SIGNAL ADAPTIVE SUBBAND DECOMPOSITION FOR ADAPTIVE ECHO CANCELLATION

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ABSTRACT

This paper introduces a new architecture for adaptive echo cancellation, where the signals involved are first decomposed in two subbands, the adaptive filtering being performed separately for each subband signal. When performing the subband decomposition such that the analysis part compacts most of echo power in one subband, and leaves almost no echo power in the other band, the adaptive filtering turns out to be more efficient than in the single channel case.

1. INTRODUCTION

This paper focuses on the problem of acoustic echo cancellation in subbands. We combine the system identification capabilities of the adaptive filter with the discriminative features in the frequency domain obtained by adaptive subband decomposition of the signals.

The adaptive decomposition in subbands has been intensively studied in connection with subband coding. It has been shown that maximizing the variance of one subband signal (optimum compaction) of a perfect reconstruction orthogonal FIR filterbank is akin to maximizing the coding gain or minimizing the distortion for any fixed bit rate.

The design of the filterbank for optimum compaction gain, in other words optimum energy compaction filter design, has received many solutions. Some methods are slow but guarantee exact optimality [1, 2]; others are fast but sometimes fail to provide the exact optimum [3, 4]. Here we need long FIR filters and therefore we use the method presented in [1]. It is based on the use of semidefinite programming (SDP) reformulation of the optimization problem.

A comprehensive survey on adaptive filtering methods is provided by [5] which we follow here in notation. The architecture using adaptive filters within a subband decomposition structure has been studied extensively (e.g. [6, 7]), but there the filterbank has had fixed parameters and it is usually designed to fulfill the sharpest separation between highpass and lowpass bands. The optimum compaction filter selects those frequencies where the input has the highest power; therefore, it can have multiple bands in passband.

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2. THE ADAPTIVE SYSTEM STRUCTURE

General configuration of an adaptive echo canceler with signal adaptive subband decomposition is shown in Figure 1. It consists of an orthogonal signal adaptive perfect reconstruction filter bank and two adaptive filters. The microphone signal $d(n)$ consists of near-end signal $e(n)$ superposed over the echo $y(n)$. Echo is to be canceled using far-end signal $x(n)$. Analysis filters are designed such that energy of the far-end signal (and also echo) on one of the two subbands is maximized; correspondingly, the energy is minimized on the other subband. The same analysis filters decompose both far-end signal $x(n)$ and the microphone signal $d(n)$ into subband signals. The adaptive filtering is performed separately for each pair of subband signals and the echo canceler output $\hat{e}(n)$ is obtained by using the corresponding synthesis filterbank.

The design and operation of the proposed structure are summarized in Table 1 and the design of a signal adaptive filter bank is reviewed in Table 2. The adaptive algorithm is an LMS variant that operates on two subbands. The filter bank is designed such that it compacts as much as possible the energy of the far-end signal into the first subband. A measure for that is the compaction gain, which measures the ratio of the powers of the first subband $x_0(n)$ and the overall signal $x(n)$. The filter bank is rebuilt whenever the estimate of compaction gain is small. When the compaction gain is very high adaptation is omitted on the second channel.

There are several advantages in using the proposed structure. When most of the energy of far-end is concentrated on one of the subbands the adaptation process is necessary only on that channel and the filter weights can even be freezed on the other channel. This

For $n = 2, 4, 6 \ldots$ iterate:

1. Decompose far-end $x(n)$ into subband signals $x_0(n)$ and $x_1(n)$ and desired output $d(n)$ into subband signals $d_0(n)$ and $d_1(n)$. Downsample subband signals by factor of two. Let $k = n/2$.

