NOTE: The cover page of this standard has been changed for administrative reasons. There are no other changes to this document.

MIL-STD-188-113
17 FEBRUARY 1987

DEPARTMENT OF DEFENSE
INTERFACE STANDARD

INTEROPERABILITY AND PERFORMANCE STANDARDS FOR ANOLOG-TO-DIGITAL CONVERSION TECHNIQUES

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DEPARTMENT OF DEFENSE

WASHINGTON, DC 20301

Interoperability and Performance Standards for Analog-to-Digital Conversion Techniques

MIL-STD-188-113

1. This Military Standard is approved and mandatory for use by all Departments and Agencies of the Department of Defense in accordance with the OUSDR&E Memorandum, dated 16 August 1983 (see appendix A).

2. Beneficial comments (recommendations, additions, deletions) and any pertinent data which may be of use in improving this document should be addressed to: Director, U.S. Army Information Systems Engineering and Integration Center, ATTN: ASBI-SST, Fort Huachuca, Arizona 85613-7300 by using the self-addressed Standardization Document Improvement Proposal (DD Form 1426) appearing at the end of this document, or by letter.

FOREWORD

1. Originally, Military Standard 188 (MIL-STD-188) covered technical standards for tactical and long-haul communications, but later evolved through revisions (MIL-STD-188A, MIL-STD-188B) into a document applicable to tactical communications only (MIL-STD-188C).

2. The Defense Communications Agency (DCA) published DCA Circulars (DCAC) promulgating standards and engineering criteria applicable to the long-haul Defense Communications System (DCS) and to the technical support of the National Military Command System (NMCS).

3. As a result of a Joint Chiefs of Staff (JCS) action, standards for all military communications are now being published in a MIL-STD-188 series of documents. The MIL-STD-188 series is subdivided into a MIL-STD-188-100 series covering common standards for tactical and long-haul communications, a MIL-STD-188-200 series covering standards for tactical communications only, and a MIL-STD-188-300 series covering standards for long-haul communications only. Emphasis is being placed on developing common standards for tactical and long-haul communications published in the MIL-STD-188-100 series.

4. This document contains technical standards and design objectives for analog-to-digital (A-D) conversion techniques to be used over both long-haul and tactical communications systems.

ACKNOWLEDGEMENT

The information in appendix F on adaptive predictive coding (APC) has been adapted from the article "The Government Standard Adaptive Predictive Coding Algorithm: APC-04" by Thomas Tremain, published in Speech Technology Magazine (February/March 1985, pages 52-62). Speech Technology Magazine is produced by Media Dimensions, Inc., 42 East 23rd Street, New York, NY 10010, and their permission to use this article is gratefully acknowledged. Any further reproduction or transmission of this material requires permission of Media Dimensions, Inc.

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1. SCOPE.

1.1 **Purpose.** The purpose of this document is to establish interoperability and performance standards for analog-to-digital (A-D) conversion techniques to be used over both long-haul (nontactical) and tactical communications systems. The standards promulgated by this document represent, in general, minimum interoperability and performance characteristics which may be exceeded in order to satisfy specific requirements.

1.2 **Content.** This standard provides electrical performance parameters and requirements for the following types of A-D conversion methods:

a. 64 kbps pulse code modulation (PCM)

b. 16 and 32 kbps continuously variable slope delta (CVSD) modulation

c. 2.4 kbps linear predictive coding (LPC)

d. 9.6 kbps adaptive predictive coding (APC)

e. 32 kbps adaptive differential pulse code modulation (ADPCM)

1.3 **Application.** This standard applies to A-D conversion techniques for use within the Department of Defense (DoD). It is to be used in the design and installation of new communications subsystems and equipment and in authorized upgrading of existing communications subsystems and equipment. However, this standard is not intended to restrict either technical advances in A-D conversion performance or the use of new A-D conversion techniques.

2. REFERENCED DOCUMENTS

2.1 Government documents.

2.1.1 Specifications, standards, and handbooks. Unless otherwise specified, the following specifications, standards, and handbooks of the issue listed in that issue of the Department of Defense Index of Specifications and Standards (DODISS) specified in the solicitation form a part of this standard to the extent specified herein.

STANDARDS

FEDERAL

FED-STD-1015	Telecommunications: Analog to Digital Conversion of Voice by 2,400 Bit/Second Linear Predictive Coding
FED-STD-1037	Glossary of Telecommunication Terms

MILITARY

MIL-STD-188-200	System Design and Engineering Standards for Tactical Communications

(Copies of specifications, standards, handbooks, drawings, and publications required by contractors in connection with specific acquisition functions should be obtained from the contracting activity or as directed by the contracting officer.)

2.2 Other publications. The following document(s) form a part of this standard to the extent specified herein. Unless otherwise specified, the issues of the documents which are DoD adopted shall be those listed in the issue of the DODISS specified in the solicitation. The issues of documents which have not been adopted shall be those in effect on the date of the cited DODISS.

NORTH ATLANTIC TREATY ORGANIZATION (NATO)
STANDARDIZATION AGREEMENTS (STANAG's)

STANAG 4198	Parameters and Coding Characteristics That Must be Common to Assure Interoperability of 2400 BPS Linear Predictive Encoded Digital Speech
STANAG 4209	The NATO Multi-Channel Tactical Digital Gateway — Standards for Analogue to Digital Conversion of Speech Signals

(Application for copies should be addressed to the Naval Publications and Forms Center, 5801 Tabor Avenue, Philadelphia, PA 19120.)

THE INTERNATIONAL TELEGRAPH AND TELEPHONE CONSULTATIVE COMMITTEE (CCITT)

CCITT Red Book, Vol. III — Fascicle III.3, Recommendation G.711: Pulse Code Modulation (PCM) of Voice Frequencies

(Application for copies should be addressed to the International Telecommunication Union, Place des Nations, CH-1211, Geneva 20, Switzerland.)

(Non-Government standards are generally available for reference from libraries. They are also distributed among non-Government standards bodies and using Federal agencies.)

2.3 **Order of precedence.** In the event of a conflict between the text of this standard and the references cited herein, the text of this standard shall take precedence.

3. DEFINITIONS

3.1 **Terms.** Definitions of terms used in this document shall be as specified in the current edition of FED-STD-1037. In addition, the following definitions are applicable for the purposes of this standard.

a. **Compression ratio.** The maximum slope voltage divided by the minimum slope voltage, measured at the output of a syllabic filter in a CVSD converter.

NOTE: Compression ratio can also be defined as the ratio of the maximum step size to the minimum step size at the output of the pulse amplitude modulator as in STANAG 4209.

b. **Duty cycle (c_d).** The mean proportion of binary '1' digits at the algorithm output where each '1' indicates a run of three consecutive bits of the same polarity in a CVSD digital signal.

c. **Exclusive OR.** Binary logic in which the output is true if either one of two input variables is true, but not if both input variables are true.

d. **Frame quantization level (Q).** In APC/SQ, the average magnitude of the residual error signal during a 25-ms frame; the frame energy used (in conjunction with the SEGQ's) to scale the one-bit-per-sample "residual error" transmitted to the receiver.

e. **Gray coded.** An ordering of the numbers from 0 to 2^n-1, where n is the number of binary digits in each number (including leading 0's) in such a way that successive numbers (and the last and first number) differ from each other by only one binary digit. The particular sequence used in the APC/SQ to encode ALPHA (n=3) is 0-1-3-2-6-7-5-4.

f. **Percent run-of-threes.** The proportion of groups of three like bits (all zeros or all ones) to the total number of bits in a CVSD digital signal. The number of groups of threes in a sequence of like bits is defined as n−2, where n is the number of like bits. For repeating sequences, the last bit of the sequence (or last two bits if alike) must be considered with the first bit of the sequence (or first two bits if alike) in determining the groups of threes.

g. **Pitch period value (TAU).** In voiced or partially voiced sounds, the time interval between air puffs emitted by relaxation of the vocal cords. In LPC and APC/SQ, the time interval is measured in units of sample interval (125 μs per sample for the LPC voice processor; 1/7600 s per sample for the APC/SQ voice processor).

h. **Pitch weighting factor (ALPHA).** A pitch gain factor in APC/SQ which represents the gain of the pitch prediction filter; a factor representing the strength of the fundamental (pitch) frequency and its harmonics during a pitch period relative to the strength of the fundamental frequency and its harmonics during the previous pitch period.

i. **Prediction coefficient.** The translated reflection coefficient which is used as a filter weight in APC and LPC synthesis.

NOTE: Reflection coefficients are more suitable for transmission than prediction coefficients. Reflection coefficients, having magnitudes less than one, result in smaller quantization errors when the coefficients are coded.

j. **Reflection coefficient.** In acoustics, the ratio between the forward and backward traveling components of pressure (p) and volume velocity (u) at a junction between contiguous lossless acoustic tubes of cross-sectional area A_i ($i = 1, 2 \ldots n$), where the tubes approximate a nonnasalized human vocal tract.

NOTE: Plane-wave acoustic propagation and optical or electrical propagation along a transmission line are analogous. Acoustic volume velocity (u) and pressure (p) are analogous to electrical current (I) and voltage (V), respectively.

k. **Segment quantization level (SEGQ).** In APC/SQ, the magnitude of the residual error signal during a segment (one tenth of a frame); the segment energy. Also, the factor by which the Q is multiplied to obtain the magnitude of the residual error signal during a segment. SEGQ is calculated in the transmitter as the ratio of the average magnitude of segment energy to the average frame energy.

l. **(21, 16) code.** An error correction code having 21 bits, of which 16 are data bits and five are error correction (check) bits.

3.2 **Abbreviations and acronyms.** The abbreviations and acronyms used in this document are defined below. Those that are common with the terms in the current edition of FED-STD-1037 have been included for the convenience of the reader.

A-D	analog-to-digital
ADPCM	adaptive differential pulse code modulation
ALP	designator for ALPHA bits
ALPHA	pitch weighting factor
AMDF	average-magnitude difference function
APC	adaptive predictive coding
APC/SQ	adaptive predictive coding with segmented quantization
bps	bits per second
c_d	duty cycle
CKT	designator for check bits
CVSD	continuously variable slope delta (modulation)
D-A	digital-to-analog
dB	decibel(s)
dBm0	power in dBm referred to or measured at 0TLP
dc	direct current
DRT	Diagnostic Rhyme Test
E	designator for error signal bits
EDL	efficiently digitally linearizable (coding characteristic)
ERR 1	first error signal from pitch-prediction filter
ERR 2	residual error signal
Hz	hertz
kbps	kilobits per second
LPC	linear predictive coding
LSB	least significant bit
MLA	modulation level analyzer
MSB	most significant bit
NATO	North Atlantic Treaty Organization
PAM	pulse amplitude modulation
PC	prediction coefficient
PCM	pulse code modulation
Q	frame quantization level
RC	reflection coefficient
SEGQ	segment quantization level
SEGQC	segment quantization level, coded
SQC	designator for segment Q bits
STANAG	Standardization Agreement (NATO)
TAU	pitch period value
XOR	exclusive OR
Z	transfer function

4. GENERAL REQUIREMENTS. Not applicable.

5. DETAILED REQUIREMENTS

5.1 **Eight-bit pulse code modulation (PCM).**

5.1.1 **General.** PCM is that form of modulation in which the input signal is sampled, quantized into discrete amplitude levels, and coded prior to transmission. For purposes of this document, a PCM converter consists of the input filter, sampler, encoder, decoder, and output filter.

5.1.2 **Reference level.** The input signal level which fully modulates the compressor/encoder over its full range of 256 levels shall be the reference level for the PCM requirements of this standard, and shall be designated 0 dBm0. The input signal shall be a 1000 Hz ± 25 Hz, preferably 1004 Hz, sinewave. Full modulation of the compressor/encoder is attained when the first occurrence of the inverted binary code corresponding to the most positive quantization interval (7903 to 8159) and the most negative quantization interval (−7903 to −8159) is noted.

5.1.3 **PCM characteristics.**

5.1.3.1 **Input and output impedances.** The analog input and output impedances for PCM converters are not standardized. These impedances depend upon the application of the converter.

5.1.3.2 **Data signaling rate.** The PCM converter shall be capable of operating at 64 kbps per channel.

5.1.3.3 **Input and output filters.** The analog input shall be bandpass filtered. The analog output shall be lowpass filtered. The attenuation provided by the input filter shall ensure that any fundamental power frequency component in the converter input is attenuated at least 15 dB relative to the insertion loss at 1000 Hz.

> NOTE: Details of input and output filters, consistent with the PCM performance requirements of this standard, will be determined in applicable equipment specifications based on validated requirements. Appendix B (Table XVIII) shows a typical input filter response. This response is for the input filter used in telephone T-1 PCM carrier equipment.

5.1.3.4 **Analog-to-digital (A-D) conversion.**

5.1.3.4.1 **Sampling rate.** The analog signal shall be amplitude sampled at a rate of 8000 samples per second.

5.1.3.4.2 **Format.** Each pulse amplitude modulated (PAM) sample shall be encoded into an 8-bit word which represents the quantized amplitude of the sample. The eight bits shall be assigned as follows: bit one shall be the polarity bit; bits two, three, and four shall be used to represent the segment; and bits five, six, seven, and eight shall indicate the level in the segment.

5.1.3.4.3 **Compressor/encoder characteristics.** The compressor/encoder shall have a 15-segment characteristic with the input step size of each segment twice that of the segment next closest to the center of the range. This segmented characteristic is a piece-wise linear approximation of a $\mu = 255$ companding law. This type of compressing/encoding is known as efficiently digitally linearizable (EDL) coding. Characteristics of EDL coding are given in appendix B.

5.1.3.4.4 **Code type and levels.** An 8-bit binary PCM code shall be used with the most significant bit (MSB) transmitted first. The input PAM signal shall be quantized into one of 256 possible amplitude levels. These 256 levels shall correspond to a relative input amplitude range of -8159 to $+8159$. Each quantization interval shall be assigned a specific 8-bit PCM code which shall be decoded as the midpoint of the interval. The digital output of the converter shall be in inverted binary form. The all-zero word which represents the most negative voltage shall be assigned the code 00000010. Tables I and II give the code and decode values for relative input amplitudes of 0 to $+8159$, and 0 to -8159, respectively.

5.1.3.4.5 **Code level stability.** The nominal code level corresponding to zero modulation (input terminated) shall statistically be midway between the least negative code level (inverted code 01111111) and the least positive code level (inverted code 11111111) and shall not vary more than ± 7 levels under all operating conditions.

5.1.3.5 **Digital-to-analog (D-A) conversion.** The expander/decoder shall convert the received PCM signal to an analog signal. The expander/decoder shall have the complementary transfer characteristics of the compressor/encoder.

5.1.3.6 **PCM converter performance.** The characteristics specified in paragraphs 5.1.3.6.1 through 5.1.3.6.6 apply when the output of a compressor/encoder is connected to the input of an expander/decoder.

> NOTE: Test signal frequencies which are exactly rational fractions of (subharmonically related to) the 8000-Hz PCM sampling rate, such as 800 Hz, 1000 Hz, and 1600 Hz, can cause nonlinear PCM distortion and, therefore, should not be used. This nonlinear PCM distortion can be avoided by changing the 1000-Hz test frequency by a few hertz.

TABLE I. Code/decode table for positive input values for PCM.

