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Appendix III (Rec. G.726)

Appendix II (Rec. G.727)

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

(05/94)

**GENERAL ASPECTS OF DIGITAL
TRANSMISSION SYSTEMS**

COMPARISON OF ADPCM ALGORITHMS

Appendix III to ITU-T Recommendation G.726
Appendix II to ITU-T Recommendation G.727

(Previously "CCITT Recommendation")



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FOREWORD

The ITU-T (Telecommunication Standardization Sector) is a permanent organ of the International Telecommunication Union (ITU). The ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Conference (WTSC), which meets every four years, establishes the topics for study by the ITU-T Study Groups which, in their turn, produce Recommendations on these topics.

The approval of Recommendations by the Members of the ITU-T is covered by the procedure laid down in WTSC Resolution No. 1 (Helsinki, March 1-12, 1993).

Appendix III to ITU-T Recommendation G.726 and Appendix II to Recommendation ITU-T G.727 were prepared by ITU-T Study Group 15 (1993-1996) and were approved on the 16 of May 1994.

NOTE

In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

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Appendix II to ITU-T Recommendation G.726 and Appendix II to ITU-T Recommendation G.727

COMPARISON OF ADPCM ALGORITHMS

(Geneva, 1994)

1 Background

During the period 1982-1990, the CCITT adopted several adaptive differential pulse code modulation (ADPCM) algorithms. First, the 32 kbit/s (ADPCM) algorithm described in Recommendation G.721 [26; 6] was approved. Later on, Recommendation G.721 was extended with Recommendation G.723 to 40 kbit/s to support voice band data modems at the rate of 9.6 kbit/s, and to 24 kbit/s to allow reduction of the bit rate in cases of network congestion [27]. Prior to the definition of Recommendation G.723, other ADPCM algorithms of performance similar to the 40 kbit/s algorithm had been incorporated in DCME designs and used in telecommunications networks. These algorithms, which may be considered by bilateral agreement, are described in COM XVIII-101 and COM XVIII-102 of the 1984-1988 study period.¹⁾ Finally, in July 1990, the CCITT combined Recommendations G.721 and G.723 and added operation at 16 kbit/s for overload situations. The combination resulted in a new Recommendation G.726. The CCITT also approved the embedded ADPCM algorithms of Recommendation G.727, which are extensions of the fixed rate ADPCM algorithms defined in Recommendation G.726.

This appendix presents a unified introduction to all these algorithms, their main features and their performance. Clause 2 gives an overview of all ADPCM algorithms that the CCITT has considered. Clause 3 reviews the principles of the algorithms of Recommendations G.726 and G.727 and COM XVIII-102. The principles of the algorithm of COM XVIII-101 are described in clause 4. The remaining clauses outline the main subjective and objective results for the performance of the various algorithms.

2 Overview of ADPCM algorithms

Figures 1 and 2 show a simplified block diagram of a G.726 encoder and decoder, respectively. Figures 3 and 4 show a simplified block diagram of a G.727 encoder and a decoder, respectively. In each set, the coder consists of a logarithmic-to-linear PCM converter, an adaptive quantizer, an inverse adaptive quantizer, and an adaptive predictor.

The PCM converter converts the A-law or μ -law PCM input signal $s(k)$ to a uniform PCM signal (k is the sampling index for a sampling period of 125 μ s). The predicted estimate of the input signal $s_e(k)$ is subtracted from the uniform PCM signal, $s_l(k)$, to yield a difference signal $d(k)$:

$$d(k) = s_l(k) - s_e(k) \quad (1)$$

The difference signal is then transformed into a logarithmic presentation with the base 2 and scaled by a scale factor $y(k)$ that is computed as described below.

In Recommendation G.726, the quantizer used is a 31-, 15-, 7- or 4-level non-uniform adaptive quantizer that stops adapting in the presence of a stationary input. This enhances the performance for voiceband data signals. In Recommendation G.727, the adaptive quantizer has 32, 16, 8 or 4 levels. Either quantizer codes the signal $d(k)$ into $I(k)$, a code word of 5, 4, 3 or 2 bits respectively, with one bit always for the sign.

¹⁾ The US patent that describes the final algorithm has some differences from the algorithm described in COM XVIII-102 [33].

