Stand-off Jamming Method in Electronic Warfare against Defense Systems

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ABSTRACT:
This article compares the quality and complexity of LPC, CELP, and MELP standard audio encoders. These standards are based on linear predictive and are used in sound (speech) processing. These standards are powerful high-quality speech coding methods that provide highly accurate estimates of audio parameters and are widely used in the commercial (mobile) and military (NATO) communications industries. To compare LPC, CELP, and MELP audio encoders in two male and female voice modes and four voice models: quiet, Audio recorded without sound by the microphone, MCE, office, and two noise models 1% and 05% were used. The simulation results show the complexity of MELP is higher than LPC and CELP in terms of both processor and memory requirements. The MELP analyzer requires 72% of its total processing time. This additional memory is, of course, due to the vector quantization tables that MELP uses for the linear spectral frequencies (LSFs) and the Fourier magnitude. Also, according to the quality comparison test using the MOS index, MELP has the highest score, followed by CELP and LPC.

KEYWORDS: Quality, Complexity, LPC, CELP, MELP.

1. INTRODUCTION

In 1966, Linear Predictive Coding (LPC) was presented and in 1978 this method was completed [1]. LPC is one of the most common audio coding methods that converts analog audio to digital at 2400 bps.

LPC is one of the powerful methods of high-quality audio encoder analysis that provides very accurate estimates of audio parameters. The way LPC works is that speech-like audio signals are produced by a noise, and sounds with frequencies have successively added to them alternately. This method is the closest approximation to the real sound.

The Code Excited Linear Predictive (CELP) was presented in 1985 [2]. CELP is a linear speech encoder programming algorithm that converts analog audio to digital audio at 4800 bits per second. This method is high quality and is used in MPEG-4 audio speech encoder.

The Mixed Excitation Linear Predictive (MELP) was registered in 1995 based on LPC. This audio speech coder was standardized in 1997[3]. This method is one of the most common audio encoding methods that converts analog audio to digital at 2400 bits per second. This method is mainly used in military applications and satellite communications, secure voice transmission, and the safety of radio communications.

2. LPC

The working method of LPC is that sound signals similar to speech are produced by noise and sounds with alternating frequencies are successively added to it. This method is the closest approximation to the real sound. LPC analyzes the speech signal by estimating the forms, removing their effects from the speech signal, and estimating the intensity and frequency of the residual noise. The process of eliminating these forms is called inverse filtering, and the remaining signal after subtracting the filtered modeled signal is called the remaining signal. Since the speech signals are different, this process is done in short pieces of the speech signal,

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which are called frames. In general, LPC compresses the speech signal at 30 to 50 frames per second.

Because LPC is often used to transmit spectrum information, it must be tolerant of transmission errors. Transferring the filter coefficients directly is undesirable because they are very sensitive to error.

In other words, a very small error can change the entire spectrum

2.1. BIT Allocation

Bit allocation of LPC frame should be according to the following table [4].

Table 1. Bit Allocation of LPC Encoder.

Bit	Voiced	Unvoiced	Bit	Voiced	Unvoiced	Bit	Voiced	Unvoiced
1	RC (l)-0	RC(l)-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	P1-0	P-0	22	RC(l)-4	RC(l)-4	40	RC(7)-2	RC(3)-7*
5	R2-0	R-0	23	RC(2)-3	RC(2)-3	41	RC(9)-0	RC(4)-6*
6	RC(l)-1	RC(l)-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(l)-2	RC(l)-2	29	RC(7)-0	RC(3)-5*3	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-3*
18	RC(2)-2	RC(2)-2	36	RC(10)-0	RC(4)-5*	54	Synch.	Synch.

3. CELP

CELP is essentially Analysis with Synthesis (AbS) meaning that coding (analysis) is performed with perceptual optimization of the decoded signal (synthesis) in a closed loop, the high complexity of CELP was initially an impractical proposition. However, many ways to speed up the coding process have been found and CELP has become a practical reality [5].

3.1. Bit Allocation

Bit allocation of CELP frame should be according to the following table [4].

Table 2. Bit Allocation of CELP Encoder.

