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Low Bit-Rate Parametric Coding

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Abstrak - Dengan kecepatan bit yang rendah akan diperoleh penghematan penggunaan lebar pita *(bandwidth)* saluran transmisi dan memori. Pada pengkodean sinyal suara manusia dibuat model tiruan penghasil suara agar diperoleh kecepatan bit yang rendah yang dikenal dengan metode *parametric coding*. Dengan metode *parametric coding* sinyal suara manusia dapat menghasilkan kecepatan bit *(bit-rate)* yang lebih rendah dibandingkan dengan metode pengkodean bentuk gelombang *(wave form coding)*. Contohnya pada metode pengkodean bentuk gelombang akan dibatasi oleh batas minimal frekuensi cuplik *(sampling frequency)* yang menurut teorema Shannon-Nyquist menyatakan agar tidak ada informasi yang hilang ketika pencuplikan sinyal, maka kecepatan pencuplikan harus minimal dua kali dari lebar pita sinyal tersebut. Dengan demikian pada pengkodean bentuk gelombang kecepatan bit terendah yang dapat dicapai adalah oleh sistem Modulasi Delta (Delta Modulation) yaitu sebesar 8 Kbit/detik. Sedangkan pada pengkodean parametrik sinyal dapat dicapai kecepatan bit yang lebih rendah yaitu sebesar 2,4 Kbit/detik oleh sistem LPC *(Linear Predictive Coding).*

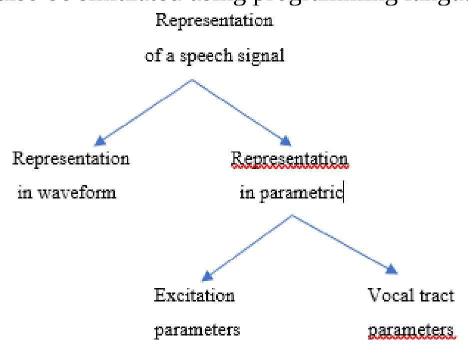
Kata Kunci: LPC, Speech Processing, Speech Compression

Abstract *- With a low bit rate will be obtained savings in the use of bandwidth transmission channels and memory. In coding the human voice signal, an artificial voice-producing model is made to obtain a low bit rate, which is known as the parametric coding method. With the parametric coding method, the human voice signal can produce a lower bit rate than the waveform coding method. For example, the waveform coding method will be limited by a minimum sampling frequency which according to the Shannon-Nyquist theorem states that so that no information is lost when sampling the signal, the sampling speed must be at least twice the bandwidth of the signal. Thus, the lowest bit rate waveform encoding that can be achieved is by the Delta Modulation system which is 8 Kbit/sec. While the signal parametric coding can achieve a lower bit rate of 2.4 Kbit/sec by the LPC (Linear Predictive Coding) system.*

Keywords*: LPC, Speech Processing, Speech Compression*

INTRODUCTION

The progress of human aids in the form of digital computers which is very extraordinary at this time makes all types of information in analog form must be converted first into digital form so that it can be processed by a digital computer. As in the human voice as a communication tool in the form of analog signals, it can be represented into two types of methods of converting analog form to binary digital form, namely waveform coding and encoding analog signal parameters (Singh, 2015), as shown in Figure 1. Waveform coding is a classic coding method that was discovered since the progress of digital computers has not been as fast as it is today, so that its manufacture still uses analog electrical circuits such as the PCM (Pulse Code Modulation), ADPCM (Adaptive PCM) sistem, and Delta Modulation (Sudaradjat, Teknik Adaptive Pada Modulasi Delta, 2019). However, after the rapid development of digital computers supported by the development of software technology, the analog signal coding system can also be simulated using programming languages.



Source: (Singh, 2015)

Figure 1. Classification of the speech signal.

