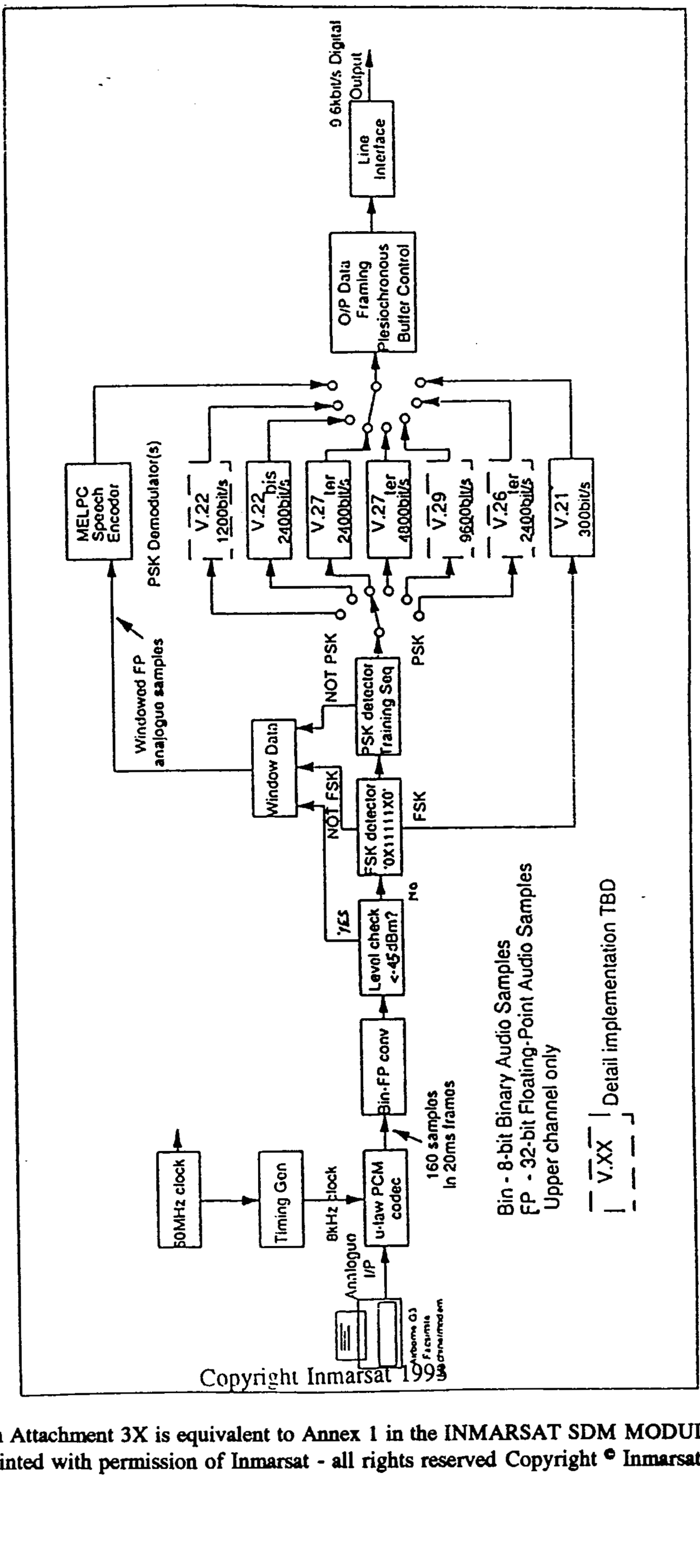
ATTACHMENT 3X (cont'd)

FIG 11: SIMPLIFIED FUNCTIONAL BLOCK DIAGRAM OF TIF ENCODER



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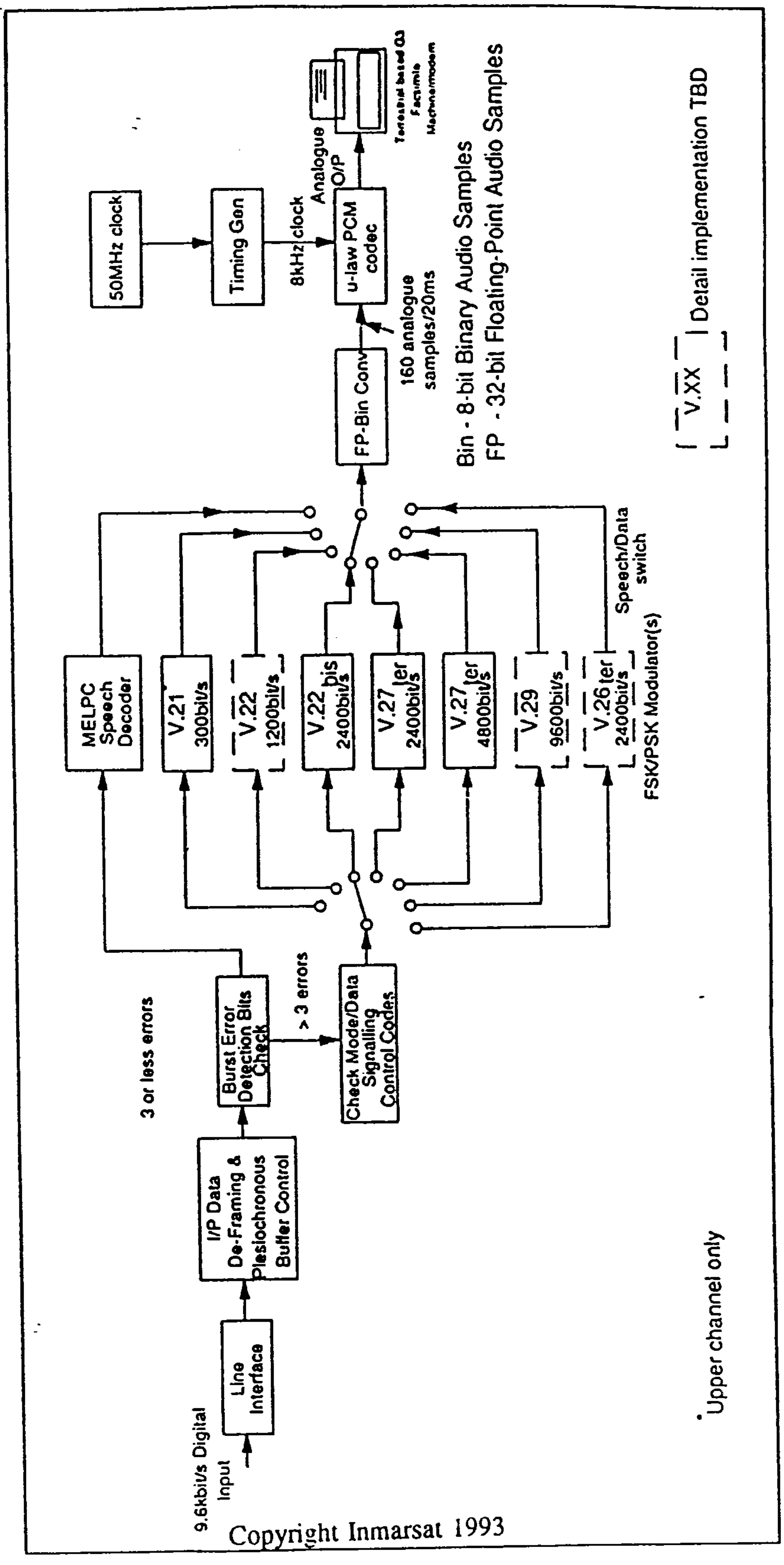
ATTACHMENT 3 X (cont'd)

