

**NOT MEASUREMENT  
SENSITIVE**

**MIL-STD-3005  
20 December 1999**

**DEPARTMENT OF DEFENSE  
TELECOMMUNICATIONS SYSTEMS  
STANDARD**

**ANALOG-TO-DIGITAL CONVERSION  
OF VOICE BY 2,400 BIT/SECOND  
MIXED EXCITATION LINEAR PREDICTION  
(MELP)**



MIL-STD-3005

FOREWARD

1. This standard is approved for use by all Departments and Agencies of the Department of Defense (DoD) and is a replacement for FIPSPUB-137, Telecommunications: Analog to Digital Conversion of Voice by 2,400 Bit/Second Linear Predictive Coding.

2. This standard contains design requirements for analog-to-digital (A-D) conversion of voice by 2,400 bit/second Mixed Excitation Linear Prediction (MELP). Adherence to this standard is required to produce interoperable systems at the defined rate and to meet or exceed the minimum performance requirements.

3. Appendix A of this document contains an example of an interoperable MELP algorithm. This information is provided as guidance only.

4. Appendix B contains guidelines for verification of all new implementations of this standard. New implementations must be verified to guarantee that the standard was correctly implemented. This verification process will determine if the standard is interoperable with other MELP implementations and will verify that the performance of the implementation meets or exceeds the performance of the MELP reference coder.

5. Beneficial comments (recommendations, additions, deletions) and any pertinent data which may be of use in improving this document should be addressed to: R224, National Security Agency, 9800 Savage Road STE 6516, Ft. Meade, Maryland 20755-6516 by using the Standardization Document Improvement Proposal (DD Form 1426) appearing at the end of this document or by letter.

## MIL-STD-3005

## CONTENTS

<u>PARAGRAPH</u>	<u>PAGE</u>
FOREWARD .....	ii
CONTENTS .....	iii
1. SCOPE .....	1
1.1 Scope .....	1
2. APPLICABLE DOCUMENTS.....	1
2.1 General .....	1
2.2 Government Documents.....	1
2.2.1 Specifications, standards, and handbooks .....	1
2.2.2 Other Government documents, drawings, and publication .....	2
2.3 Other publications .....	2
2.4 Order of precedence .....	2
3. DEFINITIONS .....	2
3.1 Terms.....	2
3.1.1 Adaptive spectral enhancement.....	2
3.1.2 Aperiodic pulses.....	2
3.1.3 Fourier magnitude modeling.....	3
3.1.4 Hamming codes.....	3
3.1.5 Jitter.....	3
3.1.6 Linear prediction coding .....	3
3.1.7 Mixed excitation .....	3
3.1.8 Prediction coefficients.....	3
3.1.9 Pulse dispersion.....	3
3.1.10 Uniform quantizer .....	3
3.1.11 Weighted Euclidean distance .....	3
3.2 Acronyms used in this standard .....	3
4. GENERAL REQUIREMENTS .....	4
5. DETAILED REQUIREMENTS .....	4
5.1 General .....	4
5.2 Analog specification.....	4
5.3 Parameter quantization and encoding .....	4
5.3.1 Pitch and overall voicing.....	4
5.3.2 Bandpass voicing .....	5
5.3.3 Gain .....	5

## MIL-STD-3005

5.3.4	Linear prediction coefficients .....	6
5.3.5	Fourier magnitudes.....	6
5.3.6	Aperiodic flag.....	6
5.3.7	Uniform quantization .....	6
5.4	Error protection .....	7
5.5	Transmission format.....	7
5.5.1	Transmission rate .....	7
5.5.2	Bit allocation .....	7
5.5.3	Bit transmission order .....	8
6.	NOTES.....	8
6.1	Intended use .....	9
6.2	Patent notice .....	9
6.3	Subject term (key word) listing.....	9

APPENDIX

	MELP ALGORITHM DESCRIPTION .....	37
A.1	SCOPE .....	37
A.1.1	Scope .....	37
A.2	APPLICABLE DOCUMENTS.....	37
A.2.1	Government Documents.....	37
A.2.2	Other publications .....	37
A.2.3	Order of precedence .....	38
A.3	DEFINITIONS .....	38
A.3.1	Terms.....	38
A.3.2	Acronyms .....	38
A.4	GENERAL REQUIREMENTS .....	38
A.5	DETAILED REQUIREMENTS .....	38
A.5.1	General .....	38
A.5.2	Encoder .....	39
A5.2.1	Low frequency removal .....	39
A5.2.2	Integer pitch calculation.....	40
A5.2.3	Bandpass voicing analysis.....	40
A5.2.4	Fractional pitch refinement .....	41
A5.2.5	Aperiodic flag.....	41
A5.2.6	Linear prediction analysis .....	41
A5.2.7	Linear prediction residual calculation.....	41
A5.2.8	Peakiness calculation .....	42
A5.2.9	Final pitch calculation .....	42
A5.2.10	Pitch doubling check .....	43
A5.2.11	Gain calculation .....	44

## MIL-STD-3005

A5.2.12	Average pitch update.....	44
A5.2.13	Quantization of prediction coefficients.....	44
A5.2.14	Pitch quantization.....	45
A5.2.15	Gain quantization .....	45
A5.2.16	Bandpass voicing quantization.....	46
A5.2.17	Fourier magnitude calculation and quantization.....	46
A5.2.18	Error protection and bit packing .....	47
A5.3	Decoder .....	47
A5.3.1	Bit unpacking and error correction .....	47
A5.3.2	Noise attenuation.....	48
A5.3.3	Parameter interpolation .....	49
A5.3.4	Mixed excitation generation.....	49
A5.3.5	Adaptive spectral enhancement.....	51
A5.3.6	Linear prediction synthesis .....	51
A5.3.7	Gain adjustment .....	52
A5.3.8	Pulse dispersion.....	52
A5.3.9	Synthesis loop control .....	52
	PERFORMANCE VERIFICATION .....	53
B.1	SCOPE .....	53
B.1.1	Scope .....	53
B.2	APPLICABLE DOCUMENTS.....	53
B.2.1	Government documents .....	53
B.2.2	Other publications .....	53
B.2.3	Order of precedence .....	53
B.3	DEFINITIONS .....	53
B.3.1	Terms.....	53
B3.1.1	A/B Test .....	53
B.3.2	Acronyms used in this appendix .....	53
B.4	GENERAL REQUIREMENTS .....	54
B.4.1	General .....	54
B.5	DETAILED REQUIREMENTS .....	54
B.5.1	Formal evaluation.....	54
B5.1.1	Intelligibility tests.....	55
B5.1.2	Quality Tests .....	56
B.5.2	Bit equivalence.....	57
<u>FIGURE</u>		
A-1	MELP decoder block diagram .....	39
<u>TABLE</u>		
I	Encode/decode table for pitch and overall voicing parameter .....	5

## MIL-STD-3005

II	MELP bit allocation .....	7
III	MELP bit transmission order .....	8
IV	Codebooks used by the LSF multi-stage quantizer for stage 1 .....	11
V	Codebooks used by the LSF multi-stage quantizer for stage 2 .....	17
VI	Codebooks used by the LSF multi-stage quantizer for stage 3 .....	20
VII	Codebooks used by the LSF multi-stage quantizer for stage 1 .....	23
VIII	Codebooks used by the Fourier magnitude vector quantizer .....	26
A-I	Filter coefficients for bandpass filter .....	50
A-II	Filter coefficients for the pulse dispersion filter .....	52
B-I	Testbed coder configurations .....	54
B-II	Intelligibility and quality test conditions.....	55
B-III	Weights and thresholds for intelligibility conditions .....	56
B-IV	Weights for quality conditions .....	57
	CONCLUDING MATERIAL.....	58

## MIL-STD-3005

## 1. SCOPE

1.1 Scope. This standard establishes interoperability and performance requirements for analog-to-digital (A-D) conversion of voice by 2,400 bit/second Mixed Excitation Linear Prediction (MELP). The requirements presented in this document must be met in order for systems to be interoperable at 2,400 bit/second. Minimum performance requirements are also provided, but may be exceeded. The performance requirements are provided in Appendix B.

## 2. APPLICABLE DOCUMENTS

2.1 General. Documents listed in this section are required in order for the document user to fully understand the guidance being provided by this standard.

2.2 Government documents.

2.2.1 Specifications, standards, and handbooks. The following specifications, standards, and handbooks form a part of this document to the extent herein. Unless otherwise specified, the issues of these documents are those listed in the issue of the Department of Defense Index of Specifications and Standards (DoDISS) and supplement thereto, cited in the solicitation.

## STANDARDS

## FEDERAL

FED-STD-1016

Telecommunications: Analog to Digital Conversion of Radio Voice by 4,800 Bit/Second Code Excited Linear Prediction (CELP)

FED-STD-1037

Glossary of Telecommunications Terms

FIPSPUB-137

Telecommunications: Analog to Digital Conversion of Voice by 2,400 Bit/Second Linear Predictive Coding

## MILITARY

MIL-STD-188-113

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

(Unless otherwise indicated, copies of the above specifications, standards, and handbooks are available from the Standardization Document Order Desk, 700 Robbins Avenue, Building 4D, Philadelphia, PA 19111-5094.)

(Copies of the Federal Information Processing Standards (FIPS) are available to Department of Defense activities from the Standardization Document Order Desk, 700 Robbins Avenue, Building 4D, Philadelphia, PA 19111-5094. Others must request copies of FIPS from the National Technical Information Service, 5285 Port Royal Road, Springfield, VA 22161-2171.)

## MIL-STD-3005

**2.2.2 Other Government documents, drawings, and publication.** The following other Government documents, drawings, and publications form a part of this document to the extent specified herein. Unless otherwise specified, the issues are those cited in the solicitation.

DoDISS

Department of Defense Index of  
Specifications and Standards

(Copies of the DoDISS are available on a yearly subscription basis either from the Government Printing Office or the DoDSSP Subscription Services, 700 Robbins Avenue, Building 4D, Philadelphia, PA 19111-5094.)

**2.3 Other publications.** The following documents form a part of this standard to the extent specified herein. Unless otherwise specified, the issues of the documents which are DoD adopted should be those listed in the issue of the DoDISS specified in the solicitation. The issues of the documents which have not been adopted should be those in effect on the date of the cited DoDISS.

## NORTH ATLANTIC TREATY ORGANIZATION (NATO)

## STANDARDIZATION AGREEMENT (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 Samples

(Application for copies should be addressed to the Naval Publications and Forms Center, 5801 Tabor Avenue, Philadelphia, PA 19120-5099.)

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

**2.4 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 should take precedence. Nothing in this document, however, supersedes applicable laws and regulations unless a specific exemption has been obtained.

## 3. DEFINITIONS

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

**3.1.1 Adaptive spectral enhancement.** This feature enhances the formant structure of the synthetic speech by use of an adaptive spectral enhancement filter that is applied to the mixed excitation.

**3.1.2 Aperiodic pulses.** Aperiodic pulses are used in the excitation model of the synthesizer when the aperiodic flag is set to 1. The aperiodic flag is set to one when the jittery voiced state is encountered during the voicing decision process. This feature is used to reduce the buzzy quality of the synthetic speech signal.

## MIL-STD-3005

3.1.3 Fourier magnitude modeling. Fourier magnitude modeling involves determining the Fourier magnitudes of the first 10 pitch harmonics of the prediction residual and vector quantizing them with 8 bits for transmission. The use of this technique improves the accuracy of the speech production model at the perceptually important lower frequencies.

3.1.4 Hamming codes. A class of linear codes used for forward error correction. These codes are used only in the unvoiced mode.

3.1.5 Jitter. Random variations introduced into the duration of a signal.

3.1.6 Linear prediction coding. A method for approximating the current speech sample by using a linear combination of past and future speech samples. This method efficiently represents a speech signal and its spectrum characteristics with a very small number of parameters when combined with an appropriate excitation signal.

3.1.7 Mixed excitation. The combination of a periodic function (such as a pulse train) and random noise for use in the excitation model. This combination is applied to sub regions of the frequency domain of the excitation signal.

3.1.8 Prediction coefficients. A set of values that are calculated using a short segment of the input speech signal and provide an estimate of the spectral properties of that signal. These values are determined by performing linear prediction analysis on the input signal. The goal of the analysis is to produce values that minimize the short term mean-squared prediction error over the input segment.

3.1.9 Pulse dispersion. Uses a fixed filter to spread the excitation energy within a pitch period.

3.1.10 Uniform quantizer. A uniform quantizer uses levels and step sizes that are distributed uniformly.

3.1.11 Weighted Euclidean distance. The euclidean distance is a distortion measure between two vectors. In this standard the euclidean distance is determined by summing the squared difference between two vectors for a select number of samples. Normally the euclidean distance is the square root of the measure described in the previous sentence.

3.2 Acronyms used in this standard. The acronyms used in this standard are defined as follows:

A-D - Analog to Digital

DoD - Department of Defense

DoDISS - Department of Defense Index of Specifications and Standards

DoDSSP - Department of Defense Single Stock Point

FEC - Forward Error Correction

LPC - Linear Prediction Coding

LSB - Least Significant Bit

LSF - Line Spectrum Frequency

## MIL-STD-3005

MELP - Mixed Excitation Linear Predictions

MSB - Most Significant Bit

MSVQ - Multi-Stage Vector Quantizer

STANAG - Standardization Agreement

#### 4. GENERAL REQUIREMENTS

Not applicable

#### 5. DETAILED REQUIREMENTS

**5.1 General.** The Mixed Excitation Linear Prediction coder is based on the traditional Linear Prediction Coder (LPC) parametric model, but also includes five additional features. They are mixed excitation, aperiodic pulses, adaptive spectral enhancement, pulse dispersion, and Fourier magnitude modeling. A MELP frame interval is 22.5 ms  $\pm 0.01$  percent in duration and contains 180 voice samples (8,000 samples/second).

**5.2 Analog specification.** The recommended analog requirements for the MELP coder are for a nominal bandwidth ranging from 100 Hz to 3800 Hz. Although the MELP coder will operate with a more band limited signal, performance degradation will result. To ensure proper operation of the MELP coder, the A-D conversion process should produce peak values of (or near) -32768 and 32767. Additionally, the coder should have unity gain, which means that the output speech level should match that of the input speech.

**5.3 Parameter quantization and encoding.** The MELP parameters which are quantized and transmitted are the final pitch ( $P_3$ ); the bandpass voicing strengths ( $Vbp_i$ ,  $i = 1, 2, \dots, 5$ ); the two gain values ( $G_1$  and  $G_2$ ); the linear prediction coefficients ( $a_i$ ,  $i = 1, 2, \dots, 10$ ); the Fourier magnitudes; and the aperiodic flag. The use of the following quantization procedures is required for interoperability among various implementations.

**5.3.1 Pitch and overall voicing.** The final pitch ( $P_3$ ), and the low band voicing strength ( $Vbp_1$ ), are quantized jointly using 7 bits, as follows. If  $Vbp_1 \leq 0.6$ , then the frame is unvoiced and the all-zero code is sent. Otherwise, the log of  $P_3$  is quantized with a 99-level uniform scalar quantizer (see 5.3.7) ranging from log20 to log160. The resulting index (range 0 to 98) is then mapped to the transmitted 7-bit codeword using the encode/decode values in table I. All 28 codes with Hamming weight of 1 or 2 are reserved for error protection. This table is also used in decoding the 7-bit pitch code to determine if a frame is voiced, unvoiced, or whether a frame erasure is indicated. A frame is determined unvoiced if the pitch code is all zero or has only one bit set. If two bits are set, then a frame erasure is indicated. Otherwise, the voiced mode is used and the pitch index is determined from the received code according to table I.

**TABLE I. Encode / decode table for pitch and overall voicing parameter.**

<b>Code</b>	<b>Index</b>	<b>Code</b>	<b>Index</b>	<b>Code</b>	<b>Index</b>	<b>Code</b>	<b>Index</b>
0x0	UNVOICED	0x20	UNVOICED	0x40	UNVOICED	0x60	ERASURE
0x1	UNVOICED	0x21	ERASURE	0x41	ERASURE	0x61	68
0x2	UNVOICED	0x22	ERASURE	0x42	ERASURE	0x62	69
0x3	ERASURE	0x23	16	0x43	42	0x63	70
0x4	UNVOICED	0x24	ERASURE	0x44	ERASURE	0x64	71
0x5	ERASURE	0x25	17	0x45	43	0x65	72
0x6	ERASURE	0x26	18	0x46	44	0x66	73
0x7	0	0x27	19	0x47	45	0x67	74
0x8	UNVOICED	0x28	ERASURE	0x48	ERASURE	0x68	75
0x9	ERASURE	0x29	20	0x49	46	0x69	76
0xA	ERASURE	0x2A	21	0x4A	47	0x6A	77
0xB	1	0x2B	22	0x4B	48	0x6B	78
0x12	ERASURE	0x32	28	0x52	54	0x72	85
0x13	5	0x33	29	0x53	55	0x73	86
0x14	ERASURE	0x34	30	0x54	56	0x74	87
0x15	6	0x35	31	0x55	57	0x75	88
0x16	7	0x36	32	0x56	58	0x76	89
0x17	8	0x37	33	0x57	59	0x77	90
0x18	ERASURE	0x38	34	0x58	60	0x78	91
0x19	9	0x39	35	0x59	61	0x79	92
0x1A	10	0x3A	36	0x5A	62	0x7A	93
0x1B	11	0x3B	37	0x5B	63	0x7B	94
0x1C	12	0x3C	38	0x5C	64	0x7C	95
0x1D	13	0x3D	39	0x5D	65	0x7D	96
0x1E	14	0x3E	40	0x5E	66	0x7E	97
0x1F	15	0x3F	41	0x5F	67	0x7F	98

5.3.2 Bandpass voicing. When  $Vbp_1 > 0.6$ , the remaining bandpass voicing strengths are quantized to 1 if their value exceeds 0.6, and quantized to 0 otherwise. There is one exception. If the quantized values of  $Vbp_i$ ,  $I = 2, 3, 4, 5$  are 0001, respectively, then  $Vbp_5$  is quantized to 0. The quantized values are transmitted using 4 bits. When  $Vbp_1 \leq 0.6$ , the bandpass voicing bits are replaced with FEC parity bits.