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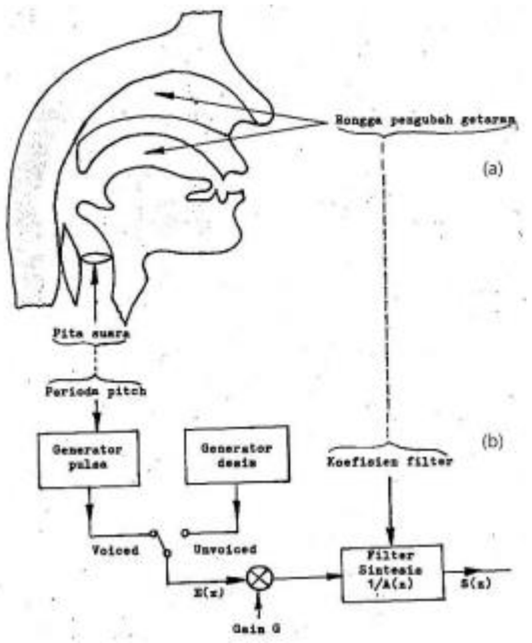
For example, the human voice signal generated by the human oral cavity can be imitated so as to produce a human voice like the original, namely by imitating the parameters that form human speech, which consists of vocal cords, cavities that change sound vibrations, voice volume, and the type of voice generated. as shown by figure 2 (Djadjat Sudaradjat, 2020).



Source : (Djadjat Sudaradjat, 2020)

Figure 2. (a) Human speech formation chart, (b) Image of a model for the formation of human speech.

The three main factors involved in the formation of an utterance such as the speech apparatus that causes vibration, the energy source, and the vibration converter cavity can be modeled as shown in Figure 2 (b). The vocal cords that cause vibrations are modeled as two types of excitation functions, namely first if the sound produced is vibrating then a pulse generator model is used that will work with a period equal to the pitch period and second if the sound produced is non-vibrating then a hiss generator model is used (random) which will work. The power source that determines the loudness of the sound is modeled as gain, while the vibration converter cavity is modeled as a digital filter. The filter is implemented on a computer in the z-region, from Figure 2(b) we get:

$\mathrm{S(z)} \; = \; \mathrm{E(z).G.}\frac{1}{\mathrm{A(z)}}$ (synthesis model) ....(1)

with :

$A(z) = \sum_{i=0}^{p} \; a(i)z^{i}$

$(\mathrm{a(i)} = \mathrm{koef.filter},\; \mathrm{a(0)} = 1)$…....…………..(2)

$$\mathrm{S(z)} \underset{\mathrm{Z}}{\overset{\mathrm{Z}^{-1}}{\Longleftrightarrow}} s(nT) \; = \; s(t)\Big|_{t\,=\,nT} \qquad \ldots(3)$$

The discrete time signal s(nT) is the inverse of the z-transformation of S(z). For the normalized discrete time period T=1, then:

$\mathrm{S(z)} \Leftrightarrow \mathrm{s(n)}.$

With the same relationship as the signal s(n) above,

$\mathrm{E(z)} \Leftrightarrow \mathrm{e(n)}$

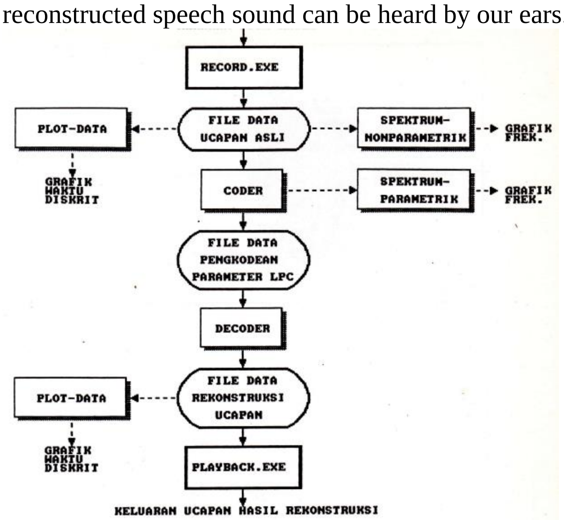
is a model of vocal cord output. If the type of sound vibrates, e(n) will be in the form of a pulse function. If the type is non-vibrating, e(n) will be a random function. The e(n) signal after amplified G is passed into the vibration converter cavity model in the form of an all-pole 1/A(z) filter to produce an s(n) signal. From equation (1), the analysis model can be obtained in the form of an equation:

$\mathrm{E(z)} \; = \; 1/\mathrm{G.S(z).A(z)}$ (analysis model)….(4)

where S(z) is the z-transformation of the voice signal s(n) as the inverse filter input (filter all-zero) A(z), the output E(z) will carry information on the parameters of the input voice signal.