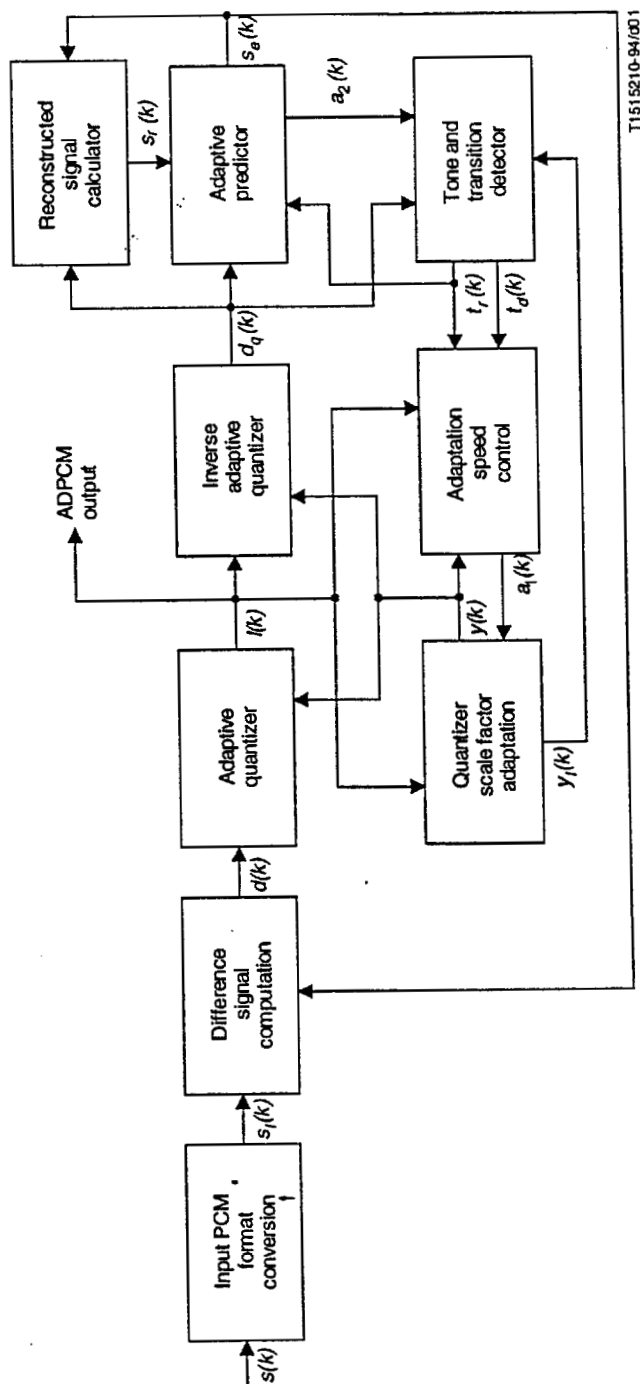
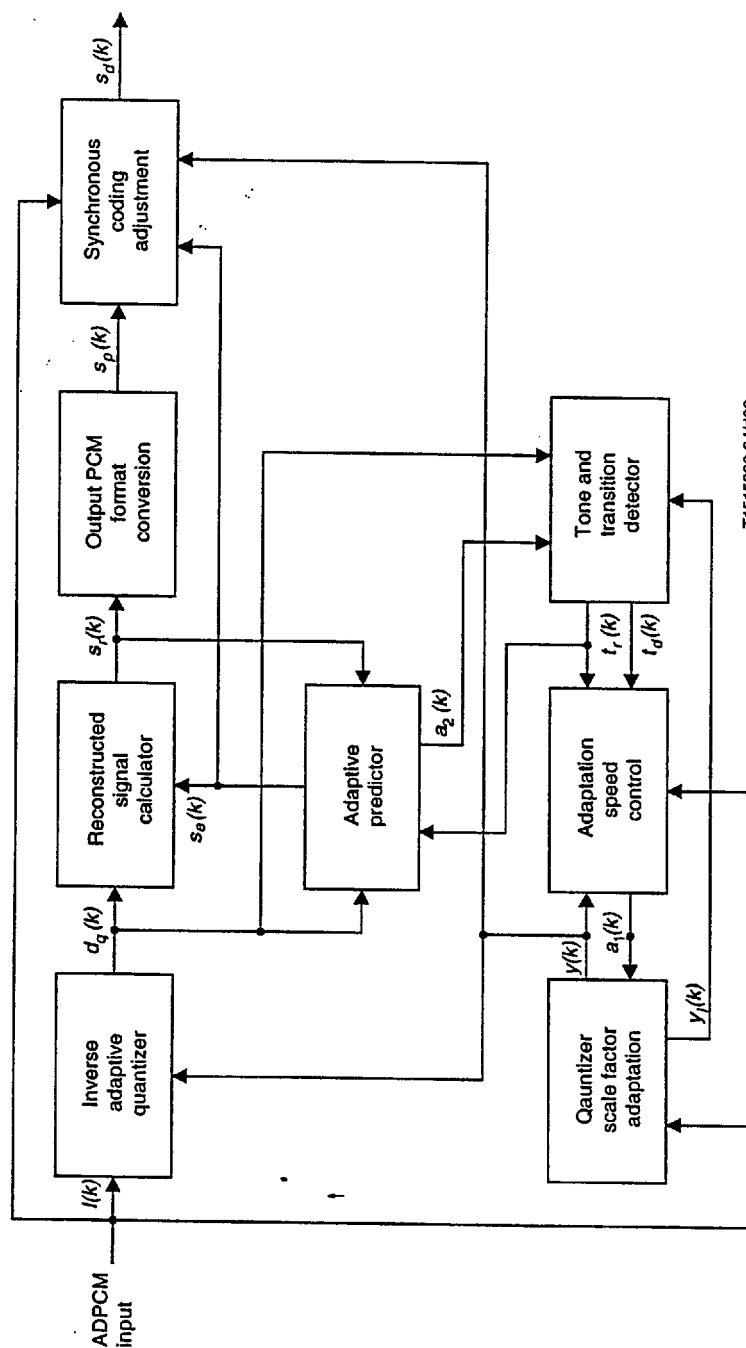