5.3.3 Gain. Two gain parameters,  $G_1$  and  $G_2$ , are transmitted each frame.  $G_2$  is quantized to 5 bits using a 32-level uniform quantizer ranging from 10.0 to 77.0 dB. The quantizer index is the transmitted codeword.  $G_1$  is quantized to 3 bits using the following adaptive algorithm. This algorithm determines if the frame is a steady state frame or a transition frame. If  $G_2$ , for the current frame, is within 5 dB of  $G_2$  for the previous frame, and  $G_1$  is within 3 dB of the average of  $G_2$  values for the current and previous frames, then the frame is steady-state and a special code (all zero) is sent to indicate that the decoder should set  $G_1$  to the mean of the  $G_2$  values for the current and previous frames. Otherwise, the frame represents a

## MIL-STD-3005

transition and  $G_1$  is quantized with a 7-level uniform quantizer ranging from 6 dB below the minimum of the  $G_2$  values for the current and previous frames to 6 dB above the maximum of those  $G_2$  values. The all-zero codeword is sent for steady state frames and a 7-bit uniform quantizer is used for transition frames. In this case, the quantizer index plus 1 is the transmitted codeword. See 5.3.7 for details on the uniform quantizer.

**5.3.4 Linear prediction coefficients.** The linear prediction coefficients are converted into line spectrum frequencies (LSF) and the resulting LSF vector is checked for monotonicity. If the vector is not monotonic it is adjusted accordingly. The LSF vector is also checked for minimum separation of 50 Hz and adjusted accordingly. The resulting LSF vector is then quantized by a multi-stage vector quantizer (MSVQ). The MSVQ codebook consists of four stages whose indices have 7, 6, 6, and 6 bits, respectively.

The quantized LSF vector,  $\hat{f}$ , is the sum of the vectors selected by the search process, with one vector selected from each stage. The MSVQ search finds the codebook vector which minimizes the square of the weighted Euclidean distance,  $d^2$ , between the unquantized and quantized LSF vectors:

$$d^2(f, \hat{f}) = \sum_{i=1}^{10} w_i (f_i - \hat{f}_i)^2, \text{ where } w_i = \begin{cases} P(f_i)^{0.3}, & 1 \leq i \leq 8 \\ 0.64P(f_i)^{0.3}, & i = 9 \\ 0.16P(f_i)^{0.3}, & i = 10 \end{cases}, \text{ EQUATION 1,}$$

$f_i$  is the  $i^{\text{th}}$  component of the unquantized LSF vector, and  $P(f_i)$  is the inverse prediction filter power spectrum evaluated at frequency  $f_i$ . The indices of the four vectors are transmitted. The code vectors and corresponding indices are provided in tables IV-VII.

**5.3.5 Fourier magnitudes.** The ten Fourier magnitudes are coded with an 8-bit vector quantizer. The index of the code vector, which minimizes the weighted Euclidean distance between the input and code vectors, is transmitted. The weights are fixed and are given by:

$$w_i = [117/(25 + 75(1 + 1.4(f_i/1000)^2)^{0.69})]^2, \quad i = 1, 2, \dots, 10, \text{ EQUATION 2,}$$

where  $f_i = 8000i/60$  is the frequency in Hz corresponding to the  $i^{\text{th}}$  harmonic for a default pitch period of 60 samples. The code vectors and corresponding indices are given in table VIII.

**5.3.6 Aperiodic flag.** The aperiodic flag is a single bit, transmitted as is. The aperiodic flag is set to 1 if  $V_{bp1} < 0.5$  and set to 0 otherwise. When set, this flag tells the decoder that the pulse component of the excitation should be aperiodic, rather than periodic.

**5.3.7 Uniform quantization.** The pitch and gain quantization processes employ uniform quantizers which operate as follows. The stepsize for an  $n$ -level quantizer ranging from  $x_1$  to  $x_2$  is  $s = (x_2 - x_1)/(n - 1)$ . The  $n$  quantizer output values are  $x_1 + i \cdot s$ ,  $i = 0, 1, \dots, n-1$ . The threshold values between levels  $I$  and  $I+1$  are  $x_1 + (0.5 + i)s$ ,  $I = 0, 1, \dots, n-2$ . The quantizer produces  $n$  indices, 0, 1, ...,  $n-1$ , which correspond to an increasing value of the parameter being quantized. For example, let  $x_1 = 1$ ,  $x_2 = 7$ , and  $n = 7$ . This gives  $s = 1$ , levels of 1, 2, ..., 7, and thresholds of 1.5, 2.5, ..., 6.5. Index 0 is assigned to input values  $x$ , for which  $x \leq 1.5$ ; index 1 is assigned to input values for which  $1.5 \leq x \leq 2.5$ ; etc.

## MIL-STD-3005

**5.4 Error protection.** Forward Error Correction (FEC) is implemented in the unvoiced mode only, when the Fourier magnitudes, bandpass voicing, and jitter bits need not be transmitted. FEC replaces those 13 bits with the parity bits of three Hamming (7,4) codes and one Hamming (8,4) code. These codes protect the first stage LSF index (7 bits) and both gain indices (8 bits); there is one spare information bit, set to 0.

The protected bits are placed into a column vector,  $u$ , which post-multiplies the parity generator matrix to produce the n-bit parity vector,  $p = [p_0 \ p_1 \ \dots \ p_{n-1}]^T$ , where n is 3 or 4. The parity generator matrix for the

Hamming (7,4) code is:  $G_{7,4} = \begin{bmatrix} 1101 \\ 1011 \\ 0111 \end{bmatrix}$ . The parity generator matrix for the Hamming (8,4) code is:

$$G_{8,4} = \begin{bmatrix} 1101 \\ 1011 \\ 0111 \\ 1110 \end{bmatrix}.$$

The 4 most significant bits (MSBs) of the first stage LSF index ( $u = [b_6 \ b_5 \ b_4 \ b_3]^T$ ) are protected by the (8,4) code, with the 4 parity bits written to the LSBs of the bandpass voicing index ( $p_0 \ p_1 \ p_2 \ p_3$ ). The remaining 3 bits of the first stage index and the spare bit ( $u = [b_2 \ b_1 \ b_0 \ 0]^T$ ) are protected with 3 parity bits written to the MSB's of the Fourier magnitude index ( $p_0 \ p_1 \ p_2$ ). The 4 MSBs of the second gain index ( $u = [b_4 \ b_3 \ b_2 \ b_1]^T$ ) are protected with 3 parity bits written to the next 3 bits of the Fourier magnitude index ( $p_0 \ p_1 \ p_2$ ). The LSB of the second gain index and the 3 bit first gain index ( $u = [b_0 \ b_2 \ b_1 \ b_0]^T$ ) are protected with 3 parity bits written to the 2 LSBs of the Fourier magnitude index ( $p_0 \ p_1$ ) and the aperiodic flag ( $p_2$ ). The parenthesized groups of parity bits show their placement in the given index, with the right-most bit having the least significance.

**5.5 Transmission format.** This section provides information on the transmission rate for the coder, the number of bits allocated for each MELP frame and the transmission order for the bits in each MELP frame.

**5.5.1 Transmission rate.** The transmission rate should be  $2,400 \text{ bits/s} \pm 0.01 \text{ percent}$ . Since all frames contain 54 bits, the frame length is  $22.5 \text{ ms} \pm 0.01 \text{ percent}$ .

**5.5.2 Bit allocation.** Table II shows how the 54 bits in an MELP frame are allocated among the parameters.

<b>TABLE II. MELP bit allocation.</b>		
<b>Parameters</b>	<b>Voiced</b>	<b>Unvoiced</b>
LSF's	25	25
Fourier Magnitudes	8	-
Gain (2 per frame)	8	8
Pitch, overall voicing	7	7
Bandpass Voicing	4	-
Aperiodic Flag	1	-
Error Protection	-	13
Sync Bit	1	1
Total Bits / 22.5 ms Frame	54	54

## MIL-STD-3005

5.5.3 Bit transmission order. Table III shows the transmission order for the 54 bits in each MELP frame for both voiced and unvoiced frames. The sync bit alternates between 0 and 1 from frame to frame.

<b>TABLE III. MELP bit transmission order.</b>								
<b>Bit</b>	<b>Voiced</b>	<b>Unvoiced</b>	<b>Bit</b>	<b>Voiced</b>	<b>Unvoiced</b>	<b>Bit</b>	<b>Voiced</b>	<b>Unvoiced</b>
1	G(2)-1	G(2)-1	19	LSF(1)-7	LSF(1)-7	37	G(1)-1	G(1)-1
2	BP-1	FEC(1)-1	20	LSF(4)-6	LSF(4)-6	38	BP-3	FEC(1)-3
3	P-1	P-1	21	P-4	P-4	39	BP-2	FEC(1)-2
4	LSF(2)-1	LSF(2)-1	22	LSF(1)-6	LSF(1)-6	40	LSF(2)-2	LSF(2)-2
5	LSF(3)-1	LSF(3)-1	23	LSF(1)-5	LSF(1)-5	41	LSF(3)-4	LSF(3)-4
6	G(2)-4	G(2)-4	24	LSF(2)-6	LSF(2)-6	42	LSF(2)-3	LSF(2)-3
7	G(2)-5	G(2)-5	25	BP-4	FEC(1)-4	43	LSF(3)-3	LSF(3)-3
8	LSF(3)-6	LSF(3)-6	26	LSF(1)-4	LSF(1)-4	44	LSF(3)-2	LSF(3)-2
9	G(2)-2	G(2)-2	27	LSF(1)-3	LSF(1)-3	45	LSF(4)-4	LSF(4)-4
10	G(2)-3	G(2)-3	28	LSF(2)-5	LSF(2)-5	46	LSF(4)-3	LSF(4)-3
11	P-5	P-5	29	LSF(4)-5	LSF(4)-5	47	AF	FEC(4)-3
12	LSF(3)-5	LSF(3)-5	30	FM-1	FEC(4)-1	48	LSF(4)-2	LSF(4)-2
13	P-6	P-6	31	LSF(1)-2	LSF(1)-2	49	FM-5	FEC(3)-3
14	P-2	P-2	32	LSF(2)-4	LSF(2)-4	50	FM-4	FEC(3)-2
15	P-3	P-3	33	FM-8	FEC(2)-3	51	FM-3	FEC(3)-1
16	LSF(4)-1	LSF(4)-1	34	FM-7	FEC(2)-2	52	FM-2	FEC(4)-2
17	P-7	P-7	35	FM-6	FEC(2)-1	53	G(1)-3	G(1)-3
18	LSF(1)-1	LSF(1)-1	36	G(1)-2	G(1)-2	54	SYNC	SYNC

NOTES:	G = Gain	BP = Bandpass Voicing
P = Pitch/Voicing		LSF = Line Spectral Frequencies
FEC = Forward Error Correction Parity Bits		FM = Fourier Magnitudes
Bit 1 = least significant bit of data set		AF = Aperiodic Flag
Highlighted Bits = 24 Most Significant MELP Bit		

## 6. NOTES

(This section contains information of a general or explanatory nature that may be helpful, but is not mandatory.)

MIL-STD-3005

6.1 Intended use. This standard specifies minimum operability and performance characteristics for analog-to-digital conversion by 2,400 bit/second MELP to be used in the design and installation of new communications subsystems and equipment and in authorized upgrading of existing communications subsystems and equipment. This standard is intended to replace FIPSPUB-137.

6.2 Patent notice. The Government has government purpose license rights under the following listed patents for the benefit of manufacturers of the item for the Government or for use in equipment to be delivered to the Government.

Awarded:

Mixed Excitation Linear Prediction with Fractional Pitch, U.S. Patent Number 5,699,477

Signal Quantizer wherein Average Level Replaces Subframe, U.S. Patent Number 5,794,180

Pending:

Multi-Stage Vector Quantization with Efficient Codebook Search

Adaptive Filter and Filtering Method for Low Bit Rate Coding

6.3 Subject term (key word) listing.

2.4 kbps

2400 bps

Analog-to-digital (A-D) conversion

Encoder/decoder, MELP

Linear prediction coefficients

Low rate

MELP

MELP analyzer

MELP synthesizer

Mixed Excitation Linear Prediction (MELP)

Voice compression

Scalar quantization

Speech coding

Speech compression

MIL-STD-3005

Vector quantization

TABLE IV. Codebooks used by the LSF multi-stage quantizer for stage 1 (component values are in Hertz).

Index	Vector	
0x0	355.243052	492.660028
0x1	640.392668	823.554944
0x2	581.990372	733.911476
0x3	694.610176	944.914844
0x4	388.049792	556.748868
0x5	620.798580	850.207244
0x6	489.979380	640.111288
0x7	472.778640	726.418032
0x8	454.914768	599.867328
0x9	553.856784	797.270016
0xa	513.719596	672.660540
0xb	602.011044	806.724744
0xc	414.160796	594.258332
0xd	429.559640	614.610428
0xe	453.999408	578.280580
0xf	482.206380	640.568984
0x10	440.262800	705.940108
0x11	620.421768	768.090716
0x12	496.800000	672.316592
0x13	676.518572	826.792700
0x14	387.165300	573.545624
0x15	501.412328	704.314976
0x16	495.552612	721.780700
0x17	611.589924	768.577188



**TABLE IV. Codebooks used by the LSF multi-stage quantizer for stage 1 (component values are in Hertz) - Continued.**

Index	Vector
0x30	262.027288 382.135568 656.530136 1474.423620 1840.272900 2051.403696 2404.401200 2679.250592 3166.215332 3389.927400
0x31	261.244876 531.776124 972.450092 1439.603064 1863.321332 2137.972540 2506.207492 2793.512032 3204.078640 3486.612756
0x32	339.674328 453.648992 751.280204 1389.907636 1635.730336 1865.774876 2173.773284 2553.385268 3065.628208 3273.235436
0x33	341.587388 661.908516 1017.171716 1394.366888 1663.345632 1920.173796 2203.118924 2548.413908 2988.712388 3297.386536
0x34	230.146968 343.667980 551.302016 859.858328 1580.424940 1832.179784 2240.921664 2617.020472 3037.2238376 3423.404416
0x35	235.039356 364.897428 689.403960 1029.605932 1665.887764 1948.938724 2346.250084 2648.456792 3069.623524 3389.224348
0x36	308.913508 449.624692 663.772816 1145.008812 1351.964772 1598.142492 1852.252004 2454.309724 2963.855000 3252.933440
0x37	257.909564 462.387712 741.203612 1052.100868 1358.228760 1622.148340 2351.586768 2675.735168 3092.801340 3413.616912
0x38	245.750260 376.103252 575.477096 922.024244 1670.456784 1903.208808 2267.835800 2608.720112 3005.315992 3260.555008
0x39	247.920760 379.176512 646.023984 1306.141840 1522.275312 2009.603136 2373.209384 2678.976628 3272.521600 3480.612912
0x3a	252.521964 397.542288 729.733388 1092.573888 1541.289920 1831.6388300 2172.991212 2432.845540 2863.515608 3353.231332
0x3b	293.760896 453.119144 952.543932 1252.249068 1524.138240 1854.489388 2240.400708 2529.295340 2980.645980 3332.247948
0x3c	194.884480 303.136856 524.001072 885.984548 1473.173076 1919.383060 2355.834416 2617.938748 3072.727588 3484.182104
0x3d	221.380896 339.827248 612.828448 974.216724 1296.283708 1802.879000 2241.073592 2652.630088 3095.005464 3428.844228
0x3e	346.441260 477.961284 659.909792 1048.696260 1321.778992 1560.564952 1860.835916 2137.948484 3028.669084 3355.727232
0x3f	306.865700 452.086676 606.287508 958.052088 1271.102660 1499.649856 2168.586976 2510.451536 2942.205140 3376.739692
0x40	367.577380 588.407212 947.277508 1401.718148 1710.042324 2095.772072 2530.043540 2877.200112 3307.829148 3553.939676
0x41	528.676400 819.913676 1280.496720 1637.511820 2066.765212 2453.835716 2824.688768 3094.627148 3371.254420 3558.843104
0x42	545.224416 669.303732 909.509712 1447.146740 1725.206608 1977.595112 2404.918460 2678.227616 3059.974176 3374.444120
0x43	502.175120 755.642512 1138.330556 1488.741484 1860.131964 2268.931212 2653.297860 2991.633600 3339.490956 3555.739556
0x44	298.315792 427.893476 578.377140 1139.800860 1526.635152 1746.979136 2250.458972 2556.079216 2919.105144 3394.443520
0x45	265.345712 679.203144 917.211964 1262.558816 1777.907428 2105.336432 2492.856668 2845.667064 3222.134752 3475.157160
0x46	459.013600 591.152844 773.066500 1211.262108 1477.162912 1731.845324 2117.279752 2387.897856 3042.563168 3318.468632
0x47	465.500052 607.671140 777.227012 1332.425448 1605.648308 1856.328328 2352.959464 2590.842552 3102.572804 3381.365096
0x48	362.053116 490.137184 745.960236 1269.985928 1492.284812 2059.254896 2441.930800 2750.527340 3227.946188 3436.200692



TABLE IV. Codebooks used by the LSF multi-stage quantizer for stage 1 (component values are in Hertz) - Continued.

