Model of Adaptive Language Synthesis Based On Cosine Conversion Furies with the Use of Continuous Fractions

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Abstract. The proposed article describes an adaptive model for the synthesis of voice signals in a digital signal processor. The use of continuous fractions in a digital signal processor is suggested. The realization of continuous fractions with the help of multicellular structures is given. This procedure is used to implement the model of the human vocal tract.

Keywords. Language Synthesis, Adaptive Synthesis, digital signal processor, Cosine Conversion Furies, Continuous Fractions, Speech synthesis system

1 Introduction

Modern voice signals recognition systems integrate technologies from such fields of modern science as signal processing, pattern recognition, natural language, and linguistics. Such systems that are widely used in signal processing have created a real boom in digital signal processing (DSP). Previously, the field was dominated by vector-oriented processors and algebraic mathematical apparatus, while the current generation of DSP relies on sophisticated statistical models and uses complex software for practical implementation. Modern voice signals recognition models are able to understand the continuous input language for dictionaries, consisting of hundreds of thousands of words in operating environments. Linear predictive analysis of voice signals is historically the most important in voice analysis technologies. The basis of this is the filter source model, which is an ideal linear filter.

2 Analytical Review of Literary and Other Sources

Linear predictive coding is most commonly used in speech analysis and synthesis, or in transmitting or storing speech signals. For this purpose, ideal cell structures are typically used to model the human vocal tract. For the first time, these structures with reflection coefficients were formulated by Markel, Gray [1], and Makhoul [2]. The model in the state space of a non-ideal cell structure with two and four factors per section for digital signal processors was analyzed in [3]. The general system of voice synthesis given in [4] is presented in Fig. 1.

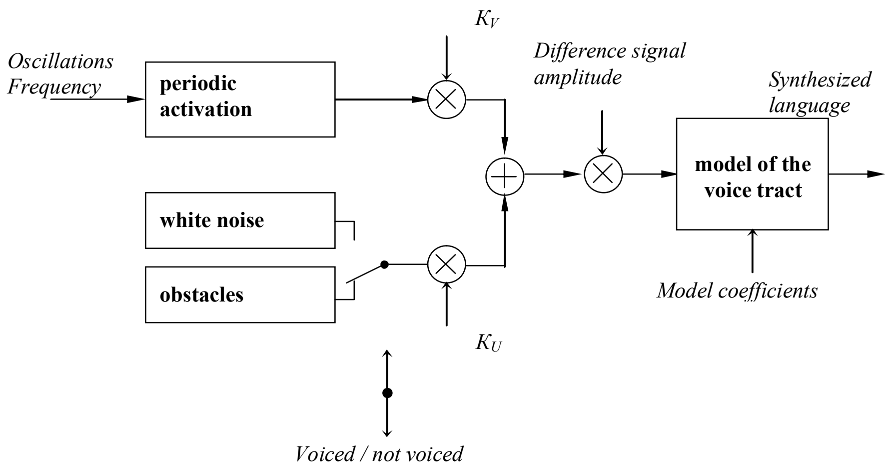

Fig. 1. Speech synthesis system

In the general case, the problem of linear prediction is as follows [9-12]. Let us have a voice signal $s(n)$, and let

$$\tilde{s}(n) = \sum_{k=1}^{p} \alpha_k s(n-k)$$

be predicted magnitude. Inaccuracy of prediction in this case is given as follows:

$$e(n) = s(n) - \tilde{s}(n) = s(n) - \sum_{k=1}^{p} \alpha_k s(n-k).$$

We usually want to minimize the error to find the best, or optimal, values $\{\alpha_k\}$. Determine the short-term average error:

$$E = \sum_n e^2(n) = \sum_n \left\{ s(n) - \sum_{k=1}^{p} \alpha_k s(n-k) \right\}^2 = \sum_n s^2(n) - \sum_n \left\{ 2s(n) \sum_{k=1}^{p} \alpha_k s(n-k) \right\} +$$
$$+ \sum_n \left\{ \sum_{k=1}^{p} \alpha_k s(n-k) \right\}^2 = \sum_n s^2(n) - 2\sum_{k=1}^{p} \alpha_k \sum_n s(n)s(n-k) + \sum_n \left\{ \sum_{k=1}^{p} \alpha_k s(n-k) \right\}^2$$