FIG 12: SIMPLIFIED FUNCTIONAL BLOCK DIAGRAM OF TIF DECODER



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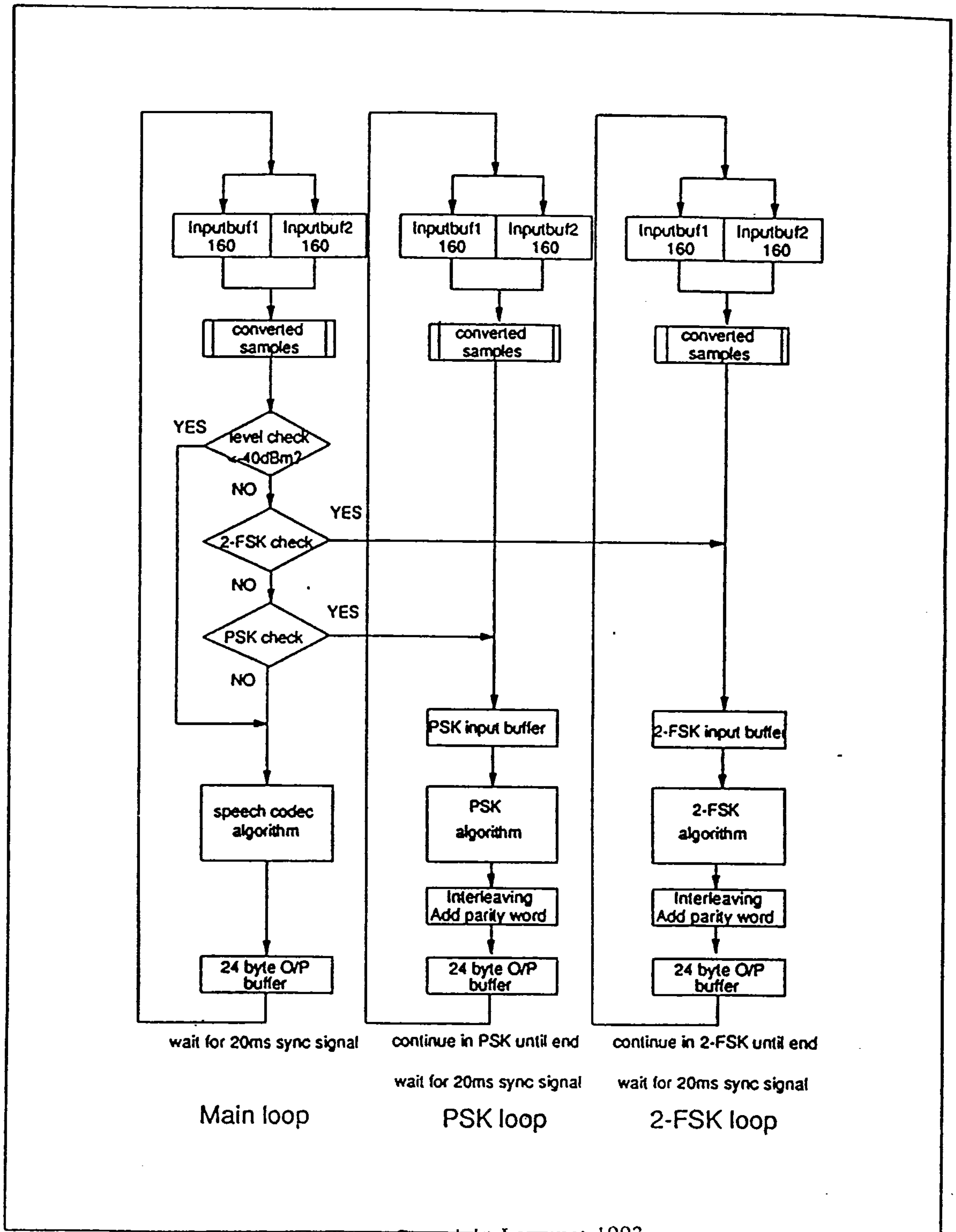
ATTACHMENT 3X (cont'd)