Index	Vector						
0x62	429.353204	649.857460	1241.313404	1625.726460	1852.896036	2164.172340	2490.051440
0x63	530.200920	767.252124	1229.137072	1718.003588	2078.513660	2358.501028	2693.879268
0x64	180.250056	286.314476	782.009868	1256.403520	1607.783616	1964.381668	2428.505580
0x65	363.046928	495.176544	932.244352	1723.729152	2069.873516	2266.036024	2598.054636
0x66	275.456260	401.919616	743.453280	1360.345704	1599.375664	1867.905052	2173.127540
0x67	497.729776	703.060684	1036.341080	1397.418484	1838.024488	2064.417060	2475.461776
0x68	218.701004	364.654668	847.953432	1455.778028	1822.566468	2271.184816	2688.881072
0x69	409.265696	722.538580	1459.908136	1792.828940	2125.621812	2424.411352	2670.935304
0x6a	458.000836	574.728436	974.289496	1430.359792	1689.177736	1976.396512	2406.885924
0x6b	522.480632	818.682484	1332.380228	1637.908688	1904.114776	2177.842412	2487.380184
0x6c	215.477808	328.983084	581.950052	997.003380	1496.216816	1965.586408	2358.623932
0x6d	211.424108	336.110144	846.609096	1295.560356	1629.676036	2078.982732	2435.617148
0x6e	301.433204	429.696220	623.869464	1219.478592	1673.674336	1872.317560	2228.603416
0x6f	227.796312	619.235776	903.737200	1186.101972	1486.952480	1881.668384	2408.921256
0x70	246.964648	410.663624	963.556232	1336.930556	1805.419368	2107.567904	2494.256680
0x71	311.054428	696.503640	1198.813572	1667.222600	2009.052656	2402.897624	2794.768432
0x72	376.233832	487.344044	957.121240	1293.941648	1542.461412	1934.271352	2279.849188
0x73	420.577340	782.720780	1157.816496	1409.439164	1730.193544	2036.551884	2542.586672
0x74	227.450880	354.217032	645.854024	1103.949568	1457.017632	1735.123748	2145.927528
0x75	298.009992	505.689304	804.054936	1230.271608	1782.344852	2092.537116	2531.424824
0x76	361.837248	481.881860	754.778976	1157.999988	1373.736592	1603.401296	1872.798780
0x77	363.145448	654.603360	851.016964	1201.750484	1524.731340	1765.555460	2194.217812
0x78	244.837812	378.566316	819.184860	1063.961972	1389.825920	1926.440476	2268.995564
0x79	289.878276	678.873488	1094.283708	1378.083076	1669.550396	2126.537152	2546.927840
0x7a	291.135372	603.970104	806.647404	1146.202408	1492.922632	1738.308380	2185.345584

**TABLE IV. Codebooks used by the LSF multi-stage quantizer for stage 1 (component values are in Hertz) - Continued.**

<b>Index</b>	<b>Vector</b>							
0x7b	477.451948	711.1119268	1120.567768	1400.126952	1652.650504	1928.385468	2281.368392	2627.063532
0x7c	197.859932	317.615912	561.566488	1004.156772	1272.079512	1786.734928	2269.843440	2576.009008
0x7d	209.857268	338.088960	742.900840	1010.365752	1329.130948	1933.476212	2274.220164	2550.770696
0x7e	371.540116	517.541568	699.734220	1056.769240	1360.004676	1581.063852	1896.072308	2138.187064
0x7f	421.325684	565.415952	733.196828	987.054432	1189.015404	1594.283500	2041.152924	2300.535092
								3176.614820
								3416.630796

**TABLE V. Codebooks used by the LSF multi-stage quantizer for stage 2 (component values are in Hertz).**

Index	Vector						
0x0	-1.083480	-11.578352	2.306316	-13.996316	-69.456300	37.766424	-72.555080
0x1	-49.649148	-46.254812	-78.096196	-42.879304	87.142444	6.807124	-84.402392
0x2	-64.871376	-74.238856	-98.4223948	73.693940	8.272440	-0.445588	-69.719076
0x3	-76.670260	-104.000124	162.718804	146.267556	35.949460	-54.029504	-165.439464
0x4	79.209324	69.866840	42.538208	-8.700224	-41.525928	-87.881980	-172.058432
0x5	46.523288	32.708904	16.3772320	87.959672	17.893436	-31.334540	-158.771560
0x6	-24.884988	89.877856	60.112892	-46.721488	-88.565924	-123.916356	-98.926960
0x7	12.216416	-10.885964	50.445364	113.287312	44.675968	7.914896	-81.805228
0x8	-18.702440	-28.96588	-107.616672	-54.971216	189.132464	132.336076	-7.024808
0x9	-56.894540	-72.024184	-52.360740	193.625260	129.735560	62.003412	-71.477448
0xa	-60.650768	-78.194956	-63.935496	64.655352	-25.341792	84.625644	73.669276
0xb	-15.112784	-31.196064	3.849784	159.4312	93.375988	58.684128	-41.109156
0xc	19.254800	29.158032	-18.931012	-97.893236	63.340576	-10.453596	-104.093976
0xd	43.597972	19.186704	57.392544	116.543328	78.867444	26.069228	-78.842272
0xe	-80.512424	6.911116	10.447620	-20.670668	122.874212	64.969424	-20.095012
0xf	15.133536	-11.533644	91.378952	186.566144	151.176812	106.171932	-10.561384
0x10	-22.857064	-45.429420	43.667084	21.242924	-51.172664	132.138704	25.815592
0x11	-34.458148	-37.767448	-50.349484	-21.896640	91.649508	17.432224	65.073496
0x12	-22.611732	-44.342752	65.095116	32.444480	-57.425188	-5.691424	138.519252
0x13	-54.683340	-52.559476	144.402736	36.354708	-66.256948	-121.447840	-47.001024
0x14	76.995752	60.130164	57.113272	18.707744	-53.600588	62.799420	-40.517780
0x15	25.816108	31.076704	124.868068	63.559872	51.002984	-27.654288	-97.466628
0x16	-45.228580	102.897240	83.947112	74.901616	49.974708	8.687204	48.971348
0x17	61.045120	182.798948	159.671368	59.777160	29.855824	-31.491100	-95.978064

TABLE V. Codebooks used by the LSF multi-stage quantizer for stage 2 (component values are in Hertz) - Continued.

Index	Vector							
0x18	49.646432	28.451656	67.128192	0.886752	-49.083656	188.228948	106.267192	13.855044
0x19	-10.121400	-10.494132	-51.702072	-102.624176	212.475096	189.695604	82.285124	56.342688
0x1a	36.358408	21.779392	42.357136	2.492964	-42.223212	230.770844	160.695752	154.000972
0x1b	-29.676572	-39.223244	-35.661060	94.983248	59.318312	2.700204	207.235004	137.851652
0x1c	122.140104	120.913744	108.201320	67.247396	59.132704	40.026908	43.540560	8.987920
0x1d	72.657632	68.934704	16.956396	25.957480	190.086812	142.948388	53.217716	2.736948
0x1e	25.878136	32.075092	-18.051520	129.615436	41.005184	141.842100	80.837536	-5.850756
0x1f	56.950488	37.665072	132.514492	135.379112	75.671416	138.633300	118.390580	101.879640
0x20	22.106900	21.059164	-34.232504	-93.793160	-155.416992	-194.590552	20.038956	-23.649856
0x21	-26.033500	-35.961000	-61.732092	-162.427040	-26.287444	-98.269260	-141.894620	95.126388
0x22	-31.289284	-49.998772	-67.781776	-161.027220	-175.279260	-225.390404	-61.840204	0.995364
0x23	-86.553744	-108.489556	-17.654176	-72.132484	-7.564812	8.047932	-86.936256	27.160092
0x24	17.779300	15.339312	-36.725424	-8.881500	-102.852904	-144.867340	-94.816980	-196.28732
0x25	-19.692492	-30.220916	-29.782548	17.903896	-49.210684	-116.668528	-256.575612	-267.358620
0x26	-19.207108	4.375076	21.642788	-48.371256	-77.052304	-161.254660	-158.684868	250.771480
0x27	-16.784044	-37.622168	5.782012	-19.570792	-88.589344	-117.076008	-236.655112	-78.102088
0x28	-33.759476	-44.175904	-74.702140	-135.044380	-188.927648	7.016552	82.302568	25.474344
0x29	-86.590864	-120.632148	-148.796704	-150.115860	-18.258660	-84.054596	2.635176	37.246548
0x2a	-70.539400	-83.361280	-55.232632	-27.094692	-100.538976	-172.089768	53.281532	-13.033652
0x2b	-51.370904	-79.509096	-141.551088	1.691600	57.508000	-53.104592	20.961464	69.160500
0x2c	-26.397896	-22.278824	-58.177544	-99.990728	-151.064636	-22.789692	-120.173196	-151.092736
0x2d	20.239624	4.006208	-28.664352	-11.823076	-26.946228	-66.325064	-59.575592	-61.081748
0x2e	-32.526320	6.534672	-84.214076	-41.573904	-42.894712	-81.510608	52.840208	-43.398352
0x2f	5.449508	-14.963136	-41.741028	103.234668	14.390108	-64.423760	100.949988	29.811640
							-101.778792	-32.993828

**TABLE V. Codebooks used by the LSF multi-stage quantizer for stage 2 (component values are in Hertz) - Continued.**

<b>Index</b>	<b>Vector</b>						
0x30	32.295632	18.407260	-5.894076	-74.483888	-152.963476	-204.044336	205.952264
0x31	18.857140	38.485396	-0.989396	-55.047384	-48.776448	-130.013268	107.102480
0x32	-0.859316	-10.029480	-5.970772	-37.158360	-112.881804	25.395820	308.705512
0x33	-74.718268	-7.140804	45.716792	5.980372	66.215188	4.203068	147.785360
0x34	66.596548	54.383604	27.843772	2.162152	-78.289776	-136.087776	83.808688
0x35	44.048796	18.914496	31.356536	6.002560	-53.040524	-33.569576	-141.531972
0x36	76.666832	98.370332	72.228980	-23.754456	-51.177288	-139.149128	-102.191076
0x37	-24.797384	74.266648	59.426828	-45.806184	45.466244	-18.249196	-27.625136
0x38	3.038084	-17.986644	-39.902620	-148.611816	-165.180916	177.990356	106.139356
0x39	-44.622868	-69.180132	-94.306064	-155.434984	79.510788	55.263412	99.390112
0x3a	8.528100	-16.482720	-20.536552	-119.745020	-107.365876	343.542148	310.162272
0x3b	-74.730600	-98.813932	-83.739676	156.816160	172.775428	133.228172	117.447612
0x3c	101.110828	100.613404	64.945400	25.583900	-51.148460	26.378476	-3.647860
0x3d	27.756364	14.300488	-39.955176	-107.269680	92.431928	47.216472	-69.001244
0x3e	71.977736	63.828088	-0.871592	-13.855768	26.229036	-35.582384	133.902408
0x3f	21.330804	14.734456	35.994576	35.047512	175.516392	109.080608	55.660600

TABLE VI. Codebooks used by the LSF multi-stage quantizer for stage 3 (component values are in Hertz).

Index	Vectors					
0x0	32.980820	12.950916	-77.521608	3.457884	21.115608	-18.522344
0x1	20.015652	21.871904	-34.912312	79.791640	42.794480	37.292348
0x2	14.547848	-5.648276	8.130888	-50.548700	-72.212136	99.232328
0x3	46.408076	31.719192	13.758460	3.196728	-60.667004	96.666596
0x4	-6.354040	-25.223476	-47.876560	-60.932144	82.686748	36.111092
0x5	-29.084848	25.116264	-32.724520	36.106892	14.532280	-62.664768
0x6	-0.834372	-21.045296	5.294624	-4.912748	-69.596788	125.862880
0x7	25.715576	39.154404	-28.547164	-23.826392	-71.987108	80.416488
0x8	-21.216668	0.919060	-54.469596	0.970748	47.454988	79.039928
0x9	-19.268628	-34.590936	16.035816	29.048544	85.936380	66.712128
0xa	-14.294308	-28.466636	24.792780	-69.392596	1.677280	112.654604
0xb	32.723716	17.351404	63.382408	13.164984	7.863884	32.270756
0xc	27.919388	2.081884	-40.560004	-85.742272	45.749008	26.365512
0xd	-103.942848	30.868692	22.905368	3.245652	-10.864848	25.740888
0xe	-13.644212	-33.497264	7.802936	-32.739608	-68.895376	-14.978252
0xf	-61.220672	-60.013976	103.693816	39.152964	37.872412	55.039632
0x10	32.505808	15.583912	1.784144	53.673772	73.051196	41.469356
0x11	11.483176	5.944724	17.110496	14.790684	6.431540	-20.200572
0x12	43.566172	18.989652	55.144036	-3.929840	-37.379720	61.812216
0x13	37.469540	28.346816	15.723080	-28.085444	-6.623116	-19.773952
0x14	1.120132	-4.159908	-14.437336	-127.887976	77.027304	45.315748
0x15	44.768968	30.467760	-6.543172	-95.704460	26.850796	-15.394560
0x16	-10.057328	2.399412	1.061956	-65.695228	20.687476	-14.959004
0x17	49.644532	65.386288	9.684648	18.181104	-14.139504	-5.888800

**TABLE VI. Codebooks used by the LSF multi-stage quantizer for stage 3 (component values are in Hertz) - Continued.**

Index	Vectors							
0x18	-16.380640	-34.278140	0.022544	56.067304	46.892096	30.245204	-28.026068	93.529428
0x19	-4.060036	-29.094112	-52.545100	79.031108	69.252092	14.267812	9.463936	-65.250772
0x1a	4.583084	-24.872664	107.840788	43.419736	-1.250280	-4.026280	-63.961776	21.494468
0x1b	2.165696	-29.151980	54.527780	94.168364	66.493356	36.958832	1.594416	-62.430724
0x1c	-11.601520	-21.077228	-18.656652	-63.311928	81.936576	29.605552	-50.989424	-103.844388
0x1d	-20.789816	23.424964	27.131436	-24.919604	70.569748	26.853608	-31.525276	-87.182472
0x1e	-49.936228	-3.446516	40.200364	-18.647372	21.569628	-58.944872	-17.371572	-22.311184
0x1f	-57.177432	62.058144	36.515704	55.561188	6.520588	6.297884	-13.632288	-52.826640
0x20	6.249736	5.520744	-30.373784	-42.147600	20.147836	-39.570052	54.020328	-16.561300
0x21	-35.205168	5.085324	-51.336656	22.956972	-28.242884	-23.552672	32.053652	-17.893740
0x22	24.945192	13.164164	11.748732	-76.478132	-4.066148	-22.276580	79.811928	50.662004
0x23	40.838864	33.272060	3.666024	12.769580	-45.868448	-68.480588	92.137976	47.209512
0x24	-13.862480	-29.68380	-16.062832	-30.689744	-21.411228	-15.875376	131.557968	101.900708
0x25	-20.078528	-17.911912	-24.694752	-20.037984	-59.345372	-122.612400	71.623648	21.364328
0x26	13.955236	-13.670672	-0.176880	-47.697124	-55.913972	-56.124192	62.862572	-2.427248
0x27	40.271076	23.012344	27.510152	-41.497056	-55.530680	-58.212060	8.071216	-61.794624
0x28	8.497868	-2.606700	-30.219332	78.619552	30.672148	61.511100	30.235552	110.460236
0x29	-20.167520	-14.445504	-9.336176	127.582768	-4.294328	-27.401844	61.786412	21.701036
0x2a	-1.647608	54.368956	37.237244	-17.980500	42.437252	27.778416	60.653436	59.946944
0x2b	13.352932	-14.668488	56.844876	39.456184	25.357476	11.232768	96.450116	63.098048
0x2c	16.214176	14.498636	-9.273052	-9.045960	-27.283232	-48.963732	39.335716	193.888504
0x2d	-47.007404	-53.423716	9.553404	0.589152	-50.564088	-1.666356	-7.432612	56.997036
0x2e	-36.631948	37.397136	27.393352	-30.540160	-93.368500	-20.053984	-17.314104	-0.911252
0x2f	-1.395788	-13.464588	93.489772	-1.615688	-30.138632	-48.064896	15.280056	66.923052

**TABLE VI. Codebooks used by the LSF multi-stage quantizer for stage 3 (component values are in Hertz) - Continued.**

<b>Index</b>	<b>Vectors</b>						
0x30	10.569172	24.809296	-60.304192	-48.704888	-38.084248	-30.716660	-41.121760
0x31	14.329408	2.787968	1.889624	17.819504	-36.893736	-131.985160	-0.567740
0x32	27.387340	7.644616	27.150636	-5.574300	-36.398996	-76.648084	-129.163304
0x33	51.258220	25.407856	15.030632	72.681664	-0.275320	-33.546500	-44.555092
0x34	-25.149456	-55.373592	-20.508880	-59.104064	39.694332	-29.855136	23.338884
0x35	-24.028792	-33.787436	7.601280	-28.737668	-9.413580	-62.113404	-56.180560
0x36	27.417444	7.533384	0.415240	-4.640872	-21.214400	-41.096728	-104.321976
0x37	34.088864	14.909740	58.394348	39.166456	2.694692	-32.094636	-33.619456
0x38	-13.082648	-30.073608	-85.362912	6.847348	-46.908688	36.769352	43.168012
0x39	-41.948776	1.387452	-52.071212	73.038768	4.876640	-17.894024	-51.727328
0x3a	-12.247768	-11.237092	-44.331148	30.028040	-39.230748	-2.865544	-119.814076
0x3b	8.651028	-16.327560	0.858176	49.272304	-33.946904	45.591748	-8.960968
0x3c	-12.528140	-28.219792	-66.117904	19.523384	2.767992	-68.743624	-86.900348
0x3d	-18.846356	-40.824100	-12.869724	6.287764	-18.501512	23.048256	-60.021784
0x3e	-65.678876	21.048360	10.027512	27.274816	12.681548	-38.586556	-102.199520
0x3f	30.881868	20.481092	5.767264	58.983840	24.991944	11.221364	42.022696

TABLE VII. Codebooks used by the LSF multi-stage quantizer for stage 4 (component values are in Hertz).