Relative Input Amplitude Interval (x) (PAM Sample)	Code (Inverted Binary)	Decode (Midpoint of Interval)
$7903 \leq x < 8159$	10000000	8031
$7647 \leq x < 7903$	10000001	7775
$7391 \leq x < 7647$	10000010	7519
$7135 \leq x < 7391$	10000011	7263
$6879 \leq x < 7135$	10000100	7007
$6623 \leq x < 6879$	10000101	6751
$6367 \leq x < 6623$	10000110	6495
$6111 \leq x < 6367$	10000111	6239
$5855 \leq x < 6111$	10001000	5983
$5599 \leq x < 5855$	10001001	5727
$5343 \leq x < 5599$	10001010	5471
$5087 \leq x < 5343$	10001011	5215
$4831 \leq x < 5087$	10001100	4959
$4575 \leq x < 4831$	10001101	4703
$4319 \leq x < 4575$	10001110	4447
$4063 \leq x < 4319$	10001111	4191
$3935 \leq x < 4063$	10010000	3999
$3807 \leq x < 3935$	10010001	3871
$3679 \leq x < 3807$	10010010	3743
$3551 \leq x < 3679$	10010011	3615
$3423 \leq x < 3551$	10010100	3487
$3295 \leq x < 3423$	10010101	3359
$3167 \leq x < 3295$	10010110	3231
$3039 \leq x < 3167$	10010111	3103
$2911 \leq x < 3039$	10011000	2975
$2783 \leq x < 2911$	10011001	2847
$2655 \leq x < 2783$	10011010	2719
$2527 \leq x < 2655$	10011011	2591
$2399 \leq x < 2527$	10011100	2463
$2271 \leq x < 2399$	10011101	2335
$2143 \leq x < 2271$	10011110	2207
$2015 \leq x < 2143$	10011111	2079
$1951 \leq x < 2015$	10100000	1983
$1887 \leq x < 1951$	10100001	1919
$1823 \leq x < 1887$	10100010	1855
$1759 \leq x < 1823$	10100011	1791
$1695 \leq x < 1759$	10100100	1727
$1631 \leq x < 1695$	10100101	1663

TABLE I. Code/decode table for positive input values for PCM.-Continued

Relative Input Amplitude Interval (x) (PAM Sample)	Code (Inverted Binary)	Decode (Midpoint of Interval)
1567 ≤ x < 1631	10100110	1599
1503 ≤ x < 1567	10100111	1535
1439 ≤ x < 1503	10101000	1471
1375 ≤ x < 1439	10101001	1407
1311 ≤ x < 1375	10101010	1343
1247 ≤ x < 1311	10101011	1279
1183 ≤ x < 1247	10101100	1215
1119 ≤ x < 1183	10101101	1151
1055 ≤ x < 1119	10101110	1087
991 ≤ x < 1055	10101111	1023
959 ≤ x < 991	10110000	975
927 ≤ x < 959	10110001	943
895 ≤ x < 927	10110010	911
863 ≤ x < 895	10110011	879
831 ≤ x < 863	10110100	847
799 ≤ x < 831	10110101	815
767 ≤ x < 799	10110110	783
735 ≤ x < 767	10110111	751
703 ≤ x < 735	10111000	719
671 ≤ x < 703	10111001	687
639 ≤ x < 671	10111010	655
607 ≤ x < 639	10111011	623
575 ≤ x < 607	10111100	591
543 ≤ x < 575	10111101	559
511 ≤ x < 543	10111110	527
479 ≤ x < 511	10111111	495
463 ≤ x < 479	11000000	471
447 ≤ x < 463	11000001	455
431 ≤ x < 447	11000010	439
415 ≤ x < 431	11000011	423
399 ≤ x < 415	11000100	407
383 ≤ x < 399	11000101	391
367 ≤ x < 383	11000110	375
351 ≤ x < 367	11000111	359
335 ≤ x < 351	11001000	343
319 ≤ x < 335	11001001	327
303 ≤ x < 319	11001010	311
287 ≤ x < 303	11001011	295

TABLE I. Code/decode table for positive input values for PCM.-Continued

Relative Input Amplitude Interval (x) (PAM Sample)	Code (Inverted Binary)	Decode (Midpoint of Interval)
$271 \leq x < 287$	11001100	279
$255 \leq x < 271$	11001101	263
$239 \leq x < 255$	11001110	247
$223 \leq x < 239$	11001111	231
$215 \leq x < 223$	11010000	219
$207 \leq x < 215$	11010001	211
$199 \leq x < 207$	11010010	203
$191 \leq x < 199$	11010011	195
$183 \leq x < 191$	11010100	187
$175 \leq x < 183$	11010101	179
$167 \leq x < 175$	11010110	171
$159 \leq x < 167$	11010111	163
$151 \leq x < 159$	11011000	155
$143 \leq x < 151$	11011001	147
$135 \leq x < 143$	11011010	139
$127 \leq x < 135$	11011011	131
$119 \leq x < 127$	11011100	123
$111 \leq x < 119$	11011101	115
$103 \leq x < 111$	11011110	107
$95 \leq x < 103$	11011111	99
$91 \leq x < 95$	11100000	93
$87 \leq x < 91$	11100001	89
$83 \leq x < 87$	11100010	85
$79 \leq x < 83$	11100011	81
$75 \leq x < 79$	11100100	77
$71 \leq x < 75$	11100101	73
$67 \leq x < 71$	11100110	69
$63 \leq x < 67$	11100111	65
$59 \leq x < 63$	11101000	61
$55 \leq x < 59$	11101001	57
$51 \leq x < 55$	11101010	53
$47 \leq x < 51$	11101011	49
$43 \leq x < 47$	11101100	45
$39 \leq x < 43$	11101101	41
$35 \leq x < 39$	11101110	37
$31 \leq x < 35$	11101111	33

TABLE I. Code/decode table for positive input values for PCM.-Continued

Relative Input Amplitude Interval (x) (PAM Sample)	Code (Inverted Binary)	Decode (Midpoint of Interval)
$29 \leq x < 31$	11110000	30
$27 \leq x < 29$	11110001	28
$25 \leq x < 27$	11110010	26
$23 \leq x < 25$	11110011	24
$21 \leq x < 23$	11110100	22
$19 \leq x < 21$	11110101	20
$17 \leq x < 19$	11110110	18
$15 \leq x < 17$	11110111	16
$13 \leq x < 15$	11111000	14
$11 \leq x < 13$	11111001	12
$9 \leq x < 11$	11111010	10
$7 \leq x < 9$	11111011	8
$5 \leq x < 7$	11111100	6
$3 \leq x < 5$	11111101	4
$1 \leq x < 3$	11111110	2
$0 \leq x < 1$	11111111	0

NOTES:
1. PAM samples are normalized to a full scale value of 8159.
2. The value of the decoder output for the first interval is zero and is referred to as "positive zero." The number of quantization intervals is 128 for positive inputs.

TABLE II. Code/decode table for negative input values for PCM.

Relative Input Amplitude Interval (x) (PAM Sample)	Code (Inverted Binary)	Decode (Midpoint of Interval)
$0 \geq x > -1$	01111111	0
$-1 \geq x > -3$	01111110	−2
Quantization interval endpoints for −3 to −7391 same as table I, except for sign.	Inverted binary code for quantization intervals same as table I, except for sign bit, which is zero (0) for negative input values.	Decode value for quantization intervals same as table I, except for sign.
$-7391 \geq x > -7647$	00000010	−7519
$-7647 \geq x > -7903$	00000001	−7775
$-7903 \geq x > -8159$	00000010	−7519

NOTES:
1. PAM samples are normalized to a full scale value of −8159.
2. The value of the decoder output for the first interval is zero and is referred to as a "negative zero."
3. The most negative quantization interval (−7903 to −8159) is assigned the same code as the interval for −7391 to −7647 to avoid an inverted code with no transitions (i.e., 00000000). This results in 127 effective quantization intervals for negative inputs. Using the inverted binary code 00000010 for the most negative interval is consistent with CCITT Recommendation G.711.

5.1.3.6.1 **Insertion loss vs. frequency characteristics.** The insertion loss relative to 1000 Hz measured with an input level of −3 dBm0 applied to the converter input shall not exceed the limits indicated in table III and shown in figure 1.

TABLE III. Insertion loss limits for PCM.

Frequency (f), Hz	Insertion loss, dB (Referenced to 1000 Hz)
$f < 300$	≥ -0.5
$300 \leq f < 3000$	−0.5 to +0.5
$3000 \leq f < 3400$	−0.5 to +3
$3400 \leq f$	≥ -0.5

5.1.3.6.2 **Idle channel noise.** The idle channel noise shall not exceed −70 dBm0 measured at the converter output.

5.1.3.6.3 **Input/output linearity.** The input versus output level characteristic of a PCM converter shall be linear within the limits shown in table IV. This measurement shall be made with a 1000-Hz ±25 Hz (preferably 1004-Hz) sine-wave test signal.

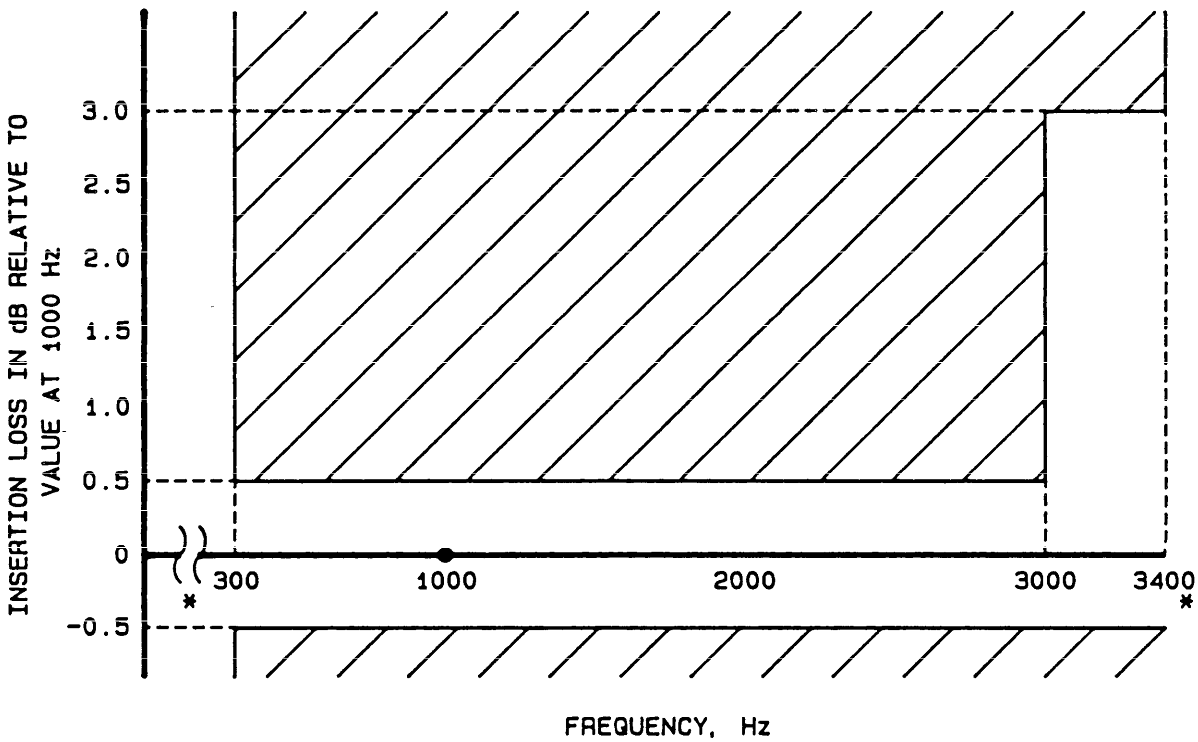
TABLE IV. Input/output linearity limits for PCM.

Input Signal, dBm0	Deviation, dB
0 thru −40	±0.5
−41 thru −53	±1.0
−54 thru −58	±3.0

5.1.3.6.4 **Signal to quantizing noise.** The total signal to quantizing noise ratio at any frequency between 300 Hz and 3400 Hz shall exceed the minimum values (C-message weighted) shown in table V. This measurement shall be made with a 1000-Hz ±25 Hz (preferably 1004-Hz) sine-wave test signal.

TABLE V. Signal to quantizing noise ratio for PCM.

Input Signal, dBm0	Test Signal to Quantizing Noise Ratio, dB
0 thru −33	33
−34 thru −43	27
−44 thru −48	22



NOTES:

1. The curve of the variations of insertion loss as a function of frequency should be normalized to pass through 0 dB at a frequency of 1000 Hz. The curve should then lie wholly between the upper and lower hatched lines.
2. The limits are for the insertion loss as measured between the analog input and analog output.
3. * These upper and lower frequency limits are not specified.

FIGURE 1. Insertion loss vs. frequency for 8-bit PCM.

5.1.3.6.5 **Nonlinear distortion.** The distortion within the band of 300 Hz to 3400 Hz shall provide at least a signal-to-second-order distortion of 35 dB and a signal-to-third-order distortion of at least 40 dB. The test signal input level shall be −13 dBm0. These requirements are based on use of the four-tone test method for determining nonlinear distortion levels. This method is described in appendix D.

5.1.3.6.6 **Envelope delay distortion.** The envelope delay distortion shall not exceed 400 microseconds in the frequency band between 500 Hz and 3000 Hz, and shall not exceed 175 microseconds in the frequency band between 1000 Hz and 2500 Hz.

5.2 **Continuously variable slope delta (CVSD) modulation.**

5.2.1 **General.** CVSD modulation is a nonlinear, sampled data, feedback system which accepts a band-limited analog signal and encodes it into binary form for transmission through a digital channel. At the receiver, the binary signal is decoded into a close approximation of the original analog signal. A typical CVSD converter consisting of an encoder and decoder is shown in figure 2.

> NOTE: Additional information on CVSD is presented in appendix C. NATO STANAG 4209 was used in developing the CVSD requirements in the following paragraphs. Some of the CVSD terminology in MIL-STD-188-113 differs from that in STANAG 4209. The terminology used for corresponding functional elements in the two documents is given in appendix C (table XXI and figure 13). MIL-STD-188-113 uses the terminology generally used in the United States.

5.2.2 **Reference level.** The decoder analog output level with the 16- and 32-kbps, 30-percent run-of-threes reference digital pattern applied to the decoder input shall be the reference level for the CVSD requirements of this standard, and shall be designated 0 dBm0 (see 5.2.3.9.1).

5.2.3 **CVSD characteristics.**

5.2.3.1 **Input and output impedances.** The analog input and output impedances for CVSD converters are not standardized. These impedances depend upon the application of the converters.

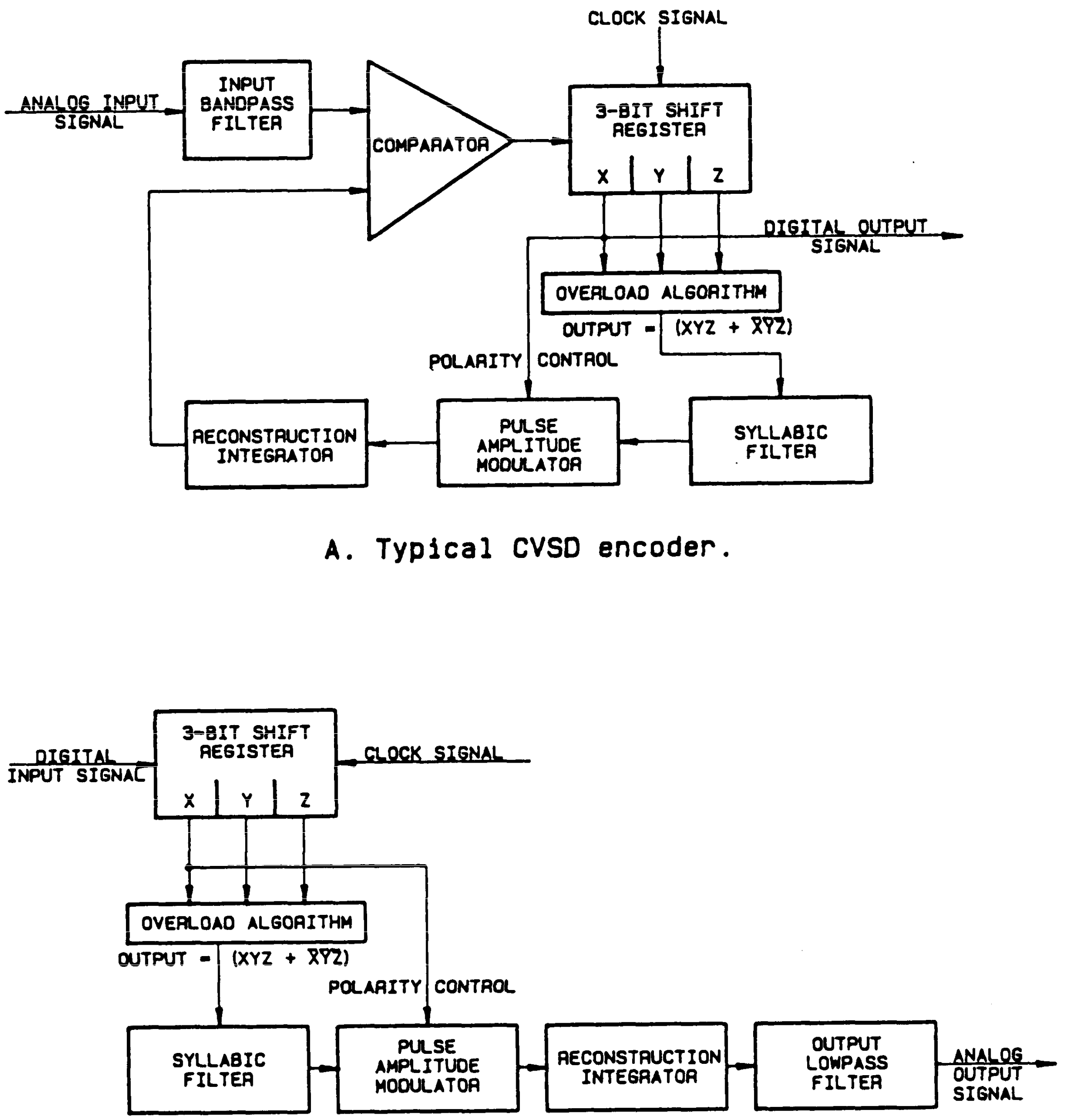
5.2.3.2 **Data signaling rates.** The CVSD converter shall be capable of operating at 16 kbps and 32 kbps.

5.2.3.3 **Input and output filters.** The analog input shall be bandpass filtered. The analog output shall be lowpass filtered.

> NOTE: Details of input and output filters, consistent with the CVSD performance requirements of this standard, will be determined in applicable equipment specifications based on validated requirements. Appendix B (Table XVIII) shows a typical input filter response. This response is for the input filter used in telephone T-1 PCM carrier equipment.

5.2.3.4 **Overload algorithm.** A 3-bit shift register shall be used for the CVSD encoder/decoder (figure 2). The overload logic shall operate on the output of this shift register using the run-of-threes coincidence algorithm. The algorithm output signal shall be a binary signal at the data signaling rate. This signal shall be true for one clock period following the detection of three like bits (all ZEROS or all ONES), and false at all other times.

5.2.3.5 **Compression ratio.** The compression ratio shall be nominally 16:1 with a maximum of 21:1 and a minimum of 12:1. The maximum slope voltage shall be measured at the output of the syllabic filter for a 30-percent run-of-threes digital pattern. The minimum slope voltage shall be measured at the output of the syllabic filter for a 0-percent run-of-threes digital pattern.



