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스펙트럼차에 기초한 LSP 추출방법

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An LSP Extraction Method Based on the Spectral Difference

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Abstract

In this paper, we propose a new method to extract the line spectrum pair (LSP) frequencies. When speech signal is analyzed by the autocorrelation method, the spectral difference in the logarithmic spectra of the model at steps p and $p-1$ oscillates. There are $p-1$ frequency points where the values of the spectral difference take on either maximum or minimum between 0 and half sampling frequency. We show that these frequencies are exactly the LSP frequencies of order $p-1$, which can be found by searching the frequencies where the spectral difference reaches either maxima or minima. Also, the LSP frequencies of order p can be obtained from this spectral difference. In this case, we derive the expression governing the pth order LSP frequencies. The efficient search for finding the LSP frequencies of order p can be done by proving the property that the LSP frequencies of order p and $p-1$ are interlaced with each other.

1 Introduction

Line spectrum pair (LSP) representation of a speech signal has been introduced by Itakura [1] as an alternation of linear predictive coding (LPC) and widely used in the speech processing areas including speech coding, synthesis, and recognition [2]. The LSP representation has better quantization property than other LPC representations, such as the log area ratio (LAR) and the partial correlation (PARCOR) coefficients. Since LSP frequencies are frequency domain parameters, the error on an LSP parameter gives rise to the spectral distortion in the neighborhood of a specific frequency at which the LSP parameter is located [3]. It has been reported experimentally [4] that the quantization bits for LSP can be reduced to 70 ~ 80% of those for PARCOR to achieve the same spectral distortion. Moreover, the LSP parameters have well-behaved dynamic range and good interpolation characteristics. The stability of LSP synthesis filter is also preserved after quantizing LSP parameters and can be checked simply.

There have been proposed several methods to extract LSP frequencies. Since the roots of the symmetric and anti-symmetric polynomials from an LPC analysis polynomial are on the unit circle and the frequency components corresponding to the roots are LSP frequencies, the procedure developed by Itakura [1] finds the zero-crossing points in frequency domain by applying Fourier transform to each polynomial. In anther way, the procedure employs the Newton-Raphson method to find roots of symmetric and anti-symmetric polynomials and converts them to LSP frequencies [5]. Soong and Juang [6], instead of applying Fourier transform over all frequencies, adopted discrete cosine transform to evaluate the roots on a fine grid on frequency domain. Kabal and Ramachandran [7] proposed the algorithm to compute LSP frequencies by finding roots iteratively for a series representation in Chebyshev polynomials. Recently, the algorithm which converts LSP frequencies from reflection coefficients and vice versa, was developed by Chan and Law [8].

In this paper, we propose a new extraction method of LSP frequencies. To begin with, we show explicitly that the spectral difference in the logarithmic spectra of the LPC models at step p and p-1 simply represents the LSP frequencies. The spectral difference oscillates in frequency axis and the number of maxima and minima is exactly p-1 between 0 and π (half sampling frequency) [9]. The LSP frequencies of order p can be easily obtained by searching the maxima and minima of the spectral difference. Also the spectral difference can be represented as a function of the LSP frequencies of order p. An efficient search of pth order LSP frequencies can be done by proving the two useful properties related to the LSP frequencies of order p and p-1. That is, one is that any LSP frequency of order $p-1$ is always different from an LSP frequency of order p. The other is that $(p-1)$th order LSP frequencies exist alternately from pth order LSP frequencies.

2 LSP Analysis

A short-time frame of speech signal is produced by pth order all-pole filter $H_p(z) = \frac{\alpha_p}{A_p(z)}$ in LPC analysis, where α_p is the filter gain corresponding to the rms of error residuals. The pth order linear prediction filter (or inverse filter) $A_p(z)$ is described as

$$A_p(z) = 1 + a_1 z^{-1} + \cdots + a_p z^{-p} \qquad (1)$$

where $\{a_1, \cdots, a_p\}$ are the linear predictive coefficients of order p. Instead of a direct form implementation of the inverse filter, it is possible to implement the filter in lattice form using the pth reflection coefficient, k_p, and the $(p$-1)th analysis filter $A_{p-1}(z)$ as follows.

$$A_p(z) = A_{p-1}(z) + k_p B_{p-1}(z), \quad (2)$$

$$zB_p(z) = k_p A_{p-1}(z) + B_{p-1}(z), \quad (3)$$

with

$$A_0(z) = zB_0(z) = 1, \quad (4)$$

where $B_p(z)$ is a pth order backward linear prediction filter which estimates the current sample based on the future samples, and has the relationship: $B_p(z) = z^{-(p+1)}A_p(z^{-1})$.