Index	Vectors						
0x0	13.864392	-19.737336	20.320452	-1.373732	-24.025380	55.403468	-33.427036
0x1	-22.288548	-34.188684	20.704048	-14.170992	28.366312	-25.985648	-43.561588
0x2	-9.949592	14.031280	2.022688	61.570748	11.537028	-20.475192	-56.247088
0x3	-64.483988	-2.641932	9.354576	-1.695224	-11.159228	1.977040	3.499672
0x4	1.859316	-18.637868	39.190940	19.354980	44.978516	5.672544	-3.360900
0x5	22.329840	-8.827276	35.918016	-26.647964	26.134024	-41.064820	4.758780
0x6	17.873364	-3.982856	44.497744	39.808760	-3.764812	-8.364812	-35.478012
0x7	7.958560	-32.041208	17.138880	37.115176	17.563628	-11.659032	33.656416
0x8	-12.145288	-24.479824	56.629472	-11.682896	-31.763652	3.683420	-20.414980
0x9	-14.054776	-35.296828	-19.199396	17.575908	-28.405600	-4.153392	7.303092
0xa	11.007008	19.022892	28.676476	8.685068	41.181528	9.850048	-35.909068
0xb	-11.754804	-0.200860	-0.397848	16.824616	41.821960	15.431596	-17.873084
0xc	4.417632	-13.858636	25.993272	3.621776	-13.282620	-54.866568	-25.740860
0xd	30.472892	2.402004	-8.414600	-14.939776	20.846288	-42.379776	44.159640
0xe	20.803664	-5.884716	-10.076072	34.678792	-1.836392	-5.648408	-57.621056
0xf	5.240548	-7.730720	-42.546312	25.064936	15.738880	6.281712	21.895068
0x10	24.895620	3.387108	14.067512	-35.964624	34.713716	56.939140	17.696072
0x11	-20.307876	32.490736	-8.269384	-37.426408	15.369516	5.518868	-40.295176
0x12	3.301884	23.180704	-24.454116	-5.660896	-46.692200	72.857908	3.808256
0x13	-36.148208	33.906036	-10.643360	27.813232	7.439476	23.088892	41.102328
0x14	41.485516	33.976276	16.557244	11.715616	1.168420	11.215132	2.419500
0x15	-34.868980	44.335072	18.371036	-9.576508	6.842808	-28.376484	-20.193584
0x16	22.181968	-4.747624	-6.587840	19.194412	13.324592	50.283068	34.604528
0x17	-23.391532	11.590768	37.601468	12.769832	-9.503604	-11.279292	54.145876

**TABLE VII. Codebooks used by the LSF multi-stage quantizer for stage 4 (component values are in Hertz) - Continued.**

Index	Vectors							
0x18	-7.408196	-30.925108	18.743584	-54.589100	40.004624	22.577360	8.870712	-11.437640
0x19	2.106444	-17.324588	-22.842364	-28.871344	-21.276340	-13.394672	23.784676	70.579396
0x1a	15.241316	17.603468	-41.962316	-4.684472	53.083624	16.845052	-13.084284	-12.831660
0x1b	-5.009304	-10.205264	-23.680948	-23.020348	55.499976	26.375932	64.954384	7.736592
0x1c	17.559028	9.416808	39.605736	23.848632	1.871332	-54.038420	27.998276	35.575168
0x1d	-6.962476	4.105532	19.244696	-23.883640	-39.505700	19.590216	53.142492	8.583808
0x1e	29.861688	7.994612	-19.893828	45.102624	8.184900	-39.195660	30.159276	-34.246888
0x1f	14.491724	-5.943524	-5.773136	28.083748	-36.821584	-25.838832	80.000780	12.158860
0x20	-5.4688828	-26.673576	-13.033748	15.823448	2.773560	29.6338816	-77.755704	29.586208
0x21	4.192508	-0.595220	-9.597436	-29.569652	-8.967564	-15.488152	-87.442240	33.505968
0x22	6.918280	5.475584	-34.010376	55.845532	3.372104	41.049620	-16.152440	1.566632
0x23	-38.947288	18.104492	-36.755712	-14.831836	14.532324	-0.120888	0.709208	-29.737616
0x24	16.827216	-8.372392	-4.047592	14.480476	48.383496	8.909460	-23.019528	74.803304
0x25	16.908200	3.1113504	-0.409796	-15.891340	30.269736	-61.639948	-30.195632	59.169420
0x26	-19.531304	-19.663028	10.757152	54.459716	-5.494000	-20.867892	18.984844	61.646372
0x27	-29.128836	-3.459292	-9.332620	17.339504	25.026592	-59.119656	54.056544	-12.055064
0x28	-15.645544	-21.784092	-4.418296	-12.523804	-16.660484	74.213892	34.056316	-0.726300
0x29	15.689516	-11.155860	-21.276536	-14.047236	-54.493236	13.666468	-30.506112	-37.156156
0x2a	-29.735320	-13.753276	3.548232	31.269112	39.668368	44.830620	-6.977028	-35.663752
0x2b	-31.082216	11.584868	8.487652	-33.036500	-13.157316	50.502976	-23.188224	-51.524568
0x2c	17.331036	-10.261824	31.780508	-23.296888	8.928572	52.597664	-44.915272	-30.676572
0x2d	7.656704	-12.769280	-34.288816	-46.295780	7.367804	0.185976	-26.244420	-33.013772
0x2e	-18.013668	-10.073172	3.763892	24.780428	-21.653296	4.704220	-13.278056	-57.884760
0x2f	-7.139380	-3.507464	-16.812212	-10.187820	11.070308	-45.039224	-26.121504	-78.043388

**TABLE VII. Codebooks used by the LSF multi-stage quantizer for stage 4 (component values are in Hertz) - Continued.**

<b>Index</b>	<b>Vectors</b>						
0x30	-0.038328	-12.205152	2.765344	43.550748	4.322620	64.041088	37.213004
0x31	-24.910772	6.883424	-12.832696	1.845308	-52.037336	32.769556	-20.558604
0x32	-1.617564	23.527380	-13.455208	32.733468	38.018240	5.814052	46.106580
0x33	-22.066160	28.427852	24.169128	9.870268	22.294164	-14.262904	-2.228472
0x34	20.856168	11.434760	-17.501460	19.356592	-23.927048	-10.080176	-29.157472
0x35	4.532676	6.949228	30.519480	-1.335668	-46.754808	-16.408320	-71.551928
0x36	-6.563488	-4.683368	-44.646952	11.974452	-5.360264	-32.144792	27.619440
0x37	-12.079848	28.782912	-5.610228	32.414792	-38.174764	-53.727672	-3.846380
0x38	9.128584	-15.964872	40.529996	-25.097428	-9.404020	33.634748	25.128176
0x39	-0.584244	-19.878520	15.734836	-26.676744	-33.555660	-67.913680	2.189504
0x3a	27.830096	3.212196	-18.081108	-53.418624	16.337124	17.790228	-0.935908
0x3b	-8.894760	17.362400	13.714624	-55.179844	3.355216	-25.666356	52.092692
0x3c	24.708572	2.926604	19.614548	11.151496	-50.921244	31.058220	38.814708
0x3d	26.751716	10.221092	20.453016	-38.210332	-40.726020	-25.178320	-9.964692
0x3e	19.678856	24.642696	-2.825972	-14.446076	-19.540088	-8.693844	1.168576
0x3f	-7.553260	16.710720	-31.982072	-28.903800	-40.696264	-28.379732	36.020044

TABLE VIII. Codebook used by the Fourier magnitude vector quantizer

Index	Vectors					
0x0	0.822998	1.496297	0.584847	1.313507	0.846008	0.614041
0x1	1.248150	1.020382	0.517184	1.489079	0.650498	0.716904
0x2	1.167392	1.468091	0.743322	0.712255	1.564293	0.721041
0x3	1.043080	1.570270	0.444422	0.9333200	1.211773	0.669870
0x4	0.759126	1.037894	1.126967	1.133186	0.789090	0.807399
0x5	0.977225	1.130888	1.044601	1.432985	0.819888	0.517787
0x6	0.666187	1.364039	1.088940	0.90231	1.110838	0.980791
0x7	0.983185	1.460579	0.732273	0.923475	1.088579	0.952291
0x8	0.697275	1.651033	0.506420	0.809603	1.113992	1.083678
0x9	0.749416	1.354976	0.731452	1.281438	1.080946	1.037335
0xa	0.976897	2.009469	0.741592	0.606862	0.644927	0.687413
0xb	0.576891	1.527863	0.657703	1.006739	1.226803	0.786733
0xc	0.517211	1.220859	0.886556	0.980153	0.884365	1.041315
0xd	0.651809	1.106789	0.885180	1.329281	1.124388	0.626778
0xe	0.614335	1.430380	0.899696	0.847168	0.624567	1.307056
0xf	0.830604	1.373081	0.785967	0.735951	0.968745	1.100706
0x10	1.084085	1.385767	0.448480	1.297458	0.770611	0.719263
0x11	1.244608	1.158451	0.615757	1.169599	1.032735	0.921809
0x12	1.313869	1.341417	0.481710	0.685601	1.086817	1.003461
0x13	1.427137	0.932544	0.595881	0.652610	1.073852	0.847480
0x14	1.027745	1.214124	0.896291	0.972030	0.841943	0.738307
0x15	0.952252	1.098463	0.690316	1.175721	0.911712	0.962678
0x16	0.902314	1.163750	1.143587	0.814860	0.923235	0.973077
0x17	1.133879	1.110463	0.687589	0.793794	1.026535	1.076900

TABLE VIII. Codebook used by the Fourier magnitude vector quantizer - Continued.

Index	Vectors						
0x18	0.623910	1.269921	0.722072	0.990256	1.183314	1.161476	0.677381
0x19	0.585861	1.189163	0.882952	1.134136	0.900218	1.263223	0.995922
0x1a	0.687616	1.562003	0.815795	0.701934	0.789752	1.204232	0.821560
0x1b	0.950686	1.329037	0.692678	0.754646	1.095995	1.446339	0.941535
0x1c	0.520026	1.293024	0.938993	0.990412	0.897114	0.914170	1.358696
0x1d	0.669622	1.291638	0.749196	1.167486	1.042641	0.721329	1.005261
0x1e	0.497926	1.222508	1.403952	0.799058	0.798533	0.793545	1.264461
0x1f	0.891705	1.342163	0.761635	0.758182	0.987397	1.008477	1.377249
0x20	1.174305	0.454482	0.432532	0.691549	1.066271	0.890505	0.845957
0x21	1.607432	0.688224	0.500617	1.288348	0.678978	0.674969	1.047764
0x22	1.739179	0.919542	0.639494	0.775935	1.021945	0.991820	0.887633
0x23	1.620174	1.211703	0.610139	1.372774	1.057336	0.618126	0.642995
0x24	1.219952	0.844191	1.016601	0.867402	1.006595	0.940309	0.893858
0x25	1.251753	1.004762	0.777132	1.203828	0.656382	0.657205	1.233246
0x26	1.348819	0.791939	1.030755	0.765314	1.014939	1.044297	0.747682
0x27	1.262759	1.109681	0.672002	1.109957	0.827608	1.272683	0.670943
0x28	1.108504	1.250712	0.620039	0.752463	1.439216	0.890828	1.018300
0x29	1.222589	1.083869	0.627894	1.150342	1.377353	0.785771	0.679900
0x2a	1.326349	1.690290	1.188837	0.595281	0.602145	0.626852	0.676343
0x2b	1.499772	1.400400	0.655881	0.892565	1.089036	0.702486	0.835749
0x2c	1.126664	1.353291	0.898265	0.897870	0.934461	1.003119	0.982470
0x2d	1.316478	1.060041	0.6667653	1.065408	0.910377	1.002121	1.339044
0x2e	0.976194	1.377380	1.071262	0.789567	1.085928	1.253806	1.017727
0x2f	1.202869	1.059008	0.962883	0.999919	0.524897	1.074531	1.091141

**TABLE VIII. Codebook used by the Fourier magnitude vector quantizer - Continued.**

Index	Vectors						
0x30	1.413976	0.836644	0.6666408	1.039645	1.088087	0.820217	0.816371
0x31	1.388093	0.807110	0.781561	0.964242	0.973557	1.138037	1.026733
0x32	1.456077	0.982162	0.557993	0.738683	1.375522	0.772422	0.644426
0x33	1.606208	1.220452	0.738208	0.744965	0.659231	0.813191	0.932141
0x34	1.126839	1.038057	0.935449	1.012686	1.080640	0.884338	1.126468
0x35	1.215351	0.845602	0.829299	1.204972	0.891515	0.934431	0.887728
0x36	1.206387	0.977863	1.051199	0.847855	0.939759	1.254109	0.970361
0x37	1.207602	0.976323	0.841307	0.870300	0.877422	0.988163	0.931714
0x38	1.095130	1.105129	0.722843	0.943180	1.211443	1.178861	0.634252
0x39	1.032014	0.998735	0.686262	1.100404	1.022352	1.222401	1.146689
0x3a	1.131931	1.489739	0.677826	0.610733	0.759219	0.808122	1.042500
0x3b	1.342504	1.076171	0.583400	0.764890	0.818097	1.293405	1.058200
0x3c	1.308520	1.219689	0.935572	0.714861	1.129251	0.856944	0.920880
0x3d	1.372592	0.887155	0.710732	1.029836	1.278897	0.650119	1.136552
0x3e	1.019956	1.381944	1.089064	0.576825	0.594933	0.752016	1.265619
0x3f	1.154864	1.069591	0.913967	0.796105	0.790255	0.854206	1.523905
0x40	0.874806	1.645517	0.834275	1.578196	0.903996	0.547038	0.565520
0x41	1.153418	1.095160	0.870564	1.655735	1.071838	0.806210	0.607539
0x42	0.830864	1.448871	1.014508	0.939034	1.574030	0.796490	0.654852
0x43	1.125819	1.396448	0.746324	1.295697	1.137447	1.301677	0.574765
0x44	0.832614	0.802739	1.148272	1.316531	1.305743	0.696667	0.940671
0x45	1.260278	0.958467	0.667246	1.406435	0.859092	0.534003	0.485425
0x46	0.887527	1.330678	1.062704	0.552689	1.098956	0.917567	0.874732
0x47	0.906302	1.326878	0.632127	0.781817	1.111568	0.753606	0.896250
0x48	0.163798	1.616260	0.520967	1.244674	0.597986	1.174228	0.802735

TABLE VIII. Codebook used by the Fourier magnitude vector quantizer - Continued.

Index	Vectors						
Index	0x49	0x4a	0x4b	0x4c	0x4d	0x4e	0x4f
0x49	0.717419	1.075445	0.997204	1.206762	1.312107	0.696587	0.745413
0x4a	0.339447	1.827655	0.684798	0.716559	0.676084	1.081139	0.894730
0x4b	0.864667	1.451421	0.725543	1.176080	1.209637	1.083631	1.163663
0x4c	0.563978	1.135516	0.835026	1.736364	1.000137	0.805360	0.828496
0x4d	1.009051	1.360317	0.668259	1.417717	1.518623	0.638014	0.446922
0x4e	0.186440	1.625134	0.835253	1.175778	0.860628	0.885086	1.011866
0x4f	0.652311	1.599127	0.599516	1.058215	0.801753	0.992815	1.146092
0x50	0.759780	1.399480	0.987329	1.024824	0.974458	0.551246	1.150852
0x51	1.069814	1.160678	0.803250	1.193140	1.112463	0.744748	1.070724
0x52	1.168517	1.6333441	0.774778	0.736824	1.089258	1.057042	0.783527
0x53	1.301817	1.572909	0.526128	0.558027	0.660410	1.078262	1.089774
0x54	0.891795	1.262297	0.911545	1.214103	0.776010	0.944986	1.138773
0x55	1.219015	1.048683	0.943178	1.150567	0.837824	0.566225	0.660764
0x56	0.925533	1.350162	1.164350	0.542695	1.227640	0.600203	1.205779
0x57	1.019130	1.292473	0.796847	0.819714	1.254208	0.522062	1.293349
0x58	0.686388	1.534513	1.023273	1.065238	0.769952	0.914897	0.958226
0x59	0.560425	1.321635	0.837563	1.105726	0.800833	0.952432	0.673960
0x5a	0.722275	1.482196	1.269134	0.695424	0.734410	1.240968	0.873750
0x5b	1.022015	1.408327	0.702067	0.966478	0.742264	1.204704	0.633885
0x5c	0.512817	1.187577	0.939995	1.273059	1.261853	0.875396	0.884657
0x5d	0.781537	1.223652	0.854824	0.972599	1.347144	0.807227	0.911694
0x5e	0.358832	1.601542	1.125781	0.730841	0.795475	0.856229	1.070817
0x5f	0.821456	1.676333	0.931807	0.740210	0.946762	0.917425	0.936539
0x60	1.133434	1.069057	0.628708	1.348112	1.024242	0.726333	0.485013
0x61	1.439495	0.401963	0.704352	1.185582	1.196889	0.894400	0.762308

TABLE VIII. Codebook used by the Fourier magnitude vector quantizer - Continued.

Index	Vectors						
0x62	1.355857	1.217795	0.623188	0.626000	1.080819	1.443039	0.690217
0x63	1.384203	0.893545	0.571820	1.422045	1.054457	0.896980	0.929532
0x64	1.272930	0.807536	0.943865	1.161822	1.109685	0.883353	0.756987
0x65	1.368426	0.939970	0.927314	1.372501	0.966845	0.661820	0.817570
0x66	1.234512	0.812394	1.092298	0.956999	0.926831	1.159350	0.560790
0x67	1.462063	1.107453	0.736919	1.182041	0.753060	0.817381	0.845724
0x68	1.154263	1.149050	0.969239	1.003195	0.709246	1.334899	0.539388
0x69	1.363427	0.856319	0.848329	1.330223	1.580245	0.593904	0.534564
0x6a	1.430748	1.277622	0.535910	1.316419	0.704018	1.139968	0.844730
0x6b	1.456831	0.852769	0.836842	1.661979	0.555156	0.659789	0.754965
0x6c	1.219340	0.941999	1.021315	1.258016	0.869648	1.086416	1.006363
0x6d	1.269435	1.217910	1.091920	1.130622	1.241678	0.548636	0.659441
0x6e	1.109102	1.109600	0.921873	1.080103	0.644065	1.551581	0.917328
0x6f	1.148823	1.115256	0.784950	1.519787	0.632186	0.880947	0.965093
0x70	1.099162	1.072237	1.059239	1.012868	1.091543	0.820111	0.632757
0x71	1.382078	1.014650	0.855020	1.088445	1.147617	1.054602	0.759230
0x72	1.331108	0.911897	0.551680	0.739304	1.080511	1.173246	0.671436
0x73	1.380841	1.130295	0.743747	0.910074	0.957719	0.986217	0.885270
0x74	1.310270	0.998818	0.989857	1.025202	0.906801	0.876459	0.895856
0x75	1.442392	0.947518	1.052543	0.888925	1.092888	0.674498	0.920650
0x76	1.173251	0.879037	1.128982	0.905439	1.302826	1.072854	0.709855
0x77	1.162547	0.939193	1.038151	0.930689	0.914292	0.877584	1.144403
0x78	0.957634	1.145630	1.008914	1.045813	1.030746	1.071616	0.801657
0x79	1.250553	1.148173	0.606277	0.991795	0.997859	0.688890	0.942568
0x7a	1.223899	1.343365	0.813470	0.600631	0.514261	1.292102	0.994158

TABLE VIII. Codebook used by the Fourier magnitude vector quantizer - Continued.