2. Update the estimate of power of far-end signal
$$\sigma_x^2(k) = \lambda\sigma_x^2(k-1) + \frac{1}{2}\left(x(n)^2 + x(n-1)^2\right)$$
and subband signals
$$\sigma_{x_i}^2(k) = \lambda\sigma_{x_i}^2(k-1) + x_i(k)^2, \quad i = 0, 1.$$

3. Compute estimate of compaction gain
$$\sigma^2(k) = \frac{\sigma_{x_0}(k)}{\sigma_x(k)}.$$

4. If the compaction gain is small recompute the filter bank filters using far-end signal (about 30 ms in length). Reinitialize also subband adaptive filter weights and subband signals so that they correspond to new filter bank filters.

5. Compute the adaptive filter output $y_0(k)$ and echo canceler output $e_0(k) = d_0(k) - y_0(k)$. Update adaptive filter weights $w_{0i}(k)$, $i = 0, 1, 2, \ldots, (M-1)/2$.

6. Compute the adaptive filter output $y_1(k)$ and echo canceler output $e_1(k) = d_1(n) - y_1(k)$. If compaction gain is not very high update also adaptive filter weights $w_1i(k)$, $i = 0, 1, 2, \ldots, (M-1)/2$.

7. Synthesize echo canceler output $\hat{e}(n)$ from subband signals $e_0(k)$ and $e_1(k)$.

Table 1: Echo canceler with signal adaptive subband decomposition

introduces more flexibility to the control of adaptation. Adaptation is partly possible even when near-end signal level is high, e.g. during double talk. Also, the lengths of the adaptive filters are halved when compared to the adaptive filters operating on fullband.

3. ARTIFICIAL EXAMPLE

We will first consider an artificial example. The purpose of this experiment is to find out what we can achieve with normalized LMS adaptive filters. Both near-end and far-end signals are AR processes of order 16 and echo is generated artificially. The echo path

1. Estimate correlation sequence r_k, $k = 0, 1, 2, \ldots, N$, of far-end signal $x(n)$.

2. Find product filter $G(z) = H(z)H(z^{-1})$ such that variance
$$\sigma^2 = r_0 + 2\sum_{k=1}^{N} r_k g_k \tag{1}$$
of subband signal $x_0(n) = H(z)x(n)$ is maximized constrained such that product filter coefficients obey Nyquist(2) ($g_0 = 1/2$, $g_{2k} = 0$, $k \geq 1$).

3. Find optimum compaction filter $H(z)$ by spectral factorization of $G(z)$.

4. Build orthogonal perfect reconstruction filter bank with $H(z)$ and $z^{-N}H(-z^{-1})$ as analysis filters and $z^{-N}H(z^{-1})$ and $H(-z)$ as synthesis filters.

Table 2: Signal adaptive filter bank design

is linear and of length 128 (16 ms), as shown in Figure 2. Spectra of the input signals and the optimum compaction filter corresponding to the far-end signal is shown in Figure 3. The far-end and near-end signals roughly correspond to vowels 'uu' and 'aa', respectively.

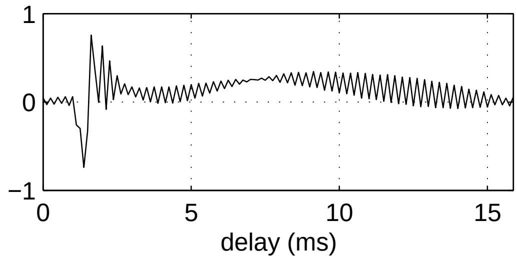

Figure 2: Artificial echo path

We simulated the performance of normalized LMS adapted echo canceler operating on one channel, operating on two fixed channels (lowpass-highpass filter bank) and two signal adapted channels. In Figure 4 we see the spectra of near-end signal and echo canceler output. As we see the spectra of the near-end signal and echo canceler outputs follow each other quite closely but there are strong peaks where there are transition bands in the corresponding filter bank filters. There is also another peak where the far-end signal is the strongest.