1 P = Pitch
2 R = RMS Amplitude
3 * = Error Control Bit
Bit 0 = least significant bit of data

Bit	Description	Bit	Description	Bit	Description	Bit	Description	Bit	Description	Bit	Description
1	PG(4)-4[4]	25	PG(3)-1	49	LSP 1-2	73	PD(l)-4	97	PG(l)-2	121	LSP 7-2
2	PD(3)-4[5]	26	PD(4)-5	50	PG(3)-2	74	CG(3)-2	98	CG(3)-4	122	CI(4)-2
3	LSP 1-1[6]	27	CG(l)-3	51	HP-1	75	LSP 7-1	99	LSP 10-2	123	PD(l)-1
4	CG(2)-4[7]	28	CI(3)-5	52	PD(3)-1	76	CI(2)-7	100	CI(4)-5	124	PG(2)-4
5	CI(3)-3[8]	29	LSP 7-O	53	CG(4)-3	77	CI(3)-O	101	CI(2)-O	125	CG(3)-3
6	CI(l)-8	30	CI(2)-1	54	LSP 8-1	78	PD(2)-5	102	PD(l)-2	126	LSP 3-1
7	PD(4)-O	31	PD(3)-7	55	PG(3)-O	79	LSP 4-1	103	LSP 5-1	127	CI(l)-7
8	LSP 8-O	32	CI(l)-O	56	CI(2)-8	80	CG(l)-O	104	SP-O[9]	128	PD(3)-2
9	PG(2)-3	33	PG(4)-O	57	PD(4)-1	81	PG(4)-3	105	PG(4)-2	129	CI(2)-6
10	CG(3)-O	34	LSP 4-3	58	CI(4)-O	82	LSP 9-1	106	CG(2)-3	130	LSP 9-2
11	PD(l)-5	35	CG(3)-1	59	LSP 3-2	83	PD(3)-6	107	LSP 2-1	131	PG(4)-1
12	LSP 3-3	36	CI(l)-5	60	PG(2)-O	84	CI(l)-4	108	PD(4)-4	132	CG(l)-1
13	CI(2)-3	37	PD(2)-O	61	PD(l)-6	85	CG(2)-1	109	CI(l)-2	133	PD(2)-4
14	CI(4)-4	38	CI(4)-1	62	CG(2)-O	86	LSP 6-2	110	PG(2)-1	134	HP-3
15	PD(2)-1	39	LSP 9-O	63	CI(3)-6	87	CI(4)-3	111	CI(3)-7	135	LSP 6-O
16	LSP 10-0	40	CI(3)-8	64	LSP 10-1	88	PG(2)-2	112	LSP 4-O	136	PG(3)-3
17	PG(l)-3	41	PG(l)-4	65	PG(l)-1	89	PD(4)-3	113	CI(2)-5	137	CI(4)-6
18	CG(4)-O	42	CG(2)-2	66	CI(4)-7	90	LSP 1-0	114	PD(l)-7	138	PD(l)-O
19	LSP 5-2	43	PD(l)-3	67	PD(3)-3	91	CG(4)-2	115	PG(l)-O	139	LSP 2-3
20	PD(3)-O	44	LSP 6-1	68	CG(l)-2	92	LSP 8-2	116	CG(4)-4	140	CG(4)-1
21	HP-O[10]	45	CI(3)-4	69	LSP 5-3	93	CI(2)-4	117	LSP 5-O	141	CI(3)-2
22	CI(l)-1	46	CI(2)-2	70	CI(l)-6	94	HP-2	118	PD(4)-2	142	LSP 4-2
23	CI(4)-8	47	CG(l)-4	71	LSP 2-O	95	PD(2)-2	119	CI(l)-3	143	PD(3)-5
24	LSP 2-2	48	PD(2)-3	72	PG(3)-4	96	LSP 3-O	120	CI(3)-1	144	SY[11]

4. MELP

MELP is based on the traditional LPC model and uses additional features such as mixed excitation, non-periodic pulses, adaptive spectrum enhancement, pulse dispersion filter, and Fourier magnitude modeling to improve performance. Adding these features allows the encoder to better match the features of the input speech. [6].