FIG 13: SIMPLIFIED TIF ENCODER OPERATION FLOWCHART



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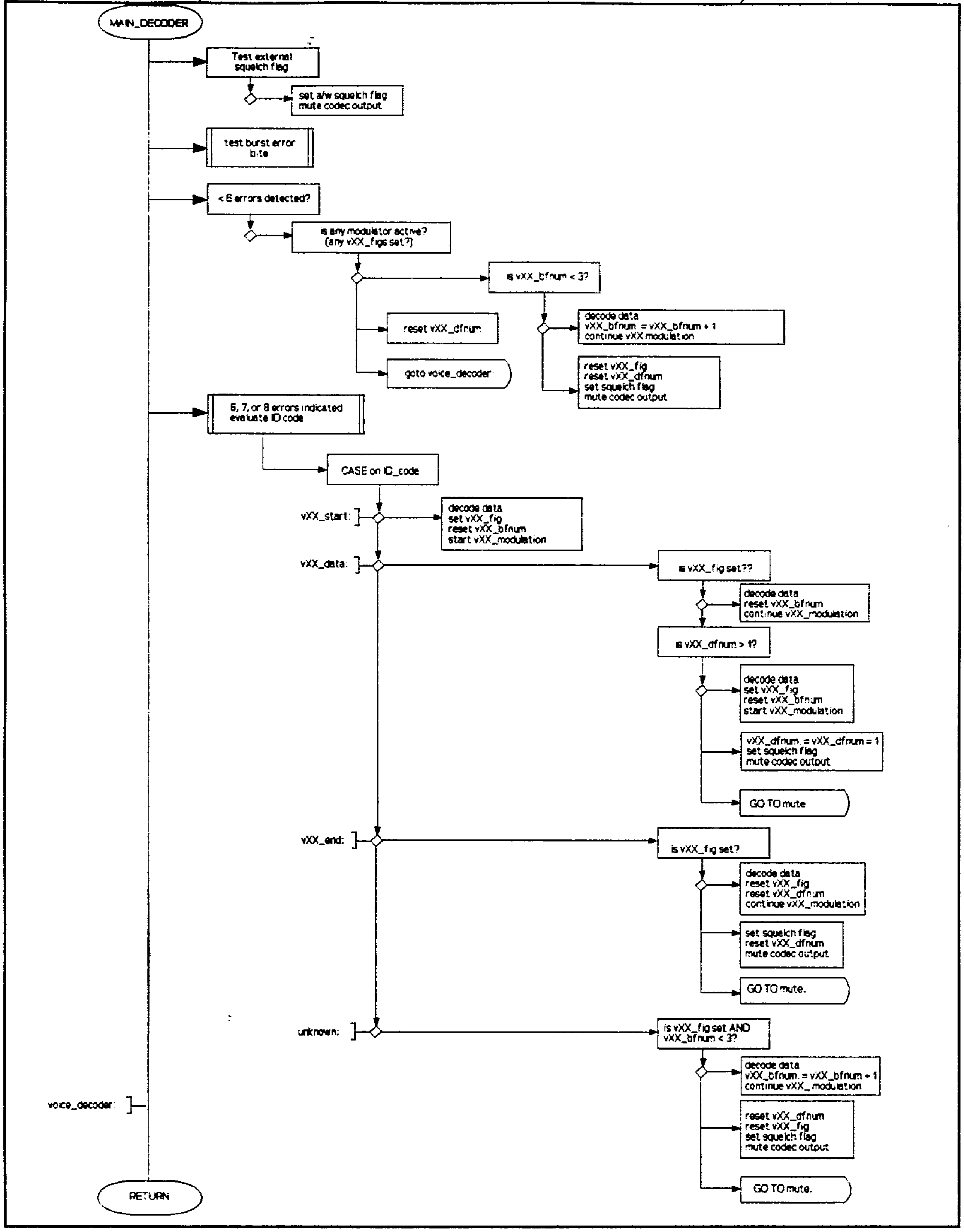
ATTACHMENT 3 X (cont'd)

FIGURE 14: SIMPLIFIED TIF DECODER DECISION LOGIC (ERROR BURST DETECTION/CODEC MUTING STRATEGY)



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ATTACHMENT 3X (cont'd)