We can minimize the error α_l for everyone $1 \le l \le p$ by differentiating E and equating the result to zero

$$\frac{\partial E}{\partial \alpha_l} = 0 = -2\sum_n s(n)s(n-l) + 2\sum_n \left\{ \sum_{k=1}^{p} \alpha_k s(n-k) \right\} s(n-l)$$

In the case of the covariance method, we will start by slightly redefining the terms

$$\sum_{n} s(n)s(n-l) = \sum_{k=1}^{p} \alpha_k \left(\sum_{n} s(n-k)s(n-l) \right) \quad \text{or} \quad c(l,0) = \sum_{k=1}^{p} \alpha_k c(k,l)$$

This equation is also known as linear prediction equation (Yule-Volcker equation). $\{\alpha_k\}$ are called linear prediction coefficients, or predictor coefficients. When calculating the equations for all values l, we can write them in a matrix form

$$\overline{c} = \underline{C}\overline{\alpha}$$

where

$$\overline{\alpha} = \begin{bmatrix} \alpha_1 \\ \alpha_2 \\ \cdots \\ \alpha_p \end{bmatrix} \qquad \underline{C} = \begin{bmatrix} c(1,1) & c(1,2) & \ldots & c(1,p) \\ c(2,1) & c(2,2) & \ldots & c(2,p) \\ \ldots & \ldots & \ldots & \ldots \\ c(p,1) & c(p,2) & \ldots & c(p,p) \end{bmatrix} \qquad \overline{c} = \begin{bmatrix} c(1,0) \\ c(2,0) \\ \cdots \\ c(p,0) \end{bmatrix}$$

To solve this equation, you need to find the inverted matrix:

$$\overline{c} = \underline{C}^{-1}\overline{\alpha}$$

This method is called the covariance method. Note that the covariance matrix is symmetric. The fastest way to find the solution to this equation is the Holetsky method (the covariance matrix is divided into lower and upper triangular matrices). Using a slightly different approach to minimize the error, we can find a solution to the linear prediction equation using the autocorrelation method

$$\overline{\alpha} = \underline{R}^{-1}\overline{r}$$

where

$$\overline{\alpha} = \begin{bmatrix} \alpha_1 \\ \alpha_2 \\ \cdots \\ \alpha_p \end{bmatrix} \qquad \underline{R} = \begin{bmatrix} r(0) & r(1) & \ldots & r(p-1) \\ r(1) & r(0) & \ldots & r(p-2) \\ \ldots & \ldots & \ldots & \ldots \\ r(p-1) & r(p-2) & \ldots & r(0) \end{bmatrix} \qquad \overline{r} = \begin{bmatrix} r(1) \\ r(2) \\ \cdots \\ r(p) \end{bmatrix}$$

The matrix of the system is symmetrical and all diagonal elements are equal, which means that the inverted matrix always exists and the solutions of the system are in the left half plane.

Autoregressive modeling using least squares prediction, or linear prediction, forms the basis of a wide range goals of signal processing and communication systems, that include adaptive filtering and control, modeling of speech and coding systems, adaptive channel alignment, parametric spectrum estimation, and identification systems.

To implement linear prediction of data or model goals, it is necessary to determine the values of linear prediction coefficients, as well as the order. Some commonly used practice model selection methods include the Akayke information criterion method, the Schwarz minimum description length method, and the Risenan prediction of least squares principle. In the original form, the first two criteria include a clear balance between the similarity of model input and the notion of fine for model complexity. Intuitively in the information criterion method, the primary purpose is to minimize the number of bits that will be required to describe the data [13-18]. When it is already possible to model the data parametrically and then encode the blocks, use the approach of allocating blocks of similar data, and then the model is fined with the additional number of bits required to encode its parameters [19-24].

However, the voice model based on cosine Fourier transform for language synthesis has better properties and use, less sensitivity to quantization effects and, as a result, produces more natural synthesized language [25-29]. The parameters of this model are the coefficients of cosine Fourier transform. This model is based on the cosine decomposition of the logarithmic short-term voice range, and the synthesis is implemented by approximate inverse Fourier cosine transformations using continuous chain fractions [30-32]. This approach is parametric and is not based on any simplifying assumptions about the voice model, because the poles as well as the zeros of the voice model are justified [33-44].