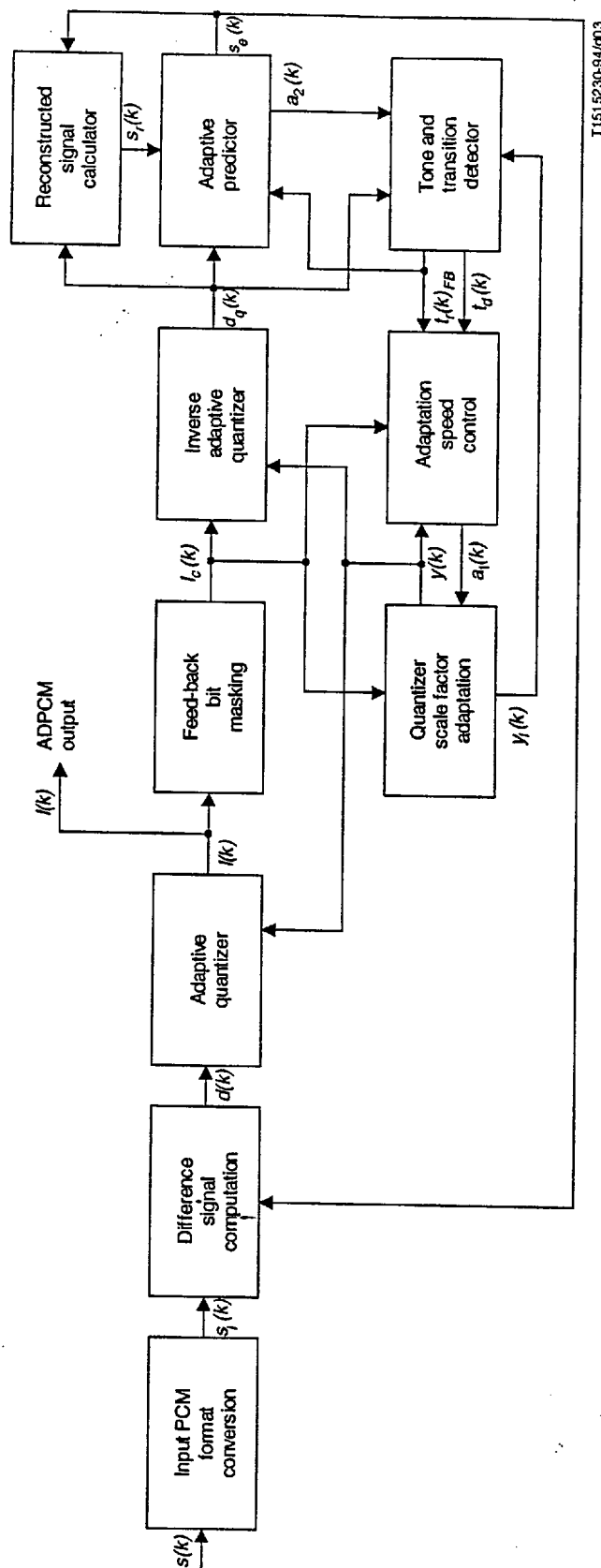