TABLE 24: TIF TRANSMISSION FRAME STRUCTURE 300bit/s USER DATA-RATE

	Bits -	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8
	Underflow																																A-5
Octets 1- 4	Normal	S-1	S-2	S-3	S-4	S-5	S-6	S-6	S-8	S-9	S10	S11	S12	S13	S14	S15	S16	S17	S18	S19	S20	S21	S22	S23	S24	S25	S26	S27	S28	S29	S30	S31	A-6
	Overflow	Signalling Control Code																															A-7
	Underflow	A-4	A-3	A-2	A-1	?	?			B-5	B-4	B-3	B-2	B-1	?									?			C-5	C-4	C-3	C-2	C-1	?	?
Octets 5- 8	Normal	A-5	A-4	A-3	A-2	A-1	?	PC2	PC1	B-6	B-5	B-4	B-3	B-2	B-1	BD8	BD7	BD6	BD5	BD4	BD3	BD2	BD1	?	PC2	PC1	C-6	C-5	C-4	C-3	C-2	C-1	?
	Overflow	A-6	A-5	A-4	A-3	A-2	A-1			B-7	B-6	B-5	B-4	B-3	B-2									B-1			C-7	C-6	C-5	C-4	C-3	C-2	C-1
	Contents	Data (Copy 1)								Data (Copy 2)						Burst Error Detection Bits								Data(Copy2)			Data (Copy 3)						
	Underflow			D-5	D-4	D-3	D-2	D-1	?	?			E-5	E-4	E-3	E-2	E-1	?	?			F-5	F-4	F-3	F-2	F-1	?	?			G-5	G-4	G-3
Octets 9-12	Normal	PC2	PC1	D-6	D-5	D-4	D-3	D-2	D-1	?	PC2	PC1	E-6	E-5	E-4	E-3	E-2	E-1	?	PC2	PC1	F-6	F-5	F-4	F-3	F-2	F-1	?	PC2	PC1	G-6	G-5	G-4
	Overflow			D-7	D-6	D-5	D-4	D-3	D-2	D-1			E-7	E-6	E-5	E-4	E-3	E-2	E-1			F-7	F-6	F-5	F-4	F-3	F-2	F-1			G-7	G-6	G-5
	Contents			Data (Copy 4)									Data (Copy 5)									Data (Copy 6)									Data(Copy7)		
	Underflow	G-2	G-1	?	?			H-5	H-4	H-3	H-2	H-1	?	?			I-5	I-4	I-3	I-2	I-1	?	?			J-5	J-4	J-3	J-2	J-1	?	?	
Octets 13-16	Normal	G-3	G-2	G-1	?	PC2	PC1	H-6	H-5	H-4	H-3	H-2	H-1	?	PC2	PC1	I-6	I-5	I-4	I-3	I-2	I-1	?	PC2	PC1	J-6	J-5	J-4	J-3	J-2	J-1	?	PC2
	Overflow	G-4	G-3	G-2	G-1			H-7	H-6	H-5	H-4	H-3	H-2	H-1			I-7	I-6	I-5	I-4	I-3	I-2	I-1			J-7	J-6	J-5	J-4	J-3	J-2	J-1	
	Contents	Data (Copy 7) cont						Data (Copy 8)									Data (Copy 9)									Data (Copy 10)							
	Underflow	K-5	K-4	K-3	K-2	K-1	?	?				L-5	L-4	L-3	L-2	L-1	?	?			M-5	M-4	M-3	M-2	M-1	?	?			N-5	N-4	N-3	N-2
Octets 17-20	Normal	PC1	K-6	K-5	K-4	K-3	K-2	K-1	?	PC2	PC1	L-6	L-5	L-4	L-3	L-2	L-1	?	PC2	PC1	M-6	M-5	M-4	M-3	M-2	M-1	?	PC2	PC1	N-6	N-5	N-4	N-3
	Overflow	K-7	K-6	K-5	K-4	K-3	K-2	K-1				L-7	L-6	L-5	L-4	L-3	L-2	L-1			M-7	M-6	M-5	M-4	M-3	M-2	M-1			N-7	N-6	N-5	N-4
	Contents	Data (Copy 11)										Data (Copy 12)									Data (Copy 13)									Data (Copy 14)			
	Underflow	N-1	?	?			O-5	O-4	O-3	O-2	O-1	?	?			P-5	P-4	P-3	P-2	P-1	?	?			Q-5	Q-4	Q-3	Q-2	Q-1	?	?		
Octets 21-24	Normal	N-2	N-1	?	PC2	PC1	O-6	O-5	O-4	O-3	O-2	O-1	?	PC2	PC1	P-6	P-5	P-4	P-3	P-2	P-1	?	PC2	PC1	Q-6	Q-5	Q-4	Q-3	Q-2	Q-1	?	PC2	PC1
	Overflow	N-3	N-2	N-1			O-7	O-6	O-5	O-4	O-3	O-2	O-1			P-7	P-6	P-5	P-4	P-3	P-2	P-1			Q-7	Q-6	Q-5	Q-4	Q-3	Q-2	Q-1		
	Contents	Data (Copy 14) cont					Data (Copy 15)									Data (Copy 16)									Data (Copy 17)								

? - Unused bit(s)

NOTE: Table 24 modified by Corrigenda Version 1.0 to INMARSAT SDM MODULE 5/VER 3.0 [Incorporating CN64]

ATTACHMENT 3 X (cont'd)

TABLE 25: TIF TRANSMISSION FRAME STRUCTURE 600bit/s USER DATA-RATE

	Bits -	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8
Octets 1- 4		S-1	S-2	S-3	S-4	S-5	S-6	S-6	S-8	S-9	S10	S11	S12	S13	S14	S15	S16	S17	S18	S19	S20	S21	S22	S23	S24	S25	S26	S27	S28	S29	S30	S31	PC2
	Contents	Signalling Control Code																															
	Underflow		A11	A10	A-9	A-8	A-7	A-6	A-5	A-4	A-3	A-2	A-1	Z	Z											B11	B10	B-9	B-8	B-7	B-6	B-5	B-4
Octets 5- 8	Normal	PC1	A12	A11	A10	A-9	A-8	A-7	A-6	A-5	A-4	A-3	A-2	A-1	Z	BD8	BD7	BD6	BD5	BD4	BD3	BD2	BD1	PC2	PC1	B12	B11	B10	B-9	B-8	B-7	B-6	B-5
	Overflow		A13	A12	A11	A10	A-9	A-8	A-7	A-6	A-5	A-4	A-3	A-2	A-1											B13	B12	B11	B10	B-9	B-8	B-7	B-6
	Contents	Data [Copy 1]														Burst Error Detection Bits								Data [Copy 2]									
	Underflow	B-3	B-2	B-1	Z	Z			C11	C10	C-9	C-8	C-7	C-6	C-5	C-4	C-3	C-2	C-1	Z	Z			D11	D10	D-9	D-8	D-7	D-6	D-5	D-4	D-3	D-2
Octets 9-12	Normal	B-4	B-3	B-2	B-1	Z	PC2	PC1	C12	C11	C10	C-9	C-8	C-7	C-6	C-5	C-4	C-3	C-2	C-1	Z	PC2	PC1	D12	D11	D10	D-9	D-8	D-7	D-6	D-5	D-4	D-3
	Overflow	B-5	B-4	B-3	B-2	B-1			C13	C12	C11	C10	C-9	C-8	C-7	C-6	C-5	C-4	C-3	C-2	C-1			D13	D12	D11	D10	D-9	D-8	D-7	D-6	D-5	D-4
	Contents						Data [Copy 3]															Data [Copy 4]											
	Underflow	D-1	Z	Z			E11	E10	E-9	E-8	E-7	E-6	E-5	E-4	E-3	E-2	E-1	Z	Z			F11	F10	F-9	F-8	F-7	F-6	F-5	F-4	F-3	F-2	F-1	Z
Octets 13-16	Normal	D-2	D-1	Z	PC2	PC1	E12	E11	E10	E-9	E-8	E-7	E-6	E-5	E-4	E-3	E-2	E-1	Z	PC2	PC1	F12	F11	F10	F-9	F-8	F-7	F-6	F-5	F-4	F-3	F-2	F-1
	Overflow	D-3	D-2	D-1			E13	E12	E11	E10	E-9	E-8	E-7	E-6	E-5	E-4	E-3	E-2	E-1			F13	F12	F11	F10	F-9	F-8	F-7	F-6	F-5	F-4	F-3	F-2
	Contents				Data [Copy 5]															Data [Copy 6]													
	Underflow	Z			G11	G10	G-9	G-8	G-7	G-6	G-5	G-4	G-3	G-2	G-1	Z	Z			H11	H10	H-9	H-8	H-7	H-6	H-5	H-4	H-3	H-2	H-1	Z	Z	
Octets 17-20	Normal	Z	PC2	PC1	G12	G11	G10	G-9	G-8	G-7	G-6	G-5	G-4	6-3	G-2	G-1	Z	PC2	PC1	H12	H11	H10	H-9	H-8	H-7	H-6	H-5	H-4	H-3	H-2	H-1	Z	PC2
	Overflow				G13	G12	G11	G10	G-9	G-8	G-7	G-6	G-5	G-4	G-3	G-2	G-1			H13	H12	H11	H10	H-9	H-8	H-7	H-6	H-5	H-4	H-3	H-2	H-1	
	Contents	Data [Copy 7]																Data [Copy 8]															
	Underflow		I11	I10	I-9	I-8	I-7	I-6	I-5	I-4	I-3	I-2	I-1	Z	Z																		
Octets 21-24	Normal	PC1	I12	I11	I10	I-9	I-8	I-7	I-6	I-5	I-4	I-3	I-2	I-1	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z
	Overflow		I13	I12	I11	I10	I-9	I-8	I-7	I-6	I-5	I-4	I-3	I-2	I-1																		
	Contents	Data [Copy 9]																															