3 The Voice Model Based on a Cosine Fourier Transform

Suppose that we have the logarithmic range $\ln\left|S\left(e^{j\omega T}\right)\right|$ of the voice data segment $\{s(n)\}$, where T – is the sampling interval, $f_s = \frac{1}{T}$ - the sampling frequency, and ω - the angular frequency. This function can be expressed using the true Fourier cosine conversion coefficients Фур'є $\{c_n\}$

$$\ln\left|S\left(e^{j\omega T}\right)\right| = \sum_{n=-\infty}^{\infty} c_n e^{-jn\omega T} . \quad (1)$$

The complex coefficients of the cosine Fourier transform of a discrete system with minimum phase stability are random and may be related to the following relations

$$\begin{aligned} g_n &= c_n, \quad n = 0, N_F/2, \\ g_n &= 2c_n, \quad 0 < n < N_F/2, \\ g_n &= 0, \quad n < 0 \end{aligned} \quad (2)$$

where N_F - the dimension of the applied FFT. A digital filter whose logarithmic correspondence approximates a function $\ln\left|S\left(e^{j\omega T}\right)\right|$ is determined by the transfer function system

$$\widetilde{S}(z) = e^{c_0} \exp \sum_{n=1}^{N_0-1} 2c_n z^{-n} = e^{c_0} \prod_{n=0}^{N_0-1} \exp\left[2c_n z^{-n}\right] \tag{3}$$

where $0 < N_0 < N_F/2$. The coefficient e^{c_0} is equal to the value of the RMS of the cosine Fourier transform model for the multiple signal. In our experiments on the voice model we used $f_s = 8kHz$, $N_F = 512$, the voice segment length is 25 *ms* with 12 *ms* overlap and $N_0 = 25$.

It follows from (3) that the system of transfer functions $\widetilde{S}(z)$ is the product of transcendental transfer functions

$$H_n(z) = e^{2c_n z^{-n}}, \quad 0 < n \le N_0 - 1 \tag{4}$$

The corresponding impulse feature is given

$$h_n(m) = \begin{cases} \dfrac{(2c_n)^i}{i!}, & m = ni, \quad i = 0,1,2,\dots \\ 0, & m \ne ni \end{cases} \tag{5}$$

This means that the system of transfer functions $\widetilde{S}(z)$ has the following form (Fig. 2)

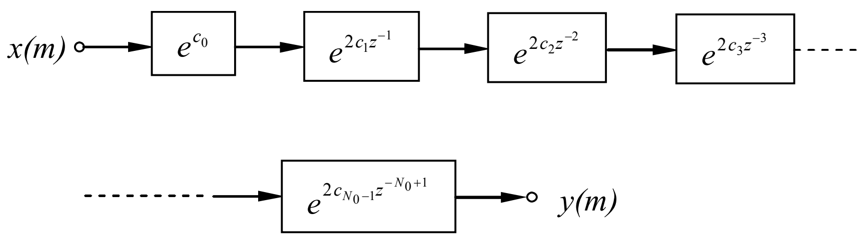

Fig. 2. Voice model of cosine Fourier transform

$$\widetilde{S}(z) = e^{c_0} \prod_{n=0}^{N_0-1} H_n(z) \tag{6}$$

To implement a transfer function $H_n(z)$ using a digital filter, it is necessary to find an approximation $H_n(z)$, that can be practically implemented. One option for approximating an exponential function in (4) is continuous chain fractions [4]. Another possibility of implementing an exponential function approximation is to use a Pade approximation. Then the system of transfer functions in a practical voice model based on a cosine Fourier transform will look like

$$\tilde{\tilde{S}}(z) = e^{c_0} \prod_{n=0}^{N_0 - 1} \tilde{H}_n(z). \qquad (7)$$

