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FIPS PUBLICATION CHANGE NOTICE

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137

PUBLICATION TITLE

FIPS PUB 137, Analog to Digital Conversion of Voice by 2,400 Bit/Second Linear Predictive Coding.

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CHANGE ITEM(S)

Specific Change

Section 1.3, Application which requires the capability for LPC-10 in all Government 2,400 bit/s synchronous digitized voice equipment is changed to make the use of and capability for LPC-10 operation **non-mandatory.**

This change will allow for the use of newer 2,400 bit/s digitized voice techniques, including Mixed Excitation Linear Prediction (MELP), that has recently been developed and will soon be standardized.

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Federal Standard 1015 has been redesignated as Federal Information Processing Standards Publication (FIPS PUB) 137. Issued by the National Institute of Standards and Technology pursuant to Section 111(d) of the Federal Property and Administrative Services Act of 1949 as amended by the Computer Security Act of 1987, Public Law 100-235

FEDERAL STANDARD 1015

ANALOG TO DIGITAL CONVERSION OF VOICE BY 2,400 BIT/SECOND LINEAR PREDICTIVE CODING

Prepared By:
National Communications System
Office Of Technology & Standards

Published By:
General Services Administration
Office Of Information Resources Management

November 28, 1984

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FEDERAL STANDARD

TELECOMMUNICATIONS: ANALOG TO DIGITAL CONVERSION OF VOICE BY 2,400 BIT/SECOND LINEAR PREDICTIVE CODING

This standard is issued by the General Service Administration pursuant to the Federal Property and Administrative Services Act of 1949, as amended.

1. Scope

1.1 Description. This standard specifies interoperability requirements relating to the conversion of analog voice to 2,400 bit/s digitized voice by Linear Predictive Coding with 10 reflection coefficients (LPC-10), and reconversion back to analog voice.

1.2 Objective. The primary objective of this standard is to facilitate the interoperability of Government communication facilities and systems that employ 2,400 bit/s digitized voice.

1.3 Application. This standard applies to all synchronous (i.e. not packetized) 2,400 bit/s digitized voice telecommunications equipment procured or leased by Federal departments and agencies. While additional analog-to-digital conversion techniques and data rates may be used, all Government synchronous 2,400 bit/s digitized voice equipment shall be capable of LPC-10 operation in conformance with this standard.

2. Related Standards

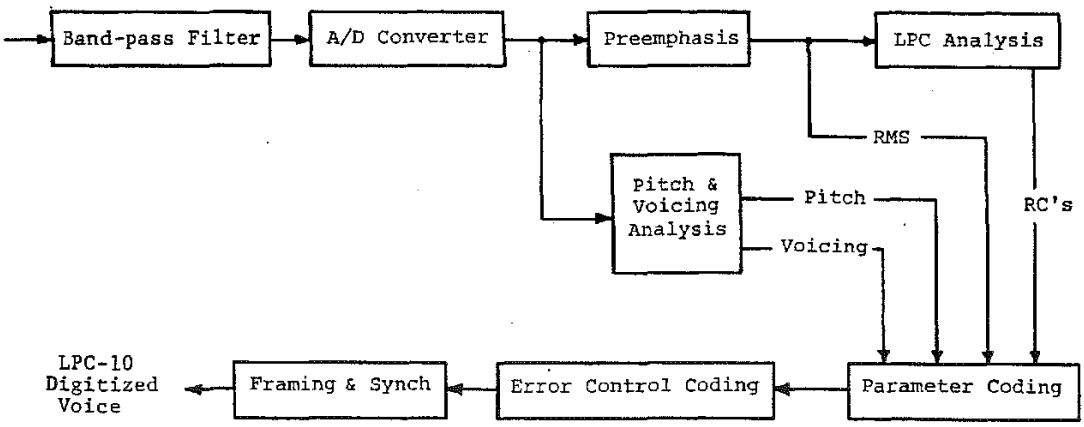
a. Military Standard 188-113: COMMON LONG HAUL/TACTICAL STANDARDS FOR ANALOG/DIGITAL CONVERSION TECHNIQUES. (Copies of this standard are available from the Naval Publications and Forms Center, 5801 Tabor Avenue, Philadelphia, PA 19120).

b. North Atlantic Treaty Organization (NATO) Standardization Agreement (STANAG) 4198: PARAMETERS AND CODING CHARACTERISTICS THAT MUST BE COMMON TO ASSURE INTEROPERABILITY OF 2400 BPS LINEAR PREDICTIVE ENCODED DIGITAL SPEECH (Controlled Distribution). (Copies of this standard are available from the Naval Publications and Forms Center, 5801 Tabor Avenue, Philadelphia, PA 19120).