Index	Vectors						
0x7b	1.434258	1.190874	0.478282	0.454840	0.731792	1.170314	0.776934
0x7c	1.075483	0.950940	1.056806	0.939311	0.767863	1.050672	0.923138
0x7d	1.152850	0.985329	0.758860	0.902903	1.268339	0.822622	0.951593
0x7e	0.934316	1.213602	1.137776	0.530840	0.583216	1.327735	0.648220
0x7f	1.096461	1.329950	1.056696	1.096312	0.518753	0.667957	1.377914
0x80	0.438479	0.986558	0.795180	1.048475	1.212747	1.336801	0.859633
0x81	1.106514	0.482598	0.612108	1.642264	0.798236	0.780622	0.825869
0x82	0.977340	0.929082	0.748423	0.688517	1.866792	0.978102	0.704420
0x83	2.061007	1.317233	0.531080	0.663650	0.466228	0.673965	0.663756
0x84	0.324483	0.566419	1.626532	0.909957	0.767043	1.218518	0.923857
0x85	1.132776	0.664024	0.658276	1.115888	1.058611	0.773592	1.143017
0x86	0.693193	0.957946	1.074530	0.991271	1.178869	1.050262	1.231718
0x87	0.981206	1.282808	0.628439	0.831905	0.871888	1.000360	1.072068
0x88	0.444708	1.320901	0.863009	0.823633	1.282252	1.175766	1.016159
0x89	0.524278	0.903183	0.842282	0.999358	1.663203	0.862494	0.990065
0x8a	0.647517	1.630489	0.674425	0.552789	1.367019	0.802523	1.078554
0x8b	1.157635	1.168651	0.661005	0.556302	1.240422	0.970912	0.829631
0x8c	0.271398	0.978008	0.699287	1.501347	0.903792	1.105029	0.960131
0x8d	0.275143	0.628587	1.143603	1.112293	1.315251	1.071179	0.889841
0x8e	0.659209	1.447333	0.894087	1.001992	0.839399	1.449942	0.796803
0x8f	0.853138	1.064738	0.855133	0.742170	0.868512	1.370756	0.989681
0x90	0.973539	1.188932	1.023116	0.931037	1.059522	0.699512	0.709015
0x91	1.226588	1.022695	0.719215	1.364611	0.676914	0.710196	1.255418
0x92	0.911359	1.344114	1.042414	0.518823	1.545711	0.394875	0.583463
0x93	1.728382	0.630457	0.384633	0.410263	0.623340	0.756137	0.963416

TABLE VIII. Codebook used by the Fourier magnitude vector quantizer - Continued.

Index	Vectors						
0x94	0.738959	0.590931	1.110580	1.101743	0.983178	0.969963	1.084357
0x95	1.244427	0.569958	1.057313	1.369461	0.723367	0.805159	1.027868
0x96	0.787743	0.905228	1.203273	0.843008	1.182612	1.029414	0.763719
0x97	1.314427	1.012982	0.695385	0.878737	0.749022	0.792771	1.048592
0x98	0.284264	1.037450	0.783668	1.045137	1.085681	0.925955	1.010468
0x99	0.182413	0.501944	0.613194	0.864945	1.046530	1.201125	1.222115
0x9a	0.399720	1.302627	1.339572	0.544091	1.113712	0.704227	0.959569
0x9b	1.312087	1.234779	1.155127	0.573204	0.603170	1.675500	0.578462
0x9c	0.347498	1.019206	0.902255	0.918363	1.078096	0.895444	1.295650
0x9d	0.228297	0.547356	1.316811	1.334579	0.850478	0.739122	1.162684
0x9e	0.450951	1.164991	1.356497	0.748838	1.098313	1.180630	0.960353
0x9f	1.101005	1.177952	1.577841	0.463778	0.535020	0.573959	0.857461
0xa0	1.008192	0.512187	0.511630	0.803656	1.363784	1.397129	1.041883
0xa1	1.333032	0.522169	0.512363	0.626419	0.780427	0.951713	1.495366
0xa2	1.680621	0.528954	0.445504	0.623836	1.316133	1.053039	0.890863
0xa3	2.235511	0.589895	0.429925	0.577054	0.680209	0.667395	0.784315
0xa4	1.092796	0.555938	1.481165	0.731586	1.165349	0.972450	0.847847
0xa5	1.276456	0.719658	0.981702	1.037082	1.176864	0.684247	0.782879
0xa6	1.228245	0.771626	1.121788	0.749934	0.884593	0.958108	1.233677
0xa7	1.469187	0.712424	0.977325	0.707326	1.237597	0.953937	0.835168
0xa8	1.079164	1.065143	1.263724	0.575489	1.628970	0.516204	0.546497
0xa9	1.337945	0.994695	0.675169	0.865744	1.373516	1.324911	0.866779
0xaa	1.511212	0.641169	1.710560	0.731522	0.590315	0.546117	0.624898
0xab	1.760845	1.236091	1.067203	0.487204	0.655720	0.713362	0.697764
0xac	1.242002	1.065400	0.986706	0.626575	1.306310	0.681929	0.717473

TABLE VIII. Codebook used by the Fourier magnitude vector quantizer - Continued.

Index	Vectors						
0xad	1.114715	0.651334	1.082160	1.095767	1.064034	1.122643	1.004475
0xae	1.244497	1.005714	1.193496	0.734583	0.598761	1.274922	1.289539
0xaf	1.408250	0.839767	1.067828	0.572007	0.672582	1.239141	0.865001
0xb0	1.250488	0.918235	1.101326	0.668242	1.365644	0.555295	0.552784
0xb1	1.631045	0.643427	0.448596	0.462433	0.721966	1.376064	1.149863
0xb2	1.535075	1.109493	1.002904	0.867410	1.515072	0.754291	0.569348
0xb3	1.968050	0.717136	1.151858	0.741070	0.764881	0.686026	0.715694
0xb4	1.413788	0.795490	1.377022	0.733397	1.230761	0.728025	0.815985
0xb5	1.339077	0.710095	1.014431	1.092420	0.988468	0.688934	1.213969
0xb6	1.172569	0.971105	1.147345	0.503528	1.077878	1.006680	0.936191
0xb7	1.346133	0.825702	1.278602	0.712776	0.990801	0.824307	0.741063
0xb8	0.838360	1.061588	1.184451	0.806865	1.445930	0.716610	1.122729
0xb9	1.276108	0.881797	1.013445	0.821855	1.264900	1.025945	1.227647
0xba	1.043635	1.011762	1.513310	0.739962	0.660585	0.587323	0.680578
0xbb	1.473979	0.627258	1.381177	0.659284	0.759202	1.049849	1.020305
0xbc	1.308547	0.926336	1.119570	0.550398	1.232051	0.605843	1.181402
0xbd	1.337645	0.769449	1.229177	0.813618	0.735630	0.883470	1.569295
0xbe	1.290794	1.093894	1.145434	0.552063	0.804350	0.769436	0.840595
0xbf	1.364932	0.913519	1.316632	0.718362	0.529225	0.723872	1.181281
0xc0	0.803741	1.388152	0.980932	1.386743	0.402886	1.229764	0.855613
0xc1	1.187083	1.328233	0.750736	1.808456	0.498683	0.428506	0.498878
0xc2	1.008780	1.634667	0.998899	1.114864	1.106568	0.814867	0.677242
0xc3	1.442123	1.578969	1.235055	1.279799	0.604508	0.459123	0.486663
0xc4	0.824699	0.832512	1.286809	0.946005	0.788899	1.581450	0.819352
0xc5	1.173084	0.852089	0.837159	1.215371	0.614107	0.664921	0.798941

TABLE VIII. Codebook used by the Fourier magnitude vector quantizer - Continued.

Index	Vectors						
0xc6	0.837980	1.195548	1.251800	1.166862	1.139151	0.630036	0.966812
0xc7	1.284059	1.360185	0.903226	1.260066	0.869280	0.794052	0.767674
0xc8	0.434909	1.062061	0.820721	0.980006	0.915903	1.352711	1.363232
0xc9	0.140705	0.369673	0.761460	1.721146	1.281940	0.814390	0.770430
0xca	0.381833	1.386068	0.895360	0.756433	0.795777	0.665474	1.098763
0xcb	0.848084	1.088554	0.824684	0.776900	0.721938	0.676660	0.800316
0xcc	0.675218	1.083363	1.015844	1.396638	0.903676	0.983177	0.888414
0xcd	0.754970	0.952752	1.127383	1.539338	0.700672	0.751028	0.640102
0xce	0.701999	1.321737	1.118923	1.317673	0.798997	1.146361	0.653062
0xcf	0.916108	1.387334	0.769658	1.200061	0.617306	1.096865	0.828575
0xd0	0.662335	1.292372	1.294051	1.134329	0.611097	0.878492	0.979876
0xd1	1.081800	1.056505	1.431825	1.568379	0.608668	0.601313	0.533137
0xd2	1.062379	1.439992	1.195124	1.101291	0.690349	0.686199	0.744970
0xd3	1.302173	1.079498	1.059178	0.961715	0.721412	0.548559	0.490001
0xd4	0.680923	1.193324	1.324460	1.076160	0.940757	1.158478	1.016504
0xd5	1.013158	0.953016	1.286560	1.218908	0.634730	1.131540	0.788277
0xd6	0.801060	0.948640	1.501946	0.737009	0.751623	1.024697	1.091762
0xd7	1.322228	1.272781	1.083638	0.879445	0.649796	0.924154	0.851784
0xd8	0.308295	1.165975	1.273902	1.153456	0.938787	0.761951	0.988352
0xd9	0.307028	0.833986	1.891761	1.166166	0.526468	0.545358	0.772629
0xda	0.724400	1.661022	1.407543	0.659395	0.621709	0.664584	0.781703
0xdb	0.899730	0.520003	2.050203	0.570193	0.525624	0.618870	0.773975
0xdc	0.564036	1.194072	1.054038	1.309234	0.682175	1.184083	1.261020
0xdd	0.705321	0.906931	1.440192	1.246568	0.945792	0.911725	0.914322
0xde	0.639259	1.118893	1.183435	0.912825	0.722814	1.107206	0.903505

TABLE VIII. Codebook used by the Fourier magnitude vector quantizer - Continued.

Index	Vectors						
0xdf	0.833210	1.220093	1.668781	0.922503	0.998916	0.696036	0.765626
0xe0	1.214346	0.962383	1.132890	0.827035	0.876470	1.255731	0.611676
0xe1	1.790710	0.452581	0.576174	0.955171	0.930091	0.858695	1.025674
0xe2	1.436801	0.888196	1.189311	0.679367	1.104152	1.351368	0.585875
0xe3	1.614636	0.783579	0.883180	1.156453	0.973477	0.848164	0.830311
0xe4	1.372060	0.499597	1.000811	0.900767	0.931695	0.898965	1.028259
0xe5	1.464549	0.744441	0.906385	0.936059	0.731646	0.716360	0.829522
0xe6	1.367898	0.866499	1.263666	1.112268	1.167850	0.937974	0.788936
0xe7	1.439445	0.795863	0.796972	1.211896	0.538767	1.255454	0.958684
0xe8	1.376381	0.568678	0.926291	0.641770	0.834535	1.808678	0.804428
0xe9	1.355563	0.460449	0.596175	1.214770	0.946022	1.352957	0.860313
0xea	1.435024	0.909544	1.041289	1.123733	0.794294	1.439099	0.648301
0xeb	1.516270	0.918985	0.927275	1.189389	0.590144	0.748748	1.248918
0xec	1.476744	0.514611	1.218581	1.065365	0.862836	0.922786	0.764142
0xed	1.315547	0.722841	1.311024	1.315785	0.913893	0.706439	0.682783
0xee	1.348712	0.858988	1.055635	1.052220	0.611431	1.171928	0.730139
0xef	1.180305	1.053248	0.878721	1.28889	0.507336	1.103290	0.705925
0xf0	1.196638	1.080029	1.301461	0.997280	0.885975	0.846955	0.887870
0xf1	1.373492	1.068929	1.077698	1.096515	0.998252	0.717247	1.234505
0xf2	1.437332	1.207382	1.145016	0.902884	0.999814	1.049959	0.751617
0xf3	1.540923	0.966825	1.074880	0.967647	0.780575	0.824665	0.721806
0xf4	1.075058	0.874560	1.223286	0.875247	1.114946	0.785841	1.030228
0xf5	1.230599	0.763269	1.223421	1.021346	0.751078	0.863709	0.928759
0xf6	1.200394	0.826452	1.374917	0.728714	0.749645	1.169742	0.809933
0xf7	1.458186	0.835033	1.120633	0.933190	0.831612	1.053974	1.060913

**TABLE VIII. Codebook used by the Fourier magnitude vector quantizer - Continued.**

<b>Index</b>	<b>Vectors</b>						
0xf8	1.087334	1.237373	1.295267	0.657919	1.094758	1.035074	0.693941
0xf9	1.183917	0.792976	1.143049	0.749982	0.626687	0.658678	0.983599
0xfa	1.694116	1.108393	1.504700	0.828677	0.586589	0.658925	0.648047
0xfb	1.363659	0.859205	1.134321	1.227826	0.519545	0.570781	0.847817
0xfc	1.202407	0.995001	1.034142	0.944991	0.778536	0.657893	1.164819
0xfd	1.325400	0.953743	1.087893	1.003095	0.754906	0.862034	1.119859
0xfe	1.321790	0.767809	1.313064	0.954328	0.816200	0.607967	0.663671
0xff	1.493994	1.117090	1.211209	1.311853	0.619853	0.708056	0.764515
						0.708914	0.718302
							0.743372

## MIL-STD-3005

## APPENDIX A

## MELP ALGORITHM DESCRIPTION

## A.1 SCOPE

A.1.1 Scope. This appendix provides a complete description of a MELP algorithm. This appendix is not a mandatory part of this standard. The information contained herein is intended for guidance only.

## A.2 APPLICABLE DOCUMENTS

A.2.1 Government documents. The documents in 2. of this standard apply to this appendix.

A.2.2 Other publications. The following documents form a part of this appendix to the extent specified.

## INSTITUTE OF ELECTRICAL AND ELECTRONICS ENGINEERS (IEEE)

"A Mixed Excitation LPC Vocoder Model for Low Bit Rate Speech Coding"  
by A.V. McCree and T.P. Barnwell III, Transactions on Speech and Audio Processing, Vol. 3, No. 4, July 1995, 242-250

"A 2.4 kbits/s MELP Coder Candidate for the New U.S. Federal Standard"  
by A. McCree, K. Truong, E.B. George, T.P. Barnwell III, and V. Viswanathan, Proceedings of IEEE ICASSP 1996, pp.200-203

"Super Resolution Pitch Determination of Speech Signals"  
by Y. Medan, E. Yair, and D. Chazan, Transactions on Signal Processing, Vol. 39, No. 1, January 1991, pp. 40-48

"The Computation of Line Spectral Frequencies Using Chebyshev Polynomials"  
by P. Kabal and R.P. Ramachandran, Transactions on Acoustics, Speech, and Signal Processing, Vol. ASSP-34, No.6, December 1986, pp.1419-1426

"Efficient Search and Design Procedures for Robust Multi-Stage VQ of LPC Parameters for 4 kb/s Speech Coding"  
by W.P. LeBlanc, B. Bhattacharya, S.A. Mahmoud, and V. Cuperman, Transactions on Speech and Audio Processing, Vol. 1, No. 4, October 1993, pp. 373-385

"New Methods for Adaptive Noise Suppression"  
by L. Arslan, A. McCree, and V. Viswanathan, Proceedings of IEEE ICASSP 1995, pp. 812-815

"MELP: The New Federal Standard at 2400 BPS"  
by L. Supplee, R. Cohn, J. Collura, A. McCree, Proceedings of IEEE ICASSP 1997, pp. 1591-1594

(Applications for copies should be addressed to IEEE Customer Service, 445 Hoes Lane, P.O. Box 1331 Piscataway, New Jersey 08855-1331, USA)

## MIL-STD-3005

## APPENDIX A

A.2.3 Order of precedence. In the event of a conflict between the text of this standard and the references stated herein, the text of this standard should take precedence.

## A.3 DEFINITIONS

A.3.1 Terms. The definitions in 3.1 of this standard apply to this appendix.

A.3.2 Acronyms. The acronyms used in this appendix are defined in 3.2 or as follows:

DC - Direct Current

DFT - Discrete Fourier Transform

FFT - Fast Fourier Transform

FIR - Finite Impulse Response

RMS - Root Mean Square

VQ - Vector Quantization

## A.4 GENERAL REQUIREMENTS

Not applicable.

## A.5 DETAILED REQUIREMENTS

A.5.1 General. The Mixed Excitation Linear Prediction coder is based on the traditional Linear Prediction Coding (LPC) parametric model, but also includes five additional features. These are: mixed excitation, aperiodic pulses, adaptive spectral enhancement, pulse dispersion, and Fourier magnitude modeling. These features are illustrated in the MELP decoder block diagram shown in Figure A-1.

