

Design of analysis/synthesis (A/S) filter banks for a subband AEC system is a complex problem involving several trade-offs/requirements: near-perfect reconstruction, low processing delays, oversampling in the subbands, low complexity. For ease of implementation, it is also desirable that the subband signals be real. Based on these considerations, we have found it convenient to use Weaver SSB A/S banks in our subband filtering structure; the latter is illustrated in Fig. 2.

The loudspeaker signal  $u(n)$  and the microphone signal  $d(n)$ , with sampling rate  $F_s$ , are each decomposed into  $B$  real subband signals by analysis filter banks based on Weaver SSB modulators. Each bank consists of  $B$  band-pass filters followed by decimators by  $K \leq B$ . The corresponding subband signals are denoted by  $u_b(m)$  and  $d_b(m)$ , where  $b = 0, \dots, B-1$  and  $m$  denotes sampling-time at the lower rate  $F'_s = F_s/K$ . In each subband, an FNTF adaptive algorithm operating at the reduced rate  $F'_s$  is used to identify the corresponding subband component of the echo path. The subband error signals  $e_b(m)$  are finally recombined by a dual synthesis bank to produce a full-band error signal  $e(n)$ , at the original rate  $F_s$ .

In our implementation, the digital spectrum  $[-\pi, \pi]$  is divided into  $B$  real subbands, with bandwidth  $\omega_\delta = \pi/B$ . The center frequency of the  $b$ th subband (positive sideband) is given by  $\omega_b = (b + \frac{1}{2})\omega_\delta$ ,  $b = 0, 1, \dots, B-1$ . Each narrow-band filter in the analysis bank is a Weaver modulator with center frequency  $\omega_b$ ; a corresponding Weaver demodulator is used in the synthesis bank (see [6] for details). In theory, subband aliasing may be avoided with critical downsampling, i.e.  $K = B$ , provided an ideal low-pass filter  $h(n)$  with cut-off  $\omega_\delta/2$  is used. In practice, because of non-ideal filters, oversampling is necessary, i.e.  $K < B$ ; we chose  $K = B/2$ . A window technique is used to design  $h(n)$ . For  $B = 16$ , the resulting A/S bank has the following properties: processing delay of 16ms, amplitude distortion within  $\pm 0.05dB$ , linear phase.

#### 4. Results and discussions

The sampling rate is set to  $F_s = 8kHz$ . Different loudspeaker signals  $u(n)$  are used in the experiments. Here, we show results for the composite source signal (CSS), a speech-like signal [7]. The echo path  $\mathcal{H}$  in Fig. 1 is simulated with a synthetic room impulse response of duration 2048 samples. To produce the microphone signal, this response is convolved with  $u(n)$  and independent white noise is added. The convergence performance is evaluated in terms of the short term power of  $e(n)$  (32ms window, dB relative to the echo level). The parameter values  $(N, M, \lambda)$  are the same for all the subband FNTFs. Following [4]:  $\lambda$  is set to a large value ( $\lambda = 1 - 1/2N$ ), an acceleration mechanism is used in (3) and the filtering part of FNTF is initially frozen (for 0.5s). Note that for  $M = 0$ , FNTF corresponds to a modified form of NLMS, while in the case  $M = N$ , FNTF corresponds to FAEST (FRLS family).

Fig. 3 shows convergence curves of the full-band FNTF for  $N = 2048$  (256ms),  $SNR = 30dB$  and different  $M$ . The top and bottom curve represent the microphone signal  $d(n)$  and the additive noise. It is seen that a large value of  $M$  ( $\geq 256$ ), is needed to obtain a performance comparable to RLS. Fig. 4 shows convergence curves of the subband FNTF for  $N = 256$  (256ms),  $SNR = 30dB$  and different  $M$ . Again, a relatively large value of  $M$  (about 128), is required to achieve RLS-like performance, despite the fact that in subbands, the effective prediction order of  $u(n)$  is smaller due to decimation.

Our study points to the following general conclusions: (1) the selection of the parameter  $M$  in FNTF can not be based strictly on the necessary AR modeling order of the source signal  $u(n)$ ; (2) the use of subband processing is not effective in reducing the ratio  $M/N$  necessary for efficient operation of FNTF with long filters; (3) subband FNTF appears to be of limited practical value for AEC in teleconference applications.

#### References

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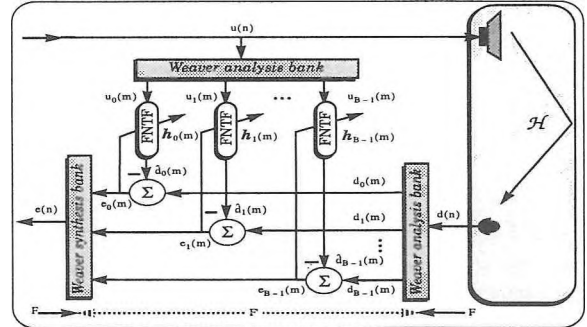


Figure 2: Weaver SSB subband structure for AEC.

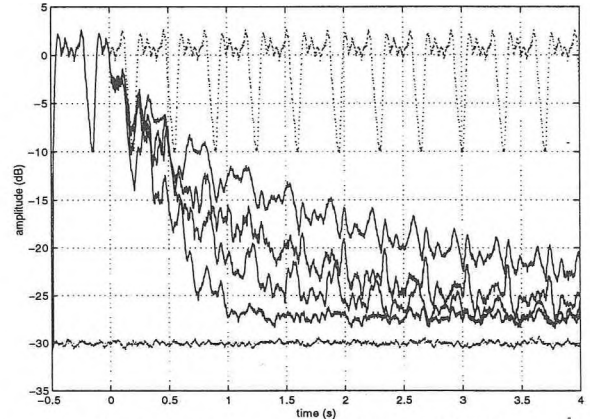


Figure 3: Convergence curves for fullband FNTF (top to bottom: echo; FNTF with  $M=0,16,256,2048$ ; noise).

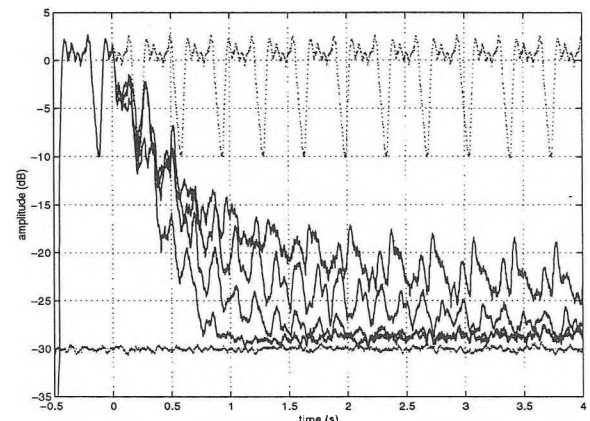


Figure 4: Convergence curves for subband FNTF (top to bottom: echo, FNTF with  $M = 0,8,128,256$ ; noise).