FIGURE 1
G.726 Encoder Block Schematic



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FIGURE 2
G.726 Decoder Block Schematic



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FIGURE 3
G.727 Encoder Block Schematic



FIGURE 4

The main difference between the fixed ADPCM algorithms of Recommendation G.726, COM XVIII-101 and COM XVIII-102 on the one side, and the embedded algorithms of Recommendation G.727 on the other side is as follows. In the embedded algorithms of Recommendation G.727, the difference between the input and the estimated signal is quantized into code words consisting of enhancement bits and core bits. The core bits are used for prediction, both in the encoder and the decoder, while the enhancement bits are used to reduce the quantization noise in the reconstructed signal. Thus, the core bits must reach the decoder to avoid mistracking, but the enhancement bits can be discarded, if such bit-dropping can alleviate congestion.

The embedded ADPCM algorithms can operate with 5-, 4-, 3- and 2-bits per sample in their feedback path (i.e. at rates of 40, 32, 24 and 16 kbit/s) with conversion to and from 64 kbit/s A-law or μ -law PCM channels. These G.727 algorithms are referenced as (x,y) pairs where x refers to the feed-forward ADPCM bits and y refers to the feedback ADPCM bits. For example, (5,2) represents 40 kbit/s embedded algorithm with two core bits, i.e. with a minimum bit rate is 16 kbit/s.

The impetus for developing Recommendation G.727 was to provide a flexible way to alleviate congestion at any point in a packet network without the need for exchanging control messages between the various nodes in the backward path of the connection (i.e. towards the transmitter). This avoids the "freeze-out" associated with fixed rate ADPCM coding, when transmission capacity is not available and the leading edge of speech bursts are clipped (i.e. the beginnings of words are chopped). This is important when the end-to-end path includes multiple nodes.

The difference between the various fixed rate ADPCM algorithms resides in the way that they accommodate 9.6 kbit/s modem signals. In Recommendation G.726, 40 kbit/s ADPCM is used for voiceband data, while 32 kbit/s ADPCM is used for speech. Accordingly, a bypass arrangement is needed so that upon detection of voiceband data, the appropriate coding is applied without affecting the coding for speech [9].

COM XVIII-101 uses the same 32 kbit/s ADPCM algorithm for speech as well as voiceband data [37]. The predictive structure, which is different from that of all other ADPCM algorithms, is composed of a 10th order adaptive zero predictor, a 4th order adaptive-pole predictor, a 16th-order fixed-pole predictor and an offset predictor. The adaptive pole filter is reserved for highly correlated signals such as speech, while the fixed pole filter is for voiceband data. The relative contribution of each filter is regulated by a set of adaptive gain coefficients. By controlling three different filters within the same structure, the algorithm treats speech signals and voiceband data modem signals up to 9.6 kbit/s with 32 kbit/s ADPCM; it does not require a change-over between speech and voiceband data at 9.6 kbit/s; the price is additional complexity. The adaptive quantizer of this algorithm operates in a 4-bit quantization mode and does not use a tone and transition detector. Figures 5 and 6 give the block diagrams of the encoder and decoder of the COM XVIII-101 algorithm.

COM XVIII-102 uses a special 32 kbit/s ADPCM algorithm that uses 5 bits/sample and is optimized for voiceband data [33; 53]. Following detection of a 2100 Hz tone, the linear PCM bit stream is down-sampled from 8 kHz to 6.4 kHz through a 100-tap symmetric finite impulse response interpolating filter. This interpolating filter introduces a flat delay of 6 ms equally distributed between the encoder and the decoder. To maintain the overall line rate of 32 kbit/s, the ADPCM coding uses 5 bits. To avoid aliasing, the input's bandwidth must be limited to 3.2 kHz. Also, a realignment from a 6.4 kHz $\times$ 5 structure to an 8 kHz $\times$ 4 structure is required. The corresponding encoder and decoder block diagrams are shown in Figures 7 and 8, respectively. In these figures, the tone detector block is assumed, because it is not described in the available documents from the algorithm developers [33; 53].

The adaptive predictor relies on the whole codeword $I(k)$ for Recommendation G.726 and the fixed rate ADPCM algorithms, and on the core codeword $I_c(k)$ for Recommendation G.727. The inverse quantizer uses the whole codeword $I_c(k)$ for fixed rate ADPCM (i.e. for Recommendation G.726 and COM XVIII-101, COM XVIII-102), and $I_c(k)$ for embedded ADPCM. Inverse quantization yields the quantized difference signal, $d_q(k)$. The signal estimate from adaptive prediction, $s_e(k)$, is added to this quantized difference signal to yield the reconstructed version, $s_r(k)$, of the input signal.