A. Typical CVSD encoder.

B. Typical CVSD decoder.

FIGURE 2. Block diagram of a typical CVSD converter.

5.2.3.6 **Syllabic filter.** The syllabic filter shall have a time constant of 5 ms ±1 ms. The step-function response of the syllabic filter shall be exponential in nature. When the output of the overload algorithm is true, a charge curve shall be applicable. When the output of the overload algorithm is false, a discharge curve shall be applicable.

5.2.3.7 **Reconstruction integrator time constant.** The reconstruction integrator shall have a time constant of 1 ms ±0.25 ms.

5.2.3.8 **Analog-to-digital (A-D) conversion.** An 800-Hz ±10 Hz signal at a 0-dBm0 level applied to the input of the encoder shall give a duty cycle (c_d) of 0.30 at the algorithm output of the encoder shown in figure 2A.

5.2.3.9 **Digital-to-analog (D-A) conversion.**

5.2.3.9.1 **Relation of output to input.** With the applicable reference digital patterns of table VI applied to the digital input of the decoder as shown in figure 3, the analog output signal shall be 800 Hz ±10 Hz at the levels shown in table VI, measured at the decoder output. These digital patterns, shown in hexadecimal form, shall be repeating sequences.

TABLE VI. Decoder reference digital patterns for CVSD.

Data Signaling Rate, kbps	Digital Pattern	Run-of-threes, Percent	Output, dBm0
16	DB492	0	−24±1
32	DB54924AB6	0	−24±1
16	FB412	30	0±1
32	FDAA10255E	30	0±1

NOTE: The order of digital pattern bit input to the decoder shall be from left to right.

5.2.3.9.2 **Conversion speed.** When the decoder input is switched from the 0-percent run-of-threes digital pattern to the 30-percent run-of-threes digital pattern, the decoder output shall reach 90 percent of its final value within 9 to 14 ms. When the decoder input is switched from the 30-percent run-of-threes digital pattern to the 0-percent run-of-threes digital pattern, the decoder output shall reach 10 percent of the 30-percent run-of-threes value within 6 to 9 ms. These values shall apply to both the 16- and 32-kbps data signaling rates. Additional information on decoder output envelope characteristics is given in appendix C.

5.2.3.10 **CVSD converter performance.** The characteristics specified in subparagraphs 5.2.3.10.1 through 5.2.3.10.7 apply to one CVSD conversion process obtained by connecting the output of an encoder to the input of a decoder (figure 3).

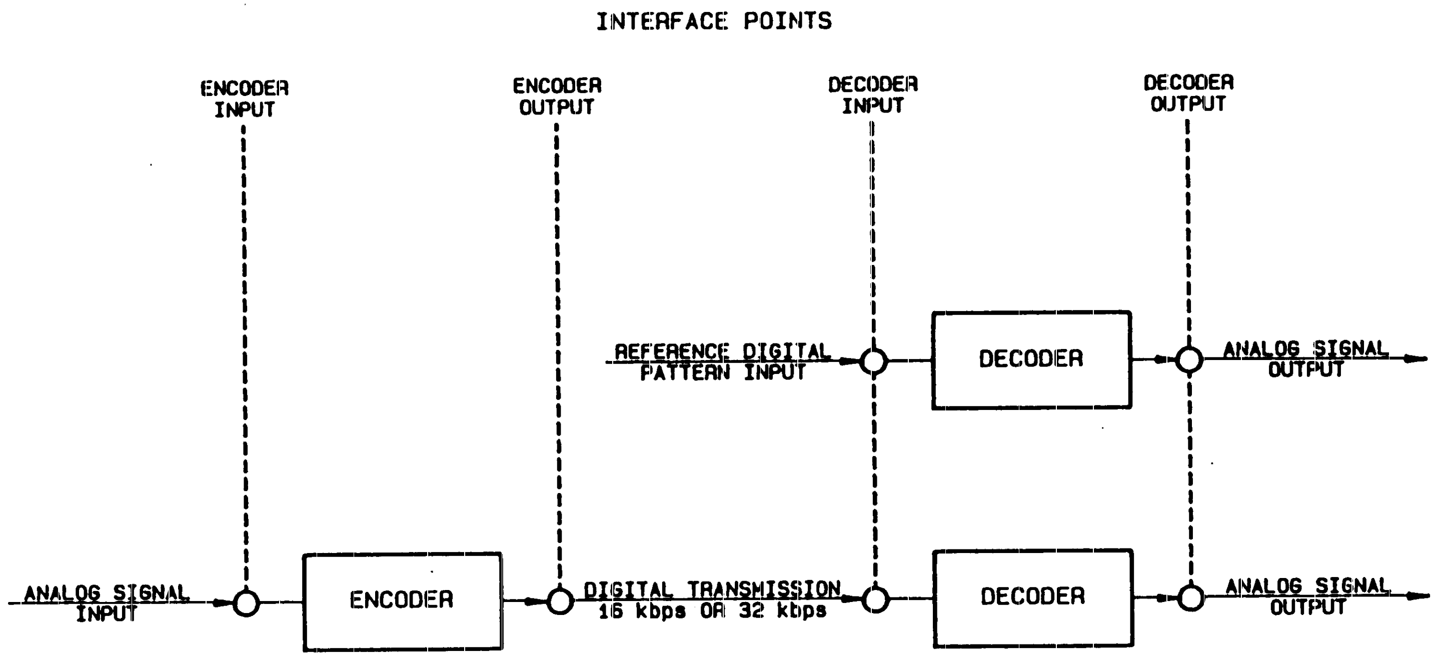


FIGURE 3. Interface diagram for CVSD converter.

NOTE: Test signal frequencies which are submultiples of the data signaling rate shall be avoided by offsetting the nominal test frequency slightly; e.g., an 800-Hz test frequency could be offset to 804 Hz. This test-frequency offset will avoid nonlinear distortion which can cause measurement difficulties when tandeming CVSD with PCM.

5.2.3.10.1 **Companding speed.** When an 800-Hz ±10 Hz sinewave signal at the encoder input is switched from −24 dBm0 to 0 dBm0, the decoder output signal shall reach 90 percent of its final value within 9 to 14 ms.

5.2.3.10.2 **Insertion loss.** The insertion loss between the encoder input and the decoder output shall be 0 dB ±2 dB with an 800-Hz ±10 Hz, 0-dBm0 input to the encoder.

5.2.3.10.3 **Insertion loss vs. frequency characteristics.** The insertion loss between the encoder input and decoder output, relative to 800 Hz ±10 Hz measured with an input level of −15 dBm0 applied to the converter input, shall not exceed the limits indicated in table VII and shown in figure 4.

TABLE VII. Insertion loss limits for CVSD.

Rate, kbps	Frequency (f), Hz	Insertion Loss, dB (Referenced to 800 Hz)
16	$f < 300$	≥ -1.5
	$300 \leq f < 1000$	-1.5 to $+1.5$
	$1000 \leq f < 2600$	-5 to $+1.5$
	$2600 \leq f < 4200$	≥ -5
	$4200 \leq f$	$\geq +25$
32	$f < 300$	≥ -1
	$300 \leq f < 1400$	-1 to $+1$
	$1400 \leq f < 2600$	-3 to $+1$
	$2600 \leq f < 3400$	-3 to $+2$
	$3400 \leq f < 4200$	≥ -3
	$4200 \leq f$	$\geq +25$

5.2.3.10.4 **Variation of gain with input level.** The variation in output level, relative to the value at −15 dBm0 input, shall be within the limits of figure 5 for an input frequency of 800 Hz ±10 Hz.

5.2.3.10.5 **Idle channel noise.** The idle channel noise shall not exceed the limits shown in table VIII when measured at the CVSD decoder output.

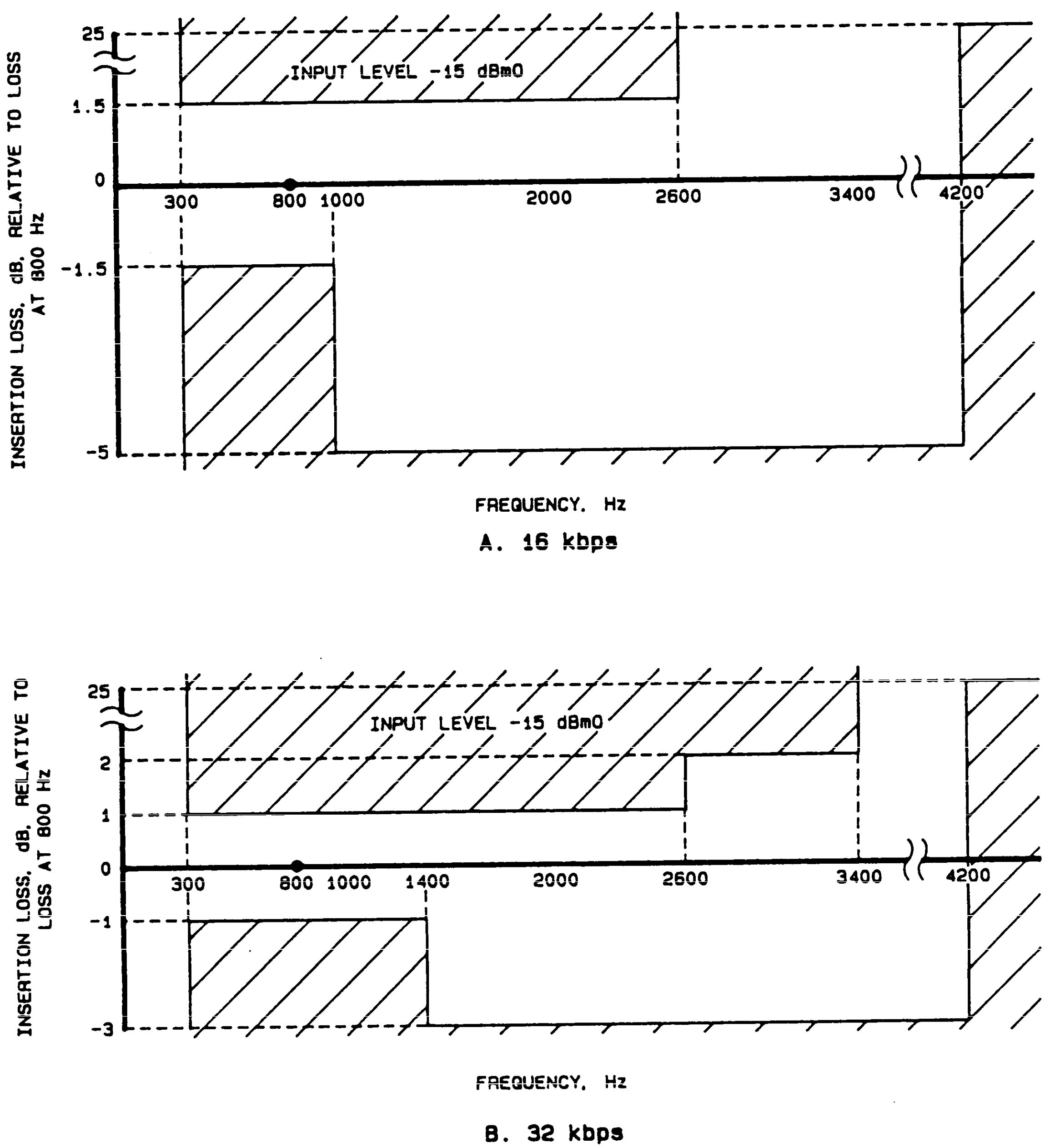


FIGURE 4. Insertion loss vs. frequency for CVSD.

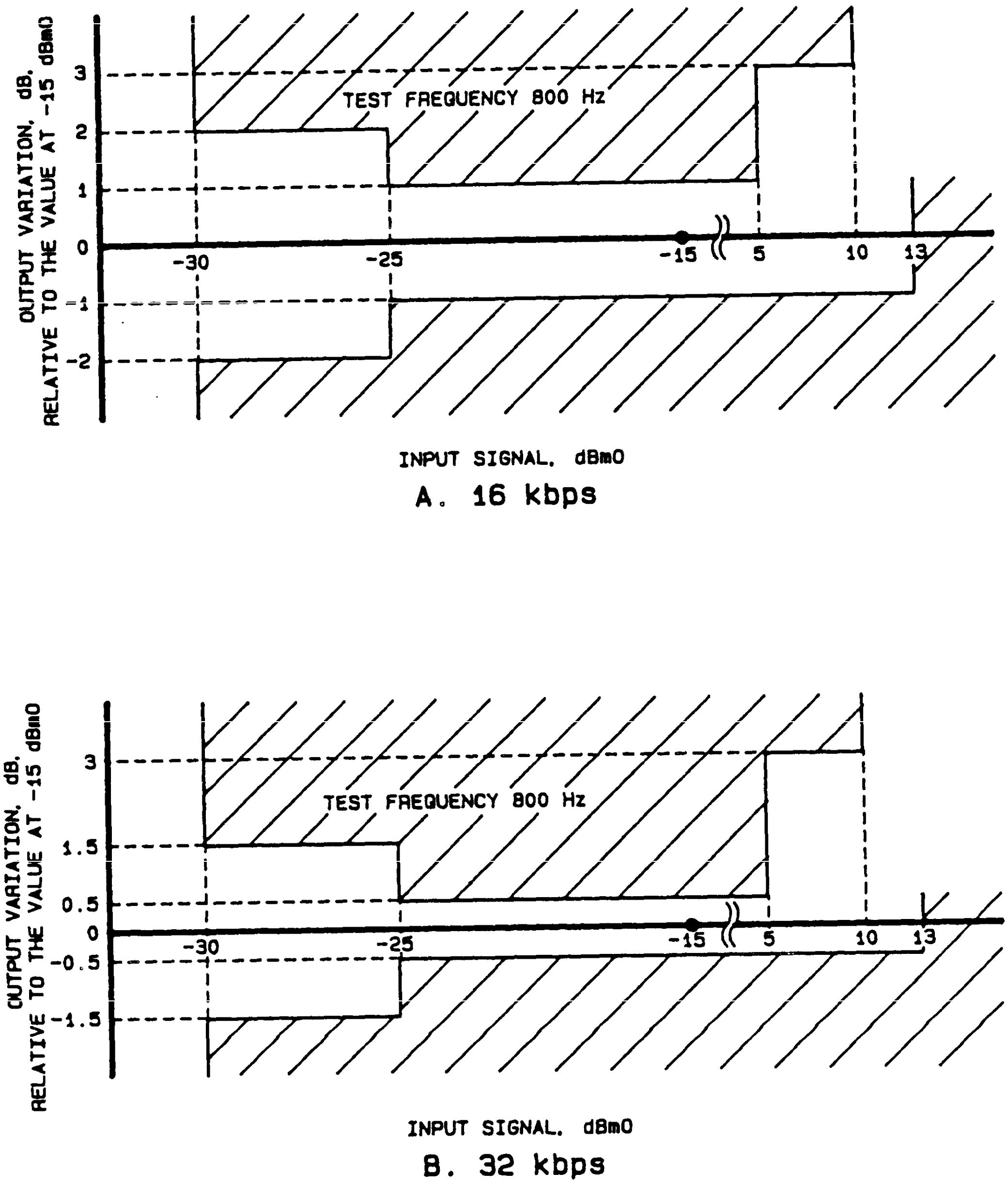


FIGURE 5. Variation of gain with input level for CVSD.

TABLE VIII. Idle channel noise limits for CVSD.

Data Signaling Rate, kbps	Idle Channel Noise, dBm0
16	−40
32	−50

5.2.3.10.6 **Variation of quantizing noise with input level.** The minimum signal to quantizing noise ratio over the input signal level range shall be above the limits of figure 6. The noise ratio shall be measured with flat weighting (unweighted) at the decoder output, with a nominal 800-Hz ±10 Hz sine-wave test signal at the encoder input.

5.2.3.10.7 **Variation of quantizing noise with frequency.** The minimum signal to quantizing noise ratio over the input frequency range shall be above the limits of figure 7. The noise ratio shall be measured with flat weighting (unweighted) at the decoder output, with a sine-wave test signal of −15 dBm0.

5.3 **Linear predictive coding (LPC).**

5.3.1 **Interoperability and performance.** The interoperability and performance relating to LPC shall comply with the applicable requirements of Federal Standard 1015.

> NOTE: NATO STANAG 4198 was used in developing the LPC-10 requirements in FED-STD-1015.

5.3.2 **Intelligibility testing.** Intelligibility testing is not standardized.

> NOTE: Procuring agencies may require intelligibility testing as part of equipment procurement. A recommended method of evaluating intelligibility is by use of the Diagnostic Rhyme Test (DRT). Information on suggested parameters for the DRT is given in appendix E, which was extracted from STANAG 4198.