Z - Unused bit(s)

NOTE: The information in Attachment 3X is equivalent to Annex 1 in the INMARSAT SDM MODULE 5/VER 3.0 [Incorporating CN64] Reprinted with permission of Inmarsat -

ATTACHMENT 3X (cont'd)

TABLE 26: TIF TRANSMISSION FRAME STRUCTURE 1.2kbit/s USER DATA-RATE

	Bits -	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8
Octets 1- 4		S-1	S-2	S-3	S-4	S-5	S-6	S-6	S-8	S-9	S10	S11	S12	S13	S14	S15	S16	S17	S18	S19	S20	S21	S22	S23	S24	S25	S26	S27	S28	S29	S30	S31	PC2
		Signalling Control Code																															
Octets 5- 8	Underflow		A23	A22	A21	A20	A19	A18	A17	A16	A15	A14	A13	A12	A11									A10	A-9	A-8	A-7	A-6	A-5	A-4	A-3	A-2	A-1
	Normal	PC1	A24	A23	A22	A21	A20	A19	A18	A17	A16	A15	A14	A13	A12	BD8	BD7	BD6	BD5	BD4	BD3	BD2	BD1	A11	A10	A-9	A-8	A-7	A-6	A-5	A-4	A-3	A-2
	Overflow		A25	A24	A23	A22	A21	A20	A19	A18	A17	A16	A15	A14	A13									A12	A11	A10	A-9	A-8	A-7	A-6	A-5	A-4	A-3
	Contents	Data [Copy 1]														Burst Error Detection Bits								Data [Copy 1] cont									
Octets 9-12	Underflow	Z	Z			B23	B22	B21	B20	B19	B18	B17	B16	B15	B14	B13	B12	B11	B10	B-9	B-8	B-7	B-6	B-5	B-4	B-3	B-2	B-1	Z	Z			C23
	Normal	A-1	Z	PC2	PC1	B24	B23	B22	B21	B20	B19	B18	B17	B16	B15	B14	B13	B12	B11	B10	B-9	B-8	B-7	B-6	B-5	B-4	B-3	B-2	B-2	Z	PC2	PC1	C24
	Overflow	A-2	A-1			B25	B24	B23	B22	B21	B20	B19	B18	B17	B16	B15	B14	B13	B12	B11	B10	B-9	B-8	B-7	B-6	B-5	B-4	B-3	B-2	B-1			C25
	Contents	Data [Copy 2]																															
Octets 13-16	Underflow	C22	C21	C20	C19	C18	C17	C16	C15	C14	C13	C12	C11	C10	C-9	C-8	C-7	C-6	C-5	C-4	C-3	C-2	C-1	Z	Z			D23	D22	D21	D20	D19	D18
	Normal	C23	C22	C21	C20	C19	C18	C17	C16	C15	C14	C13	C12	C11	C10	C-9	C-8	C-7	C-6	C-5	C-4	C-3	C-2	C-1	Z	PC2	PC1	D24	D23	D22	D21	D20	D19
	Overflow	C24	C23	C22	C21	C20	C19	C18	C17	C16	C15	C14	C13	C12	C11	C10	C-9	C-8	C-7	C-6	C-5	C-4	C-3	C-2	C-1			D25	D24	D23	D22	D21	D20
	Contents	Data [Copy 3]																								Data [Copy 4]							
Octets 17-20	Underflow	D17	D16	D15	D14	D13	D12	D11	D10	D-9	D-8	D-7	D-6	D-5	D-4	D-3	D-2	D-1	Z	Z			E23	E22	E21	E20	E19	E18	E17	E16	E15	E14	E13
	Normal	D18	D17	D16	D15	D14	D13	D12	D11	D10	D-9	D-8	D-7	D-6	D-5	D-4	D-3	D-2	D-1	Z	PC2	PC1	E24	E23	E22	E21	E20	E19	E18	E17	E16	E15	E14
	Overflow	D19	D18	D17	D16	D15	D14	D13	D12	D11	D10	D-9	D-8	D-7	D-6	D-5	D-4	D-3	D-2	D-1			E25	E24	E23	E22	E21	E20	E19	E18	E17	E16	E15
	Contents	Data [Copy 4] cont																			Data [Copy 5]												
Octets 21-24	Underflow	E12	E11	E10	E-9	E-8	E-7	E-6	E-5	E-4	E-3	E-2	E-1																				
	Normal	E13	E12	E11	E10	E-9	E-8	E-7	E-6	E-5	E-4	E-3	E-2	E-1	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z
	Overflow	E14	E13	E12	E11	E10	E-9	E-8	E-7	E-6	E-5	E-4	E-3	E-2	E-1																		
	Contents	Data [Copy 5] cont																															