4.1. Bit Allocation

Bit allocation of MELP frame should be according to the following table [6].

Table 3. Bit Allocation of MELP Encoder.

4 PG(n)-i = Adaptive Code Gain
5 PD(n)-i = Adaptive Code Index
6 LSP j-i = Line Spectral Parameter (LSP),
where j = LSP number
7 CG(n)-i = Fixed, Stochastically-derived Code Gain
8 CI(n)-i = Fixed, Stochastically-derived Code Index
9 SP = Expansion Bit
10 HP-i = Parity
11 SY = Synchronization Bit
Note: i = bit number, with O being the least significant bit
n = subframe number

Bit	Voiced	Unvoiced	Bit	Voiced	Unvoiced	Bit	Voiced	Unvoiced
1	G[12](2)-1	G(2)-1	19	LSF(1)-7	LSF(1)-7	37	G(1)-1	G(1)-1
2	BP[14]-1	FEC[13](1)-1	20	LSF(4)-6	LSF(4)-6	38	BP-3	FEC(1)-3
3	P[15]-1	P-1	21	P-4	P-4	39	BP-2	FEC(1)-2
4	LSF[16](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[17]	FEC(4)-3
12	LSF(3)-5	LSF(3)-5	30	FM[18]-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

5. COMPARISON

Audio encoder standards LPC, CELP, and MELP were thoroughly reviewed. It is necessary to compare their performance in terms of quality, Intelligibility, Communicability, Recognizability, and complexity for two different types of speech (male and female) in order to conclude which one performs better.

5.1. Quality

For quality testing, we use MOS[19] for benign noise conditions [7]. Quality testing is often used to supplement or replace comprehensibility testing. It provides a picture of the listeners' personal opinions about the signal sent by the communication systems or processed by the algorithms under test.

MOS test has been done in four audio noise conditions and two-channel conditions. The two error environments tested are: a 1% random bit error channel and a 0.5% random block error channel. Block error contains 50% error in a 35 ms block. Q-H250 is Audio recorded without sound by the microphone

"MCE" is a mobile command environment. The office is recorded in a modem office. Quiet is a soundless environment [8].

The MOS test results for LPC, CELP, and MELP audio encoders are shown in Figs. 1 to 4.

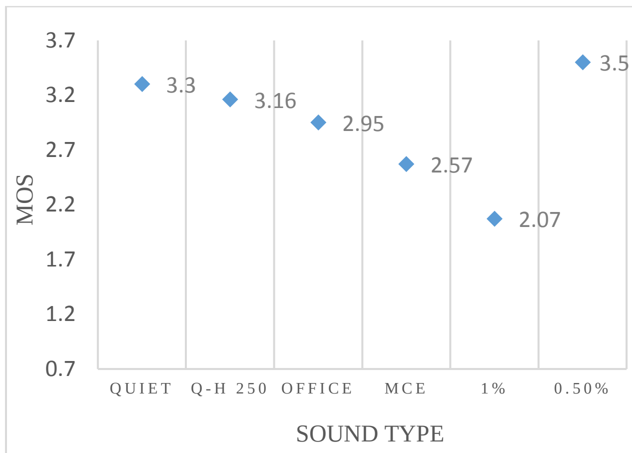

Fig. 1. MOS Test Result for LPC

12 Gain
13 Forward Error Correction Parity Bits
14 Band pass Voicing
15 Pitch voicing
16 Line Spectral Frequencies
17 Aperiodic Flag
18 Fourier Magnitude
Note: Bit 1 = least significant bit of data set
19 Mean Opinion Score

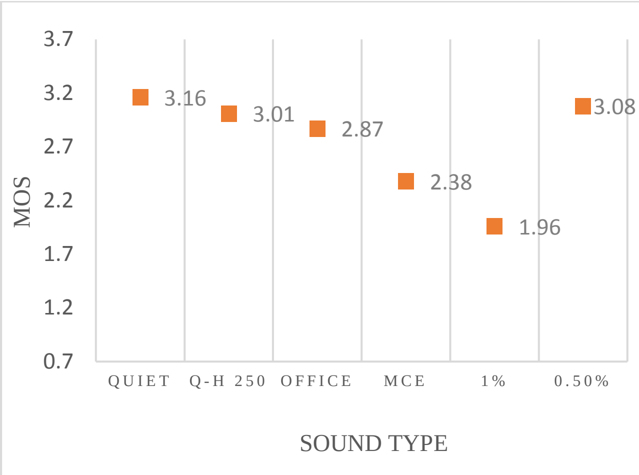

Fig. 2. MOS Test Result for CELP.