In this example the compaction gain of the under-

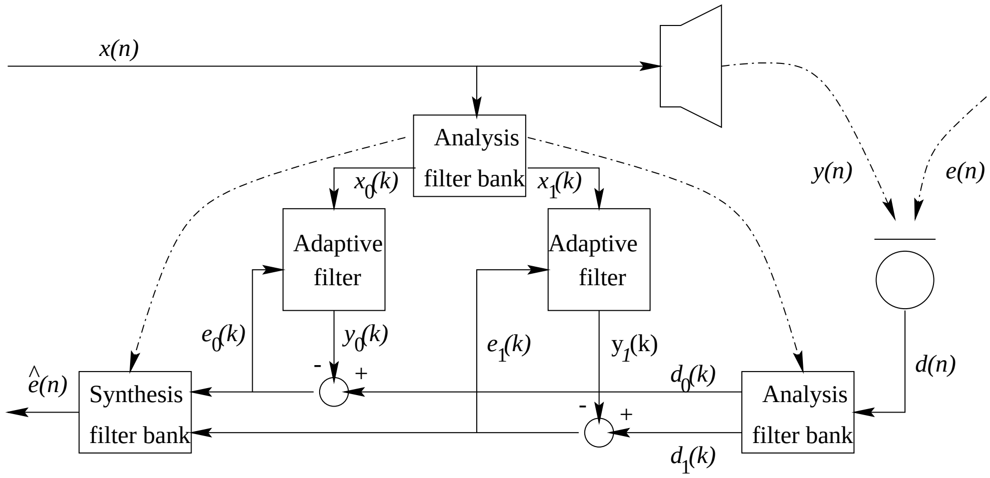

Figure 1: General configuration of adaptive echo canceler with signal adaptive subband decomposition. Far-end signal is denoted by $x(n)$ near-end signal by $e(n)$, echo by $y(n)$ and echo canceler output by $\hat{e}(n)$ Filter banks adapt to statistics of far-end signal $x(n)$.

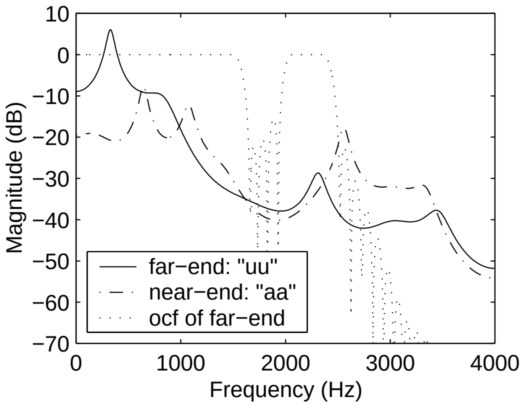

Figure 3: Spectra of far-end and near-end signal and optimum compaction filter (ocf) corresponding to far-end signal.

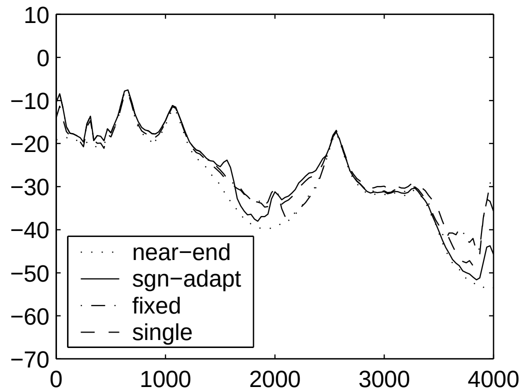

Figure 4: Spectra of original near-end signal and echo canceler outputs with different echo canceler structures: conventional, operating in two fixed subbands and operating in signal adapted subbands.

lying compaction filter is very high (1.9991). Therefore, we tried what happens if echo cancellation is completely omitted on the weaker channel. The resulting spectrum of the echo canceler output is shown in Figure 5. The results indicate that cancelation of echo on weaker channel is not always necessary.

4. SIMULATIONS

The proposed echo cancellation structure has been simulated using real test signals recorded in a halted car during single talk and in a moving car (100 km/h) during double talk (male far-end speaker, female near-end speaker). In the first case there is no strong near-end signal. In the second case the near-end signal level is pretty strong and it consists of car noise and speech (female voice). We will consider the two cases separately.