Z - Unused bit(s)

ATTACHMENT 3 X (cont'd)

TABLE 27: TIF TRANSMISSION FRAME STRUCTURE 2.4kbit/s USER DATA-RATE

Octets	Bits →	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8
Octets 1- 4		S-1	S-2	S-3	S-4	S-5	S-6	S-6	S-8	S-9	S10	S11	S12	S13	S14	S15	S16
		Signalling Control Code															
Octets 5- 8	Underflow		A45	A44	A47	A46	A41	A40	A43	A42	A37	A36	A39	A38	A33		
	Normal	PC1	A46	A45	A48	A47	A42	A41	A44	A43	A38	A37	A40	A39	A34	BD8	BD7
	Overflow		A47	A46	A49	A48	A43	A42	A45	A44	A39	A38	A41	A40	A35		
	Contents	Data (Copy 1)														Burst Error Detection Bits	
Octets 9-12	Underflow	A26	A21	A20	A23	A22	A17	A16	A19	A18	A13	A12	A15	A14	A-9	A-8	A11
	Normal	A27	A22	A21	A24	A23	A18	A17	A20	A19	A14	A13	A16	A15	A10	A-9	A12
	Overflow	A28	A23	A22	A25	A24	A19	A18	A21	A20	A15	A14	A17	A16	A11	A10	A13
	Contents	Data (Copy 1) cont															
Octets 13-16	Underflow	B41	B40	B43	B42	B37	B36	B39	B38	B33	B32	B35	B34	B29	B28	B31	B30
	Normal	B42	B41	B44	B43	B38	B37	B40	B39	B34	B33	B36	B35	B30	B29	B32	B31
	Overflow	B43	B42	B45	B44	B39	B38	B41	B40	B35	B34	B37	B36	B31	B30	B33	B32
	Contents	Data (Copy 2) cont															
Octets 17-20	Underflow	B-9	B-8	B11	B10	B-5	B-4	B-7	B-6	?	B-1	B-3	B-2	?			C45
	Normal	B10	B-9	B12	B11	B-6	B-5	B-8	B-7	B-2	B-1	B-4	B-3	?	PC2	PC1	C46
	Overflow	B11	B10	B13	B12	B-7	B-6	B-9	B-8	B-2	B-1	B-4	B-3	B-5			C47
	Contents	Data (Copy 2) cont													Data (Copy 3)		
Octets 21-24	Underflow	C28	C31	C30	C25	C24	C27	C26	C21	C20	C23	C22	C17	C16	C19	C18	C13
	Normal	C29	C32	C31	C26	C25	C28	C27	C22	C21	C24	C23	C18	C17	C20	C19	C14
	Overflow	C30	C33	C32	C27	C26	C29	C28	C23	C22	C25	C24	C19	C18	C21	C20	C15
	Contents	Data (Copy 3) cont															

Octets	Bits →	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8
Octets 1- 4		S17	S18	S19	S20	S21	S22	S23	S24	S25	S26	S27	S28	S29	S30	S31	PC2
Octets 5- 8	Underflow							A32	A35	A34	A29	A28	A31	A30	A25	A24	A27
	Normal	BD6	BD5	BD4	BD3	BD2	BD1	A33	A36	A35	A30	A29	A32	A31	A26	A25	A28
	Overflow							A34	A37	A36	A31	A30	A33	A32	A27	A26	A29
	Contents							Data (Copy 1) cont									
Octets 9-12	Underflow	A10	A-5	A-4	A-7	A-6	?	A-1	A-3	A-2	?			B45	B44	B47	B46
	Normal	A11	A-6	A-5	A-8	A-7	A-2	A-1	A-4	A-3	?	PC2	PC1	B46	B45	B48	B47
	Overflow	A12	A-7	A-6	A-9	A-8	A-2	A-1	A-4	A-3	A-5			B47	B46	B49	B48
	Contents											Data (Copy 2)					
Octets 13-16	Underflow	B25	B24	B27	B26	B21	B20	B23	B22	B17	B16	B19	B18	B13	B12	B15	B14
	Normal	B26	B25	B28	B27	B22	B21	B24	B23	B18	B17	B20	B19	B14	B13	B16	B15
	Overflow	B27	B26	B29	B28	B23	B22	B25	B24	B19	B18	B21	B20	B15	B14	B17	B16
	Contents																
Octets 17-20	Underflow	C44	C47	C46	C41	C40	C43	C42	C37	C36	C39	C38	C33	C32	C35	C34	C29
	Normal	C45	C48	C47	C42	C41	C44	C43	C38	C37	C40	C39	C34	C33	C36	C35	C30
	Overflow	C46	C49	C48	C43	C42	C45	C44	C39	C38	C41	C40	C35	C34	C37	C36	C31
	Contents																
Octets 21-24	Underflow	C12	C15	C14	C-9	C-8	C11	C10	C-5	C-4	C-7	C-6	?	C-1	C-3	C-2	?
	Normal	C13	C16	C15	C10	C-9	C12	C11	C-6	C-5	C-8	C-7	C-2	C-1	C-4	C-3	?
	Overflow	C14	C17	C16	C11	C10	C13	C12	C-7	C-6	C-9	C-8	C-2	C-1	C-4	C-3	C-5
	Contents																