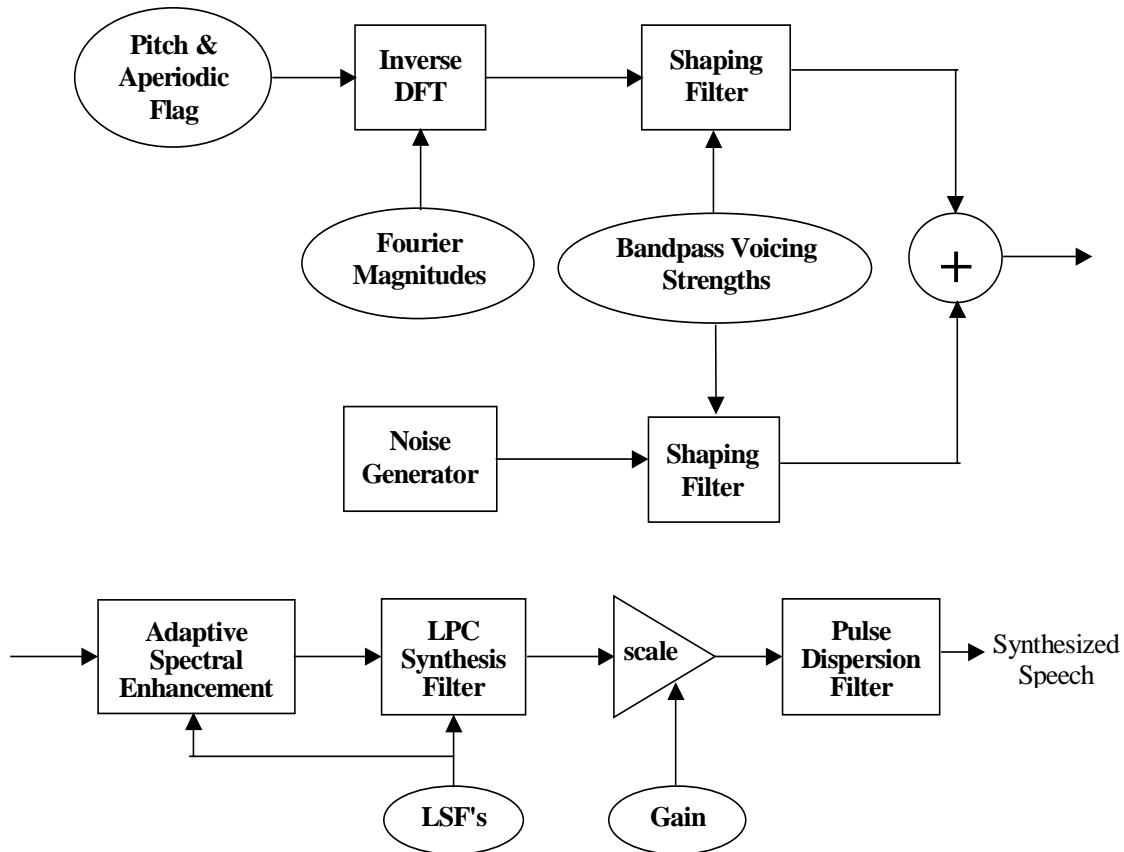
The mixed excitation is implemented using a multi-band mixing model. This model can simulate frequency-dependent voicing strength using an adaptive filtering structure implemented with a fixed filter bank. The primary effect of this mixed excitation is to reduce the buzz usually associated with LPC vocoders, especially in broadband acoustic noise.

When the input speech is voiced, the MELP coder can synthesize using either periodic or aperiodic pulses. Aperiodic pulses are used most often during transition regions between voiced and unvoiced segments of the speech signal. This feature enables the decoder to reproduce erratic glottal pulses without introducing tonal sounds.

The adaptive spectral enhancement filter is based on the poles of the linear prediction synthesis filter. Its use enhances the formant structure of the synthetic speech and improves the match between the synthetic and natural bandpass waveforms. It also gives the synthetic speech a more natural quality.

## MIL-STD-3005

## APPENDIX A

**Figure A- 1. MELP decoder block diagram.**

Pulse dispersion is implemented using a fixed filter based on a spectrally-flattened triangle pulse. This filter spreads the excitation energy within a pitch period, reducing some of the harsh quality of the synthetic speech.

The first ten Fourier magnitudes are determined from the peaks of the Fourier transform of the prediction residual signal. The information in these coefficients improves the accuracy of the speech production model at the perceptually-important lower frequencies. This increases the quality of the synthetic speech, particularly for male speakers and when background noise is present.

**A.5.2 Encoder.** Input speech is encoded by performing the following steps in the order given.

**A5.2.1 Low frequency removal.** The first step in the encoding process is to remove any low frequency energy which may be present in the input signal. This is accomplished with a 4<sup>th</sup> order Chebychev type II highpass filter, having a cutoff frequency of 60 Hz and a stopband rejection of 30 dB. The filter output is referred to as the input speech signal throughout the following encoder description.

## MIL-STD-3005

## APPENDIX A

A buffer containing the most recent samples of the input speech signal is maintained in the encoder. One of these samples is designated the last sample in the current frame. The buffer extends beyond this sample into the past and future to contain the samples needed for the encoding process. The last sample in the current frame serves as a reference point for many of the encoder calculations.

**A5.2.2 Integer pitch calculation.** For this pitch calculation, the input speech signal is first processed with a 1 kHz, 6<sup>th</sup> order Butterworth lowpass filter. The integer pitch value,  $P_1$ , is the value of,  $\tau$ ,  $\tau = 40, 41, \dots, 160$ , for which the normalized autocorrelation function,  $r(\tau)$ , is maximized. This function is defined by:

$$r(\tau) = \frac{c_\tau(0, \tau)}{\sqrt{c_\tau(0, 0)c_\tau(\tau, \tau)}}, \text{ EQUATION A-1,}$$

$$\text{where } c_\tau(m, n) = \sum_{k=-\lfloor \tau/2 \rfloor - 80}^{\lfloor \tau/2 \rfloor + 79} s_{k+m}s_{k+n}, \text{ EQUATION A-2,}$$

and  $\lfloor \tau/2 \rfloor$  represents truncation to an integer value. The center of the pitch analysis window is at sample  $s_0$  in equation A-2. For the integer pitch calculation, this window is centered on the last sample in the current frame. The lowpass filter output is sample  $s_0$  when its input is the last sample in the current frame. The time index  $k$  in the autocorrelation preserves the pitch analysis window alignment around its center point; the normalization compensates for changing signal amplitudes. The final pitch calculation (see A5.2.9) extends the pitch range to a lag of 20 samples.

**A5.2.3 Bandpass voicing analysis.** This portion of the encoder determines the five bandpass voicing strengths,  $V_{bp_i}$ ,  $i = 1, 2, \dots, 5$ . It also refines the integer pitch measurement and the corresponding normalized autocorrelation value. The bandpass voicing analysis begins by filtering the input speech signal into five frequency bands. These filters are 6<sup>th</sup> order Butterworth, with passbands of 0-500, 500-1000, 1000-2000, 2000-3000, and 3000-4000 Hz.

A refined pitch measurement is made using the 0-500 Hz filter output signal. This measurement is centered on the filter output produced when its input is the last sample in the current frame. Two pitch candidates are considered in this refinement, namely the integer pitch values,  $P_1$ , from the current and previous frames. For each candidate, equation A-1 is used to perform an integer pitch search over lags from 5 samples shorter to 5 samples longer than the candidate, and a fractional pitch refinement (see A5.2.4) is performed around the optimum integer pitch lag. This produces two fractional pitch candidates and their corresponding normalized autocorrelation values. The candidate having the higher normalized autocorrelation is selected as the fractional pitch,  $P_2$ . The corresponding normalized autocorrelation,  $r(P_2)$ , is saved as the lowest band voicing strength,  $V_{bp_1}$ .  $P_2$  is saved for use in determining the voicing strength for the remaining frequency bands. It is also used in the final pitch calculation (see A5.2.9) and gain calculation (see A5.2.11).

For each remaining band, the bandpass voicing strength is the larger of  $r(P_2)$  as determined by the fractional pitch procedure for the bandpass signal and the time envelope of the bandpass signal, where  $r(P_2)$  for the time envelope is first decremented by 0.1 to compensate for an experimentally observed bias (due to the smoothness of the time envelope signals). The envelopes are calculated by full-wave rectification followed by a smoothing filter. This filter consists of a zero at DC in cascade with a complex pole pair at 150 Hz with a radius of 0.97. For each

## MIL-STD-3005

## APPENDIX A

calculation of  $r(P_2)$ , the analysis window is centered on the last sample in the current frame, as was the case for the first band.

**A5.2.4 Fractional pitch refinement.** This procedure, which is used at several places in the encoding process, utilizes an interpolation formula to increase the accuracy of an input pitch value. This value is first rounded to the nearest integer. Assume that this integer has a value of  $T$  samples. The interpolation formula presumes that  $r(\tau)$  has a maximum between lags of  $T$  and  $T+1$ . Hence,  $c_T(0, T-1)$  and  $c_T(0, T+1)$  are computed and compared to determine if the maximum is more likely to fall between  $T$  and  $T+1$  or between  $T-1$  and  $T$ . If  $c_T(0, T-1) > c_T(0, T+1)$ , then the maximum probably falls between  $T-1$  and  $T$  and the pitch,  $T$ , is decremented by one prior to interpolation. The fractional offset,  $\Delta$ , is then computed by the interpolation equation:

$$\Delta = \frac{c_T(0, T+1)c_T(T, T) - c_T(0, T)c_T(T, T+1)}{c_T(0, T+1)[c_T(T, T) - c_T(T, T+1)] + c_T(0, T)[c_T(T+1, T+1) - c_T(T, T+1)]}, \text{EQUATION A-3,}$$

where  $c_T(m, n)$  is defined by equation A-2. In some cases, this formula produces an offset outside the range of 0.0 to 1.0, so the offset is clamped between -1 and 2. The fractional pitch is  $T + \Delta$  and is clamped between 20 and 160.

The normalized autocorrelation at the fractional pitch value is given by:

$$r(T + \Delta) = \frac{(1 - \Delta)c_T(0, T) + \Delta c_T(0, T+1)}{\sqrt{c_T(0, 0)[(1 - \Delta)^2 c_T(T, T) + 2\Delta(1 - \Delta)c_T(T, T+1) + \Delta^2 c_T(T+1, T+1)]}}, \text{EQUATION A-4.}$$

Equations A-3 and A-4 produce the fractional offset and corresponding normalized autocorrelation which would be obtained if the input signal had been linearly interpolated to obtain values between the actual sampling times.

**A5.2.5 Aperiodic flag.** The aperiodic flag is set to 1 if  $V_{bp_1} < 0.5$  and set to 0 otherwise. The  $V_{bp_1}$  value determined by bandpass voicing analysis (see A5.2.3) is used for this comparison. When set, this flag tells the decoder that the pulse component of the excitation should be aperiodic, rather than periodic. A5.3.1 describes the use of the aperiodic flag.

**A5.2.6 Linear prediction analysis.** A 10<sup>th</sup> order linear prediction analysis is performed on the input speech signal using a 200 sample (25 ms) Hamming window centered on the last sample in the current frame. The traditional autocorrelation analysis procedure is implemented using the Levinson-Durbin recursion. In addition, a bandwidth expansion coefficient of 0.994 (15 Hz) is applied to the prediction coefficients,  $a_i$ ,  $i = 1, 2, \dots, 10$ , where each coefficient is multiplied by 0.994<sup>i</sup>.

**A5.2.7 Linear prediction residual calculation.** The linear prediction residual signal is calculated by filtering the input speech signal with the prediction filter whose coefficients were determined by linear prediction analysis (see A5.2.6). The residual window is centered on the last sample in the current frame, and is made wide enough for use by the final pitch calculation (see A5.2.9).

## MIL-STD-3005

## APPENDIX A

**A5.2.8 Peakiness calculation.** The peakiness of the residual signal is calculated over a 160 sample window centered on the last sample in the current frame. The peakiness value is the ratio of the L2 norm to the L1 norm of the residual signal,  $r_n$ , in the window:

$$\text{peakiness} = \frac{\sqrt{\frac{1}{160} \sum_{n=1}^{160} r_n^2}}{\frac{1}{160} \sum_{n=1}^{160} |r_n|}, \text{ EQUATION A-5.}$$

If the peakiness exceeds 1.34, then the lowest band voicing strength,  $Vbp_1$ , is forced to 1.0. If the peakiness exceeds 1.6, then the lowest three band voicing strengths,  $Vbp_i, i=1,2,3$ , are all forced to 1.0. This is the only use of the peakiness measure.

**A5.2.9 Final pitch calculation.** The final pitch measurement uses the lowpass filtered residual signal, where the filter is a 6<sup>th</sup> order Butterworth, with a 1 kHz cutoff. Equation A-1 is used to perform an integer pitch search over lags from 5 samples shorter to 5 samples longer than  $P_2$ , rounded to the nearest integer. This measurement is centered on the filter output produced when its input is the last residual sample in the current frame. A fractional pitch refinement (see A5.2.4) is then made around the optimum integer pitch lag. This produces tentative values for the final pitch,  $P_3$ , and for the corresponding normalized autocorrelation,  $r(P_3)$ .

If  $r(P_3) \geq 0.6$ , the pitch doubling check procedure (see A5.2.10) is performed on the filtered residual, using  $P_3$  as the candidate pitch, and doubling threshold  $D_{th} = 0.75$  if  $P_3 \leq 100$ , or  $D_{th} = 0.5$  otherwise. The doubling check procedure may produce new values for  $P_3$  and  $r(P_3)$ .

The else action for the preceding if is as follows. A fractional pitch refinement around  $P_2$  is performed using the input speech signal. This measurement is centered on the last sample in the current frame and produces new values for  $P_3$  and  $r(P_3)$ . If  $r(P_3) < 0.55$ , then  $P_3$  is replaced by  $P_{avg}$ , the long-term average pitch (see A5.2.12). Otherwise, the pitch doubling check procedure is performed on the input speech signal, using  $P_3$  as the candidate pitch, and doubling threshold  $D_{th} = 0.9$  if  $P_3 \leq 100$ , or  $D_{th} = 0.7$  otherwise. The doubling check procedure may produce new values for  $P_3$  and  $r(P_3)$ .

Finally, if  $r(P_3) < 0.55$ , then  $P_3$  is replaced by  $P_{avg}$ .

The following pseudo code shows the final pitch algorithm:

```

inputs: the input speech signal; the residual signal; P2; Pavg
outputs: P3, cor_P3

fresid buffer = filter the residual with a 1 kHz Butterworth
P3 = best integer pitch on fresid over the range P2-5 to P2+5
P3, cor_P3 = frac_pitch(fresid, P3)
if (cor_P3 >= 0.6)

```

## MIL-STD-3005

## APPENDIX A

```

Dth = 0.5
if (P3 <= 100) Dth = 0.75
P3, cor_P3 = double_ck(fresid, P3, Dth)
else
    P3, cor_P3 = frac_pitch(input, P2)
    if (cor_P3 < 0.55)
        P3 = Pavg
    else
        Dth = 0.7
        if (P3 <= 100) Dth = 0.9
        P3, cor_P3 = double_ck(input, P3, Dth)
    endif
endif
if (cor_P3 < 0.55) P3 = Pavg

```

**A5.2.10 Pitch doubling check.** The pitch doubling check procedure looks for and corrects pitch values which are multiples of the actual pitch. This procedure takes a signal, a candidate pitch  $P_c$ , and a doubling threshold  $D_{th}$ , and returns the checked pitch  $P_c$ , and the corresponding correlation,  $r(P_c)$ . All fractional pitch calculations are made using the signal given to the doubling check procedure.

This procedure begins with a fractional pitch refinement around  $P_c$ . This produces tentative values for  $P_c$  and  $r(P_c)$ . Next, the largest value of  $k$  is found for which  $r(P_c/k) > D_{th}r(P_c)$ , where  $(P_c/k) \geq 20$  and  $k = 8, 7, \dots, 2$ .  $r(P_c/k)$  is calculated in two steps: 1) a fractional pitch refinement around  $P_c/k$ , producing  $P_k$ ; and 2) a double verification, if  $P_k < 30$ . If such a  $k$  is found, then a fractional pitch refinement around  $P_k$  is performed, producing new values for  $P_c$  and  $r(P_c)$ .

Finally, if  $P_c$  is less than 30 samples, then double verification is performed.

The following pseudo code shows the pitch double check procedure:

```

inputs: signal; P; Dth
outputs: Pc, cor_Pc

Pc, cor_Pc = frac_pitch(signal, P)
for (k=8; k>=2; k--)
    Pk = Pc/k
    if (Pk >= 20)
        Pk, cor_Pk = frac_pitch(signal, Pk)
        if (Pk < 30) cor_Pk = double_ver(Pk, cor_Pk)
        if (cor_Pk > Dth * cor_Pc)
            Pc, cor_Pc = frac_pitch(signal, Pk)
            break
        endif
    endif
endfor

```

## MIL-STD-3005

## APPENDIX A

```
if (Pc < 30) cor_Pc = double_ver(Pc, cor_Pc)
```

For inputs  $P$  and  $r(P)$ , the double verification procedure returns the smaller of  $r(P)$  and  $r(2P)$ , where  $r(2P)$  is determined by the fractional pitch procedure around  $2P$ . The use of double verification in the double check procedure provides robustness against spurious short pitch values.

**A5.2.11 Gain calculation.** The input speech signal gain is measured twice per frame using a pitch-adaptive window length. This length is identical for both gain measurements and is determined as follows. When  $Vbp_1 > 0.6$ , the window length is the shortest multiple of  $P_2$  which is longer than 120 samples. If this length exceeds 320 samples, it is divided by 2. When  $Vbp_1 \leq 0.6$ , the window length is 120 samples. The gain calculation for the first window produces  $G_1$  and is centered 90 samples before the last sample in the current frame. The calculation for the second window produces  $G_2$  and is centered on the last sample in the current frame. The gain is the RMS value, measured in dB, of the signal in the window,  $s_n$ :

$$G_i = 10 \log_{10} \left( 0.01 + \frac{1}{L} \sum_{n=1}^L s_n^2 \right), \text{EQUATION A-6},$$

where  $L$  is the window length. The 0.01 term prevents the log argument from going too close to zero. If a gain measurement is less than 0.0, it is clamped to 0.0. The gain measurement assumes that the input signal range is -32768 to 32767 (see 5.2).