To obtain the LSP representation of order p, the lattice filter of order $(p+1)$ can be extended from the pth order filter by setting the $(p+1)$th reflection coefficient k_{p+1} to the two extreme values of ± 1. It means that the vocal tract at the sound source of the lossless tube model is completely closed or completely open and the power loss in the vocal tract is zero. Thus a symmetric polynomial $P_{p+1}(z)$ and an anti-symmetric polynomial $Q_{p+1}(z)$ are obtained as follows.

$$P_{p+1}(z) = A_p(z) + B_p(z), \quad (5)$$

$$Q_{p+1}(z) = A_p(z) - B_p(z). \quad (6)$$

Real zeros of the polynomials $P_{p+1}(z)$ and $Q_{p+1}(z)$ are at $z = -1$ and $z = +1$, respectively, and all the other zeros are complex. These complex zeros determine the LSP frequencies of order p. $P_{p+1}(z)$ and $Q_{p+1}(z)$ have the following important properties:

(1) All zeros of $P_{p+1}(z)$ and $Q_{p+1}(z)$ lie on the unit circle.

(2) Zeros of $P_{p+1}(z)$ and $Q_{p+1}(z)$ are interlaced with each other.

(3) Minimum phase property of $A_p(z)$ is easily preserved after quantization of the zeros of $P_{p+1}(z)$ and $Q_{p+1}(z)$.

Since the zeros of $P_{p+1}(z)$ and $Q_{p+1}(z)$ are on the unit circle, they can be expressed $\{e^{j\theta_i}\}$ $(i = 1, 2, \cdots, p)$, and these $\{\theta_i\}$ are called the LSP frequencies of order p.

To calculate $\{\theta_i\}$ from $\{a_i\}$, $A_p(z)$ is firstly converted to $P_{p+1}(z)$ and $Q_{p+1}(z)$, and then a search procedure is required in the frequency axis after applying the fast Fourier transform (FFT) or discrete cosine transform (DCT) to $P_{p+1}(z)$ and $Q_{p+1}(z)$. Another method projects the polynomials of (5) and (6) on the real axis with $x = \cos\theta = \frac{z+z^{-1}}{2}$, and then the roots of the projected polynomials are approximately found and converted into the LSP frequencies with $\theta = \cos^{-1} x$.

The synthesis filter $H_p(z)$ also can be obtained from the polynomials of $P_{p+1}(z)$ and $Q_{p+1}(z)$ as follows:

$$H_p(z) = \frac{\alpha_p}{A_p(z)} = \frac{\alpha_p}{1 + \frac{(P_{p+1}(z)-1)+(Q_{p+1}(z)-1)}{2}}. \quad (7)$$

By using the LSP frequencies, we can construct the synthesis filter without LPC coefficients.

3 Extraction of LSP Frequencies Using Spectral Difference

3.1 Derivation of Extraction Method

When a speech signal is analyzed by the autocorrelation method of order p and p-1, respectively, spectral difference (SD) of order p is defined as the difference between the spectra of the two all-pole filter. SD of order p becomes

$$E_p(\theta) = \ln\Big[\frac{\alpha_p^2}{|A_p(e^{j\theta})|^2}\Big] - \ln\Big[\frac{\alpha_{p-1}^2}{|A_{p-1}(e^{j\theta})|^2}\Big] \quad (8)$$

$$= \ln\Big[\frac{\alpha_p^2}{\alpha_{p-1}^2}\Big] - 2\ln\Big[\Big|\frac{A_p(e^{j\theta})}{A_{p-1}(e^{j\theta})}\Big|\Big] \quad (9)$$

$$= \ln\Big[\frac{\alpha_p^2}{\alpha_{p-1}^2}\Big] + 2\ln\Big[\Big|\frac{A_{p-1}(e^{j\theta})}{A_p(e^{j\theta})}\Big|\Big] \quad (10)$$

where α_p^2 and α_{p-1}^2 are the minimum error powers of order p and p-1, respectively. Dividing (2) by $A_{p-1}(z)$ results in

$$\frac{A_p(z)}{A_{p-1}(z)} = 1 + k_p \frac{B_{p-1}(z)}{A_{p-1}(z)}. \quad (11)$$