3. Requirements

3.1 Typical Block Diagrams

3.1.1 Transmitter. The block diagram of a typical Linear Predictive Coding (LPC) transmitter is shown below.



As shown in the diagram above, speech is band-pass filtered prior to analog-to-digital (A/D) conversion. After A/D conversion and preemphasis, Linear Predictive Coding (LPC) analysis is accomplished to determine reflection coefficients (RCs). Also, root-mean-square (RMS) amplitude is calculated. Additionally, after A/D conversion, pitch and voicing analysis is accomplished to determine whether to treat the LPC coded frame as voiced or otherwise. If voiced, pitch is determined. If not voiced, a determination is made whether the frame is to be considered as unvoiced or in voicing transition. Next, the RMS amplitude, reflection coefficients, and pitch and voicing are coded; transmission error control coding is added as applicable; a synchronization bit is added; and a 54-bit frame is formed.

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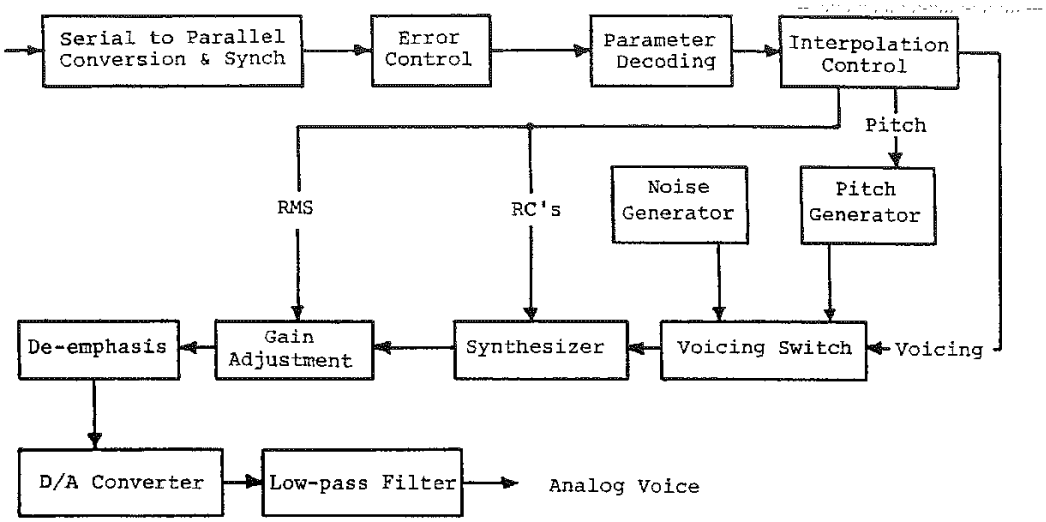
3.1.2 Receiver. The block diagram of a typical Linear Predictive Coding (LPC) receiver is shown below.



As shown in the diagram above, the Linear Predictive Coding (LPC) receiver's digital input is first unframed and error detection and forward error correction are performed, as applicable. Parameter decoding is then performed, along with interpolation and other enhancements, to obtain the proper voicing decision, pitch (for voiced frames), reflection coefficients, and RMS amplitude. Of these parameters, the voicing decision is forwarded to the voicing switch function that chooses the random noise generator (unvoiced) or the pitch generator (voiced) as the synthesizer input. Likewise, the correct pitch is forwarded to the pitch generator (if the frame is determined to be voiced), the reflection coefficients are forwarded to the synthesizer function, and the RMS amplitude is forwarded to the gain adjustment function. Lastly, the output of the gain adjustment function is de-emphasized, converted from digital to analog by a D/A converter, and low-pass filtered. (Note that some filtering is accomplished as a part of the de-emphasis function).

3.2 Input Conditioning

3.2.1 Band-pass Filtering. The LPC transmitter input passband should be essentially flat from 100-3600 Hz. A typically used input filter has 3 dB attenuation points at 100 and 3,600 Hz, less than 1 dB of inband ripple, and attenuations at 4,000 and 4,400 Hz of 23 dB and 46 dB, respectively.

3.2.2 A/D Conversion. Analog-to-digital (A/D) conversion shall use an 8 kHz ±0.1 percent sampling frequency and have a dynamic range of at least 12 bits.

3.2.3 Preemphasis. Preemphasis shall use the first order digital transfer function $1 - 0.9375\ z^{-1}$.

3.3 Transmission Format

3.3.1 Transmission Rate. The transmission rate shall be 2400 bits/s ±.01 percent. Since all frames contain 54 bits, the frame length is 22.5 ms ±.01 percent.