4 Decomposition Approximation Using Continuous Chain Fractions

The exponential function expressed by a decomposition into a continuous fraction can be represented as the following decomposition [5], [6], [7]:

$$e^x = \frac{1}{1-}\frac{x}{1+}\frac{x}{2-}\frac{x}{3+}\ldots\frac{x}{+2-}\frac{x}{2s-1},\ldots \qquad (8)$$

where the parameter is $x = 2c_n z^{-n}$. The accuracy of the approximation of the voice model depends not only on the number of cosine Fourier transform coefficients in (3), but also on the number of members of a continuous fraction in (8), that is, on the length of a continuous fraction to be determined with s. A finite chain fraction for a function e^x can also be expressed by a set of real functions that approximate an exponential function with increasing accuracy

$$e^x \cong \frac{1}{1}, \frac{1}{1-x}, \frac{2+x}{2-x}, \frac{6+2x}{6-4x+x^2}, \frac{12+6x+x^2}{12-6x+x^2}, \cdots \qquad (9)$$

These functions are known as Pade approximations of an exponential function. It is recommended to use an odd number of elements of a continuous fraction in (8). This leads to an approximation of an exponential function by a rational function with equal degrees of polynomials in the numerator and denominator in (9). These are the approximations chosen

$$\begin{aligned}
&\tilde{H}_1(z) = \frac{2+x}{2-x}, \quad \tilde{H}_2(z) = \frac{12+6x+x^2}{12-6x+x^2}, \\
&\tilde{H}_3(z) = \frac{120+60x+12x^2+x^3}{120-60x+12x^2-x^3}, \\
&\tilde{H}_4(z) = \frac{1680+840x+180x^2+20x^3+x^4}{1680-840x+180x^2-20x^3+x^4}
\end{aligned} \qquad (10)$$

where z is the variable z-transformation and $x = 2c_n z^{-n}$. In the general case, to achieve a better approximation, we can use decompositions of rational functions by taking more suitable fractions.

$$v_{2s+1}(m) = -\frac{c_n}{2s-1} v_{2s}(m-n),$$
$$v_{2s}(m) = -\frac{c_n}{2s-1} v_{2s-1}(m-n) + v_{2s+1}(m),$$
$$\vdots$$
$$v_5(m) = -\frac{c_n}{3} v_4(m-n) + v_6(m),$$
$$v_4(m) = -\frac{c_n}{3} v_3(m-n) + v_5(m),$$
$$\begin{aligned} v_3(m) &= -c_n v_2(m-n) + v_4(m), \\ v_2(m) &= 2c_n v_1(m-n) + v_3(m), \\ v_1(m) &= x(m) + v_2(m), \\ y(m) &= v_1(m). \end{aligned} \tag{11}$$

5 Structure of adaptive synthesis

As noted above, the approximation error for e^x is determined by the number of elements of a continuous fraction to decompose an exponential function into a continuous fraction. This error further depends on the magnitudes of the modules of the true Fourier cosine transform coefficients $|c_n|$. On the basis of a statistical analysis of the Fourier cosine coefficients for the description of the voice model of a male loudspeaker, the following estimation was made in relation to the stability of the system and the well-defined safety limit for transfer functions $H_n(z)$ in equation (5). From the above it follows that the functions $H_n(z)$ can be approximated as follows:

$$\begin{aligned} n &= 1,2,3 && \Rightarrow \tilde{H}_3(z) \\ n &= 4,5 && \Rightarrow \tilde{H}_2(z) \\ n &= 6,7,\ldots,25 && \Rightarrow \tilde{H}_1(z) \end{aligned}$$

It is more effective in relation to the total error of approximation and in relation to saving the number of arithmetic operations required for the practical implementation of voice modeling to use the adaptive structure of a continuous fraction. The number of corresponding cells (Fig. 3) can be selected according to the magnitudes of the cosine Fourier transform coefficients. The following adaptive empirical rule can be used:

$$\textit{for } |c_n| \le 0.3 \textit{ two cells - match } \tilde{H}_1(z),$$

$$\textit{for } |c_n| \le 0.5 \textit{ four cells - match } \tilde{H}_2(z),$$

$$\textit{for } |c_n| \le 1 \quad \textit{six cells - match } \widetilde{H}_3(z),$$

$$\textit{for } |c_n| > 1 \quad \textit{eight cells - match } \widetilde{H}_4(z).$$

For example, the voice model of the stationary part (24 *ms)* of the "e" vowel sound is used