? - Unused bit(s)

NOTE: Table 27 modified by Corrigenda Version 1.0 to INMARSAT SDM MODULE 5/VER 3.0 [Incorporating CN64]

ATTACHMENT 3X (cont'd)

TABLE 28: TIF TRANSMISSION FRAME STRUCTURE 4.8kbit/s USER DATA-RATE

	Bits -	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8
Octets 1-4		S-1	S-2	S-3	S-4	S-5	S-6	S-6	S-8	S-9	S10	S11	S12	S13	S14	S15	S16	S17	S18	S19	S20	S21	S22	S23	S24	S25	S26	S27	S28	S29	S30	S31	PC2
	Contents															Signalling Control Code																	
Octets 5-8	Underflow	D89	D90	D91	D92	D93	D94	D95		D83	D84	D85	D86	D87	D88									D81	D82	D73	D74	D75	D76	D77	D78	D79	D80
	Normal	D90	D91	D92	D93	D94	D95	D96	PC1	D84	D85	D86	D87	D88	D89	BD8	BD7	BD6	BD5	BD4	BD3	BD2	BD1	D82	D83	D74	D75	D76	D77	D78	D79	D80	D81
	Overflow	D91	D92	D93	D94	D95	D96	D97		D85	D86	D87	D88	D89	D90									D83	D84	D75	D76	D77	D78	D79	D80	D81	D82
	Contents							Data										Burst Error Detection Bits										Data					
Octets 9-12	Underflow	D65	D66	D67	D68	D69	D70	D71	D72	D57	D58	D59	D60	D61	D62	D63	D64	D49	D50	D51	D52	D53	D54	D55	D56	D41	D42	D43	D44	D45	D46	D47	D48
	Normal	D66	D67	D68	D69	D70	D71	D72	D73	D58	D59	D60	D61	D62	D63	D64	D65	D50	D51	D52	D53	D54	D55	D56	D57	D42	D43	D44	D45	D46	D47	D48	D49
	Overflow	D67	D68	D69	D70	D71	D72	D73	D74	D59	D60	D61	D62	D63	D64	D65	D66	D51	D52	D53	D54	D55	D56	D57	D58	D43	D44	D45	D46	D47	D48	D49	D50
	Contents																Data																
Octets 13-16	Underflow	D33	D34	D35	D36	D37	D38	D39	D40	D25	D26	D27	D28	D29	D30	D31	D32	D17	D18	D19	D20	D21	D22	D23	D24	D-9	D10	D11	D12	D13	D14	D15	D16
	Normal	D34	D35	D36	D37	D38	D39	D40	D41	D26	D27	D28	D29	D30	D31	D32	D33	D18	D19	D20	D21	D22	D23	D24	D25	D10	D11	D12	D13	D14	D15	D16	D17
	Overflow	D35	D36	D37	D38	D39	D40	D41	D42	D27	D28	D29	D30	D31	D32	D33	D34	D35	D25	D21	D22	D23	D24	D25	D26	D11	D12	D13	D14	D15	D16	D17	D18
	Contents					Data											Data																
Octets 17-20	Underflow	D-1	D-2	D-3	D-4	D-5	D-6	D-7	D-8							‡	‡																
	Normal	D-2	D-3	D-4	D-5	D-6	D-7	D-8	D-9	P49	P50	P51	P52	P53	P54	‡	D-1	P41	P42	P43	P44	P45	P46	P47	P48	P33	P34	P35	P36	P37	P38	P39	P40
	Overflow	D-3	D-4	D-5	D-6	D-7	D-8	D-9	D10							D-1	D-2																
	Contents					Data															Error protection bits												
Octets 21-24	Underflow																																
	Normal	P25	P26	P27	P28	P29	P30	P31	P32	P17	P18	P19	P20	P21	P22	P23	P24	P-9	P10	P11	P12	P13	P14	P15	P16	P-1	P-2	P-3	P-4	P-5	P-6	P-7	P-8
	Overflow																																
	Contents																Error protection bits																

‡ - Unused bit(s)

NOTE: Table 28 modified by Corrigenda Version 1.0 to INMARSAT SDM MODULE 5/VER 3.0 [Incorporating CN64]

ATTACHMENT 3 X (cont'd)

TABLE 29: TIF TRANSMISSION FRAME STRUCTURE 7.2kbit/s USER DATA-RATE

	Bits -	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8
Octets 1- 4		S-1	S-2	S-3	S-4	S-5	S-6	S-6	S-8	S-9	S10	S11	S12	S13	S14	S15	S16	S17	S18	S19	S20	S21	S22	S23	S24	S25	S26	S27	S28	S29	S30	S31	PC2
	Contents	Signalling Control Code																															
Octets 5- 8	Underflow		143	142	141	140	139	138	137	136	135	134	133	132	131									130	129	128	127	126	125	124	123	122	121
	Normal	PC1	144	143	142	141	140	139	138	137	136	135	134	133	132	BD8	BD7	BD6	BD5	BD4	BD3	BD2	BD1	131	130	129	128	127	126	125	124	123	122
	Overflow		145	144	143	142	141	140	139	138	137	136	135	134	133									132	131	130	129	128	127	126	125	124	123
	Contents	Data														Burst Error Detection Bits								Data									
Octets 9-12	Underflow	120	119	118	117	116	115	114	113	112	111	110	109	108	107	106	105	104	103	102	101	100	99	98	97	96	95	94	93	92	91	90	89
	Normal	121	120	119	118	117	116	115	114	113	112	111	110	109	108	107	106	105	104	103	102	101	100	99	98	97	96	95	94	93	92	91	90
	Overflow	122	121	120	119	118	117	116	115	114	113	112	111	110	109	108	107	106	105	104	103	102	101	100	99	98	97	96	95	94	93	92	91
	Contents	Data																															
Octets 13-16	Underflow	88	87	86	85	84	83	82	81	80	79	78	77	76	75	74	73	72	71	70	69	68	67	66	65	64	63	62	61	60	59	58	57
	Normal	89	88	87	86	85	84	83	82	81	80	79	78	77	76	75	74	73	72	71	70	69	68	67	66	65	64	63	62	61	60	59	58
	Overflow	90	89	88	87	86	85	84	83	82	81	80	79	78	77	76	75	74	73	72	71	70	69	68	67	66	65	64	63	62	61	60	59
	Contents	Data																															
Octets 17-20	Underflow	56	55	54	53	52	51	50	49	48	47	46	45	44	43	42	41	40	39	38	37	36	35	34	33	32	31	30	29	28	27	26	25
	Normal	57	56	55	54	53	52	51	50	49	48	47	46	45	44	43	42	41	40	39	38	37	36	35	34	33	32	31	30	29	28	27	26
	Overflow	58	57	56	55	54	53	52	51	50	49	48	47	46	45	44	43	42	41	40	39	38	37	36	35	34	33	32	31	30	29	28	27
	Contents	Data																															
Octets 21-24	Underflow	24	23	22	21	20	19	18	17	16	15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	Z	Z						
	Normal	25	24	23	22	21	20	19	18	17	16	15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	Z	EP6	EP5	EP4	EP3	EP2	EP1
	Overflow	26	25	24	23	22	21	20	19	18	17	16	15	14	13	12	11	10	9	8	7	6	5	4	3	2	1						
	Contents	Data																										Error protection bits					