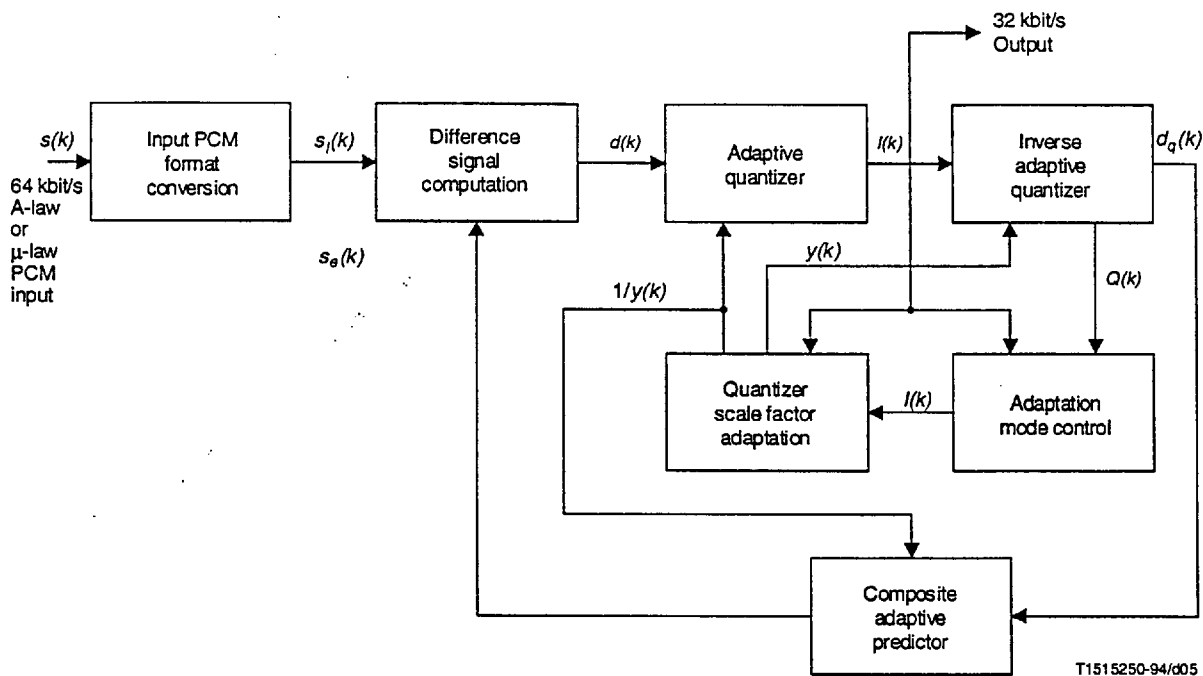


FIGURE 5
COM XVIII-101 Encoder Block Schematic