The length of the adaptive filter is 256 and the length of the filter bank filters is 80. Echo return loss enhancement (ERLE) was computed and the echo canceler output was listened to with all the simulation results.

The simulation results in halted car were not very promising; on the contrary, the ordinary normalized LMS algorithm gave the best results in terms of ERLE as shown in Figure 6. Decomposition into subbands and canceling the parts of the echo independently distorts the echo signal but does not attenuate it. Echo canceler output is not at all pleasant to hear.

On the other hand, the simulation results in moving car during double talk were promising. It was possible to benefit from packing most of the echo on one subband and canceling it there. In this experiment ERLE does not describe the echo cancelers performance, since the level of the near-end signal is rather strong. The fil-

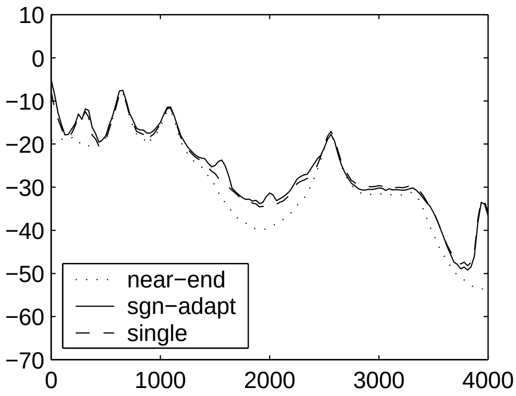

Figure 5: Spectra of the original near-end signal and echo canceler output with two echo canceler structures: conventional and operating on stronger subband.

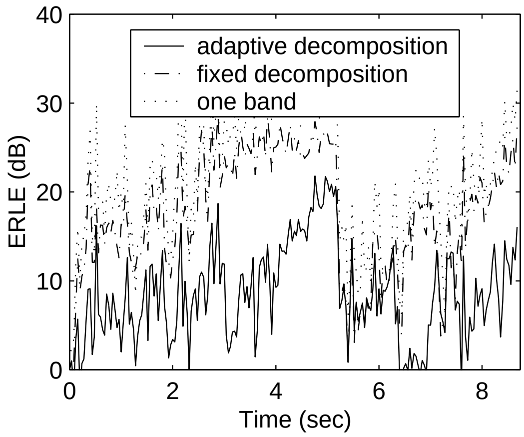

Figure 6: Echo return loss enhancement computed over a window of length 40 ms with different echo canceler structures: conventional, operating in two fixed subbands and operating in signal adapted subbands.

ter bank filters were recomputed whenever compaction gain dropped below 1.65. Values of the compaction gain during simulations are shown in Figure 7. Compaction gain is actually highest when there is only car noise in far-end.

5. CONCLUSIONS

We have proposed a new structure for adaptive echo cancellation where the signals involved are first decomposed into two subbands according to the frequency content of the far-end signal and then the echo cancellation is performed independently on both channels. This introduces flexibility to the control of adaptation. The results indicate small improvement in performance when the level of the near-end signal is strong.

We used rather short filter for building the signal adapted filter bank, since their design is a computationally demanding task. Simulations with artificially created signals have shown that the performance improves as the compaction filter order increases.

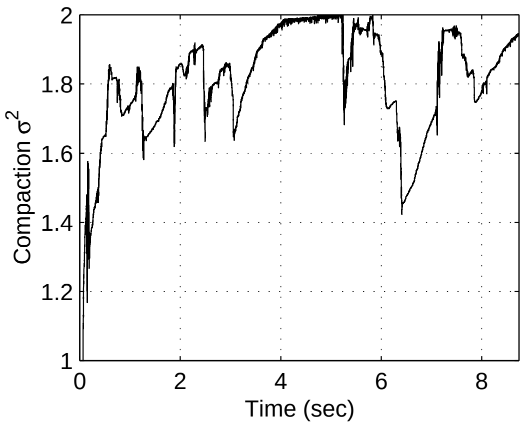

Figure 7: Value of compaction gain during echo cancellation.

6. REFERENCES

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