Z - Unused bit(s)

ATTACHMENT 3X (cont'd)

TABLE 30: TIF TRANSMISSION FRAME STRUCTURE 9.6kbit/s USER DATA-RATE (preamble frame)

	Bits -	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8
Octets 1- 4	Contents	S-1	S-2	S-3	S-4	S-5	S-6	S-6	S-8	S-9	S10	S11	S12	S13	S14	S15	S16	S17	S18	S19	S20	S21	S22	S23	S24	S25	S26	S27	S28	S29	S30	S31	Z
		Signalling Control Code																															
Octets 5- 8	Contents	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	BD8	BD7	BD6	BD5	BD4	BD3	BD2	BD1	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z
																Burst Error Detection Bits																	
Octets 9-12		Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z
Octets 13-16		Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z
Octets 17-20		Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z
Octets 21-24		Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z

Z - Unused bit(s)

ATTACHMENT 3 X (cont'd)

TABLE 31: TIF TRANSMISSION FRAME STRUCTURE 9.6kbit/s USER DATA-RATE (data frame)

	Bits -	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8
Octets 1- 4		192	191	190	189	188	187	186	185	184	183	182	181	180	179	178	177	176	175	174	173	172	171	170	169	168	167	166	165	164	163	162	161
	Contents	Data																															
Octets 5- 8		160	159	158	157	156	155	154	153	152	151	150	149	148	147	146	145	144	143	142	141	140	139	138	137	136	135	134	133	132	131	130	129
	Contents	Data (cont)																															
Octets 9-12		128	127	126	125	124	123	122	121	120	119	118	117	116	115	114	113	112	111	110	109	108	107	106	105	104	103	102	101	100	99	98	97
	Contents	Data (cont)																															
Octets 13-16		96	95	94	93	92	91	90	89	88	87	86	85	84	83	82	81	80	79	78	77	76	75	74	73	72	71	70	69	68	67	66	65
	Contents	Data (cont)																															
Octets 17-20		64	63	62	61	60	59	58	57	56	55	54	53	52	51	50	49	48	47	46	45	44	43	42	41	40	39	38	37	36	35	34	33
	Contents	Data (cont)																															
Octets 21-24		32	31	30	29	28	27	26	25	24	23	22	21	20	19	18	17	16	15	14	13	12	11	10	9	8	7	6	5	4	3	2	1
	Contents	Data (cont)																															

? - Unused bit(s)

ATTACHMENT 3X (cont'd)

TABLE 32: TIF TRANSMISSION FRAME STRUCTURE 9.6kbit/s USER DATA-RATE (postamble frame)

	Bits -	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8
Octets 1- 4		S-1	S-2	S-3	S-4	S-5	S-6	S-6	S-8	S-9	S10	S11	S12	S13	S14	S15	S16	S17	S18	S19	S20	S21	S22	S23	S24	S25	S26	S27	S28	S29	S30	S31	S-1
	Contents	Signalling Control Code [Copy 1]																															
Octets 5- 8		S-2	S-3	S-4	S-5	S-6	S-7	S-8	S-9	S10	S11	S12	S13	S14	S15	BD6	BD7	BD6	BD5	BD4	BD3	BD2	BD1	S16	S17	S18	S19	S20	S21	S22	S23	S24	S25
	Contents	Signalling Control Code [Copy 2]														Burst Error Detection Bits								Signalling Control Code [Copy 2] cont									
Octets 9-12		S26	S27	S28	S29	S30	S31	S-1	S-2	S-3	S-4	S-5	S-6	S-7	S-8	S-9	S10	S11	S12	S13	S14	S15	S16	S17	S18	S19	S20	S21	S22	S23	S24	S25	S26
	Contents							Signalling Control Code [Copy 3]																									
Octets 13-16		S27	S28	S29	S30	S31	S-1	S-2	S-3	S-4	S-5	S-6	S-7	S-8	S-9	S10	S11	S12	S13	S14	S15	S16	S17	S18	S19	S20	S21	S22	S23	S24	S25	S26	S27
	Contents						Signalling Control Code [Copy 4]																										
Octets 17-20		S28	S29	S30	S31	S-1	S-2	S-3	S-4	S-5	S-6	S-6	S-8	S-9	S10	S11	S12	S13	S14	S15	S16	S17	S18	S19	S20	S21	S22	S23	S24	S25	S26	S27	S28
	Contents					Signalling Control Code [Copy 5]																											
Octets 21-24		S29	S30	S31	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z	Z

Z - Unused bit(s)

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