MODIFIED CEPSTRAL ANALYSIS FOR ACCURATE ESTIMATION OF ECHO PARAMETERS IN TELECOMMUNICATION NETWORKS

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ABSTRACT

A modified cepstral analysis for accurate estimation of the echo delay and the echo loss in a telecommunication system is presented. It is based on the optimization of a parametric transformation of the observed signal energy spectrum. Simulation results that show the effectiveness and the accuracy of the proposed method are reported and discussed.

1. INTRODUCTION

The presence of echo signals in telecommunication systems has always played an important role in the degree of quality provided by telephone-type networks when long distance links or satellite links are involved. In recent years, the adoption of coding techniques that introduce a processing delay in the transmitted signal has made echo a relevant effect also for the case of short-distance networks. In this context, an estimate of echo parameters is an important task both for the manteinance of the network and for an objective assessment of the provided transmission quality.

When performing echo measurements two parameters assume a key role: the delay of the echo signal with respect to the incident signal and the attenuation introduced by the echo path (echo loss).

A simplified model of a long-distance telephone channel between A and B can be sketched as reported in Fig.1 [1]. In order to perform the echo measurement, a suitably chosen test signal $x(\cdot)$ is injected from T into the two-wire line of the telephone channel by means of a hybrid device H1. If a reflection of $x(\cdot)$ occurs at the hybrid H3, a distant echo is generated and is observed in R. Furthermore, also a near echo, composed of leakage through the hybrid H1, reflection along the two-wire path between H1 and H2 and leakage through the hybrid

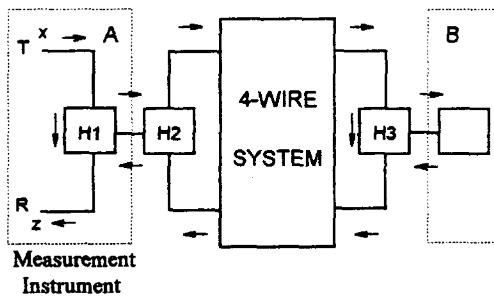

Fig.1 Simplified model of a telephone channel

H2 is observed in R [1]. The received signal $z(\cdot)$ acquired in R can be expressed as:

$$z(t) = e_0(t) + e_1(t) + w(t) \tag{1}$$

where $e_0(\cdot)$ is the near echo, $e_1(\cdot)$ is the distant echo and $w(\cdot)$ accounts for the noise and can be modeled as a wide-sense stationary white gaussian noise. In order to estimate the delay of the distant echo $e_1(\cdot)$ from observation of the received signal $z(\cdot)$ several techniques can be adopted [2]. Among these, a great interest is in cepstral analysis, both for its accuracy and for its implementation simplicity [2],[3]. However, severe limitations arise when the echo delay has short duration or the echo loss is large. In this paper, a modified cepstral analysis is introduced in order to overcome such limitations. It is based on a generalized whitening transformation depending on a parameter that can be chosen in such a way to optimize the peak detection in the cepstral domain. Furthermore, a signal processing technique for the estimation of the echo loss is presented.

The outline of the paper is as follows. In Section 2 the echo delay estimation procedure is described, while in Section 3 the estimation of the echo loss is presented.

In Section 4 simulation results are reported: they show the effectiveness and the accuracy of the proposed technique with respect to known estimation procedures.

2. ECHO DELAY ESTIMATION

To the aim of estimating the echo delay, the impulse response of the echo path can be well-approximated with a Dirac distribution properly delayed and attenuated. The near echo $e_0(\cdot)$ and the distant echo $e_1(\cdot)$ can be expressed as $e_0(t) = a_0 \cdot x(t-\tau_0)$ and $e_1(t) = a_1 \cdot x(t-\tau_1)$ respectively, where $x(\cdot)$ is the transmitted test signal, a_0, a_1 are the amplitudes of the echo signals with respect to $x(\cdot)$ and τ_0, τ_1 are the corresponding arrival times. As a consequence, the observed signal $z(\cdot)$ can be written as:

$$z(t) = a_0 \cdot x(t-\tau_0) + a_1 \cdot x(t-\tau_1) + w(t). \quad (2)$$

For the sake of conciseness the effect of noise is neglected in the following. Under such assumption, if the arrival time τ_0 of the near echo $e_0(\cdot)$ is assumed to be equal to the time origin $t=0$, eq. (2) becomes:

$$z(t) = x(t) + a \cdot x(t-\tau), \quad (3)$$

where $\tau = \tau_1 - \tau_0$ is the echo delay and a is the amplitude of $e_1(\cdot)$ with respect to $e_0(\cdot)$. In order to estimate the echo delay τ a cepstral analysis can be performed. The power cepstrum $z_c(u)$ of the observed signal $z(t)$ can be obtained by computing its energy spectrum $|Z(f)|^2$, performing a whitening transformation $g\{\cdot\}$, and by evaluating the inverse Fourier Transform (FT) of $g\{|Z(f)|^2\}$. If the whitening transformation is suitably chosen, $z_c(u)$ presents peaks of decreasing amplitude at $k \cdot \tau$, $k=1,2,\ldots$ [3]. Thus, the detection of the highest peak in the cepstral domain provides an estimate of the echo delay τ.

In the classical definition of power cepstrum the adopted whitening transformation is the logarithmic function. However, other interesting transformations have been proposed in the literature, since the logarithm may be too a severe spectral whitener for many applications. In [2] the envelope of the logarithmic derivative is considered, while in [3] the square root function is also proposed. In this paper, a modified parametric whitening transformation:

$$g\{y(\cdot), \beta\} = y^{\beta}(\cdot) \quad (4)$$

that depends on the parameter β is introduced, and the modified cepstral sequence $z_c(\cdot)$ provided by the inverse FT of $g\{|Z(f)|^2, \beta\} = |Z(f)|^{2\beta}$ is considered.

If transformation (4) is applied to the energy spectrum of the observed signal (3), one obtains:

$$|Z|^{2\beta} = |X|^{2\beta}\left(1 + a^2 + 2a \cdot \cos\omega\tau\right)^{\beta}. \quad (5)$$

When $a << 1$, a^2 can be neglected in eq. (5) and remembering that $(1+x)^a \cong 1 + a \cdot x$ if $x << 1$, $|Z|^{2\beta}$ can be well-approximated by the following expression:

$$|Z|^{2\beta} \cong |X|^{2\beta}(1 + 2a \cdot \beta \cdot \cos\omega\tau) \quad (6)$$

Thus, when an echo is present, $|Z|^{2\beta}$ contains a periodic component with period equal to $1/\tau$ and its inverse FT $z_c(u)$ exhibits a peak at $u = \tau$ with amplitude proportional to $2a \cdot \beta$. Hence, β should be chosen as large as possible in order to maximize the amplitude of the useful peak.

For larger values of a, a^2 can be no longer neglected in (5). In this case, by considering a series expansion, eq. (6) becomes:

$$|Z|^{2\beta} \cong |X|^{2\beta}\left(1+a^2\right)^{\beta}\left(1 + k\beta\cos\omega\tau + \frac{k^2}{2}\beta(\beta-1)\cos^2\omega\tau\right) \quad (8)$$

where $k = 2a/\left(1+a^2\right)$. By remembering that $\cos^2\vartheta = \frac{1+\cos 2\vartheta}{2}$, it can be seen that $z_c(u)$ presents a peak at $u=\tau$ with amplitude proportional to $k \cdot \beta$ and a second peak at $u = 2\tau$ with amplitude proportional to $\frac{k^2}{4}\beta \cdot (\beta - 1)$. In order to obtain a dominant peak at $u = \tau$, it should be:

$$k \cdot \beta > \frac{k^2}{4}\beta \cdot (\beta-1), \quad (9)$$

i.e. β should satisfy the following condition:

$$\beta < 1 + 2 \cdot (1+a^2)/a. \quad (10)$$

Since the function $(1+a^2)/a$ exhibits a minimum equal to 2 for $a = 1$, then it should be $\beta < 5$.

Finally, when $a \cong 1$, $z_c(u)$ may exhibits many other peaks at multiples of τ with amplitude depending on a and β.

In order to obtain relatively high values of the amplitude of the useful peak when $a << 1$ and avoid the presence of spurious peaks higher than the useful one when $a \cong 1$, a good trade off consists in choosing $\beta = 2$. In this case, in fact, $|Z|^{2\beta}$ can be written as:

$$|Z|^{2\beta} = |X|^{2\beta}\left(1 + 2k \cdot \cos\omega n_0 + k^2 \cdot \cos^2 \omega n_0\right) \quad (11)$$

and it can be easily seen that the useful peak in $u = \tau$ is higher than the peak at $u = 2\tau$ also for $a \geq 1$.