If we define

$$R_p(e^{j\theta}) = \frac{B_{p-1}(e^{j\theta})}{A_{p-1}(e^{j\theta})}, \quad (12)$$

and substitute (12) into (9) with the well known equation of $\alpha_p^2 = (1-k_p^2)\alpha_{p-1}^2$, $E_p(\theta)$ can be written as

$$E_p(\theta) = \ln(1-k_p^2) - 2\ln[|1 + k_p R_p(e^{j\theta})|]. \quad (13)$$

Comparing $R_p(e^{j\theta})$ with the polynomials of $P_p(e^{j\theta})$ and $Q_p(e^{j\theta})$, the frequencies θ's of $R_p(e^{j\theta}) = -1$ and $R_p(e^{j\theta}) = +1$ are equivalent to the zeros of $P_p(e^{j\theta})$ and $Q_p(e^{j\theta})$, respectively. Thus, the roots of $R_p(e^{j\theta}) = \pm 1$ are the LSP frequencies of order p-1. Since $R_p(e^{j\theta})$ is a unit gain all-pass filter, we can have

$$R_p(e^{j\theta}) = e^{-j\Psi_p(\theta)}, \quad (14)$$

where $\Psi_p(\theta)$ is the negative phase function of $R_p(\theta)$.

If we let $z_i = r_i e^{j\theta_i}$ $(i = 1, \cdots, p-1)$ be a root of $A_{p-1}(z)$, $\Psi_p(\theta)$ can be expressed as

$$\Psi_p(\theta) = p\theta + 2\sum_{i=1}^{p-1} \tan^{-1} \frac{r_i \sin(\theta - \theta_i)}{1 - r_i\cos(\theta - \theta_i)}. \quad (15)$$

And the group delay of $R_p(e^{j\theta})$ defined as the slope of $\Psi_p(\theta)$ is

$$\frac{\partial \Psi_p(\theta)}{\partial \theta} = 1 + \sum_{i=1}^{p-1} \tan^{-1} \frac{1 - r_i^2}{1 - 2r_i\cos(\theta - \theta_i) + r_i^2}, \quad (16)$$

and if the stability of the filter $A_{p-1}(z)$ is assumed, all $r_i < 1$ means $\frac{\partial\Psi_p(\theta)}{\partial\theta} > 1$. Moreover $\Psi_p(0) = 0$ and $\Psi_p(\pi) = p\pi$. Therefore, $\Psi_p(\theta)$ is a monotonic increasing function and the number of points which make $R_p(e^{j\theta}) = \pm 1$ for $0 < \theta < \pi$, is $p-1$. These are the LSP frequencies of order $p-1$.

The values of $E_p(\theta)$ at these points $\{\theta_{p-1,1}, \cdots, \theta_{p-1,p-1}\}$ are given by

$$E_p(\theta_{p-1,i}) = (-1)^i \ln\frac{1-k_p}{1+k_p}, \quad \text{for } 1 \le i \le p-1. \quad (17)$$

Also, $E_p(\theta)$ is bounded by $|\ln\frac{1-k_p}{1+k_p}|$ and has the equiripple property [9].

To extract the LSP frequencies of order p-1, we firstly analyze speech signal by the autocorrelation method and calculate SD of order p by applying FFT. The minima and maxima of $E_p(\theta)$ given by (17) are searched in the frequency axis.