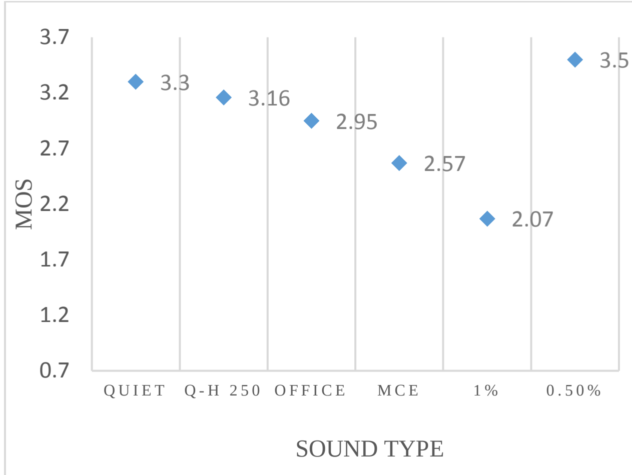

Fig. 3. MOS Test Result for MELP.

Fig. 4. MOS Comparison.

Relative coder ranking is easily seen in Figure 4. In all environments, MELP shows the highest MOS score, followed by CELP, and LPC.

MELP and LPC coders scored higher overall for male speakers than female speakers. Only in the 0.5% block error condition did the female MELP score exceed the male score, but this variance was within the standard error. The CELP coder, in contrast, scored higher overall for female speakers than for male speakers. This was especially bad in the office environment. CELP on QH250 also showed significantly higher scores for female speakers [8]. Table 4 shows the simulation results obtained from each of the standards in different modes.

Table 4. MOS simulation results.

Bit	Quiet	Q-H250	Office	MCE	1%	. 5%
MELP	3.30	3.16	2.95	2.57	2.07	3.50
CELP	3.16	3.01	2.87	2.38	1.96	3.08
LPC	2.20	1.98	2.08	1.09	.98	2.31

6. COMPLEXITY

Complexity was measured using MIPS[20], read only memory (ROM) and random access memory (RAM) measurements.

Tab5: Complexity Comparison [8].

Coder	MIPS	RAM	ROM
MELP	20.43	98.2K	128K
CELP	17.0	14.8K	128K
LPC	8.7	12.93K	128K

As Table 5 shows, MELP complexity exceeds, LPC, and CELP in both processor and memory requirements. The MELP analyzer requires 72% of its total processing. These additional memory requirements are due to vector quantization tables which MELP uses for both line spectral frequencies (LSFs) and Fourier magnitudes [8].

SUPPORT

Adapted from Saeed Talati's doctoral thesis at comprehensive Imam Hossein University entitled "Recognition of digital audio steganography in LPC10, CELP, and MELP audio encoder standards".

7. CONCLUSION

In this article, Standard audio Encoders LPC, CELP, and MELP are compared in the two areas of quality and complexity. These audio coding techniques are powerful audio coding standards that are widely used in the mobile, commercial and military industries (official NATO standard).

The quality comparison test using different sounds is given in figure 4. The obtained results show that MELP has the highest score, followed by CELP and LPC.

Quality comparison using the MOS index shows that MELP has the highest score, followed by CELP and LPC.

The complexity comparison test using different voices is shown in Table 4. The obtained results show that the complexity of MELP is higher than LPC and CELP in terms of both processor and memory requirements. The MELP analyzer requires 72% of its total processing time. This additional memory is, of course, due to the vector quantization tables that MELP uses for the linear spectral frequencies (LSFs) and the Fourier magnitude.

20 million instructions per second

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