3. ECHO LOSS ESTIMATION

In order to estimate the echo loss, the received signal $z(\cdot)$ is observed in a measurement period D with duration T_D. The echo loss can be defined as:

$$A = \frac{P_x}{P_{e1}} \quad (12)$$

where P_x is the energy of the transmitted test signal, which can be considered known, while P_{e1} is the power of the distant echo e_1. From (2) it can be seen that the following relationship holds:

$$P_{e1} = P_z + a_0^2 P_x - 2a_0 C_{zx} \quad (13)$$

where P_z is the power of $z(\cdot)$, a_0 is the amplitude of the near echo and $C_{zx} = \frac{1}{T_D}\int_D x(t) \cdot z(t) \cdot dt$. Notice that P_z can be estimated from the observed signal $z(\cdot)$, while a_0 can be directly measured if a calibration stage is performed before the measurement. Since a common time origin for the transmitted and the received signal can be provided inside the measurement instrument, also C_{zx} can be easily estimated. By substituting the estimates of P_z, a_0 and C_{zx} in (13) and from (12), an estimate $\hat{A}$ of the echo loss can be obtained as:

$$\hat{A} = \frac{P_x}{\hat{P}_z + \hat{a}_0^2 P_x - 2\hat{a}_0 \hat{C}_{zx}} \quad (14)$$

where the superscript " ^ " denotes an estimated value.

4. EXPERIMENTAL RESULTS

In order to verify the effectiveness of the proposed technique a number of simulations have been performed. In the simulations different digital filters that model the behaviour of the hybrid circuits of Fig.1 and that introduce adjustable echo delay and loss have been used. Furthermore, both computer generated noise and real-life disturbances at various levels have been considered. For the sake of conciseness, only some meaningful results are reported in the following.

In the first example an impulse of duration of about 30 ms with spectrum flat between 300 and 800 Hz and nearly null elsewhere has been used as transmitted test signal. The echo signal has been generated by means of a real-life hybrid impulse response previously acquired. Computer generated white gaussian noise at Signal to Noise Ratio (SNR) equal to 60 dB with respect to the useful test signal has been added. The echo delay τ has been varied in the range 10-200 ms, with steps of 5 ms, while the echo loss has been fixed to 36 dB. For each value of τ, 40 different realizations of the superimposed computer-generated noise have been considered. The echo delay has been estimated by means of a cepstral analysis. In particular, three definitions of cepstrum have been considered:

a) $z_{CL} = FT^{-1}\left\{\log(|Z(f)|^2)\right\}$,

b) $z_{CR} = FT^{-1}\left\{|Z(f)|\right\}$,

c) $z_{CP} = FT^{-1}\left\{|Z(f)|^4\right\}$.

Notice that a) and b) represent classical definitions found in the literature [3], while c) is the proposed modified cepstrum. In Fig. 2(a) and 2(b) the mean and standard deviation of the echo delay estimation error are reported for the above definitions of power cepstrum. It can be seen that severely distorted estimates are obtained by adopting definitions a) and b), while the mean estimation error is null if the proposed modified cepstrum is adopted.

In the second example, the same test signal has been used. However, the delay τ has been chosen equal to 14 ms and the SNR equal to 60 dB. The echo loss has been varied beween 0 and 40 dB with steps equal to 3 dB. In Fig. 3 the mean (expressed in dB) of the echo loss estimation error is reported. It can be seen that the estimation error is within ± 1dB as far as the attenuation of the distant echo is higher than 25 dB.

In the third example, the measurement conditions are the same of the previous one, but in this case the echo loss has been fixed to 20 dB, while the Echo to Noise Ratio (ENR), defined as the ratio between the power of the echo signal to the power of the underlying noise, has been varied between 0 and 50 dB with steps equal to 3 dB. In Fig. 4 the mean of the estimation error of the echo loss is reported. It can be seen that the mean estimation error is negligible if the ENR is higher than 25 dB.

5. CONCLUSIONS

A modified cepstral analysis for accurate echo parameter estimation in telecommunication systems has been presented. Simulation results that show the effectiveness and the accuracy of the method have been reported and discussed.