Next, we show that the LSP frequencies of order p can also be obtained from SD of order p. Combining (2) and (3) gives

$$A_{p-1}(z) = \frac{A_p(z) - zk_pB_p(z)}{1 - k_p^2}. \quad (18)$$

By substituting (18) into (10), we can rewrite $E_p(\theta)$ as

$$E_p(\theta) = -\ln(1-k_p^2) + 2\ln[|1 - k_p e^{j\theta} R_{p+1}(e^{j\theta})|]. \quad (19)$$

This equation is similar to (13) at which $R_p(e^{j\theta})$ instead of $R_{p+1}(e^{j\theta})$ appears. From the similar interpretation that $R_p(e^{j\theta}) = \pm 1$ is related to the LSP frequencies of order

p-1, the LSP frequencies $\{\theta_{p,1},\cdots,\theta_{p,p}\}$ of order p are the roots of $R_{p+1}(e^{j\theta}) = \pm 1$ for $0 < \theta < \pi$. Hence, $E_p(\theta)$ at the LSP frequencies of order p is expressed as

$$E_p(\theta_{p,i}) = \ln \frac{1-(-1)^i 2k_p \cos\theta_{p,i} + k_p^2}{1-k_p^2}, \quad \text{for } 1 \le i \le p. \tag{20}$$

Another search finding the frequencies given by (20) can bring forth the LSP frequencies of order p. To efficiently find the LSP frequencies of order p, the following properties are proved.

3.2 Ordering Properties of LSP Frequencies

The first property we will prove is that any LSP frequency of order p-1 is always different from an LSP frequency of order p.

Property 1 *A set of LSP frequencies $\{\theta_{p,1},\cdots,\theta_{p,p}\}$ of order p does not include any LSP frequencies $\{\theta_{p-1,1},\cdots,\theta_{p-1,p-1}\}$ of order $p-1$.*

This property is expressed as

$$\theta_{p-1,i} \neq \theta_{p,j} \quad \text{for } 1 \le i \le p-1, 1 \le j \le p. \tag{21}$$

Since $0 < \theta_{p,i} < \pi$ for $1 \le i \le p$, $\cos\theta_{p,i} \neq +1$. And thus, the value of log function of (20) can not be equal to $(1-(-1)^i k_p)^2$. Therefore, we can obtain

$$E_p(\theta_{p,i}) \neq (-1)^i \ln \frac{1-k_p}{1+k_p}, \quad \text{for } 1 \le i \le p. \tag{22}$$

The right side of (22) is identical to that of (17). For any $\theta_{p,i}$, the value of SD at this frequency is different from that of any LSP frequency of order $p-1$ □.

From this result, we can say inductively that an LSP frequency obtained from the analysis of any order is different from LSP frequencies extracted from the analysis of different orders. This property is considered as an intrinsic one appearing only in the set of LSP frequencies among many alternations of LSP representation.

Property 2 *As the similar property that zeros of $P_{p+1}(z)$ and $Q_{p+1}(z)$ are interlaced with each other, the LSP frequencies of order p and $p-1$ are also interlaced with each other.*

This gives the relationship:

$$0 < \theta_{p,1} < \theta_{p-1,1} < \theta_{p,2} \cdots < \theta_{p-1,p-1} < \theta_{p,p} < \pi. \tag{23}$$

(19) can be rewritten in the phase function $\Psi_{p+1}(\theta)$ of $R_{p+1}(\theta)$ as

$$E_p(\theta) = -\ln(1-k_p^2) + \ln[1-2k_p\cos(\theta-\Psi_{p+1}(\theta))+k_p^2]. \tag{24}$$

After substituting (15) into (24), the partial derivative of $E_p(\theta)$ for θ is given by

$$\frac{\partial E_p(\theta)}{\partial\theta} = \frac{2k_p \sin(\theta-\Psi_{p+1}(\theta))(1-\frac{\partial\Psi_{p+1}(\theta)}{\partial\theta})}{1-2k_p\cos(\theta-\Psi_{p+1}(\theta))+k_p^2}. \tag{25}$$

Since the denominator of (25) is always positive for $0 < \theta < \pi$ and $\frac{\partial\Psi_{p+1}(\theta)}{\partial\theta} > 1$, the sign change of $\frac{\partial E_p(\theta)}{\partial\theta}$ at $\theta_{p,i}$ is equal to the sign change of $(-1)^{i+1}k_p$.