RESEARCH METHODOLOGY

The process flow diagram for the LPC vocoder sistem simulation is shown in Figure 3. At first, the source voice signal is recorded, then a digital data file is created with the name RECORD.EXE and called the original speech data file. This original speech data can be seen in the time domain named PLOT-DATA, and in the frequency domain named NONPARAMETRIC SPECTRUM. Furthermore, the speech data series is processed with the CODER program to produce parameters that have been encoded into digital form, namely filter coefficients, pitch period, V/UV, and Gain. The coding process is carried out on each frame of the input signal, so that a code for the parameters of the filter coefficient, pitch period, V/UV, and Gain is generated for each frame. The DECODER program is used to reconstruct the speech data again based on the parameters stored in the LPC parameter encoding data file, and the reconstruction results are recorded back in a speech reconstruction data file. To get back the analog output form, PLAYBACK.EXE software is used which will convert the digital data in the speech reconstruction data file into analog data. After going through the LPF filter and speaker, the reconstructed speech sound can be heard by our ears.



Source: (Sudaradjat, Pemrosesan Sinyal Suara Dengan Metoda LPC, 1993)

Figure 3. Simulation Process Flowchart.

To test the sound quality, the MOS (Mean Opinion Score) technique is used, which is to calculate the average of the number of speakers, listeners, and other conditions used in the study as shown in table 1.

Table 1. Sound Quality Test Conditions

Objective test method		Mean Opinion Score (MOS)
Score	Quality scale	Impairment
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

Sumber: (Sudaradjat, Pemrosesan Sinyal Suara Dengan Metoda LPC, 1993)

RESULTS AND DISCUSSION

To test the sound quality on a vocoder sistem with a certain bit-rate, it is necessary to plan in advance the design of the coding sistem and the synthetic filter used. In this research, a coding sistem with bit-rate is designed as shown in table 2, with the design of the synthetic filter used is a filter with a lattice structure. The results of the test for the sound of the *ekor-ikan-biru* sentence are shown in table 3, the sound of the *aneka-sate-ikan* sentence in table 4, the sound of the *petugas-sekolah* sentence in table 5, and the sound of the *biru-api* sentence in table 6. In general, the results of the quality test The sound sistem in tables 3, 4, 5 and 6 can be said that the voice signal coding sistem with the parametric encoding method has an intelligible sound quality and the minimum bit-rate limit that can be used is around 2.4 Kbit/second. In contrast to the waveform coding sistem, which has almost the same sound quality but with a higher bit-rate, which is 8 Kbit/sec.

Table 2. Calculation of bit-rate with a frame length of 30 seconds and a sampling frequency of 8 Khz

Parameter	Jml bit tiap frame		
k_1	6	9	12
k_2	4	7	10
k_3 sampai k_4	2x3	2x6	2x9
k_5 sampai k_{10}	6x2	6x5	6x8
Pitch	7	7	7
V/UV	1	1	1
Gain	5	5	5
Sinkronisasi	1	1	1
Jumlah total bit	42	72	102
Bit-rate (KBit/det)	1,4	2,4	3,4

Source: (Sudaradjat, Pemrosesan Sinyal Suara Dengan Metoda LPC, 1993)

