13.5 Suppose a speech waveform is modeled with vocal tract poles and zeros, and consider the problem of estimating the speech in the presence of additive noise. Propose a (linear) iterative method, as a generalization of the LMAP algorithm for all-pole estimation of Section 13.4, that estimates both the poles and zeros, as well as the speech, simultaneously. *Hint:* Maximize the *a posteriori* probability density $p(\underline{a}, \underline{b}, \underline{x}/\underline{y})$. The vectors $\underline{a}$ and $\underline{b}$ represent the pole and zero polynomial coefficients, respectively, and the vectors $\underline{x}$ and $\underline{y}$ represent (for each analysis frame) the clean and noisy speech, respectively.

13.6 In this problem, you explore the use of a masking threshold in a suppression filter $h_s[n]$ that results in perceived noise that is an attenuated version of the original noise. In this formulation, given an original noisy signal $y[n] = x[n] + b[n]$, the desired signal can be written as $d[n] = x[n] + \alpha b[n]$ where α is the noise suppression scale factor.

(a) Show that an estimate of the short-time power spectrum of the noise error $\alpha b[n] - h_s[n] * b[n]$ is given by Equation (13.20).

(b) Derive the suppression filter range, Equation (13.21), for which the short-time noise error power spectrum estimate falls below the masking threshold $T(pL, \omega)$.

(c) Discuss tradeoffs in the degree of noise attenuation and the speech distortion that can be obtained through the parameter α. Compare this suppression tradeoff with that of the standard Wiener filter.

13.7 Show that with FBS synthesis in Section 13.6.2, the operation equivalent to filtering an STFT by $P\left(n, \frac{2\pi}{N}k\right)$ along temporal trajectories is a filtering by a single time-invariant linear filter consisting of a sum of bandpass filters, i.e.,

$$y[n] = x[n] * \left[\frac{1}{Nw[0]} \sum_{k=0}^{N-1} \left\{ w[n] * P\left(n, \frac{2\pi}{N}k\right) \right\} e^{j\frac{2\pi}{N}kn} \right].$$

13.8 Suppose that we apply the filter $P(n, \omega)$ along each time-trajectory of the STFT magnitude of a sequence $x[n]$, as in Section 13.6.3. Show that applying the filter-bank summation (FBS) method with discretized frequencies $\omega_k = \frac{2\pi}{N}k$ results in the input sequence $x[n]$ being modified by the time-varying filter of Equation (13.28).

13.9 Consider a signal $y[n]$ of the form $y[n] = x[n] * g[n]$ where $g[n]$ represents a linear time-invariant distortion of a desired signal $x[n]$. In this problem you explore different formulations of the STFT of $y[n]$.

(a) Given $y[n] = x[n] * g[n]$, show that

$$Y(n, \omega) = (g[n]e^{-j\omega n}) * X(n, \omega)$$

where the above convolution is performed with respect to the time variable n. Then argue that the two block diagrams in Figure 13.14 are equivalent.