5.4 **Adaptive predictive coding (APC).**

5.4.1 **General.** Figure 8 is a block diagram of a typical transmitter for the adaptive predictive coding with segmented quantization (APC/SQ) system. The input speech signal is filtered and converted to digital. The long-term dc bias is then removed and the signal is preemphasized by a second-order filter. The pitch period value (TAU) and the pitch weighting factor (ALPHA) are calculated. The adaptive coding analysis uses a fourth-order predictive coder. The APC/SQ algorithm uses a two-spin iterative procedure for finding improved frame quantization levels (Q's) for use in generating the residual error signal (ERR 2 (FINAL)). In a typical receiver (figure 9), decoding and error detection is accomplished by direct table lookup. The error signal is generated based on the frame Q value and the 10-segment Q (SEGQ) values. The reflection

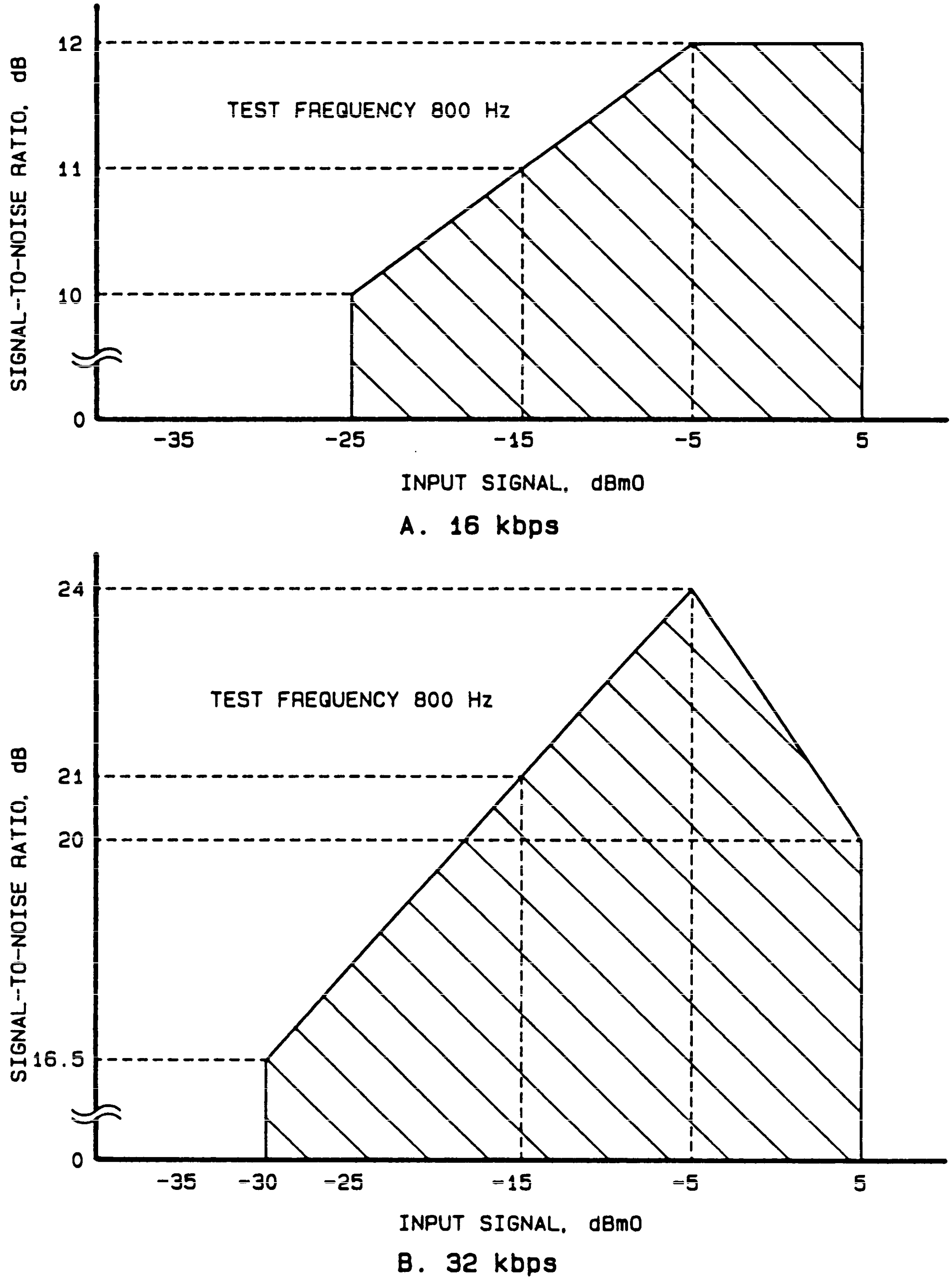
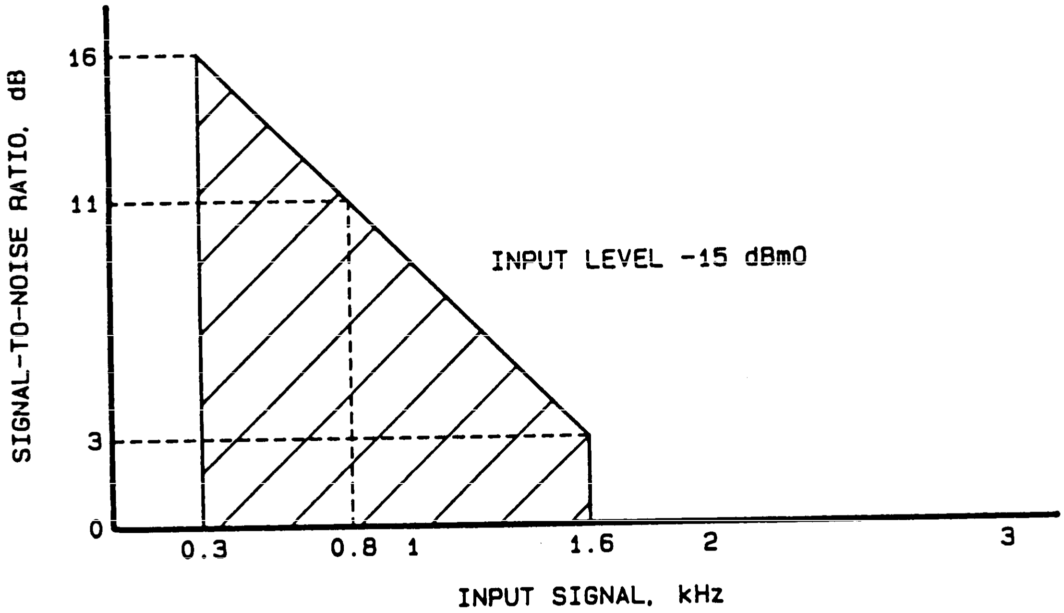
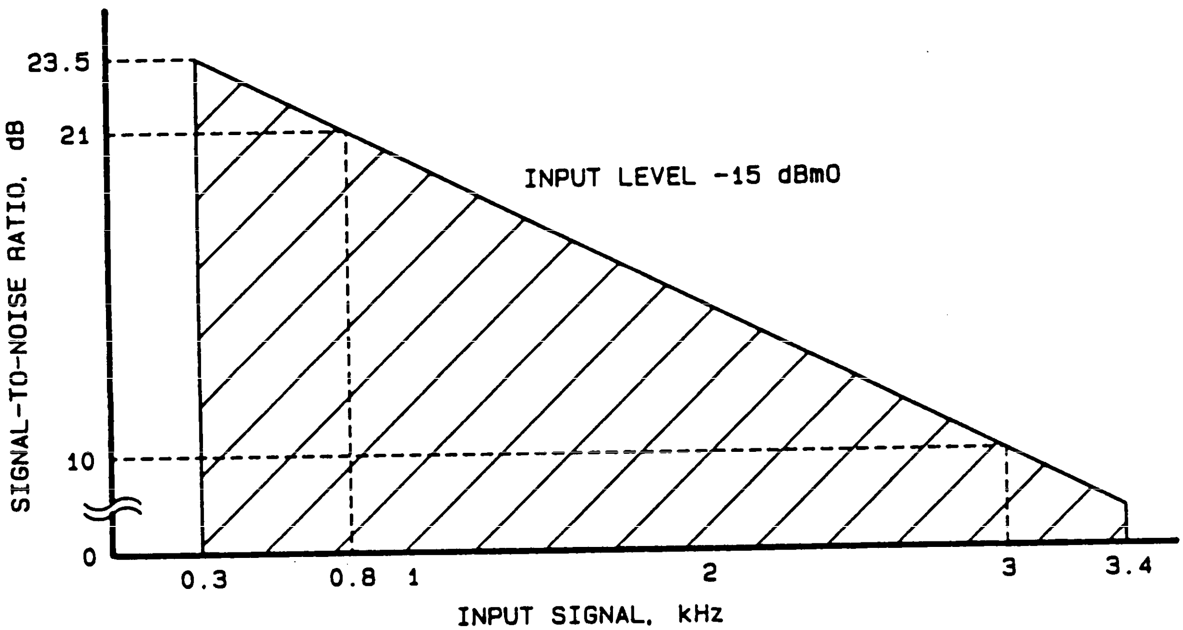


FIGURE 6. Signal to quantizing noise ratio vs. input level for CVSD.

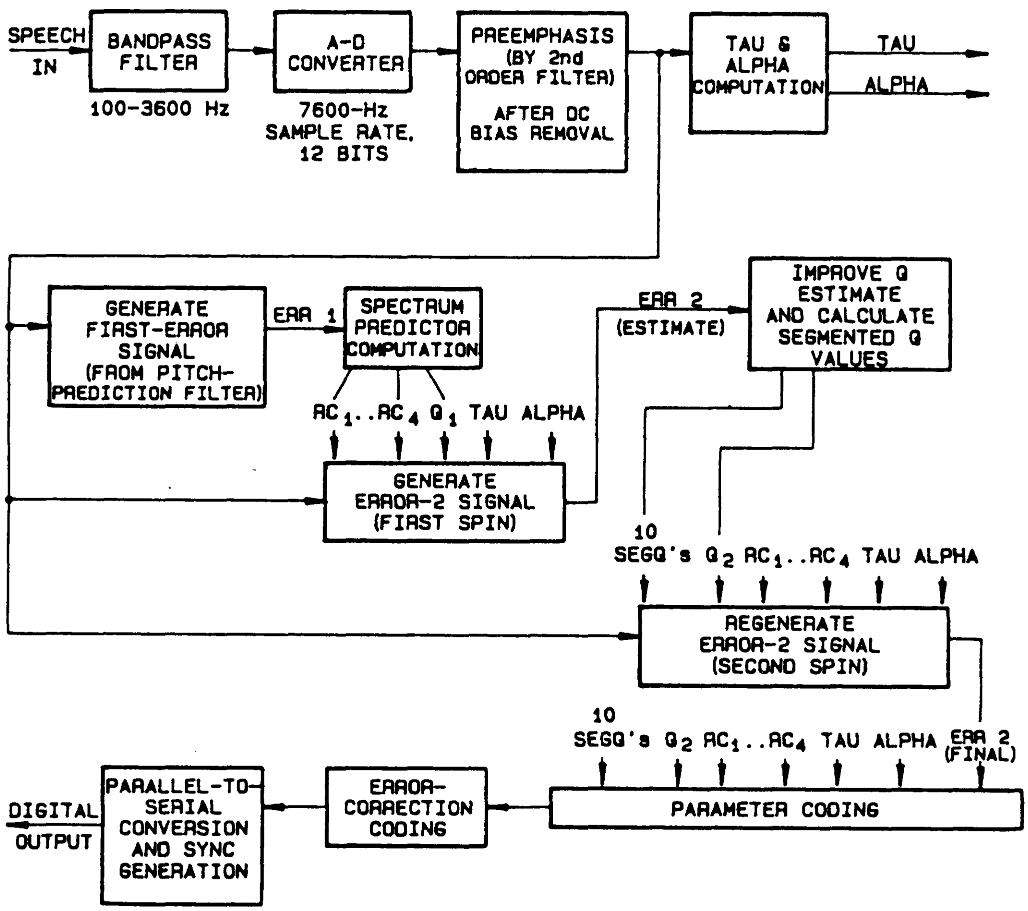


A. 16 kbps



B. 32 kbps

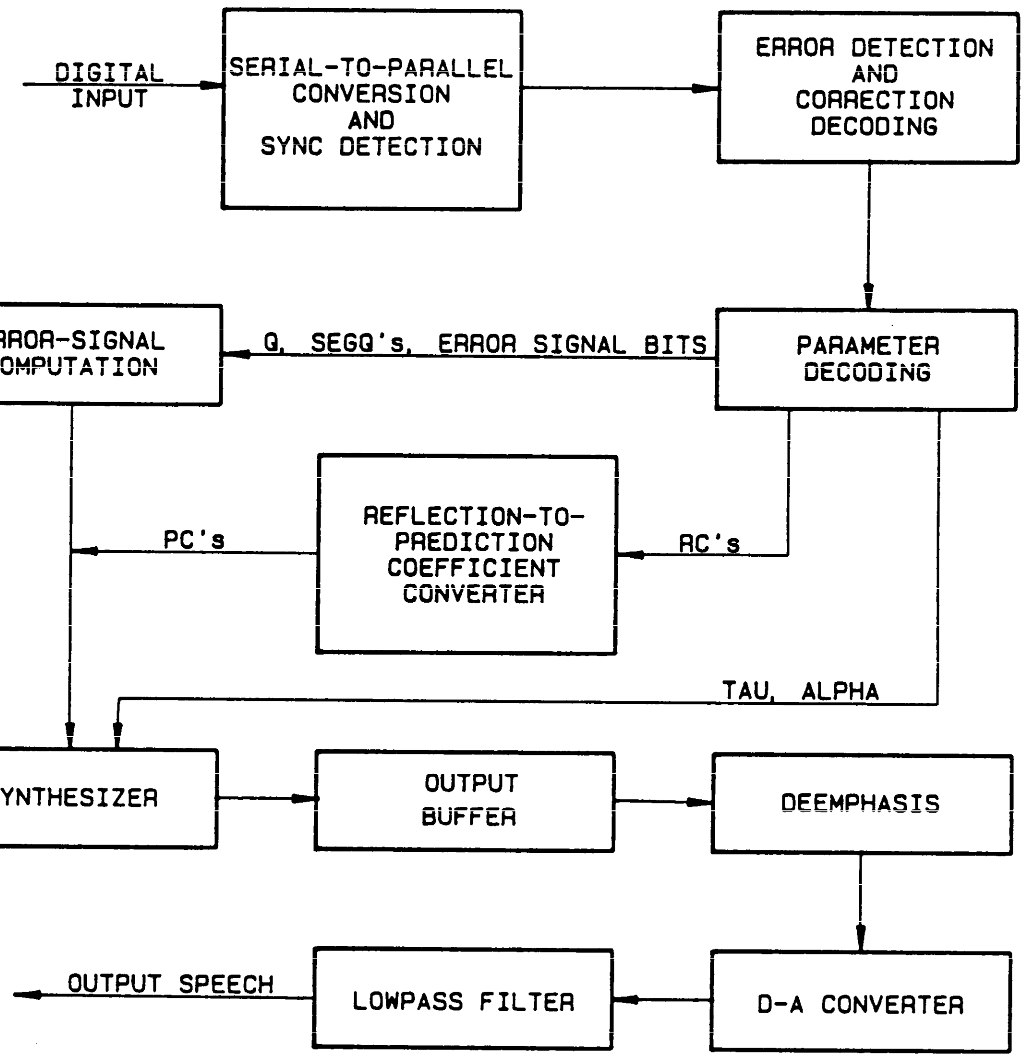
FIGURE 7. Signal to quantizing noise ratio vs. frequency for CVSD.



LEGEND:

TAU is pitch period value.
ALPHA is pitch weighting factor.
ERR 1 is first signal generated from pitch-prediction filter.
ERR 2 is residual error signal which is generated in two iterations (spins). A first estimate of the residual error signal is generated by the first spin; the final residual error is generated by the second spin.
RC is reflection coefficient.
Q is frame quantization level.
Q_1 is first estimate of Q.
Q_2 is second estimate of Q.
SEGQ is segment quantization level.

FIGURE 8. Typical APC/SQ transmitter.



LEGEND:

TAU is pitch-period value.
ALPHA is pitch-weighting factor.
PC is prediction coefficient.
RC is reflection coefficient.
Q is frame quantization level.
SEGQ is segment quantization level.

FIGURE 9. Typical APC/SQ receiver.

coefficients (RC's) are translated to prediction coefficients (PC's). The synthesizer is a fourth-order linear predictor with pitch period loops.

NOTE: Additional information on APC is presented in appendix F.

5.4.2 **APC/SQ transmitter.**

5.4.2.1 **Input filtering.** The APC/SQ transmitter input shall be bandpass filtered. The passband should be essentially flat from 100 Hz to 3600 Hz.

NOTE: Details of the input filter, consistent with the APC requirements of this standard, will be determined in applicable equipment specifications based on validated requirements. Appendix B (Table XVIII) shows a typical input filter response. This response is for the input filter used in telephone T-1 PCM carrier equipment.

5.4.2.2 **Analog-to-digital (A-D) conversion.** The A-D converter shall use a 7600 Hz ±0.1 percent sampling frequency and have a dynamic range of at least 12 bits.

5.4.2.3 **Long-term dc bias removal.** The long-term dc bias shall be removed from the digitized speech signal prior to preemphasis.

5.4.2.4 **Preemphasis.** Preemphasis shall use a second-order filter with the digital transfer function $1 - Z^{-1} + 0.5Z^{-2}$.

5.4.2.5 **Analysis.** Subparagraphs 5.4.2.5.1 through 5.4.2.5.5 present the analysis performed by the APC transmitter. For convenience in using the tables, both encoding and decoding of the APC parameters are presented here.

5.4.2.5.1 **Pitch prediction.**

5.4.2.5.1.1 **Pitch period value (TAU).** Sixty values of TAU shall be calculated in the closed interval 20 to 156, with 20 values per octave. The value to be transmitted is selected and encoded with 6 bits. A transmitted 1 corresponds to a period of 20 and a transmitted 60 corresponds to a period of 156. The relationship between the coded pitch value, the pitch period, and the pitch frequency is shown in table IX. This table is also used for decoding in the receiver. TAU shall be coded to ZERO when the encoded ALPHA is ZERO.