Table 3. MOS for sentences *ekor-ikan-biru*

N	SCORE		
	1,4 Kbit/det	**2,4 Kbit/det**	**3,4 Kbit/det**
1	2	2	2
2	2	2	3
3	2	3	3
4	3	3	3
5	2	2	2
6	2	3	3
7	2	2	3
8	3	2	3
9	2	2	2
10	2	3	3
11	2	2	2
12	3	2	3
13	3	3	3
14	3	4	4
15	3	4	4
16	2	4	4
17	2	4	3
18	2	4	3
19	3	4	4
20	3	4	4
21	3	4	4
22	2	3	3
23	3	4	3
24	3	4	3
25	3	2	3
26	2	3	3
27	2	3	3
28	2	3	3
29	2	2	2
30	2	3	3
31	3	3	3

32	2	3	3
$\overline{X}$	2,41	3,00	3,03

Source: (Sudaradjat, Pemrosesan Sinyal Suara Dengan Metoda LPC, 1993)

Table 4. MOS for sentences *aneka-sate-ikan*

N	SCORE		
	1,4 Kbit/det	**2,4 Kbit/det**	**3,4 Kbit/det**
1	2	2	3
2	2	2	3
3	2	2	3
4	2	3	3
5	2	2	2
6	2	3	3
7	2	3	3
8	2	3	3
9	2	2	3
10	2	2	3
11	3	2	2
12	3	2	3
13	3	3	3
14	3	4	3
15	3	4	4
16	2	4	4
17	2	4	3
18	2	4	3
19	2	4	3
20	3	4	3
21	3	3	3
22	3	3	3
23	3	4	3
24	3	4	3
25	3	2	3
26	2	3	3
27	2	3	3
28	2	3	3
29	2	2	3
30	2	3	3
31	3	3	3
32	2	3	3
$\overline{X}$	2,37	2,97	3,00

Source: (Sudaradjat, Pemrosesan Sinyal Suara Dengan Metoda LPC, 1993)

Table 5. MOS for sentences *petugas-sekolah*

N	SCORE		
	1,4 Kbit/det	**2,4 Kbit/det**	**3,4 Kbit/det**
1	2	2	3
2	2	2	3
3	2	2	3
4	2	2	3
5	2	2	2
6	2	3	3
7	2	3	3
8	3	3	3
9	2	2	2
10	2	3	3
11	2	2	2
12	3	2	3
13	3	3	3
14	3	4	3
15	3	4	4
16	2	4	4
17	2	4	3
18	2	3	3
19	3	3	3
20	3	4	3
21	3	4	4
22	2	3	3
23	3	4	3
24	3	4	3
25	3	3	3
26	2	3	3
27	2	3	3
28	2	3	4
29	2	3	2
30	2	3	3
31	3	3	3
32	2	3	3
$\overline{X}$	2,37	3,00	3,00

Source: (Sudaradjat, Pemrosesan Sinyal Suara Dengan Metoda LPC, 1993)

Table 6. MOS for sentences *biru-api*

N	SCORE		
	1,4 Kbit/det	**2,4 Kbit/det**	**3,4 Kbit/det**
1	2	2	3
2	2	3	3
3	3	3	3
4	3	3	3
5	2	2	2
6	2	3	3
7	2	2	3
8	3	2	3
9	2	2	2
10	2	3	3
11	2	2	2
12	2	2	3
13	3	3	3
14	3	4	4
15	3	4	4
16	3	4	4
17	2	4	3
18	2	4	3
19	3	4	3
20	2	4	4
21	3	4	4
22	2	3	3
23	3	4	3
24	3	4	3
25	3	2	3

26	3	3	3
27	3	3	3
28	2	3	4
29	2	2	2
30	2	3	3
31	2	3	3
32	2	3	3
$\overline{X}$	2,44	3,03	3,06

Source: (Sudaradjat, Pemrosesan Sinyal Suara Dengan Metoda LPC, 1993)

CONCLUSION

Parametric encoding method on human voice signal can produce lower bit-rate (2.4 Kbit/sec) than waveform encoding method (8 Kbit/sec). With digital technology, it is possible to create an imitation model of a human speech sound-producing system. In order to obtain a lower bite-rate without compromising sound quality, it can be further investigated by using mock models that allow for use such as the selection of adaptive frame length, the type of digital filter used, or a more precise pitch algorithm.

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