**A5.2.12 Average pitch update.** The long-term average pitch,  $P_{avg}$ , is updated with a simple smoothing procedure. If  $r(P_3) > 0.8 > 0.8$  and  $G_2 > 30\text{dB}$ , then  $P_3$  is placed into a buffer containing the three most recent strong pitch values,  $p_i$ ,  $i = 1, 2, 3$ . Otherwise, all three pitch values in the buffer are moved toward a default pitch,  $P_{default} = 50$  samples, according to:

$$p_i = 0.95p_i + 0.05P_{default}, i = 1, 2, 3, \text{EQUATION A-7}.$$

The average pitch is then updated as the median of the three values in the buffer.  $P_{avg}$  is used in the final pitch calculation (see A5.2.9).

**A5.2.13 Quantization of prediction coefficients.** First, the linear prediction coefficients  $a_i$ ,  $i = 1, 2, \dots, 10$ , are converted into LSFs. Next, a process which forces the LSF components to be in ascending order with a minimum separation of 50 Hz is performed. This process begins by checking all adjacent pairs of the LSF components and swapping any pair not in ascending order. This step is repeated as many as ten times, if necessary. The minimum separation criterion is then applied by correcting each pair,  $f_i$  and  $f_{i+1}$ , for which  $d = f_{i+1} - f_i$  is less than 50 Hz,  $\Delta_{min}$ , as shown in the following pseudo code. The LSF components and frequency-related constants are in Hertz; scaling in other implementations may differ. The minimum separation process is repeated ten times.

```
Dmin = 50
for (I=1; I<10; I++)
    d = f[I+1] - f[I]
```

## MIL-STD-3005

## APPENDIX A

```

if (d < dmin)
    s1 = s2 = (dmin-d)/2
if (I == 1 and f[I] < dmin) s1 = f[I]/2
else if (I > 1)
    tmp = f[I] - f[I-1]
    if      (tmp < dmin)   s1 = 0
    else if (tmp < 2*dmin) s1 = (tmp-dmin)/2
endif
if (I == 9 and f[I+1] > 4000-dmin) s2 = (4000-f[I+1])/2
else if (I < 9)
    tmp = f[I+2] - f[I+1]
    if      (tmp < dmin)   s2 = 0
    else if (tmp < 2*dmin) s2 = (tmp-dmin)/2
endif
f[I+1] = f[I+1] + s2
endif
endfor

```

The resulting LSF vector,  $f$ , is then quantized using a MSVQ. The MSVQ codebook consists of four stages of 128, 64, 64, and 64 levels respectively. The quantized vector,  $\hat{f}$ , is the sum of the vectors selected by the search process, with one vector selected from each stage. The MSVQ search finds the codebook vector which minimizes the square of the weighted Euclidean distance,  $d^2$ , between the unquantized and quantized LSF vectors:

$$d^2(f, \hat{f}) = \sum_{i=1}^{10} w_i (f_i - \hat{f}_i)^2, \text{ where } w_i = \begin{cases} P(f_i)^{0.3}, & 1 \leq i \leq 8 \\ 0.64P(f_i)^{0.3}, & i = 9 \\ 0.16P(f_i)^{0.3}, & i = 10 \end{cases}, \text{ EQUATION A-8,}$$

$f_i$  is the  $i^{\text{th}}$  component of the unquantized LSF vector, and  $P(f_i)$  is the inverse prediction filter power spectrum evaluated at frequency  $f_i$ . The search procedure is an M-best approximation to a full search, in which the M=8 best code vectors from each stage are saved for use with the next stage. The process to ensure ascending order and minimum separation (described in the first part of this section) is then applied to the quantized LSF vector. The resulting vector is used in the Fourier magnitude calculation (see A5.2.17).

**A5.2.14 Pitch quantization.** The final pitch value,  $P_3$ , is quantized on a logarithmic scale with a 99-level uniform quantizer ranging from 20 to 160 samples. These pitch values are then mapped to a 7-bit codeword using a look-up table, as shown in section 5.3.1. The all-zero codeword represents the unvoiced state, and is sent if  $Vbp_1 \leq 0.6$ . All 28 codewords with Hamming weight of 1 or 2 are reserved for error protection. The uniform quantizer details are described in 5.3.7.

**A5.2.15 Gain quantization.** The two gain values are quantized as follows.  $G_2$  is quantized with a 5-bit uniform quantizer ranging from 10 to 77 dB.  $G_1$  is quantized to 3 bits using the following adaptive algorithm. If  $G_2$  for the current frame is within 5 dB of  $G_2$  for the previous frame, and  $G_1$  is within 3 dB of the average of the  $G_2$  values for the current and previous frames, then the frame is steady-state and a special code (all zero) is

## MIL-STD-3005

## APPENDIX A

sent to indicate that the decoder should set  $G_1$  to the mean of the  $G_2$  values for the current and previous frames. Otherwise, the frame represents a transition and  $G_1$  is quantized with a 7-level uniform quantizer ranging from 6 dB below the minimum of the  $G_2$  values for the current and previous frames to 6 dB above the maximum of those  $G_2$  values. The quantizer range is clamped to 10 and 77 dB. The uniform quantizer details are described in 5.3.7. Pseudo code for the adaptive quantization of  $G_1$  is shown below.

```
If (|G2 - G2p| < 5.0 and |G1 - 0.5 * (G2 + G2p)| < 3.0)
    quantizer_index = 0
else
    gain_max = max(G2p, G2) + 6.0
    gain_min = min(G2p, G2) - 6.0
    if (gain_min < 10.0) gain_min = 10.0
    if (gain_max > 77.0) gain_max = 77.0
    quantizer_index values 1 to 7 are determined by quantizing G1 with a 7-level,
    uniform quantizer ranging from gain_min to gain_max
endif
```

**A5.2.16 Bandpass voicing quantization.** When  $V_{bp1} \leq 0.6$  (unvoiced), the remaining voicing strengths,  $V_{bp_i}, i = 2,3,4,5$ , are quantized to 0. When  $V_{bp1} > 0.6$ , the remaining voicing strengths are quantized to 1 if their value exceeds 0.6, and quantized to 0 otherwise. There is one exception. If the quantized values of  $V_{bp_i}, i = 2,3,4,5$  are 0001, respectively, then  $V_{bp5}$  is quantized to 0.

**A5.2.17 Fourier magnitude calculation and quantization.** This analysis measures the Fourier magnitudes of the first 10 pitch harmonics of the prediction residual generated by the quantized prediction coefficients. It uses a 512-point Fast Fourier Transform (FFT) of a 200 sample window centered at the end of the frame. First, a set of quantized predictor coefficients is calculated from the quantized LSF vector (see A5.2.13). Then the residual window is generated using the quantized prediction coefficients. Next, a 200 sample Hamming window is applied, the signal is zero-padded to 512 points, and the complex FFT is performed. Finally, the complex FFT output is transformed into magnitudes, and the harmonics are found with a spectral peak-picking algorithm.

The peak-picker finds the maximum within a width of  $512/\hat{P}_3$  frequency samples centered around the initial estimate for each pitch harmonic, where  $\hat{P}_3$  is the quantized pitch. This width is truncated to an integer. The initial estimate for the location of the  $I^{\text{th}}$  harmonic is  $512i/\hat{P}_3$ . The number of harmonic magnitudes searched for is limited to the smaller of 10 or  $\hat{P}_3/4$ . These magnitudes are then normalized to have an RMS value of 1.0. If fewer than 10 harmonics are found, the remaining magnitudes are set to 1.0.

The 10 magnitudes are quantized with an 8-bit vector quantizer. The codebook is searched using a perceptually weighted Euclidean distance, with fixed weights that emphasize low frequencies over higher frequencies. The weights are given by:

## APPENDIX A

$$w_i = \left[ \frac{117}{25 + 75 \left( 1 + 1.4 \left( \frac{f_i}{1000} \right)^2 \right)^{0.69}} \right]^2, i = 1, 2, \dots, 10, \text{ EQUATION A-9},$$

where  $f_i = 8000i / 60$  is the frequency in Hz corresponding to the  $i^{\text{th}}$  harmonic for a default pitch period of 60 samples. The weights are applied to the squared difference between the input Fourier magnitudes and the codebook values.

**A5.2.18 Error protection and bit packing.** Table II in 5.5.2 shows the bit allocation for the MELP coder. To improve performance in channel errors, the unused coder parameters for the unvoiced mode are replaced with forward error correction. Three Hamming (7,4) codes and one Hamming (8,4) code are used. The (7,4) code corrects single bit-errors, while the (8,4) code in addition detects double bit-errors. The (8,4) code is applied to the 4 most significant bits (MSBs) of the first MSVQ index, and the 4 parity bits are written over the bandpass voicing. The remaining 3 bits of the first MSVQ index along with a reserved bit (set to zero), are covered by a (7,4) code with the resulting 3 parity bits written to the MSBs of the Fourier series VQ index. The 4 MSBs of the  $G_2$  codeword are protected with 3 parity bits which are written to the next 3 bits of the Fourier magnitudes. Finally, the LSB of the second gain index and the 3 bit  $G_1$  codeword are protected with 3 parity bits written to the 2 LSBs of the Fourier magnitudes and the aperiodic flag.

The bit transmission order is given in 5.5.3.

**A5.3 Decoder.** The channel data is decoded by performing the following steps in the order given.

**A5.3.1 Bit unpacking and error correction.** The received bits are unpacked from the channel and assembled into the parameter codewords. Parameter decoding is different for voiced and unvoiced modes. The pitch is decoded first, since it contains the mode information. If the pitch code is all-zero or has only one bit set, then the unvoiced mode is used. If two bits are set, a frame erasure is indicated. Otherwise, the pitch value is decoded and the voiced mode is used.

In the unvoiced mode, the (8,4) Hamming code is decoded to correct single bit errors and detect double errors. If an uncorrectable error is detected, a frame erasure is indicated. Otherwise, the (7,4) Hamming codes are decoded, correcting single errors but without double error detection.

If any erasure is detected in the current frame, by the Hamming code, by the pitch code, or directly signaled from the channel, then a frame repeat mechanism is implemented. All of the parameters for the current frame are replaced with the parameters from the previous frame. In addition, the first gain term is set equal to the second gain term so that no gain transitions are allowed.

If an erasure is not indicated, the remaining parameters are decoded. The LSFs are checked for ascending order and minimum separation as described in A5.2.13. In the unvoiced mode, default parameter values are used for the pitch, jitter, bandpass voicing, and Fourier magnitudes. The pitch value is set to 50 samples, the jitter is set to

## MIL-STD-3005

## APPENDIX A

25%, all of the bandpass voicing strengths are set to 0, and the Fourier magnitudes are set to 1. In the voiced mode,  $Vbp_1$  is set to 1; jitter is set to 25% if the aperiodic flag is a 1; otherwise jitter is set to 0%. The bandpass voicing strength for the upper four bands is set to 1 if the corresponding bit is a 1; otherwise the voicing strength is set to 0. There is one exception. If 0001 is received for  $Vbp_i$ ,  $i = 2,3,4,5$ , respectively, then  $Vbp_5$  is set to 0.

When the special all-zero code for the first gain parameter,  $G_1$ , is received, some errors in the second gain parameter,  $G_2$ , can be detected and corrected. This correction process provides improved performance in channel errors. The decoding for the two gain parameters is shown in the following pseudo code.

```

Inputs: G1_index, G2_index
outputs: G1, G2
internal: G2p, G2p_error

G2 = decode(G2_index)           32 levels; range: 10 to 77 dB
if (G1_index == 0)
    if (|G2 - G2p| > 5)
        if (G2p_error == 0)
            G2 = G2p
        endif
        G2p_error = 1
    else
        G2_index probably correct
        G2p_error = 0
    endif
    G1 = 0.5 * (G2 + G2p)       mean of G2 and G2p
else
    G1 = decode(G1_index)       7 levels; range: min(G2,G2p)-6 to max(G2,G2p)+6
    G2p_error = 0
endif
G2p = G2                         (above range is clamped to 10 to 77 dB)
                                    save for use as past value

```

**A5.3.2 Noise attenuation.** For quiet input signals, a small amount of gain attenuation is applied to both decoded gain parameters using a power subtraction rule. This attenuation is a simplified, frequency invariant case of the Smoothed Spectral Subtraction noise suppression method.

Before determining the attenuation for the first gain term,  $G_1$ , a background noise estimate,  $G_n$ , is updated as follows. If  $G_1 > G_n + C_{up}$  then  $G_n = G_n + C_{up}$ . If  $G_1 < G_n - C_{down}$  then  $G_n = G_n - C_{down}$ . Otherwise,  $G_n = G_1$ .  $C_{up} = 0.0337435$  and  $C_{down} = 0.135418$ , so that the noise estimator moves up by 3 dB per second and down by 12 dB per second for the gain update rate of 88.9 updates per second. The noise estimate is clamped between 10 and 80. Noise estimation is disabled for repeated frames to prevent repeated attenuation. The background noise estimate is also used in the adaptive spectral enhancement calculation (see A5.3.5).

Gain  $G_1$  is then modified by subtracting a (positive) correction term,  $G_{att}$ , given in dB by

$$G_{att} = -10 \log_{10} \left( 1 - 10^{0.1[G_n + 3 - G_1]} \right), \text{EQUATION A-10},$$

## MIL-STD-3005

## APPENDIX A

where  $G_n$  is the background noise estimate (in dB), and  $G_1$  is the first gain term (in dB). The correction is clamped to a maximum value of 6 dB to avoid fluctuations and signal distortion. To ensure that the attenuation is applied only to quiet signals, the  $G_n$  value as used in equation A-10 is clamped at an upper limit of 20 dB.

The noise estimation and gain modification steps are then repeated for the second gain term,  $G_2$ . Noise estimation and gain attenuation are disabled for repeated frames.

**A5.3.3 Parameter interpolation.** All MELP synthesis parameters are interpolated pitch-synchronously for each synthesized pitch period. The interpolated parameters are the gain (in dB), LSFs, pitch, jitter, Fourier magnitudes, pulse and noise coefficients for mixed excitation, and spectral tilt coefficient for the adaptive spectral enhancement filter. Gain is linearly interpolated between the second gain of the prior frame,  $G_{2p}$ , and the first gain of the current frame,  $G_1$ , if the starting point,  $t_0$ ,  $t_0 = 0, 1, \dots, 179$ , of the new pitch period is less than 90; otherwise, gain is interpolated between  $G_1$  and  $G_2$ . Normally, the other parameters are linearly interpolated between the past and current frame values. The interpolation factor, int, for these parameters is based on the starting point of the new pitch period:

$$\text{int} = t_0 / 180, \text{ EQUATION A-11.}$$

There are two exceptions to this interpolation procedure. First, if there is an onset with a high pitch frequency, pitch interpolation is disabled and the new pitch is immediately used. This condition is met when  $G_1$  is more than 6 dB greater than  $G_{2p}$  and the current frame's pitch period is less than half the prior frame's pitch period. The second exception also involves a gain onset. If  $G_2$  differs from  $G_{2p}$  by more than 6 dB, then the LSF's, spectral tilt, and pitch are interpolated using the interpolated gain trajectory as a basis, since the gain is transmitted twice per frame and has a more accurate interpolation path. In this case, the interpolation factor is given by

$$\text{int} = \frac{G_{\text{int}} - G_{2p}}{G_2 - G_{2p}}, \text{ EQUATION A-12,}$$

where  $G_{\text{int}}$  is the interpolated gain. This interpolation factor is then clamped between 0 and 1.

**A5.3.4 Mixed excitation generation.** The mixed excitation is generated as the sum of the filtered pulse and noise excitations. The pulse excitation,  $e_p(n)$ ,  $n = 0, 1, \dots, T - 1$ , is computed using an inverse Discrete Fourier Transform of one pitch period in length:

$$e_p(n) = \frac{1}{T} \sum_{k=0}^{T-1} M(k) e^{j2\pi nk/T}, \text{ EQUATION A-13.}$$

The pitch period,  $T$ , is the interpolated pitch value plus the jitter times the pitch, where the jitter is the interpolated jitter strength times the output of a uniform random number generator between -1 and 1. This pitch period is rounded to the nearest integer and clamped between 20 and 160. All of the phases for the pulse excitation are set to zero, hence  $M(k)$  is real. Since  $e_p(n)$  is real, the magnitudes obey:

## MIL-STD-3005

## APPENDIX A

$$M(T-k) = M(k), k = 1, 2, \dots, L, \text{EQUATION A-14},$$

where  $L = T/2$  if  $T$  is even, and  $L = (T-1)/2$ , if  $T$  is odd. The DC term,  $M(0)$ , is set to 0. Magnitude terms  $M(k), k = 1, 2, \dots, 10$ , are set to the interpolated values of the Fourier magnitudes, and any magnitudes not otherwise specified are set to 1. To prevent rapid changes at the start of the pitch period, the pulse excitation is circularly shifted by ten samples of delay so the main excitation pulse occurs at the tenth sample of the period. The pulse is then multiplied by the square root of the pitch to give a unity RMS signal, and then multiplied by 1000 to give a nominal signal level.

The noise is generated by a uniform random number generator with an RMS value of 1000, and range of -1732 to 1732.

The pulse and noise excitation signals are then filtered and summed to form the mixed excitation. The pulse filter for the current frame is given by the sum of all the bandpass filter coefficients for the voiced frequency bands, while the noise filter is given by the sum of the bandpass filter coefficients for the unvoiced bands. These filter coefficients are interpolated pitch synchronously. The bandpass filter coefficients for each of the five bands are given in table A-I.