The second partial derivative of $E_p(\theta)$ at $\theta = \theta_{p-1,j}$ is

$$\frac{\partial^2 E_p(\theta)}{\partial\theta^2}\Big|_{\theta=\theta_{p-1,j}} = \frac{2k_p(-1)^j(1-\frac{\partial\Psi_{p+1}(\theta)}{\partial\theta}|_{\theta=\theta_{p-1,j}})^2}{1-2(-1)^j k_p + k_p^2}. \tag{26}$$

Similarly, the sign change of the second partial derivative of $E_p(\theta)$ at $\theta_{p-1,j}$ is the same to that of $(-1)^j k_p$. The results can be summarized as

$$sgn(\frac{\partial E_p(\theta)}{\partial\theta}|_{\theta=\theta_{p,j}}) = sgn((-1)^{j+1}k_p), \quad \text{for } 1 \le j \le p, \tag{27}$$

$$sgn(\frac{\partial^2 E_p(\theta)}{\partial\theta^2}|_{\theta=\theta_{p-1,j}}) = sgn((-1)^j k_p), \quad \text{for } 1 \le j \le p-1, \tag{28}$$

$$\frac{\partial E_p(\theta)}{\partial\theta}|_{\theta=\theta_{p-1,j}} = 0, \text{for } 1 \le j \le p-1, \tag{29}$$

and in addition,

$$0 < \theta_{p-1,1} < \theta_{p-1,2} < \cdots < \theta_{p-1,p-1} < \pi, \tag{30}$$

$$0 < \theta_{p,1} < \theta_{p,2} < \cdots < \theta_{p,p} < \pi, \tag{31}$$

where $sgn(x)$ means positive if $x > 0$, negative if $x < 0$, and zero otherwise. If we assume $k_p < 0$ and p is even, the region where $\theta_{p,i}$ exists should be

$$\begin{aligned} 0 < \quad & \theta_{p,l} < \theta_{p-1,1}, \quad \theta_{p-1,2} < \theta_{p,l} < \theta_{p,3}b, \\ & \cdots, \\ \theta_{p-1,p-2} < \quad & \theta_{p,l} < \theta_{p-1,p-1}, \quad \text{for } l = \text{odd}, \end{aligned} \tag{32}$$

and

$$\begin{aligned} \theta_{p-1,1} < \quad & \theta_{p,l} < \theta_{p-1,2}, \quad \theta_{p-1,3} < \theta_{p,l} < \theta_{p,4}, \\ & \cdots, \\ \theta_{p-1,p-1} < \quad & \theta_{p,l} < \pi, \quad \text{for } l = \text{even}. \end{aligned} \tag{33}$$

To satisfy both (32) (33) and (31) simultaneously, the property of (23) must be held. The case of $k_p < 0$ can be proved similarly □.

In summary, we first apply the LPC autocorrelation method to the signal and SD is calculated from the model spectra by applying FFT to the linear predictive coefficients of order p and p-1. Next, maxima and minima of SD are marked. The LSP frequencies of order p are successively searched as frequencies satisfying (20) by the property 2. The i-th LSP frequency of order p is found by searching the interval between (i-1)-th and i-th LSP frequencies of order p-1, where 0 and half sampling frequency are assumed as the frequencies of order p-1 at i = 0 and $i = p$, respectively.

4 Experiments on the Extraction of LSP Frequencies

In order to verify the proposed extraction method derived so far and show its usefulness, we firstly conduct a numerical experiment using synthetic signal. A synthetic signal is generated from (7). The zero-mean white Gaussian noise is used as an excitation source. The LSP frequencies of order 8 are assigned as $\frac{5000i}{9}$Hz (i = 1, $\cdots$, 8), which means that the synthetic signal is noisy. The data record length for this simulation is 300 for each LSP extraction that is corresponding to 30ms interval with a sampling rate of 10kHz. 300 realizations of LSP extraction are done for each FFT point. For each LSP frequency from 1 to 8, the deviation from the theoretical LSP frequency for each iteration is computed and averaged over 300 realizations. Fig. 1(a) shows a part of synthetic time samples. The averaged square deviation is plotted in Fig. 1(b) against FFT point. FFT point is varied from 2^5 to

2^{12}. The dotted line indicates the value obtained from Newton-Raphson method in [5] with the desired number of accurate significant digit of 6. Since the frequency spacing is less than $\frac{5000}{8}$Hz, FFT point is greater than or equal to 32 points. All the analyses are failed to extracting LSP frequencies when 16-point FFT is applied to the extraction method.