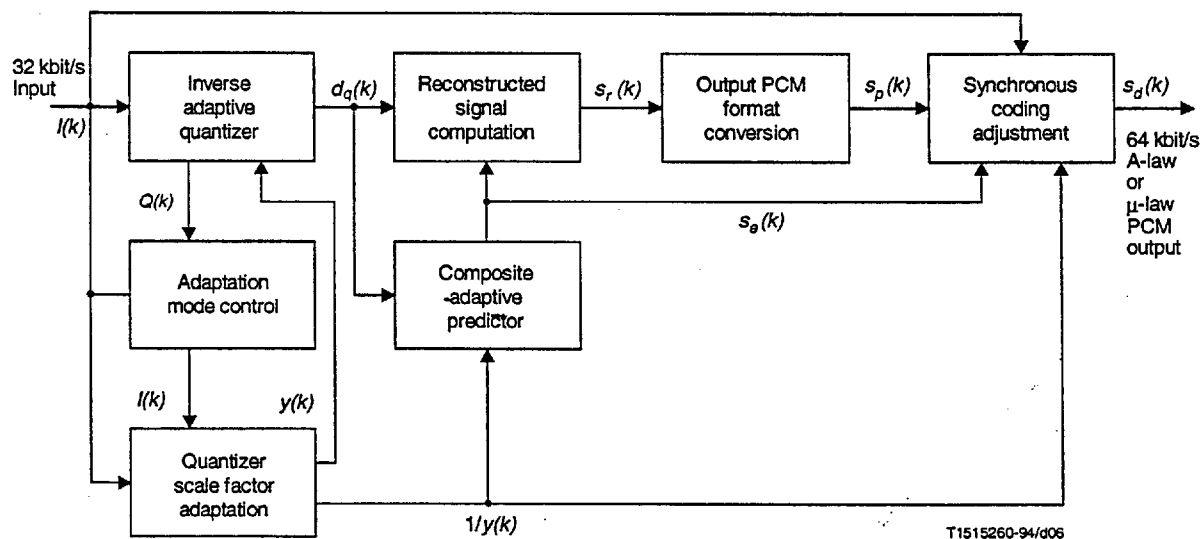
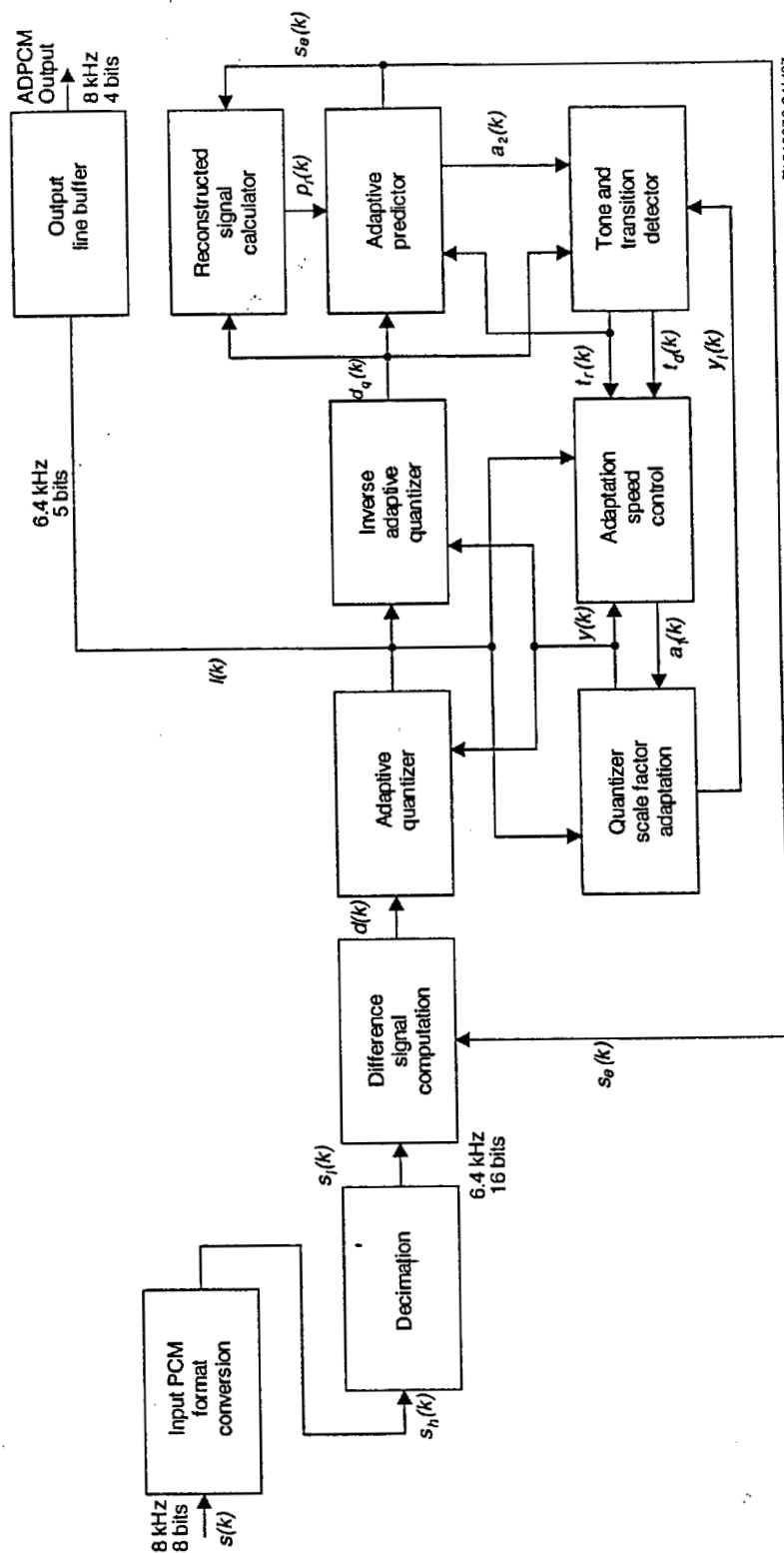