**TABLE A-I. Filter coefficients for bandpass filter.**

<b>0-500 Hz</b>	<b>500-1000 Hz</b>	<b>1000-2000 Hz</b>	<b>2000-3000 Hz</b>	<b>3000-4000Hz</b>
-0.00302890	-0.00249420	-0.00231491	0.00231491	0.00554149
-0.00701117	-0.00282091	0.00990113	0.00990113	-0.00981750
-0.01130619	0.00257679	0.02086129	-0.02086129	0.00856805
-0.01494082	0.01451271	-0.00000000	0.00000000	-0.00000000
-0.01672586	0.02868120	-0.03086123	0.03086123	-0.01267517
-0.01544189	0.03621179	-0.02180695	-0.02180695	0.02162277
-0.01006619	0.02784469	0.00769333	-0.00769333	-0.01841647
0.00000000	0.00000000	-0.00000000	-0.00000000	0.00000000
0.01474923	-0.04079870	-0.01127245	0.01127245	0.02698425
0.03347158	-0.07849207	0.04726837	0.04726837	-0.04686914
0.05477206	-0.09392213	0.10106105	-0.10106105	0.04150730
0.07670890	-0.07451087	-0.00000000	0.00000000	-0.00000000
0.09703726	-0.02211575	-0.17904543	0.17904543	-0.07353666
0.11352143	0.04567473	-0.16031428	-0.16031428	0.15896026
0.12426379	0.10232715	0.09497157	-0.09497157	-0.22734513
0.12799355	0.12432599	0.25562154	0.25562154	0.25346255

## APPENDIX A

**TABLE A-I. Filter coefficients for bandpass filter - Continued.**

<b>0-500 Hz</b>	<b>500-1000 Hz</b>	<b>1000-2000 Hz</b>	<b>2000-3000 Hz</b>	<b>3000-4000Hz</b>
0.12426379	0.10232715	0.09497157	-0.09497157	-0.22734513
0.11352143	0.04567473	-0.16031428	-0.16031428	0.15896026
0.09703726	-0.02211575	-0.17904543	0.17904543	-0.07353666
0.07670890	-0.07451087	-0.00000000	0.00000000	-0.00000000
0.05477206	-0.09392213	0.10106105	-0.10106105	0.04150730
0.03347158	-0.07849207	0.04726837	0.04726837	-0.04686914
0.01474923	-0.04079870	-0.01127245	0.01127245	0.02698425
0.00000000	-0.00000000	-0.00000000	-0.00000000	0.00000000
-0.01006619	0.02784469	0.00769333	-0.00769333	-0.01841647
-0.01544189	0.03621179	-0.02180695	-0.02180695	0.02162277
-0.01672586	0.02868120	-0.03086123	0.03086123	-0.01267517
-0.01494082	0.01451271	-0.00000000	0.00000000	-0.00000000
-0.01130619	0.00257679	0.02086129	-0.02086129	0.00856805
-0.00701117	-0.00282091	0.00990113	0.00990113	-0.00981750
-0.00302890	-0.00249420	-0.00231491	0.00231491	0.00554149

A5.3.5 Adaptive spectral enhancement. The adaptive spectral enhancement filter is applied to the mixed excitation signal. This filter is a tenth order pole/zero filter, with an additional first-order tilt compensation. Its coefficients are generated by bandwidth expansion of the linear prediction filter transfer function,  $A(z)$ , corresponding to the interpolated LSFs. The transfer function of the enhancement filter,  $H_{ase}(z)$ , is given by:

$$H_{ase}(z) = \frac{A(\alpha z^{-1})}{A(\beta z^{-1})} * (1 + \mu z^{-1}), \text{ where } \begin{aligned} \alpha &= 0.5p \\ \beta &= 0.8p \end{aligned} \quad \text{EQUATION A-15},$$

and tilt the coefficient  $\mu$  is first calculated as  $\max(0.5k_1, 0)$ , then interpolated, then multiplied by  $p$ , the signal probability. The first reflection coefficient,  $k_1$ , is calculated from the decoded LSFs. By the MELP predictor coefficient sign convention,  $k_1$  is usually negative for voiced spectra. The signal probability  $p$  is estimated by comparing the current interpolated gain,  $G_{int}$ , to the background noise estimate  $G_n$  using the formula:

$$p = \frac{G_{int} - G_n - 12}{18}, \quad \text{EQUATION A-16}.$$

This signal probability is clamped between 0 and 1.

A5.3.6 Linear prediction synthesis. The synthesis uses a direct form filter, with the coefficients corresponding to the interpolated LSFs.

## MIL-STD-3005

## APPENDIX A

**A5.3.7 Gain adjustment.** Since the excitation is generated at an arbitrary level, the speech gain must be introduced to the synthesized speech. The correct scaling factor,  $S_{\text{gain}}$ , is computed for each synthesized pitch period of length T by dividing the desired RMS value ( $G_{\text{int}}$  must be converted from dB) by the RMS value of the unscaled synthetic speech signal  $\hat{s}_n$ :

$$S_{\text{gain}} = \frac{10^{G_{\text{int}}/20}}{\sqrt{\frac{1}{T} \sum_{n=1}^T \hat{s}_n^2}}, \text{ EQUATION A-19.}$$

To prevent discontinuities in the synthesized speech, this scale factor is linearly interpolated between the previous and current values for the first ten samples of the pitch period.

**A5.3.8 Pulse dispersion.** The pulse dispersion filter is a 65<sup>th</sup> order FIR filter derived from a spectrally-flattened triangle pulse. The coefficients are listed in table A-II.

<b>TABLE A-II. Filter coefficients for the pulse dispersion filter.</b>				
<b>Samples 1-13</b>	<b>Samples 14-26</b>	<b>Samples 27-39</b>	<b>Samples 40-52</b>	<b>Samples 53-65</b>
-0.17304259	0.24325127	0.07343483	0.02968464	0.00019707
-0.01405709	-0.01767043	-0.00518645	-0.01247640	-0.02825247
0.01224406	-0.00018612	0.01298488	0.01854666	0.01720989
0.11364226	0.05869485	0.02928440	0.00076184	-0.06004292
0.00198199	-0.00327456	-0.01989405	-0.07749640	-0.07076744
0.00000658	0.00607395	0.01216758	0.01244697	0.00914347
0.04529633	0.02753924	0.01180979	-0.02721777	0.06082730
-0.00092027	-0.03351673	-0.38924775	0.07266098	0.01805528
-0.00103078	0.00602189	0.00720325	0.00472008	-0.00318634
0.02552787	0.01436539	-0.01154561	0.03526439	0.03444110
-0.06339257	0.82854582	0.08426287	0.02674603	0.00026302
-0.00122031	0.00033165	-0.00355720	-0.00744038	-0.01053809
0.01412525	-0.00360180	0.02151233	0.02582623	0.02165922

**A5.3.9 Synthesis loop control.** After processing each pitch period, the decoder updates  $t_0$  by adding T, the number of samples in the period just synthesized. If  $t_0 < 180$ , synthesis for the current frame continues from the parameter interpolation step (see A5.3.3). Otherwise, the decoder buffers the remainder of the current period which extends beyond the end of the current frame and subtracts 180 from  $t_0$  to produce its initial value next frame.

## MIL-STD-3005

## APPENDIX B

## PERFORMANCE VERIFICATION

## B.1 SCOPE

**B.1.1 Scope.** This appendix is a mandatory part of this standard. The information contained herein is intended for compliance. All new implementations of the MELP coder must be tested to verify that their performance meets or exceeds that of the MELP reference coder. This appendix provides guidelines for verifying the performance of a MELP implementation. Two methods of verification are presented.

## B.2 APPLICABLE DOCUMENTS

**B.2.1 Government documents.** Not applicable.

**B.2.2 Other publications.** The following documents form a part of this appendix to the extent specified.

## ANSI Standard

S3.2-1989

American National Standard Method for  
Measuring the Intelligibility of Speech  
over Communications Systems

(Applications for copies should be addressed to ANSI Customer Service, 11 West 42<sup>nd</sup> Street, New York, New York 10036, USA)

## INSTITUTE OF ELECTRICAL AND ELECTRONICS ENGINEERS (IEEE)

"IEEE Recommended Practice for Speech Quality Measurements"  
by IEEE Subcommittee on Subjective Measurements, Transactions on Audio and Electroacoustics, Vol. 17, 1969, pp. 227-246

(Applications for copies should be addressed to IEEE Customer Service, 445 Hoes Lane, P.O. Box 1331 Piscataway, New Jersey 08855-1331, USA)

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

## B.3 DEFINITIONS

**B.3.1 Terms.** Terms used in this appendix are defined in section 3.1 of the standard or as follows.

**B.3.1.1 A/B test.** An A/B test is a direct paired forced choice comparison test that is used to assess the quality of one voice coder against another voice coder.

**B.3.2 Acronyms used in this appendix.** Acronyms used in this section are either defined in 3.2 of the standard or as follows.

## MIL-STD-3005

## APPENDIX B

ANSI - American National Standards Institute

DRT - Diagnostic Rhyme Test

PCM - Pulse Code Modulation

#### B.4 GENERAL REQUIREMENTS

**B.4.1 General.** New implementations of MELP must be verified to assure that their performance meets or exceeds the performance of the MELP reference coder. A new implementation must also meet the same performance standards when interoperating with the MELP reference coder. Testing is accomplished through formal evaluation of intelligibility and quality or by showing bit equivalence between the new implementation and a verified MELP implementation. Both verification methods evaluate an implementation over an extensive set of conditions.

#### B.5 DETAILED REQUIREMENTS

**B.5.1 Formal evaluation.** The intelligibility of a new implementation is evaluated using the DRT. DRT performance thresholds have been set and are discussed later in this section. Quality is evaluated using a direct paired forced choice comparison test, i.e., an A/B test. Performance thresholds for the quality tests have also been set and are based on the percent preference for the new implementation over the MELP reference coder. The Federal Standard CELP coder is also included in the quality test to broaden the context of the test and to include a known difference from MELP. Table B-I summarizes the coder configurations which are evaluated for intelligibility and quality. In table B-I, the "Implementation → MELP Reference" and "MELP Reference → Implementation" cases are "cross-coder" configurations which test interoperability. Table B-II shows the intelligibility and quality test conditions. Six talkers (3 male, 3 female) are used for each condition.

**TABLE B-I. Tested coder configurations.**

<b>Intelligibility (Encoder → Decoder)</b>	<b>Quality (Encoder → Decoder)</b>
Implementation → Implementation	Implementation → Implementation
Implementation → MELP Reference	Implementation → MELP Reference
MELP Reference → Implementation	MELP Reference → Implementation
	MELP Reference → MELP Reference
	CELP → CELP

## MIL-STD-3005

## APPENDIX B

<b>TABLE B-II. Intelligibility and quality test conditions.</b>	
<b>Intelligibility Condition (microphone)</b>	<b>Quality Condition (microphone)</b>
Quiet (Dynamic)	Quiet (Dynamic)
Quiet (H250)	Quiet (H250)
Office (STU-III)	Office (STU-III)
E3A AWACS (R215)	E3A AWACS (R215)
HMMWV (H250)	HMMWV (H250)
MCE Field Shelter (M87)	MCE Field Shelter (M87)
Car (STU-III cellular)	Car (STU-III cellular)
0.5% Block Error Rate in Quiet (Dynamic)	0.5% Block Error Rate in Quiet (Dynamic)
1.0% Bit Error Rate in Quiet (Dynamic)	1.0% Bit Error Rate in Quiet (Dynamic)
CVSD → Coder in Quiet (Dynamic)	CVSD → Coder in Quiet (Dynamic)
CVSD → Coder → CVSD in Quiet (Dynamic)	CVSD → Coder → CVSD in Quiet (Dynamic)
M2 Bradley (M138)	
CH47 helicopter (M87)	
F-15 jet (M101)	
P3C Orion plane (EV985)	

B5.1.1 Intelligibility tests. The DRT will be performed in accordance with ANSI standard S3.2-1989 and will be scored with eight listeners. The combined talker score determined by the test lab must meet or exceed the corresponding threshold score for each condition and for the weighted combination of all conditions. Table B-III shows the weight and threshold score for each condition. The implementation's combined score is calculated by multiplying its individual score for each condition by the corresponding weight given in table B-III. The results are then summed to produce the combined result. This process is used for each of the three coder configurations evaluated for intelligibility. The threshold score for each condition is based on a one-tail 99.5% confidence interval; the combined threshold score is based on a one-tail 99% confidence interval.

## MIL-STD-3005

## APPENDIX B

**TABLE B-III. Weights and thresholds for intelligibility conditions.**

<b>Intelligibility condition (microphone)</b>	<b>Weight</b>	<b>Threshold score</b>
Quiet (Dynamic)	0.100	90.80
Quiet (H250)	0.100	88.77
Office (STU-III)	0.0667	89.34
E3A AWACS (R215)	0.0667	85.10
HMMWV (H250)	0.0667	61.88
MCE Field Shelter (M87)	0.0667	87.28
Car (STU-III cellular)	0.0667	82.48
M2 Bradley (M138)	0.0667	61.33
CH47 helicopter (M87)	0.0667	63.27
F-15 jet (M101)	0.0667	75.06
P3C Orion plane (EV985)	0.0667	82.98
0.5% Block Error Rate in Quiet (Dynamic)	0.050	89.58
1.0% Bit Error Rate in Quiet (Dynamic)	0.050	87.47
CVSD → Coder in Quiet (Dynamic)	0.050	84.07
CVSD → Coder → CVSD in Quiet (Dynamic)	0.050	81.33
Combined (weighted)	1.000	82.634

B5.1.2 Quality tests. The quality of the three coder configurations involving the new implementation (see table B-I) is compared with the quality of the MELP reference coder using a direct paired forced choice comparison (A/B) test. Quality is measured in each condition by the percent preference taken over all talkers and listeners. For each coder configuration involving the new implementation, the percent preferred must meet or exceed the threshold for each condition and for the weighted combination of all conditions.

The threshold for individual conditions is 45.84%, i.e., each coder configuration involving the new implementation must have a preference percentage of 45.84% or more in each condition. This threshold is equivalent to saying that the new implementation must be selected at least 440 times out of the 960 comparisons between the new implementation and the reference in each condition. The threshold is based on a one-tail 99.5% confidence interval.

The combined percent preferred is calculated for each of the three coder configurations by summing the weighted individual condition percentages using the weights listed in table B-IV. The combined threshold percentage is 48.71%, i.e., each of the three coder configurations involving the new implementation must have a combined preference percentage of 48.71% or more. The combined threshold is based on a one-tail 99% confidence interval.

## MIL-STD-3005

## APPENDIX B

<b>TABLE B-IV. Weights for quality conditions.</b>	
<b>Quality condition (microphone)</b>	<b>Weight</b>
Quiet (Dynamic)	0.245
Quiet (H250)	0.105
Office (STU-III)	0.070
E3A AWACS (R215)	0.070
HMMWV (H250)	0.070
MCE Field Shelter (M87)	0.070
Car (STU-III cellular)	0.070
0.5% Block Error Rate in Quiet (Dynamic)	0.075
1.0% Bit Error Rate in Quiet (Dynamic)	0.075
CVSD → Coder in Quiet (Dynamic)	0.075
CVSD → Coder → CVSD in Quiet (Dynamic)	0.075

**B.5.2 Bit equivalence.** A lower cost alternative for implementation verification is accomplished by demonstrating that the new implementation is bit equivalent to a verified MELP implementation. Bit equivalent means that given the same input, the new implementation's encoder produces the same bitstream as produced by the encoder of the verified MELP implementation. Also given the same bitstream, the new implementation's decoder produces the same 16 bit PCM samples as produced by the decoder of the verified MELP implementation. This equivalence must be shown for all intelligibility material and quality material used in the formal evaluation.

MIL-STD-3005

CONCLUDING MATERIAL

Custodians:

Army – CR  
Navy – EC  
Air Force – 02  
NSA – NS

Preparing Activity:

NSA – NS  
(Project TCSS-0044)

Review Activities:

Army – AM  
Navy – MC, NC, TD  
Air Force – 11, 13, 19, 93  
DISA – DC1

# STANDARDIZATION DOCUMENT IMPROVEMENT PROPOSAL

## INSTRUCTIONS

1. The preparing activity must complete blocks 1, 2, 3, and 8. In block 1, both the document number and revision letter should be given.
2. The submitter of this form must complete blocks 4, 5, 6, and 7, and send to preparing activity.
3. The preparing activity must provide a reply within 30 days from receipt of the form.

NOTE: This form may not be used to request copies of documents, nor to request waivers, or clarification of requirements on current contracts. Comments submitted on this form do not constitute or imply authorization to waive any portion of the referenced document(s) or to amend contractual requirements.

**I RECOMMEND A CHANGE:**
**1. DOCUMENT NUMBER**  
MIL-STD-3005

**2. DOCUMENT DATE (YYYYMMDD)**  
19991220

**3. DOCUMENT TITLE** ANALOG-TO-DIGITAL CONVERSION OF VOICE BY 2,4000 BIT/SECOND MIXED EXCITATION LINEAR PREDICTION (MELP)

**5. REASON FOR RECOMMENDATION**
**6. SUBMITTER**

a. NAME <i>(Last, First, Middle Initial)</i>	b. ORGANIZATION
c. ADDRESS <i>(Include Zip Code)</i>	d. TELEPHONE <i>(Include Area Code)</i> (1) Commercial (2) AUTOVON <i>(if applicable)</i>
<b>7. DATE SUBMITTED</b> <b>(YYYYMMDD)</b>	

**8. PREPARING ACTIVITY**

a. NAME NATIONAL SECURITY AGENCY	b. TELEPHONE <i>(Include Area Code)</i> (1) Commercial (301) 688-3586	(2) AUTOVON 644-3586
c. ADDRESS <i>(Include Zip Code)</i> NATIONAL SECURITY AGENCY 9800 SAVAGE ROAD STE 6516 FT. MEADE, MD 20755-6516	<b>IF YOU DO NOT RECEIVE A REPLY WITHIN 45 DAYS, CONTACT:</b> Defense Standardization Program Office (DLSC-LM) 8725 John J. Kingman road, Suite 2533, Ft. Belvoir, VA 22060-2533 Telephone (703) 767-6888	