Secondly, we synthesize the voiced speech whose LSP frequencies are obtained from the voiced frame of Fig. 3(a). The analysis order is 14 and the LSP frequencies are the same shown in Fig. 3(e). The pitch period is set to 8ms (80 sample points). The synthesized speech is also generated as the above procedure with the excitation of a pulse train and displayed as Fig. 2(a). The smallest difference of LSP frequencies is 87.89Hz in this experiment. To extract LSP frequencies correctly, FFT point should be greater than 64. The failure occurs when FFT point is less than or equal to 64. Also the reasonable accuracy is guaranteed when 1024-point FFT is applied to the proposed method as shown in Fig. 2(b).

Next, a real speech is used in this experiment. A speech signal spoken by a male is low-pass filtered with a cut-off frequency of 4.7kHz and sampled at 10kHz. A voiced frame of 30ms is analyzed by the autocorrelation method of order 14. The LPC coefficients corresponding to the analysis order of 14 and 13 are obtained from the Levison-Durbin recursion simultaneously during the analysis of order 14. The speech signal used in this experiment is shown in Fig. 3(a). The spectral difference of order 14 and 13 obtained from the LPC coefficients by taking 1024 point FFTs are shown in Fig.3(b) and 3(c), respectively. SD of order 13, which is the difference between Fig. 3(b) and 3(c), is displayed in Fig. 3(d). Notice that the number of maxima and minima of SD is 13 and the frequencies at these points become the LSP frequencies of order 13. Fig.3(e) shows the LSP frequencies of both the order 13 and 14. The upper part and the lower part of the figure represents the the LSP frequencies of order 14 and the LSP frequencies of order 13, respectively. If we observe each set of LSP frequencies, we find that the LSP frequencies in one set are interlaced with the other set as is explained in section 3.2.

5 Conclusions

We proposed a new method to extract LSP frequencies. This method is based on the spectral difference between the log spectra of each linear predictive model of order p and $p-1$, and can give the LSP frequencies of two orders simultaneously. Also, we proved two interesting properties related to LSP frequencies. One is that the set of LSP frequencies of order p does not include any $(p-1)$th order LSP frequency. The other is that the LSP frequencies of order p and $p-1$ are interlaced with each other. From three experiment including synthetic noise, synthetic voiced sound, and real voiced sound, we confirmed the derivation of the proposed extraction method.

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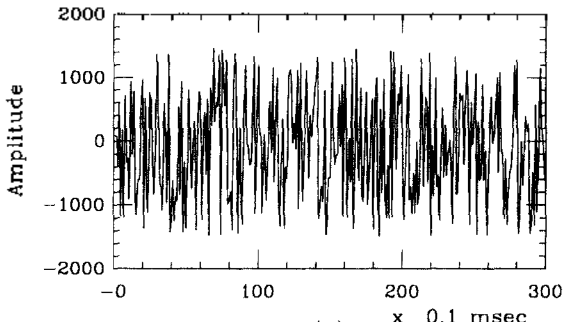
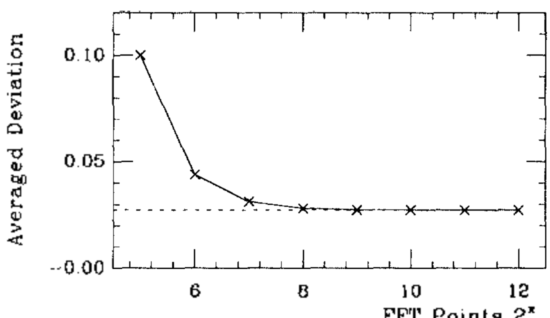
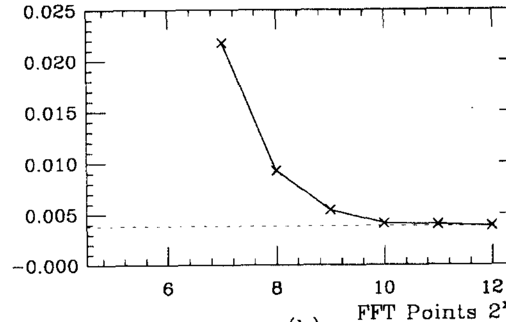


(a)



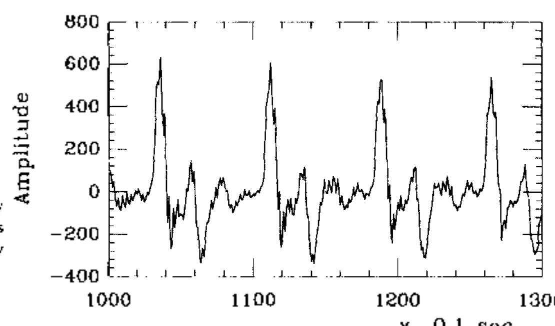
(b)

Figure 1: Performance of the proposed extraction method for noisy signal. (a) Time samples. (b) Average deviation against FFT points where the dotted line indicates the averaged deviation obtained by Newton-Raphson method.



