acoustic horn $h(t)$:

$$v(t) = s(t) * h(t). \tag{6.36}$$

Our goal is to recover $s(t)$ without knowing $h(t)$, a problem that is sometimes referred to as "blind deconvolution." The approach in discrete time is to find an estimate of the horn $\hat{h}[n]$, invert its frequency response to form the inverse filter $\hat{H}^{-1}(\omega)$, and then apply inverse filtering to obtain an estimate of $s[n]$. (The notation "hat" denotes an estimate, and not a cepstrum as in this chapter.) We will accomplish this through a homomorphic filtering scheme proposed by Stockham, Cannon, and Ingebretsen [18] and shown in Figure 6.25.

(a) Our first step is to window $v[n]$ with a sliding window $w[n]$ so that each windowed segment is given by

$$v_i[n] = v[n + iL]w[n]$$

where we slide $v[n]$ L samples at a time under the window $w[n]$. If $w[n]$ is "long and smooth relative to $s[n]$," argue for and against the following convolutional approximation:

$$\begin{aligned} v_i[n] &= w[n](s[n + iL] * h[n]) \\ &\approx (w[n]s[n + iL]) * h[n] \\ &= s_i[n] * h[n] \end{aligned} \tag{6.37}$$

where $s_i[n] = w[n]s[n + iL]$ and where we have ignored any shift in $h[n]$.

(b) Determine the complex cepstrum of $v_i[n]$ (from the approximation in Equation (6.37)) and argue why we cannot separate out $s[n]$ using the homomorphic deconvolution method that requires liftering the low-quefrency region of the complex cepstrum.

(c) Suppose that we average the complex logarithm (Fourier transform of complex cepstrum) over many segments, i.e.,

$$V_{avg} = \frac{1}{M}\sum_{i=1}^{M} \log[V_i(\omega)].$$

Give an expression for V_{avg} in terms of $S_{avg} = \frac{1}{M}\sum_{i=1}^{M} \log[S_i(\omega)]$ and $\log[H(\omega)]$. Suppose that S_{avg} = constant. Describe a procedure for extracting $H(\omega)$ to within a gain factor.

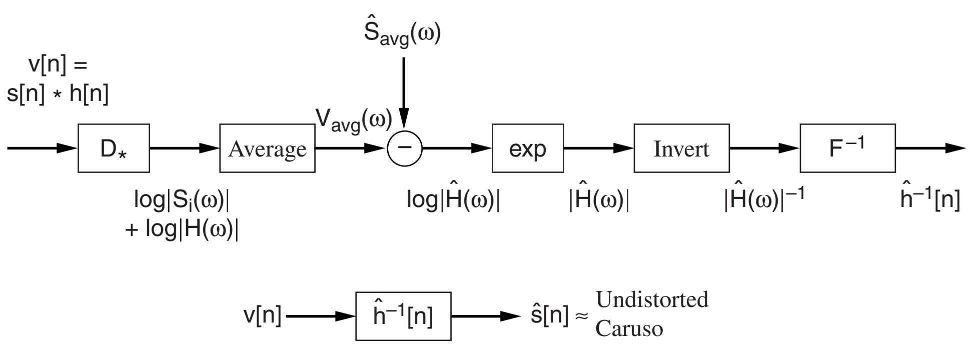

Figure 6.25 Restoration of Caruso based on homomorphic filtering.