$c_0 = -0.491$ - logarithm of the value of the difference signal

$c_1 = 0.700$ $\quad$ $c_2 = 0.354$ $\quad$ $c_3 = -0.026$ $\quad$ $c_4 = 0.205$

$c_5 = 0.159$ $\quad$ $c_6 = -0.027$ $\quad$ $c_7 = -0.310$ $\quad$ $|c_{8\ldots25}| < 0.3$

Using the empirical rule indicated, the voice model of the cosine Fourier transform is presented in Fig. 3 can be built:

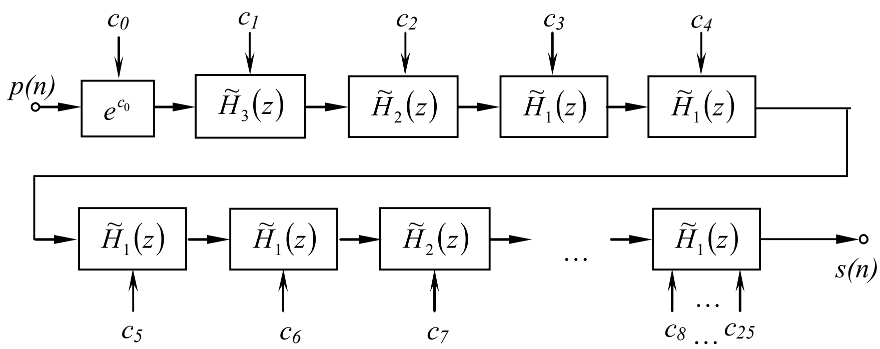

Fig. 3. Voice model of cosine Fourier transform of the stationary part of the vowel "e": $p(n)$ is activated signal; $s(n)$ is synthesized voice signal

In the practical implementation of transcendental transfer functions $H_n(z)$ the following numerical results were obtained:

Table 1. The value of the transcendental transfer function $H_1(z) = e^{z^{-1}}$

n	$H_n(z)$ - exact values	$H_n(z)$ - approximate values
1	1.00000000000000	0.99993896484375
2	1.00000000000000	0.99993896484375
3	0.50000000000000	0.49999648242188
4	0.16666666666667	0.16666549414062
5	0.08333333333333	0.06092749023438
6	0.00138888888889	0.00003051757812

n	$H_n(z)$ - exact values	$H_n(z)$ - approximate values
7	0.00019841269841	0.00030517578125
8	0.00002480158730	-0.0012207031250
9	0.00000271636432	-0.00003051757812

We will also present our numerical results in the following diagram (Fig. 4).

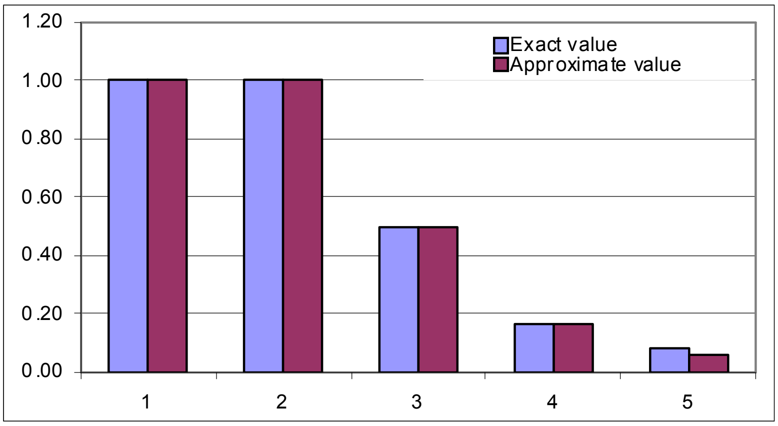

Fig. 4. The value of the transcendental transfer function $H_1(z) = e^{z^{-1}}$

6 Conclusions

Voice modeling based on cosine Fourier transform is in fact related to spectral synthesis of voice signals, and is not based on any simplifying a priori considerations about the language reproduction system. It also contains information about the range of the activated voice path.

The voice modeling procedure based on Fourier cosine transforms requires more arithmetic operations than approaches based on linear predictive coding, but the structure of the digital filter can be optimized.

Continuous fractions offer an interesting tool not only in language synthesis. A high-order approximation of algebraic transcendental functions can be used in biological and industrial modeling systems. The direct implementation of continuous fractions further enables the implementation of multi-chamber structures.

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