5.4.2.5.1.2 **Pitch weighting factor (ALPHA).** ALPHA shall be calculated and Gray coded to three bits as shown in table X. The decoding of ALPHA shall be in accordance with this table.

5.4.2.5.2 **Reflection coefficients (RC's).** Four RC's shall be calculated and log-area-ratio coded to 5 bits as shown in table XI. The decoding of the RC's shall be in accordance with this table.

NOTE: The sign convention for RC's biases RC_1 (for voiced sounds) toward +1.

TABLE IX. Relation of coded pitch value to pitch period and pitch frequency for APC/SQ.

Code	Pitch Period	Pitch Frequency	Code	Pitch Period	Pitch Frequency	Code	Pitch Period	Pitch Frequency
1	20	380	21	40	190	41	80	95
2	21	362	22	42	181	42	84	90
3	22	345	23	44	173	43	88	86
4	23	330	24	46	165	44	92	83
5	24	317	25	48	158	45	96	79
6	25	304	26	50	152	46	100	76
7	26	292	27	52	146	47	104	73
8	27	281	28	54	141	48	108	70
9	28	271	29	56	136	49	112	68
10	29	262	30	58	131	50	116	66
11	30	253	31	60	127	51	120	63
12	31	245	32	62	123	52	124	61
13	32	238	33	64	119	53	128	59
14	33	230	34	66	115	54	132	58
15	34	224	35	68	112	55	136	56
16	35	217	36	70	109	56	140	54
17	36	211	37	72	106	57	144	53
18	37	205	38	74	103	58	148	51
19	38	200	39	76	100	59	152	50
20	39	195	40	78	97	60	156	49

TABLE X. ALPHA coding/decoding.

ALPHA	Code	Decode
0.000 thru 0.516	0	0.406
0.517 thru 0.725	1	0.616
0.726 thru 0.885	3	0.825
0.886 thru 0.975	2	0.945
0.976 thru 1.025	6	1.000
1.026 thru 1.115	7	1.055
1.116 thru 1.275	5	1.175
1.276 thru 1.500	4	1.375

NOTES:
1. A received TAU of zero will cause the decoded ALPHA to be set at zero.
2. ALPHA's greater than 1.5 will cause both ALPHA and TAU to be encoded all zeros.

TABLE XI. RC coding/decoding.

Reflection Coefficients	Code	Index	Decode
−.9999 thru −.9844	10001	−15	−.9844
−.9843 thru −.9688	10010	−14	−.9688
−.9687 thru −.9531	10011	−13	−.9531
−.9530 thru −.9375	10100	−12	−.9375
−.9374 thru −.9063	10101	−11	−.9218
−.9062 thru −.8750	10110	−10	−.8906
−.8749 thru −.8281	10111	−9	−.8438
−.8280 thru −.7656	11000	−8	−.7812
−.7655 thru −.6875	11001	−7	−.7187
−.6874 thru −.6094	11010	−6	−.6406
−.6093 thru −.5313	11011	−5	−.5625
−.5312 thru −.4219	11100	−4	−.4688
−.4218 thru −.3125	11101	−3	−.3593
−.3124 thru −.2032	11110	−2	−.2500
−.2031 thru −.0938	11111	−1	−.1406
−.0937 thru +.0937	00000	+0	+.0313
.0938 thru .2031	00001	+1	+.1406
.2032 thru .3124	00010	+2	+.2500
.3125 thru .4218	00011	+3	+.3593
.4219 thru .5312	00100	+4	+.4688
.5313 thru .6093	00101	+5	+.5625
.6094 thru .6874	00110	+6	+.6406
.6875 thru .7655	00111	+7	+.7187
.7656 thru .8280	01000	+8	+.7812
.8281 thru .8749	01001	+9	+.8438
.8750 thru .9062	01010	+10	+.8906
.9063 thru .9374	01011	+11	+.9218
.9375 thru .9530	01100	+12	+.9375
.9531 thru .9687	01101	+13	+.9531
.9688 thru .9843	01110	+14	+.9688
.9844 thru .9999	01111	+15	+.9844

NOTE: Coded values are the index numbers expressed in binary "two's complement."

5.4.2.5.3 **Frame quantization level (Q).** The Q shall be calculated and coded to five bits using table XII. The decoding of the Q's shall be in accordance with this table.

TABLE XII. Q coding/decoding.

Q	Code	Decode
0, 1	0	1
2, 3	1	2
4, 5	2	4
6, 7	3	6
8, 9	4	8
10, 11	5	10
12, 13	6	12
14, 15	7	14
16, 17	8	16
18, 19	9	18
20, 21	10	20
22, 23	11	22
24, 25	12	24
26 thru 29	13	27
30 thru 33	14	31
34 thru 37	15	35
38 thru 43	16	40
44 thru 51	17	47
52 thru 59	18	55
60 thru 69	19	64
70 thru 81	20	75
82 thru 95	21	90
96 thru 111	22	103
112 thru 129	23	120
130 thru 149	24	138
150 thru 173	25	160
174 thru 201	26	186
202 thru 235	27	216
236 thru 275	28	253
276 thru 323	29	296
324 thru 379	30	348
≥ 380	31	408

5.4.2.5.4 **Segment quantization levels, coded (SEGQC's).** A segment quantization level (SEGQ) shall be generated and coded as a 2-bit parameter for each of the 10 segments (20 bits total). Table XIII gives the coding and decoding for the SEGQ's. The receiver linearly interpolates between segment Q's on a per-sample basis. The last nine samples are not interpolated.

TABLE XIII. SEGQ coding/decoding.

SEGQ	Code	Decode
0.00 thru 0.49	0	0.40
0.50 thru 0.99	1	0.70
1.00 thru 1.59	2	1.30
≥1.60	3	1.80

5.4.2.5.5. **Error signal.** The residual error signal (ERR 2(FINAL)) shall be generated in two iterations (spins) to improve accuracy. The error signal shall be quantized to one bit per sample (E0 through E189). Bit E9, and every 19th bit thereafter (10 bits total), shall be deleted so only 180 bits per frame are transmitted.

5.4.2.6 **Frame format.** An APC frame shall consist of 240 bits. These bits shall be allocated as shown in table XIV and assigned as shown in table XV.

TABLE XIV. Bit allocation for APC/SQ frame.

Error signal		180 bits
Speech parameters (as follows)		54 bits
Reflection coefficients (RC's)	20 bits	
Pitch period value (TAU)	6 bits	
Pitch weighting factor (ALPHA)	3 bits	
Frame quantization level (Q)	5 bits	
Segment quantization levels (SEGQC's)	20 bits	
Error control (check)		5 bits
Synchronization		1 bit
	TOTAL	240 bits

TABLE XV. Bit assignment for APC/SQ frame.

Position	Parameter Bit	Position	Parameter Bit	Position	Parameter Bit
1	E8	41	E62	81	E116
2	E7		E61		E115
3	E6		E60		E114
4	RC1-0		RC3-0		ALP-1
5	E5	45	E59	85	E141
6	E4		E58		E140
7	E3		E57		E139
8	RC1-1		RC3-1		ALP-2
9	E2		E84		E138
10	E1	50	E83	90	E137
	E0		E82		E136
	RC1-2		RC3-2		TAU-0
	E27		E81		E135
	E26		E80		E134
15	E25	55	E79	95	E133
	RC1-3		RC3-3		TAU-1
	E24		E78		E160
	E23		E77		E159
	E22		E76		E158
20	RC1-4	60	RC3-4	100	TAU-2
	E21		E103		E157
	E20		E102		E156
	E19		E101		E155
	RC2-0		RC4-0		TAU-3
25	E46	65	E100	105	E154
	E45		E99		E153
	E44		E98		E152
	RC2-1		RC4-1		TAU-4
	E43		E97		E179
30	E42	70	E96	110	E178
	E41		E95		E177
	RC2-2		RC4-2		TAU-5
	E40		E122		E176
	E39		E121		E175
35	E38	75	E120	115	E174
	RC2-3		RC4-3		Q-0
	E65		E119		E173
	E64		E118		E172
	E63		E117		E171
40	RC2-4	80	ALP-0	120	RC4-4

TABLE XV. Bit assignment for APC/SQ frame.-Continued

Position	Parameter Bit	Position	Parameter Bit	Position	Parameter Bit
121	E18	161	E72	201	E126
122	E17		E71		E125
123	E16		E70		E124
124	Q-1		SQC0-1		SQC5-1
125	E15	165	E69	205	E151
126	E14		E68		E150
127	E13		E67		E149
128	Q-2		SQC1-0		SQC6-0
129	E12		E94		E148
130	E11	170	E93	210	E147
	E10		E92		E146
	Q-3		SQC1-1		SQC6-1
	E37		E91		E145
	E36		E90		E144
135	E35	175	E89	215	E143
	Q-4		SQC2-0		SQC7-0
	E34		E88		E170
	E33		E87		E169
	E32		E86		E168
140	CKT-0	180	SQC2-1	220	SQC7-1
	E31		E113		E167
	E30		E112		E166
	E29		E111		E165
	CKT-1		SQC3-0		SQC8-0
145	E56	185	E110	225	E164
	E55		E109		E163
	E54		E108		E162
	CKT-2		SQC3-1		SQC8-1
	E53		E107		E189
150	E52	190	E106	230	E188
	E51		E105		E187
	CKT-3		SQC4-0		SQC9-0
	E50		E132		E186
	E49		E131		E185
155	E48	195	E130	235	E184
	CKT-4		SQC4-1		SQC9-1
	E75		E129		E183
	E74		E128		E182
	E73		E127		E181
160	SQC0-0	200	SQC5-0	240	SYNC

TABLE XV. Bit assignment for APC/SQ frame.-Continued

NOTES:
1. E = error signal; RC = reflection coefficient; ALP = ALPHA, the pitch weighting factor; Q = frame quantization level; CKT = check bit; SQC = segment Q, coded.
2. Bits are transmitted in ascending order, 1 through 240.
3. Error signal is composed of bits E0 through E189.
4. For speech parameters, bit 0 = LSB.

5.4.2.7 **Error control.** Five bits shall be allocated to error correction coding using a (21, 16) code. Table XVI shows the bits to be protected. The data bits to be protected are ordered as shown in figure 10, and a check bit is generated for each of the five fields shown. Check bits 4, 3, 2, 1, and 0 are set to 1 if the data bits in their respective fields are odd parity. In the decoding process, the procedure is repeated, and the calculated check bits are exclusive OR'ed (XOR'ed) with the received check bits. The five-bit result of the XOR is used as an index to table XVII. This table gives the bit position in the data word of figure 10 to be corrected.

TABLE XVI. Data bits to be protected for APC/SQ.

Parameter	Bits protected
RC1	Most significant bit
RC2	Two most significant bits
RC3	Two most significant bits
RC4	Two most significant bits
ALPHA	Most significant bit
Q	Two most significant bits
TAU	All six bits

5.4.2.8 **Synchronization.** Bit 240 of each frame shall be a synchronization bit. The synchronization bit shall alternate between ZERO and ONE from frame to frame, starting with ZERO.

5.4.2.9 **Transmission rate.** The transmission rate shall be 9600 bps ±0.01 percent. The frame length shall be 25.0 milliseconds, which gives 40 frames per second.

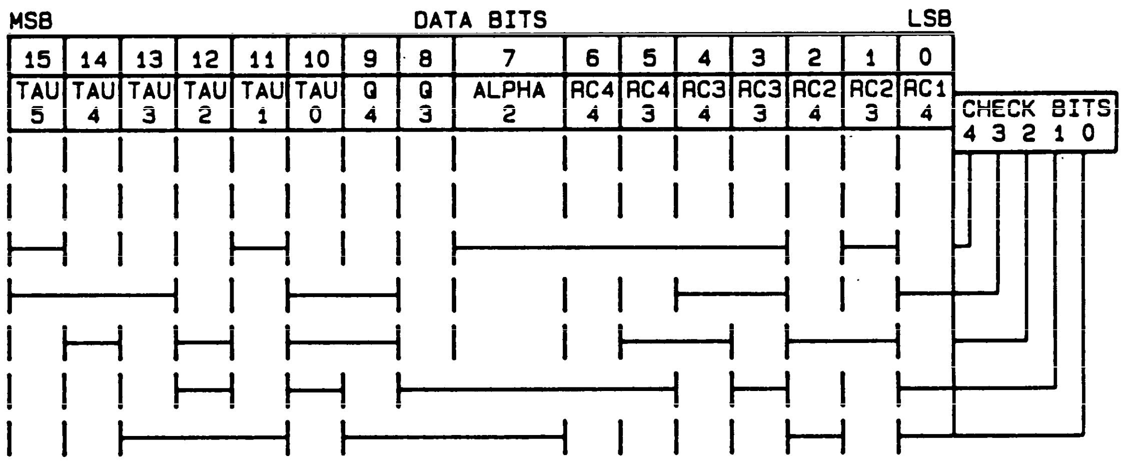
5.4.3 **APC/SQ receiver.**

5.4.3.1 **Parameter decoding.** The 240 bits per frame shall be decoded with appropriate error correction to generate pitch, ALPHA, Q, RC's, and segment Q values.

5.4.3.2 **Error signal computation.** The error signal shall be generated from the frame Q, 10-segment Q values, and error-signal bits.

5.4.3.3 **Speech generation.** The error signal shall be used to excite the synthesizer to reconstruct the speech waveform. The synthesizer shall be a fourth-order linear predictor with pitch period loop.

5.4.3.4 **Deemphasis.** The deemphasis transfer function shall be
$\frac{1}{1 - Z^{-1} + 0.5\, Z^{-2}}$, the inverse of the preemphasis filter.



NOTE: The five check bits constitute the 5-bit parity word.

FIGURE 10. Ordering of data bits to be protected for APC/SQ.

TABLE XVII. Error correction decoding for APC/SQ.

Parity Word	Invert Bit	Parity Word	Invert Bit
0	none	16	none
1	none	17	11
2	none	18	6
3	8	19	7
4	none	20	1
5	2	21	none
6	none	22	5
7	12	23	none
8	none	24	15
9	13	25	none
10	none	26	3
11	0	27	none
12	14	28	4
13	9	29	none
14	10	30	none
15	none	31	none

NOTE: The contents of this table give the bit position (in the 16-bit word of figure 10) of the error to be corrected by bit inversion.

5.4.3.5 **Digital-to-analog (D-A) conversion.** The digital output of the deemphasis filter shall be converted to an analog signal by a converter having the characteristics stated in 5.4.2.2.

5.4.3.6 **Output filtering.** The analog output of the D-A converter shall be lowpass filtered.

> NOTE: Details of the output filter, consistent with the APC requirements of this standard, will be determined in applicable equipment specifications based on validated requirements.

5.5 **Adaptive differential pulse code modulation (ADPCM).** Under consideration.

> NOTE: Algorithms for ADPCM have been selected and equipment using these algorithms is in use. ADPCM requirements will be incorporated in the next revision of this standard.

6. NOTES.

6.1 **Intended use.** This standard specifies minimum operability and performance characteristics for analog-to-digital conversion to be used in the design and installation of new communications subsystems and equipment and in authorized upgrading of existing communications subsystems and equipment. The PCM and CVSD requirements of this standard are applicable to converters and, accordingly, do not include special considerations which may be necessary when the converter is used in a multiplexer application. The requirements herein are for the following data rates and techniques:

a. 64 kbps pulse code modulation (PCM).

b. 16 and 32 kbps continuously variable slope delta (CVSD) modulation.

c. 2.4 kbps linear predictive coding (LPC).

d. 9.6 kbps adaptive predictive coding (APC).

e. 32 kbps adaptive differential pulse code modulation (ADPCM).

6.2 **Subject term (key word) listing.**

Adaptive differential pulse code modulation (ADPCM)
Adaptive predictive coding (APC)
Adaptive predictive coding with segmented quantization (APC/SQ)
Algorithm, APC/SQ
Analog-to-digital (A-D) conversion
APC/SQ analysis
APC/SQ receiver
APC/SQ transmitter
Compressor/encoder, PCM
Continuously variable slope delta (CVSD) modulation
CVSD converter
Diagnostic Rhyme Test (DRT)
Digital-to-analog (D-A) conversion
Efficiently digitally linearizable (EDL) coding
Encoder/decoder, CVSD
Encoder/decoder, PCM
Expander/decoder, PCM
Linear predictive coding (LPC)
PCM converter
Pulse code modulation (PCM)

APPENDIX A

UNDER SECRETARY OF DEFENSE MEMORANDUM

10. SCOPE.

10.1 **General.** This appendix contains information related to MIL-STD-188-113 on the mandatory use of military telecommunications standards in the MIL-STD-188- series.

10.2 **Purpose.** The referenced memorandum makes MIL-STD-188-113 mandatory for use by all Departments and Agencies of the Department of Defense (DoD).

10.3 **Application.** Appendix A is a mandatory part of this standard.

20. REFERENCED DOCUMENTS.

20.1 **Government documents.** The following document forms a part of this appendix to the extent specified:

Military Memorandum

Under Secretary of Defense (Research and Engineering) memorandum, 16 August 1983.

30. DEFINITIONS. Not applicable.

40. GENERAL REQUIREMENTS. The memorandum herein requires that all relevant standards in the -188 series be mandatory for use within the DoD.