3.3.2 Bit Allocation. The allocation of the 54 bits in an LPC frame shall be as shown in the following table.

	Voiced	Nonvoiced
Pitch & Voicing	7	7
RMS Amplitude	5	5
RC(1)	5	5
RC(2)	5	5
RC(3)	5	5
RC(4)	5	5
RC(5)	4	
RC(6)	4	
RC(7)	4	
RC(8)	4	
RC(9)	3	
RC(10)	2	
Error Control		20
Synchronization	1	1
Unused		1
Total	54	54

Note: RC = Reflection Coefficient
Nonvoiced = Unvoiced or In Voicing Transition

3.3.3 Bit Assignment. The assignment of bits within an LPC frame shall be as shown in the following table.

Bit	Voiced	Nonvoiced	Bit	Voiced	Nonvoiced	Bit	Voiced	Nonvoiced
1	RC(1)-0	RC(1)-0	19	RC(3)-3	RC(3)-3	37	RC(8)-1	R-6*
2	RC(2)-0	RC(2)-0	20	RC(4)-2	RC(4)-2	38	RC(5)-1	RC(1)-6*
3	RC(3)-0	RC(3)-0	21	R-3	R-3	39	RC(6)-1	RC(2)-6*
4	P-0	P-0	22	RC(1)-4	RC(1)-4	40	RC(7)-2	RC(3)-7*
5	R-0	R-0	23	RC(2)-3	RC(2)-3	41	RC(9)-0	RC(4)-6*
6	RC(1)-1	RC(1)-1	24	RC(3)-4	RC(3)-4	42	P-5	P-5
7	RC(2)-1	RC(2)-1	25	RC(4)-3	RC(4)-3	43	RC(5)-2	RC(1)-7*
8	RC(3)-1	RC(3)-1	26	R-4	R-4	44	RC(6)-2	RC(2)-7*
9	P-1	P-1	27	P-3	P-3	45	RC(10)-1	Unused
10	R-1	R-1	28	RC(2)-4	RC(2)-4	46	RC(8)-2	R-7*
11	RC(1)-2	RC(1)-2	29	RC(7)-0	RC(3)-5*	47	P-6	P-6
12	RC(4)-0	RC(4)-0	30	RC(8)-0	R-5*	48	RC(9)-1	RC(4)-7*
13	RC(3)-2	RC(3)-2	31	P-4	P-4	49	RC(5)-3	RC(1)-8*
14	R-2	R-2	32	RC(4)-4	RC(4)-4	50	RC(6)-3	RC(2)-8*
15	P-2	P-2	33	RC(5)-0	RC(1)-5*	51	RC(7)-3	RC(3)-8*
16	RC(4)-1	RC(4)-1	34	RC(6)-0	RC(2)-5*	52	RC(9)-2	RC(4)-8*
17	RC(1)-3	RC(1)-3	35	RC(7)-1	RC(3)-6*	53	RC(8)-3	R-8*
18	RC(2)-2	RC(2)-2	36	RC(10)-0	RC(4)-5*	54	Synch.	Synch.

NOTES:
P = Pitch
R = RMS Amplitude
RC = Reflection Coefficient
* = Error Control Bit
Bit 0 = least significant bit of data
Bit 5 = least significant bit of error control
Order of transmission is from bit 1 to bit 54
Nonvoiced = Unvoiced or In Voicing Transition

3.3.4 Synchronization. The synchronization bit shall alternate between ZERO and ONE from frame to frame.

3.4 Pitch and Voicing

3.4.1 Encoding. Pitch and voicing information shall be coded as a seven bit field. For error protection purposes, unvoiced frames shall be coded as seven ZERO bits and frames in voicing transition shall be coded as seven ONE bits. For voiced frames, one of 60 selected pitch values (51-400 Hz) shall be selected and coded as shown in the following table. Note that the pitch period is the sampling frequency (8,000 Hz) divided by the pitch frequency.

Pitch Freq	Pitch Period	Code	Pitch Freq	Pitch Period	Code	Pitch Freq	Pitch Period	Code	Pitch Freq	Pitch Period	Code
51	156	76	83	96	78	138	58	26	235	34	46
53	152	101	87	92	74	143	56	58	242	33	38
54	148	100	91	88	75	148	54	56	250	32	39
56	144	108	95	84	73	154	52	60	258	31	7
57	140	104	100	80	77	160	50	52	266	30	15
59	136	106	103	78	69	167	48	54	276	29	14
61	132	98	105	76	85	174	46	50	286	28	30
63	128	114	108	74	81	184	44	51	296	27	22
65	124	112	111	72	83	190	42	49	308	26	23
67	120	113	114	70	82	200	40	53	320	25	21
69	116	97	118	68	86	205	39	37	333	24	29
71	112	99	121	66	84	210	38	45	348	23	25
74	108	67	125	64	92	216	37	41	364	22	27
77	104	71	129	62	88	222	36	43	381	21	11
80	100	70	133	60	90	228	35	42	400	20	19

3.4.2 Decoding. The following table shall be used in decoding the seven-bit pitch and voicing field to determine if a frame is unvoiced (U), is in voicing transition (T), is invalid (I), or is voiced. If voiced, the decoded value shall be used as the pitch period (8,000 Hz sampling frequency divided by pitch frequency).

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