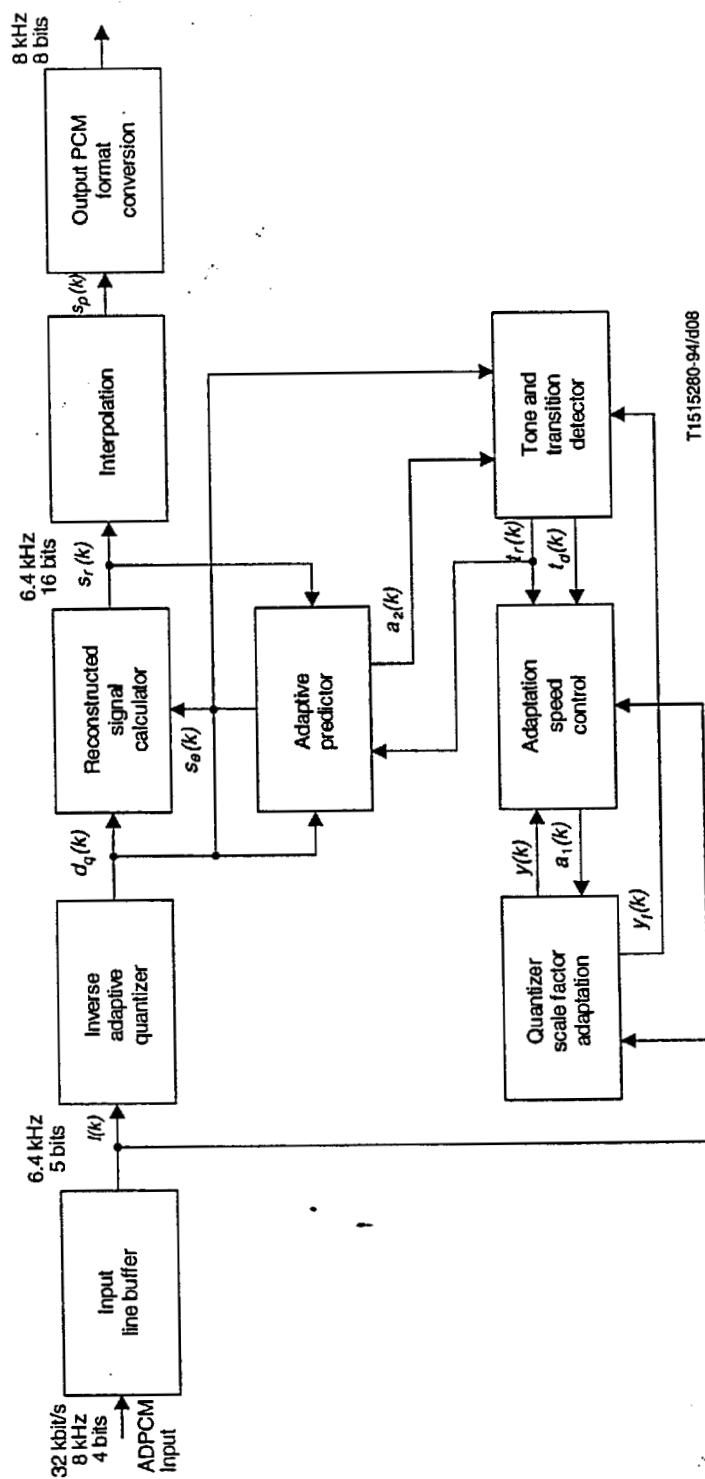
FIGURE 6
COM XVIII-101 Decoder Block Schematic



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NOTE - The tone detector block is assumed; it is neither described in [33] nor in [53].

FIGURE 7
COM XVIII-102 Encoder Block Schematic



NOTE – The tone detector block is assumed; it is neither described in [33] nor in [53].

FIGURE 8
COM XVIII-102 Decoder Block Schematic

The adaptation of both the quantizer and inverse quantizer depends on all the bits of $I(k)$ in the fixed rate algorithms, and on the core bits, $I_c(k)$ in Recommendation G.727. The quantizer output $I(k)$ takes all the corresponding non-zero values, while the adaptation operates on all possible values since the associated all-zeros codeword may result from transmission errors.

The decoder at both the receiver and the transmitter includes a structure identical to the feedback portion of the encoder. In addition, the receiver for both Recommendations G.726 and G.727 transcodes the bit stream from uniform PCM back to A-law or μ -law PCM for synchronous coding adjustment. This operation is based on all the received bits and prevents cumulative distortion on synchronous tandem codings (ADPCM-PCM-ADPCM) under certain conditions. This is achieved by selecting the PCM output codes that eliminate the quantizing distortion in the next ADPCM encoding stage [38].

The algorithm of COM XVIII-102 does not have a synchronous coding adjustment block as in Recommendations G.726, G.727 and COM XVIII-101; therefore, it does not exhibit the synchronous tandem property described in 5.2.

Clause 5 recapitulates the encoder principles and explains the differences among the various algorithms of Recommendations G.726 and G.727 and COM XVIII-102. Discussion of the algorithm of COM XVIII-101, whose structure is different from the other algorithms, is the subject of clause 6.