THE UNDER SECRETARY OF DEFENSE
WASHINGTON, D.C. 20301

RESEARCH AND ENGINEERING

16 AUG 1983

MEMORANDUM FOR ASSISTANT SECRETARY OF THE ARMY (INSTALLATIONS, LOGISTICS & FINANCIAL MANAGEMENT)
ASSISTANT SECRETARY OF THE NAVY (SHIPBUILDING & LOGISTICS)
ASSISTANT SECRETARY OF THE AIR FORCE (RESEARCH DEVELOPMENT & LOGISTICS)
COMMANDANT OF THE MARINE CORPS
DIRECTOR, DEFENSE COMMUNICATIONS AGENCY
DIRECTOR, NATIONAL SECURITY AGENCY

SUBJECT: Mandatory Use of Military Telecommunications Standards in the MIL-STD-188 Series

On May 10, 1977, Dr. Gerald Dinneen, then Assistant Secretary of Defense(C^3I), issued the following policy statement regarding the mandatory nature of the MIL-STD-188 series telecommunications standards:

"...standards as a general rule are now cited as 'approved for use' rather than 'mandatory for use' in the Department of Defense.

This deference to the judgment of the designing and procuring agencies is clearly appropriate to standards dealing with process, component ruggedness and reliability, paint finishes, and the like. It is clearly not appropriate to standards such as those in the MIL-STD-188 series which address telecommunication design parameters. These influence the functional integrity of telecommunication systems and their ability to efficiently interoperate with other functionally similar Government and commercial systems. Therefore, relevant military standards in the 188 series will continue to be mandatory for use within the Department of Defense.

To minimize the probability of misapplication of these standards, it is incumbent upon the developers of the MIL-STD-188 series to insure that each standard is not only essential but of uniformly high quality, clear and concise as to application, and wherever possible compatible with existing or proposed national, international and Federal telecommunication standards. It is also incumbent upon the users of these standards to cite in their procurement specifications only those standards which are clearly necessary to the proper functioning of the device or systems over its projected lifetime."

This statement has been reviewed by this office and continues to be the policy of the Department of Defense.

R. D. DeLauer

APPENDIX B

GENERAL INFORMATION ON PCM

10. SCOPE.

10.1 **Purpose.** This appendix provides general information in support of the PCM requirements in MIL-STD-188-113.

10.2 **Application.** This appendix is intended for guidance only. The information contained herein is not a mandatory part of this standard.

20. REFERENCED DOCUMENTS. Not applicable.

30. DEFINITIONS.

30.1 **Terms.** For purposes of this appendix, definitions of terms shall be as specified in this standard and in the current edition of FED-STD-1037.

30.2 **Abbreviations and acronyms.** For purposes of this appendix, definitions of abbreviations and acronyms shall be as specified in this standard and in the current edition of FED-STD-1037.

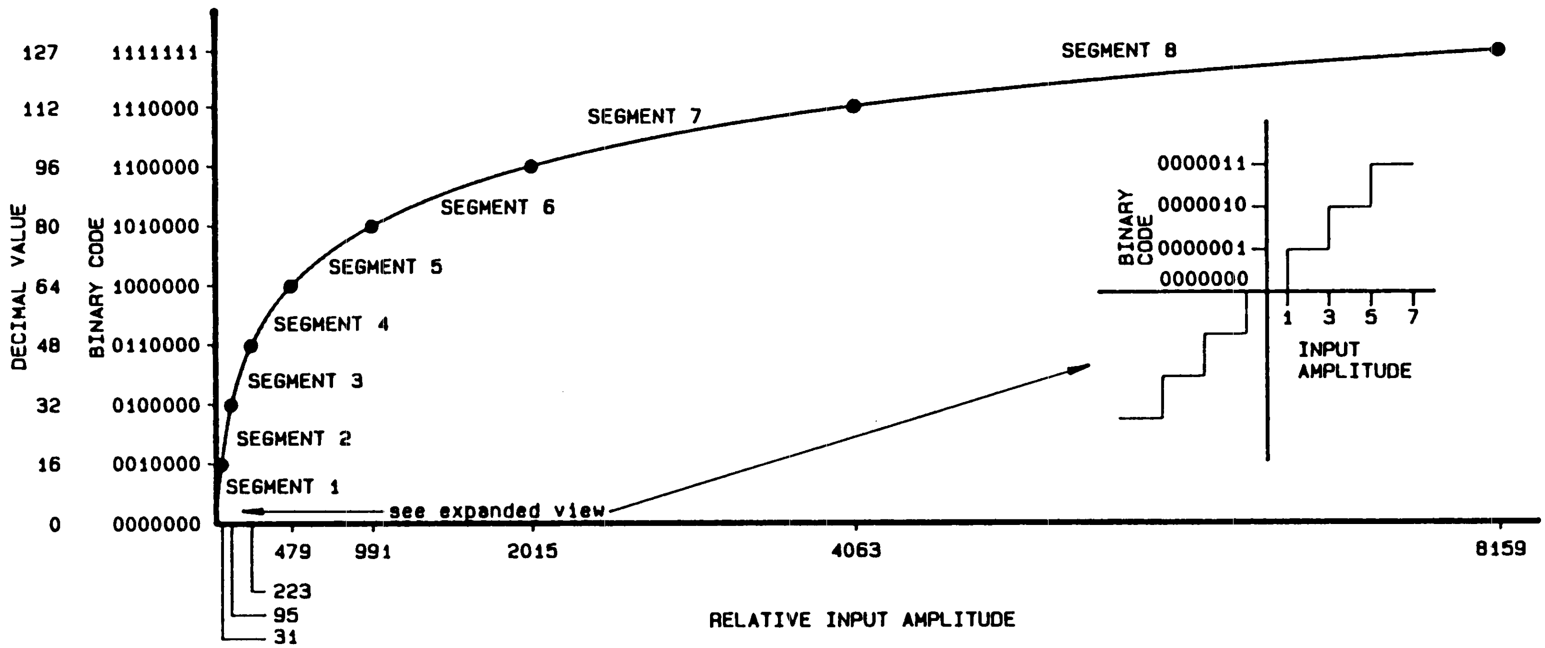
40. GENERAL DESCRIPTIONS.

40.1 **PCM input filter characteristics.** Table XVIII shows a typical input filter response. This response is for the input filter used in telephone T-1 PCM carrier equipment.

TABLE XVIII. Typical input filter response for PCM.

Frequency (Hz)	Attenuation (dB)
40	32
60	18
80	8
100	3
200	0
500	0
1000	0
1500	0
2000	0
2500	0
3000	0
3600	3
3800	13
4000	23
4200	33
4400	46

40.2 **Characteristics of efficiently digitally linearizable (EDL) coding.** Figure 11 gives the positive half of the EDL coding characteristic. Tables XIX and XX present the detailed structure of EDL coding, including the number and size of quantization intervals for each segment, and the decimal value of the binary code.



NOTES:

1. Only the positive half of the coding characteristic is shown.
2. The binary code shown is for the first quantization interval in each segment. For Segment 8, the code is shown for both the first and last intervals. The polarity bit is not shown.
3. Segment 1 of the negative half of the coding characteristic is colinear with Segment 1 shown; therefore, these two segments are considered as one. This makes a total of 15 segments.
4. The inverted form of the binary code, which is used for transmission, is not shown.

FIGURE 11. EDL coding characteristic for PCM

TABLE XIX. Structure of EDL coding for positive input values for PCM.

Segment	Relative Quantization Interval Size	Number of Quantization Intervals	Relative Input Amplitude (PAM Sample)	Binary Code			Decimal Value of Code
				Polarity	Segment	Level in Segment	
8	256	16	7903 to 8159 through 4063 to 4319	0	111	1111 through 0000	+127 through +112
7	128	16	3935 to 4063 through 2015 to 2143	0	110	1111 through 0000	+111 through + 96
6	64	16	1951 to 2015 through 991 to 1055	0	101	1111 through 0000	+ 95 through + 80
5	32	16	959 to 991 through 479 to 511	0	100	1111 through 0000	+ 79 through + 64
4	16	16	463 to 479 through 223 to 239	0	011	1111 through 0000	+ 63 through + 48
3	8	16	215 to 223 through 95 to 103	0	010	1111 through 0000	+ 47 through + 32
2	4	16	91 to 95 through 31 to 35	0	001	1111 through 0000	+ 31 through + 16
1	2	15	29 to 31 through 1 to 3	0	000	1111 through 0001	+ 15 through + 1
	1	1	0 to 1	0	000	0000	0

TABLE XIX. Structure of EDL coding for positive input values for PCM.-Continued

NOTES:
1. The relative input amplitude for the first and last quantization interval in each segment is shown. For segment 1, the input amplitude for the first, second, and last quantization interval is given.
2. The progression through the 16 code levels in each segment (0000 thru 1111) is in normal binary sequence.
3. All bits are inverted for transmission.
4. PAM samples are normalized to a full scale value of 8159.
5. The value of the decoder output for the first interval is zero and is referred to as a "positive" zero. The number of quantization intervals is 128 for positive inputs.

TABLE XX. Structure of EDL coding for negative input values for PCM.

Segment	Relative Quantization Interval Size	Number of Quantization Intervals	Relative Input Amplitude (PAM Sample)	Binary Code			Decimal Value of Code
				Polarity	Segment	Level in Segment	
	1	1	0 to −1	1	000	0000	0
1	2	15	−1 to −3 through −29 to −31	1	000	0001 through 1111	−1 through −15
2	4	16	−31 to −35 through −91 to −95	1	001	0000 through 1111	−16 through −31
3	8	16	−95 to −103 through −215 to −223	1	010	0000 through 1111	−32 through −47
4	16	16	−223 to −239 through −463 to −479	1	011	0000 through 1111	−48 through −63
5	32	16	−479 to −511 through −959 to −991	1	100	0000 through 1111	−64 through −79
6	64	16	−991 to −1055 through −1951 to −2015	1	101	0000 through 1111	−80 through −95
7	128	16	−2015 to −2143 through −3951 to −4063	1	110	0000 through 1111	−96 through −111
8	256	16	−4063 to −4391 through −7903 to −8159	1	111	0000 through 1101	−112 through −127

TABLE XX. Structure of EDL coding for negative input values for PCM.-Continued

NOTES:
1. The relative input amplitude for the first and last quantization interval in each segment is shown. For segment 1, the input amplitude for the first, second, and last quantization interval is given.
2. The progression through the 16 code levels in each segment (0000 thru 1111) is in normal binary sequence.
3. All bits are inverted for transmission.
4. PAM samples are normalized to a full scale value of −8159.
5. The most negative quantization interval (−7903 thru −8159) is assigned the same code as the interval for −7391 to −7647 to avoid an inverted code with no transitions, i.e., 00000000. This results in 127 effective quantization intervals for negative inputs. Using the code 11111101 for the most negative interval is consistent with CCITT Recommendation G.711.
6. The value of the decoder output for the first interval is zero and is referred to as a "negative" zero.

APPENDIX C

GENERAL INFORMATION ON CVSD MODULATION

10. SCOPE.

10.1 **Purpose.** This appendix provides general information in support of the CVSD requirements in MIL-STD-188-113.

10.2 **Application.** This appendix is intended for guidance only. The information contained herein is not a mandatory part of this standard.

20. REFERENCED DOCUMENTS.

20.1 **Government documents.** The following document forms a part of this appendix to the extent specified:

Military Standard

MIL-STD-188-200	System Design and Engineering Standards for Tactical Communications

20.2 **Other publications.** The following document forms a part of this appendix to the extent specified:

NATO Standardization Agreement

STANAG 4209	The NATO Multi-Channel Tactical Digital Gateway — Standards for Analogue to Digital Conversion of Speech Signals

30. DEFINITIONS.

30.1 **Terms.** For purposes of this appendix, definitions of terms shall be as specified in this standard and in the current edition of FED-STD-1037.

30.2 **Abbreviations and acronyms.** For purposes of this appendix, definitions of abbreviations and acronyms shall be as specified in this standard and in the current edition of FED-STD-1037.

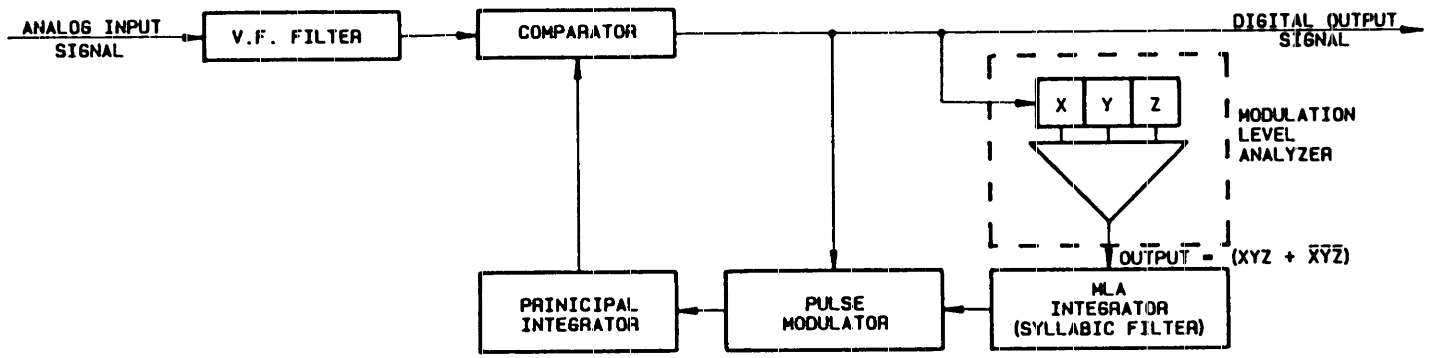
30.3 **Terminology comparisons.** Some of the CVSD terminology in MIL-STD-188-113 differs from that found in STANAG 4209. Table XXI provides a comparison of the terms that differ in the illustrated block diagrams for the CVSD converter (figure 2 for MIL-STD-188-113, and the diagram extracted from STANAG 4209 shown in figure 12 of this appendix). MIL-STD-188-113 uses the terminology generally used in the United States.

TABLE XXI. CVSD terminology comparisons.	
MIL-STD-188-113 Terminology	STANAG 4209 Terminology
3-bit shift register, overload algorithm	Collectively called modulation level analyzer (MLA)
Syllabic filter	MLA integrator (syllabic filter)
Pulse amplitude modulator	Pulse modulator
Reconstruction integrator	Principal integrator
Input bandpass filter (encoder) Output lowpass filter (decoder)	V.F. filter for both coder and decoder

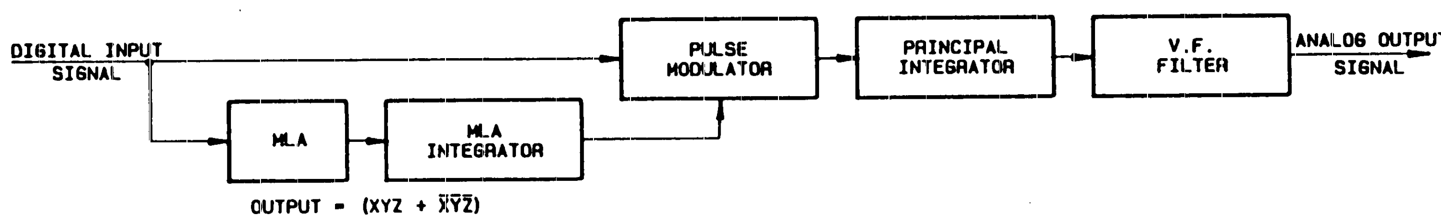
40. GENERAL DESCRIPTIONS. Information in subparagraphs 40.1 through 40.3.6 has been extracted from MIL-STD-188-200 (paragraph 4.6.2.2 and appendix D).

40.1 **Delta modulation.** Delta modulation is an A-D conversion technique resulting in a form of digital pulse modulation. A delta modulator periodically samples the amplitude of a band-limited analog signal and the amplitude differences of two adjacent samples are coded into n-bit code words. This nonlinear, sampled-data, feedback system then transmits the encoded bit stream through a digital channel. At the receiving end, an integrating network converts the delta-modulated bit stream, through a decoding process, into a close approximation of the original analog signal.

40.2 **CVSD converter.** (See figure 2.) A typical CVSD converter consists of an encoder and a decoder. The analog input signal of the CVSD encoder is band-limited by the input bandpass filter. The CVSD encoder compares the band-limited analog input signal with an analog feedback approximation signal generated at the reconstruction integrator output. The digital output signal of the encoder is the output of the first register in the "run-of-three" counter. The digital output signal is transmitted at the clock (sample) rate and will equal "1" if the analog input signal is greater than or equal to the analog feedback signal at the instant of sampling. For this value of the digital output signal, the pulse amplitude modulator (PAM) applies a positive feedback pulse to the reconstruction integrator; otherwise, a negative pulse is applied. This function is accomplished by the polarity control signal, which is equal to the digital encoder output signal. The amplitude of the feedback pulse is derived by means of a 3-bit shift register, logic sensing for overload, and a syllabic lowpass filter. When a string of three consecutive ONES or ZEROS appears at the digital output, a discrete voltage level is applied to the syllabic filter, and the positive feedback pulse amplitude increases until the overload string is broken. In such an event, ground potential is fed to the filter by the overload algorithm, forcing a decrease in the amplitude of the slope voltage out of the syllabic filter. The encoder and decoder have identical characteris-



A. Block schematic of the coder.



B. Block schematic of the decoder.

FIGURE 12. Block diagram of a CVSD converter.

tics except for the comparator and filter functions. The CVSD decoder consists of the input bandpass filter, shift register, overload algorithm, syllabic filter, PAM and reconstruction integrator used in the encoder, and an output lowpass filter. The decoder performs the inverse function of the encoder and regenerates speech by passing the analog output signal of the reconstruction integrator through the lowpass filter. Other characteristics optimize the CVSD modulation technique for voice signals. These characteristics include:

a. Changes in the slope of the analog input signal determine the step-size changes of the digital output signal.

b. The feedback loop is adaptive to the extent that the loop provides continuous or smoothly incremental changes in step size.

c. Companding is performed at a syllabic rate to extend the dynamic range of the analog input signal.

d. The reconstruction integrator is of the exponential (leaky) type to reduce the effects of digital errors.