(b)

Figure 2: Performance of the proposed extraction method for a synthetic voiced speech signal. (a) Time samples. (b) Average deviation against FFT points where the dotted line indicates the averaged deviation obtained by Newton-Raphson method.



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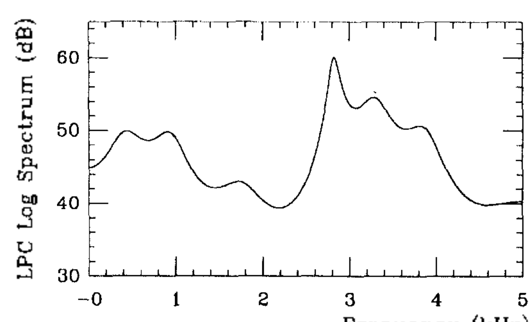

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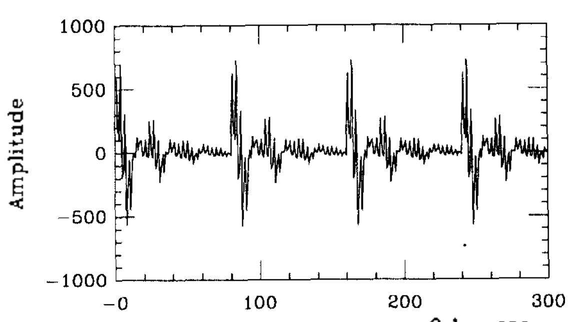

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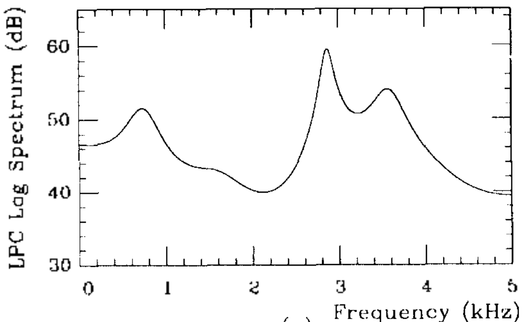

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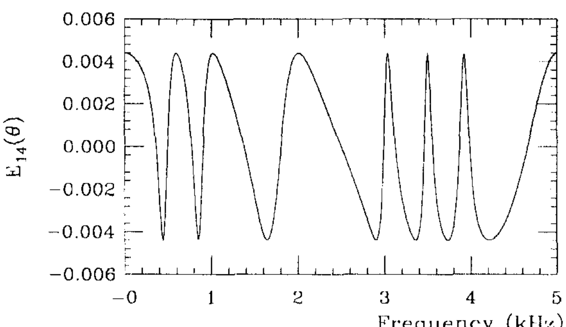

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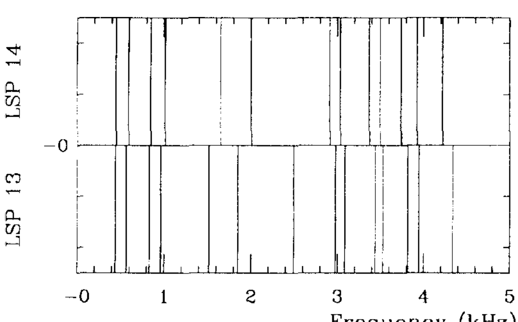

(c)

Figure 3: An experimental results for a real speech signal. (a) A speech segment /o/ pronounced by male speaker. (b) LPC logarithmic speech spectrum of the analysis order of 14. (c) LPC logarithmic speech spectrum of the analysis order of 13. (d) Spectral difference function of order 14. (e) LSP frequencies of order 14 (upper) and 13 (lower).