The proposed technique has been implemented in a Digital Signal Processor (DSP) based instrument. A Texas Instruments TMS 320C31 running at 60 ns instruction cycle provided with 2 kword of 32-bit static random access memory has been adopted. The instrument is also equipped with 14-bit A/D and D/A converters. With the adopted processor the algorithm requires approximately 20% of the available processing time for real-time measurement.

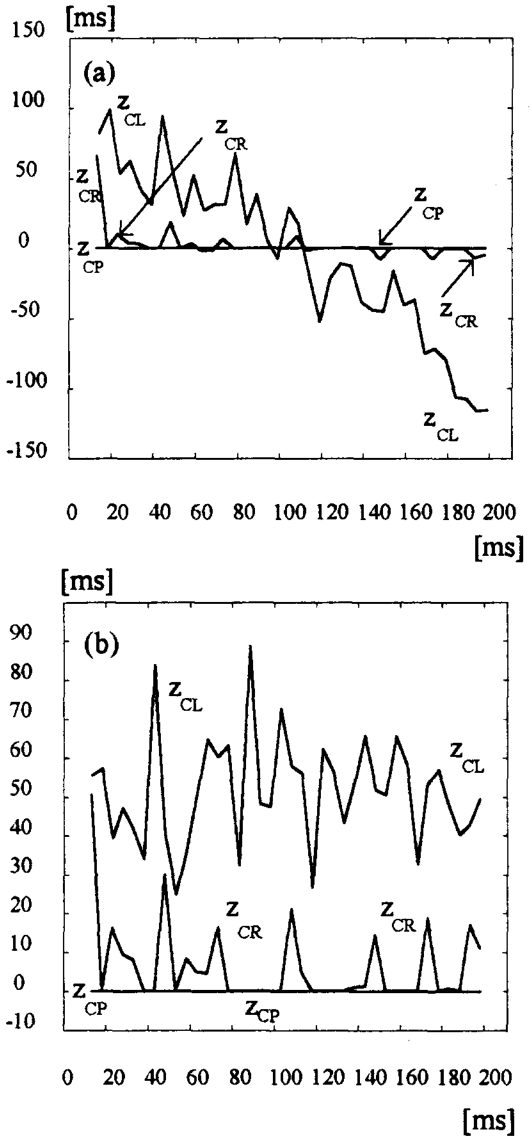

Fig. 2 Echo delay estimation error as a function of the echo delay. (a) Mean error (b) Error standard deviation.

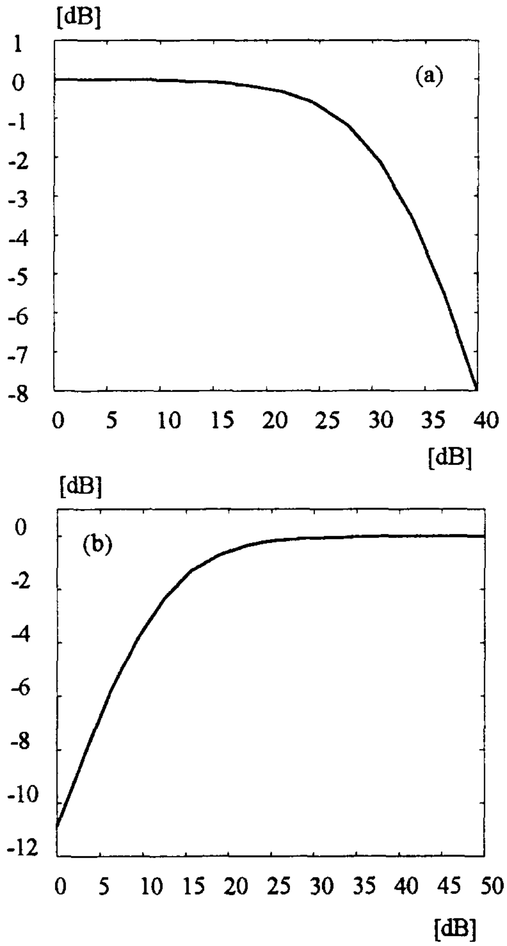

Fig. 4. Mean estimation error of the echo loss as a function of the echo loss (a) and of the Echo-to-Noise-Ratio (b).

6. REFERENCES

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