50. DETAILED DESCRIPTIONS. The characteristics described in subparagraphs 50.1 through 50.9 are in addition to those specified in 5.2.3 of this standard and are for guidance only.

50.1 **Input bandpass filter.** (See figure 2A.) The input filter provides band-limiting and is typically a second- or higher-order filter.

50.2 **Comparator.** (See figure 2A.) The comparator compares the band-limited analog input signal from the filter with the output signal of the reconstruction integrator. This comparison produces the digital error signal input to the 3-bit shift register. The transfer characteristic of the comparator is such that the difference between the two input signals causes the output signal to be driven to saturation in the direction of the sign of the difference.

50.3 **Three-bit shift register.** (See figure 2.) The 3-bit shift register acts as a sampler which clocks the digital error signal from the comparator at the specified data signaling rate and stores the current samples and two previous samples of the error signal. The digital output signal is a binary signal having the same polarity as the input signal from the comparator at the time of the clock signal. The digital output signal is also the digital output of the encoder and is referred to as the baseband signal. Further processing for transmission, such as conditioned diphase modulation, may be applied to the baseband signal. It is necessary that the inverse of any such processing be accomplished and the baseband signal restored before the CVSD decoding process is attempted.

50.4 **Overload algorithm.** (See figure 2.) The overload algorithm operates on the output of the 3-bit shift register (X, Y, Z) using the run-of-threes coincidence algorithm, so that the algorithm output equals $(XYZ + \overline{XYZ})$. The output signal is a binary signal at the clock signaling rate and is true for one clock period following the detection of three like bits, and false at all other times.

50.5 **Syllabic filter.** (See figure 2.) The syllabic filter acts as a lowpass filter for the output signal from the overload algorithm. The slope-voltage output of the syllabic filter is the modulating input to the PAM. The step-function response of the syllabic filter is related to the syllabic rate of speech, is independent of the sampling rate, and is exponential in nature. When the overload algorithm output is true, a charging curve is applicable. When this output is false, a discharging curve is applicable.

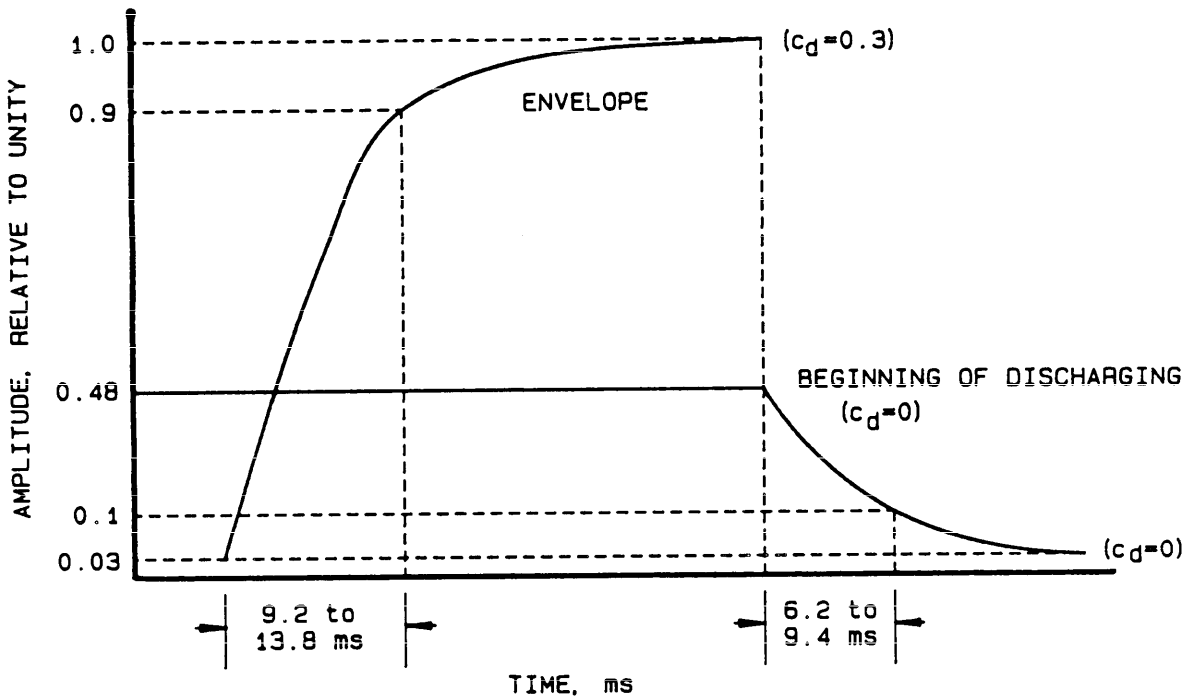
50.6 **Pulse amplitude modulator (PAM).** (See figure 2.) The PAM operates with two input signals: the output signal from the syllabic filter, and the digital signal from the 3-bit shift register. The syllabic filter output signal determines the amplitude of the PAM output signal and the signal from the 3-bit shift register is the polarity control that determines the direction, plus or minus, of the PAM output signal. The phrase "continuously variable" in CVSD is derived from the way the PAM output signal varies almost continuously.

50.7 **Reconstruction integrator.** (See figure 2.) The reconstruction integrator operates on the output signal of the PAM to produce an analog feedback signal to the comparator (or an output signal to the output lowpass filter in the receiver) that is an approximation of the analog input signal.

50.8 **Output lowpass filter.** (See figure 2B.) The output filter is a lowpass filter having a frequency response that typically has an asymptotic rolloff with a minimum slope of 40 dB per octave, and a stopband rejection that is 45 dB or greater. The same output filter characteristic is used for encoder digital output signals of either 16 kbps or 32 kbps.

50.9 **Typical CVSD decoder output envelope characteristics.** (Refer to decoder in figure 2B.) For a resistance/capacitance circuit in the syllabic filter with time constants of 5 ms for both charging and discharging, the envelope characteristics of the signal at the decoder output are shown in figure 13. For the case of switching the signal at the decoder input from the 0-percent run-of-threes digital pattern to the 30-percent run-of-threes digital pattern, the characteristic of the decoder output signal follows the resistance/capacitance charge curve. Note that the number of time constants required to reach the 90-percent charge point is 2.3, which gives a nominal charge time of 11.5 ms.

When switching the other way (from the 30-percent pattern to the 0-percent pattern), the amplitude at the beginning of discharging is, at the first moment of switching, higher (by a factor of 16) than the final value which is reached asymptotically. The final value equals −24 dBm0, i.e., 0.03. Therefore, the amplitude at the beginning of discharging is 0.48 (percent run-of-threes = 0). Note that the number of time constants required to reach the 10-percent point on the discharge curve is 1.57, which gives a nominal discharge time of 7.8 ms.



NOTES:
1. C_d is duty cycle.
2. Figure is half of the envelope.

FIGURE 13. Typical envelope characteristics of the decoder output signal for CVSD.

APPENDIX D

GENERAL DESCRIPTION OF THE FOUR-TONE METHOD FOR NONLINEAR DISTORTION TESTING

10. SCOPE.

10.1 **Purpose.** This appendix provides general information in support of the requirements in MIL-STD-188-113.

10.2 **Application.** This appendix is intended for guidance only. The information contained herein is not a mandatory part of this standard.

20. REFERENCED DOCUMENTS. Not applicable.

30. DEFINITIONS.

30.1 **Terms.** For purposes of this appendix, definitions of terms shall be as specified in this standard and in the current edition of FED-STD-1037.

30.2 **Abbreviations and acronyms.** For purposes of this appendix, definitions of abbreviations and acronyms shall be as specified in this standard and in the current edition of FED-STD-1037.

40. GENERAL DESCRIPTIONS.

40.1 **General.** This method is used for making multitone nonlinear, or intermodulation, distortion measurements when the distortion is specified in terms of second- and third-order limits. The measurement is accomplished with an intermodulation distortion measurement set that features four-tone capability.

40.2 **Method of measurement.** Four-tone nonlinear distortion is measured by applying two pairs of tones (tone pair A and tone pair B) at the input of the system or subsystem under test and measuring second-order (B−A and B+A) components and third-order (2B−A) components at the output. All four tones are applied at specified equal levels. In a typical measurement setup, the first pair of tones is separated by 6 Hz ±1 Hz and centered at 860 Hz ±1 Hz. The second pair of tones is separated by 16 Hz ±1 Hz, and centered at 1380 Hz ±1 Hz. The second-order products measured are the combined outputs of two filters, one having a bandpass of 503 Hz to 537 Hz, and the other having a bandpass of 2223 Hz to 2257 Hz. The third-order products measured are the output of a filter having a bandpass of 1877 Hz to 1923 Hz.

APPENDIX E

SUGGESTED PARAMETERS FOR THE DIAGNOSTIC RHYME TEST

10. SCOPE.

10.1 **Purpose.** This appendix provides general information in support of the requirements in MIL-STD-188-113.

10.2 **Application.** This appendix is intended for guidance only. The information contained herein is not a mandatory part of this standard.

20. REFERENCED DOCUMENTS.

20.1 **Other publications.** The following document forms a part of this appendix to the extent specified:

NATO Standardization Agreement

STANAG 4198 — Parameters and Coding Characteristics That Must be Common to Assure Interoperability of 2400 bps Linear Predictive Encoded Digital Speech

30. DEFINITIONS.

30.1 **Terms.** For purposes of this appendix, definitions of terms shall be as specified in this standard and in the current edition of FED-STD-1037.

30.2 **Abbreviations and acronyms.** For purposes of this appendix, definitions of abbreviations and acronyms shall be as specified in this standard and in the current edition of FED-STD-1037.

40. GENERAL DESCRIPTIONS. The following description was extracted from STANAG 4198.

40.1 **Method of measurement.** The voice intelligibility of the voice processor is measured using the Diagnostic Rhyme Test (DRT-IV). For the DRT, English/American and French versions are to be used, and the talkers and listeners are to be familiar with the language in each case. The input analog tapes to be used for the English/American DRT and the minimum acceptable scores, which should be obtained from an independent contractor, are given in table XXII.

TABLE XXII. Parameters for the DRT.

Acoustic Environment	Talkers	Tapes	Bit Error Rate	Microphone	Minimum Acceptable Score
Quiet	6M	E-1-A E-1-B	0	Dynamic	86
Office	3M	G-4-A	0	Dynamic	84
Shipboard (Saipan)	3M	K6-1.2-A	0	H250	85
Aircraft (P-3C)	3M	K7-1.2-A	0	EV985	82
Jeep	3M	K8-1.2-A	0	H250	82
Tank	3M	K9-1.2-A	0	EV985	82
Quiet	3M	E-1-A	2%	Dynamic	82

APPENDIX F

GENERAL INFORMATION ON APC

10. SCOPE.

10.1 **Purpose.** This appendix provides general information in support of the APC requirements in MIL-STD-188-113.

10.2 **Application.** This appendix is intended for guidance only. The information contained herein is not a mandatory part of this standard.

20. REFERENCED DOCUMENTS.

20.1 **Other publications.** The following document forms a part of this appendix to the extent specified:

"The Government Standard Adaptive Predictive Coding Algorithm: APC-04" by Thomas Tremain (Speech Technology, Feb/Mar 1985, pages 52-62).

30. DEFINITIONS.

30.1 **Terms.** For purposes of this appendix, definitions of terms shall be as specified in this standard and in the current edition of FED-STD-1037.

30.2 **Abbreviations and acronyms.** For purposes of this appendix, definitions of abbreviations and acronyms shall be as specified in this standard and in the current edition of FED-STD-1037.

40. GENERAL DESCRIPTIONS. The information in the following subparagraphs was adapted from the reference in paragraph 20.1.

40.1 **General.** The adaptive predictive coder segmented quantizer (APC/SQ) algorithm is used in the implementation of voice systems operating at 9.6 kbps. The APC/SQ algorithm offers the advantage of higher speech intelligibility and quality than can be achieved by algorithms that operate at 2.4 kbps, while maintaining digital transmission rates that can be readily transmitted over nearly all telephone lines. The APC/SQ algorithm improves on the performance of an LPC algorithm principally by deriving and transmitting an error signal (called the residual) that is used to improve the all-pole synthesis in the receiver. The residual is made up of three parts:

a. An energy scale or frame quantization level (called Q) that applies to the entire 25-ms frame;

b. Ten scale factors or segment quantization levels (called SEGQ's) that track the variation of residual energy during the frame; and

c. 180 bits that indicate whether the residual is plus or minus.

The general functional aspects of the transmitter and receiver are described in subparagraphs 40.2 and 40.3, respectively.

40.2 **Transmitter.** A functional block diagram of the APC/SQ transmitter is shown in figure 14, and a block diagram of the complete transmitter is shown in figure 8. The input speech signal is conditioned for processing by bandpass filtering, A-D converting, removing the long-term dc bias, and preemphasizing the signal. A pitch value (TAU) is calculated for all frames using the same average-magnitude difference function (AMDF) method that is used for LPC-10. Since the residual is used to excite the LPC filter in the receiver instead of the noise (buzz-hiss) generator, the voiced/unvoiced decision and the accurate pitch tracking associated with the standard LPC algorithm is not needed. The signal labeled ERR 1 in figure 8 is generated from the pitch-prediction filter. Reflection coefficients RC_1, RC_2, RC_3, and RC_4, and a first estimate of Q (labeled Q_1 in figure 8), together with the pitch gain or pitch weighting factor (ALPHA), are used to generate, by the first spin, a first estimate of the residual (labeled ERR 2 (ESTIMATE) in figure 8) by comparison of the predicted sample value with the preemphasized input sample value. Final values of Q (labeled Q_2 in figure 8) and the variation of Q (labeled 10 SEGQ's in figure 8) are calculated and used together with RC_1 to RC_4, TAU, and ALPHA to generate, by the second spin, the final residual error (labeled ERR 2 (FINAL) in figure 8). All the final parameters are then encoded, error-correction coding is added, the synchronization signal is added, and the data are shifted out serially. In the feedback loop at the bottom of figure 14 that depicts the generation of the residual error signal, the output signal (S_{out}) is equal to the input signal (S_{in}) plus a quantizing noise. The generating function of the transmitter can be summarized by the following steps:

(1) Preemphasize the input speech;
(2) Determine the pitch period, TAU, using the AMDF function;
(3) Derive the pitch gain factor, ALPHA;
(4) Filter the preemphasized signal through a pitch-prediction filter based on TAU and ALPHA;
(5) Derive four LPC reflection coefficients (RC's) from the resulting error signal using the covariance method;
(6) Convert RC's to prediction coefficients (PC's) and calculate Q and the SEGQ's;
(7) Using Q, SEGQ's, pitch, pitch gain, and PC's, generate the quantized residual error as depicted in the feedback loop at the bottom of figure 14; and
(8) Code the values for TAU, ALPHA, RC's, Q, SEGQ's, and the residual error for transmission using a modified (21, 16) Hamming code to protect the most significant bits of ALPHA, Q, and the RC's.

40.3 **Receiver.** A functional block diagram of the APC/SQ receiver is shown in figure 15, and a block diagram of the complete receiver is shown in figure 9. In the receiver (figure 15), the residual is used to excite two prediction filters in series. The first is a filter that predicts the sample amplitudes of the vocal tract spectrum (i.e., nonpitch) portion of the voice signal based on the four previously predicted samples and the PC's. The second filter supplies the pitch harmonic information based on TAU and ALPHA. Finally, the signal is deemphasized by means of a second-order filter that is the inverse of the preemphasis filter used in the transmitter. The block diagram of the complete receiver shown in figure 9, taken together with the explanation of the transmitter, is self-explanatory.