3 Principles of Recommendations G.726 and G.727 and COM XVIII-102

3.1 Adaptive Prediction and Reconstruction of the Signal

The primary function of the adaptive predictor is to compute the signal estimate $s_e(k)$ from the quantized difference signal $d_q(k)$ and past values of the reconstructed signal $s_r(k)$. The predictor has the form of an autoregressive moving average (ARMA) filter whose frequency spectrum fits a wide range of input voice-band signals.

The signal estimate is computed from the reconstructed signal $s_r(k)$ as:

$$s_e(k) = \sum_{i=1}^2 a_i^{k-1} s_r(k-i) + \sum_{i=1}^6 b_i^{k-1} d_q(k-i) \quad (2)$$

with

$$s_r(k) = s_e(k) + d_q(k) \quad (3)$$

and

$$d_q(k) = d(k) + e(k) = Q \left[d \frac{(k)}{y} (k) \right] y(k) \quad (4)$$

where

a_1^k, a_2^k are the autoregressive coefficients for sample k ;

$b_i^k, i = 1, \dots, 6$ the moving average coefficients for sample k ;

$d(k)$ is the *difference signal at sample k* ;

$d_q(k)$ is the quantized difference signal at sample k ;

$e(k)$ is the quantization error at sample k ;

$Q[x]$ is the normalized quantizer output for the input x ;

$= |x| - y(k)$ in the logarithmic domain;

$y(k)$ is the the scale factor error at sample k .

The starting values are: $d(0) = s_o(0) = s_r(0) = 0$ and $d_q(k) = 0$ for $k < 0$. Equations (2)-(4) are similar for both Recommendations G.726 and G.727 with the exception that the quantization error $e(k)$ in Recommendation G.727 includes the effects of using $I_c(k)$ instead of $I(k)$. As explained above, $d_q(k)$ was obtained from $I_c(k)$ through inverse quantization.

With B = delay operator such that $Bd_q(k) \triangleq d_q(k-1)$, we can combine (2) and (3):

$$s_e(k) = \frac{\left(a_1^{k-1} + b_1^{k-1}\right) B + \left(a_2^{k-1} + b_2^{k-1}\right) B^2 + \dots + b_6 B^6}{1 - a_1^{k-1} B - a_2^{k-1} B^2} \cdot d_q(k) \quad (5)$$

From (1), (3) and (5) we can write:

$$\left(1 + \sum_{i=1}^6 b_i^{k-1} B^i\right) d(k) = \left(1 - a_1^{k-1} B - a_2^{k-1} B^2\right) s_l(k) + \left[\left(a_1^{k-1} + b_1^{k-1}\right) B + \left(a_2^{k-1} + b_2^{k-1}\right) B^2 + \dots + b_6 B^6\right] e(k) \quad (6)$$

The update equations for the autoregressive and moving average coefficients of the ARMA (2,6) predictor follow the same simplified gradient algorithm used in Recommendation G.721 [39; 45].

$$a_1^k = (1 - 2^{-8}) a_1^{k-1} + (3 \cdot 2^{-8}) \text{sgn}[p(k)] \text{sgn}[p(k-1)] \quad (7)$$

$$a_2^k = (1 - 2^{-7}) a_2^{k-1} + 2^{-7} \left[\text{sgn}[p(k)] \text{sgn}[p(k-2)] - f(a_1^{k-1}) \text{sgn}[p(k)] \text{sgn}[p(k-1)] \right] \quad (8)$$

where $p(k)$ is an MA (6) process defined as:

$$p(k) = d_q(k) + s_{ez}(k) = \left(1 + b_1^{k-1} B + \dots + b_6^{k-1} B^6\right) d_q(k) \quad (9)$$

and

$$f(a_1) = \begin{cases} 4a_1, & |a_1| \leq 2^{-1} \\ 2 \text{sgn}(a_1), & |a_1| > 2^{-1} \end{cases}$$

From equation (5), the frequency response of the predictor is given by:

$$H(w) = \frac{\left(a_1^{k-1} + b_1^{k-1}\right) e^{-j\omega T} + \left(a_2^{k-1} + b_2^{k-1}\right) e^{-j2\omega T} + \dots + b_6^{k-1} e^{-j6\omega T}}{1 - a_1^{k-1} e^{-j\omega T} - a_2^{k-1} e^{-j2\omega T}} \quad (10)$$

$$j = \sqrt{-1}, 0 \leq \omega \leq \frac{\pi}{T} \text{ and } T = 125 \mu s$$