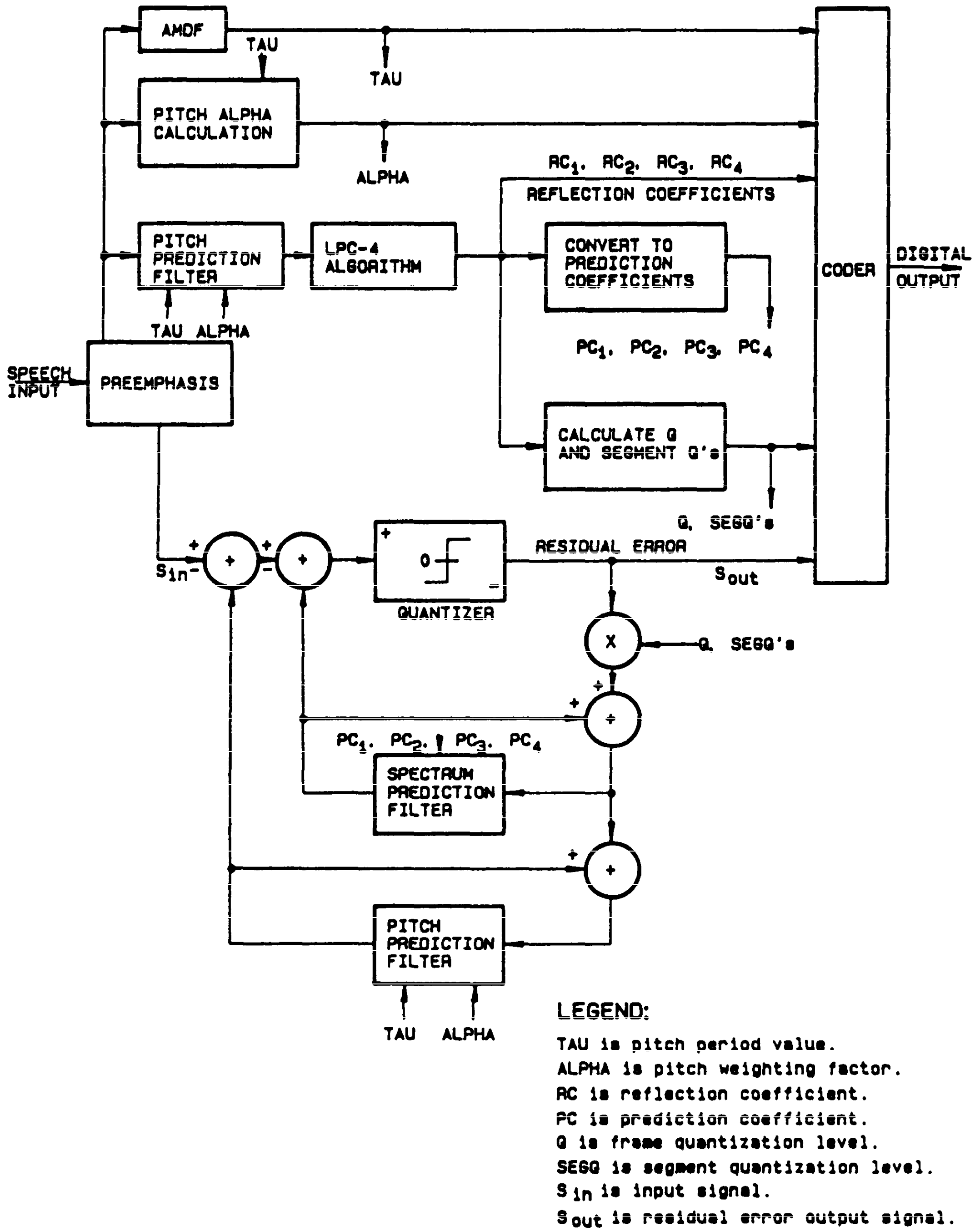
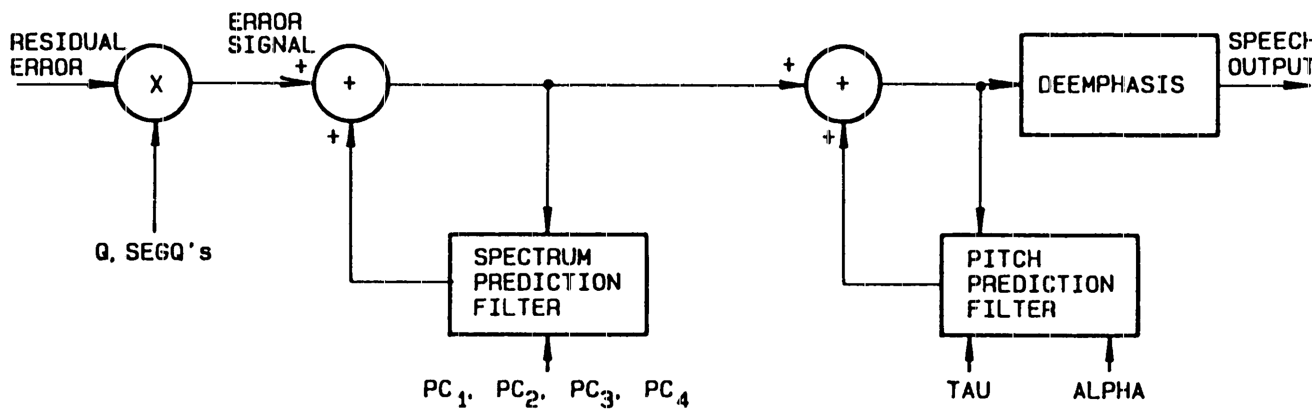


FIGURE 14. Functional diagram of the APC/SQ transmitter.



LEGEND:

TAU is pitch period value.
ALPHA is pitch weighting factor.
PC is prediction coefficient.
Q is frame quantization level.
SEGQ is segment quantization level.

FIGURE 15. Functional diagram of the APC/SQ receiver.

50. DETAILED DESCRIPTIONS. The information in the following subparagraphs was adapted from the reference in paragraph 20.1.

50.1 **Transmitter.** Refer to figures 8 and 14 for the detailed descriptions that follow. The input speech is bandpass filtered through a sharp cutoff filter having a frequency response that is typically the same as the response for the input filter used in telephone T-1 PCM carrier equipment (see table XVIII). The input speech is then A-D converted at 12 bits per sample and 7600 samples per second, and the samples are stored in a double-buffered input array. The long-term dc bias is removed using the transfer function depicted in figure 16. The speech signal is then preemphasized using the transfer function in the following equation:

$$1 - Z^{-1} + 0.5\ Z^{-2}$$

All subsequent processing is based on the resulting signal and is performed on a frame-by-frame basis. Each frame consists of 25 ms of speech (190 samples, ±2 to allow for clock variations). The pitch period is calculated using the average-magnitude difference function (AMDF) and the same pitch values calculated by the LPC method are calculated for the APC.

> NOTE: The transmitted pitch values are those determined by the AMDF minimum rather than the smoothed values used in the standard LPC.

The pitch period, TAU, is then used in the following equation to obtain the pitch gain, ALPHA:

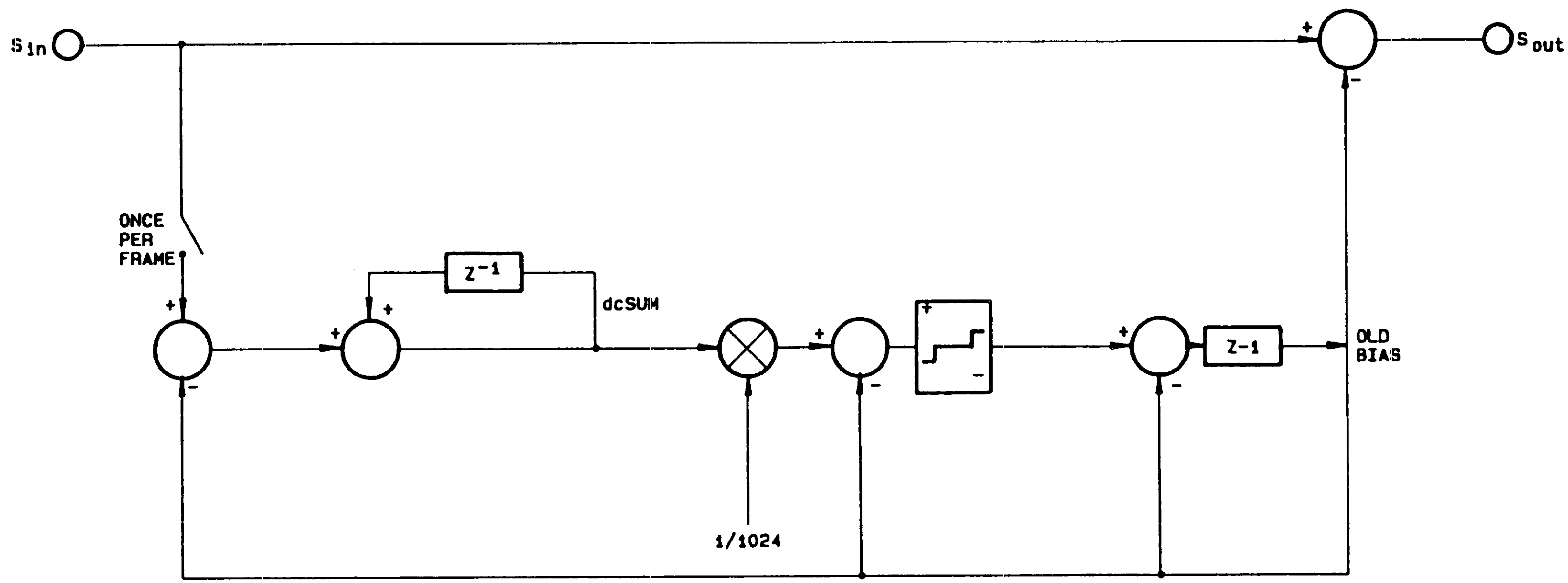
$$\text{ALPHA} = \frac{\sum_{n=1}^{n=N} S_n S_{n-\text{TAU}}}{\sum_{n=1}^{n=N} S^2_{n-\text{TAU}}}$$

where S_n is the amplitude of the nth sample and N is the number of samples in the frame period.

The speech signal is passed through a pitch-prediction filter to obtain the signal (labeled ERR 1 in figure 8) for input to the spectrum analyzer using the following equation:

$$\text{ERR 1} = S_n - \text{ALPHA}\ S_{n-\text{TAU}}$$

The spectrum analyzer performs the spectrum predictor computation (figure 8) and uses a fourth-order LPC algorithm (LPC-4 ALGORITHM block in figure 14). Matrix load uses no block scaling of the ERR 1 signal, but is calculated using double precision. The matrix load and invert routines for the DoD Standard LPC-10 algorithm are also used for this algorithm. However, the covariance matrix load, which is determined every 25 ms, was modified by using high-frequency correction. A first estimate of the residual scale factor or frame quantization level (labeled Q_1 in



LEGEND:
S_{in} is input signal.
S_{out} is output signal.
Z is transfer function.

Figure 16. Long-term bias removal transfer function for APC/SQ.

figure 8) is obtained from the four reflection coefficients (RC_1, RC_2, RC_3, and RC_4) by the following equation:

$$Q_1 = 0.6875 \sqrt{\frac{R(o) \prod_{n=1}^{n=4} (1-RC_n^2)}{1 \text{ FRAME}}}$$

where R(o) is the energy in the preemphasized speech signal (obtained from the matrix load routine) and n is the number of the RC.

The RC's are converted to prediction coefficients (PC_1, PC_2, PC_3, and PC_4) by the same recursion as is used for the DoD Standard LPC-10 algorithm. For example, the recursion for the fourth-order predictor is as follows:

First calculation: $P_1^{(1)} = RC$

Second calculation:
$$P_2^{(2)} = RC_2$$
$$P_1^{(2)} = P_1^{(1)} - RC_2P_1^{(1)}$$

Third calculation:
$$P_3^{(3)} = RC_3$$
$$P_2^{(3)} = P_2^{(2)} - RC_3P_1^{(2)}$$
$$P_1^{(3)} = P_1^{(2)} - RC_3P_2^{(2)}$$

Fourth calculation:
$$P_4^{(4)} = RC_4$$
$$P_3^{(4)} = P_3^{(3)} - RC_4P_1^{(3)}$$
$$P_2^{(4)} = P_3^{(2)} - RC_4P_2^{(3)}$$
$$P_1^{(4)} = P_3^{(1)} - RC_4P_3^{(3)}$$

$$PC_1 = P_1^{(4)},\ PC_2 = P_2^{(4)},\ PC_3 = P_3^{(4)},\ PC_4 = P_4^{(4)}$$

The final values of Q, SEGQ's, and the residual error are calculated in two steps:

(1) Q, ALPHA, TAU, and the PC's are used to implement the feedback loop shown in figure 14 and obtain a residual error function (labeled ERR 2 (ESTIMATE) in figure 8) from the first spin of the feedback loop. ERR 2 (ESTIMATE) (one point for each sample) is used to calculate the final Q and SEGQ's.
(2) The feedback loop is spun again using the final Q and SEGQ's to obtain the final (1-bit-per-sample) residual error (labeled ERR 2 (FINAL) in figure 8).

The formula used for calculating Q is

$$Q = 0.5\ (Q_1 + C < |\ \text{ERR 2 (ESTIMATE)}_n\ | >)$$

where n = 0, 1, 2, . . . 189; ERR 2 $(\text{ESTIMATE})_n$ indicates the frame average of the absolute magnitude of ERR 2 $(\text{ESTIMATE})_n$; and C equals 190/256.

Ten segment Q's are calculated as a ratio:

$$\text{SEGQ} = \frac{2\left(\sum_{N} \mid \text{ERR 2 (ESTIMATE)}_n \mid\right)}{Q}$$

where $N = 1, 2, \ldots 10$; $n = 1, 2, \ldots 19$; $\sum_{N} \mid \text{ERR 2 (ESTIMATE)}_n \mid$ is the sum of the residual for the Nth segment; and Q is the frame energy level.

Q is quantized and coded to five bits (see table XII) and the SEGQ's are coded to two bits each (see table XIII). The frame rms (Q) and SEGQ's are decoded (as indicated in tables XII and XIII, respectively) and multiplied together to obtain the segment Q values, Q_R:

$$Q_R = Q_N Q, \quad R = N = 1, 2, \ldots 10.$$

Finally, a ΔQ_R is obtained for each of eleven segments that run between the halfway points of the previously defined segments:

$$\Delta Q_0 = \frac{1}{9}(Q_1 - Q_0)$$

$$\Delta Q_R = \frac{1}{19}(Q_{R+1} - Q_R), \quad R = 1, 2, \ldots 9$$

$$\Delta Q_{10} = 0$$

where Q_0 is the segment Q of the last segment of the previous frame; ΔQ_0 applies to the first nine samples in the frame; ΔQ_{10} applies to the last nine samples of the frame; and ΔQ_R applies to the 19 samples between midpoints of the previously defined segments (i.e., samples 10 to 28; 29 to 47; etc.).

During the second spin, the value for the residual for each sample (Q_n) is calculated by the following equation:

$$Q_n = Q_{n-1} + \Delta Q_R$$

except that the center residual error sample of each segment is set to zero in order to save 10 transmission bits. The final residual error for transmission is set to +1 if the error is positive and −1 if the error is negative.

> NOTE: 180 bits are required since the center residual error for the 10 segments is not coded.

In the analysis performed by the transmitter, 60 values of TAU are calculated in the closed interval 20 to 156, with 20 values per octave. The value to be transmitted is selected and encoded with 6 bits. A transmitted 1 corresponds to a period of 20 samples, and a transmitted 60 corresponds to a period of 156 samples. (See table IX for the relationship between the coded pitch value, the pitch period, and the pitch frequency.) TAU is coded to zero when the encoded ALPHA is zero. Encoding of the remaining parameters in the APC analysis is presented in the tables as referenced in the following discussion. The pitch gain (ALPHA) is Gray coded as shown in table X. The reflection coefficients (RC's) are log-area-ratio coded as indicated in table XI. The error signal, frame quantization (Q), and segment quantization (SEGQ) are coded as described above and in tables XII and XIII.

An APC frame consists of 240 bits and the frame length is 25 ms. The frame bits are allocated as shown in table XIV and the bits encoded and assigned to the 240-bit frame for TAU, ALPHA, Q, SEGQ, RC's, error signal, synchronization, and forward error correction as shown in table XV. Five bits are allocated to forward error corrections (labeled CKT in table XV) using a (21, 16) code. (Refer to table XVI for the bits to be protected.) The ordering of the data bits to be protected is shown in figure 10 and a check bit is generated for each of the five fields shown. The check bits are set to 1 if the data bits in their respective fields are odd parity. The 240th bit (labeled SYNC in table XV) is the synchronization bit and alternates between 1 and 0 for successive frames.

50.2 **Receiver.** In the receiver (refer to figures 9 and 15), error correction is achieved by generating the check bits based on the received bits as described above for the transmitter, and both calculated check bits and received bits undergoing an exclusive OR (XOR) operation. The 5-bit result of the XOR is used as the index to table XVII, which gives the bit position (in the data word shown in figure 10) of the error to be corrected.

The decoding of the APC parameters, for convenience, is presented in the same tables as the encoding for the transmitter. TAU (to the 60 values) is decoded as shown in table IX, ALPHA as shown in table X, and the RC's are decoded as shown in table XI. Q and SEGQ are decoded as shown in tables XII and XIII, respectively. All decoding and error detection are accomplished by direct table lookup.

The error signal is generated based on the 180 residual error bits, the frame Q, and the 10-segment Q values. The error signal is interpolated between adjacent segmented Q values. The RC's are translated to PC's using the same algorithm used in the transmitter and described in the previous section on the transmitter.

The signal is deemphasized using the filter that is the inverse of that used for preemphasis in the transmitter:

$$\frac{1}{1 - Z^{-1} + 0.5\ Z^{-2}}$$

This deemphasis restores the speech spectrum balance and provides a high-frequency roll-off to the quantizing noise spectrum. The digital output of the deemphasis filter drives a D-A converter, the output of which is passed through a lowpass filter.

> NOTE: The Government standard APC-04 algorithm utilizes a synthesizer that is a direct-form transversal filter with 16-bit single precision accumulation of products; the input excitation level (error signal) is scaled up to reduce the quantizing and round-off effects in the filter. The PC's, scale factor, pitch, ALPHA, Q, and SEGQ's are updated on a frame basis. The PC's are scaled so that the largest is between 2^{14} and 2^{15}. The output is double-buffered to meet real-time input/output requirements of the special-purpose signal processor used to implement the algorithm.

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