ECE 537 Fundamentals of Speech Processing
Problem Set 2

UNIVERSITY OF ILLINOIS
Department of Electrical and Computer Engineering

Assigned: Monday, 8/29/2022; Due: Monday, 9/5/2022
Reading: Homer Dudley, "The Vocoder—Electrical Re-creation of Speech," and J.C.R. Licklider, "A Duplex Theory of Pitch Perception"

1. Suppose that, in the style of the vocoder, we have a voiced excitation signal

$$e[n] = \sum_{m=-\infty}^{\infty} \delta[n - mN_0]$$

We want to filter it through ten bandpass filters $H_l(\omega)$ ($1 \le l \le 10$), then scale each band by amplitude A_l, in order to match the energy of each band in the original speech signal $s[n]$. To be more precise, we want it to be the case that, for each l,

$$\sum_{k=0}^{N_0-1} \left| H_l\left(\frac{2\pi k}{N_0}\right)\right|^2 |S_k|^2 = \sum_{k=0}^{N_0-1} |A_l|^2 | \left| H_l\left(\frac{2\pi k}{N_0}\right)\right|^2 |E_k|^2, \tag{1}$$

where S_k are the discrete-time Fourier series coefficients of $s[n]$, and E_k are the Fourier series coefficients of $e[n]$.

(a) (1 point) Suppose that the filters are ideal bandpass filters with bandwidth $\frac{2\pi b}{N_0}$ and center frequency $\frac{2\pi a}{N_0}$ for some integers a and b, in other words:

$$H\left(\frac{2\pi k}{N_0}\right) = \begin{cases} 1 & a - \frac{b}{2} \le k < a + \frac{b}{2} \\ 1 & N_0 - a - \frac{b}{2} < k \le N_0 - a + \frac{b}{2} \\ 0 & \text{otherwise} \end{cases}$$

Find positive real numbers A_l in terms of S_k so that Eq.(1) is satisfied.

(b) (1 point) In 1940, Fourier analyzers were not very cheap. Instead, most spectral analysis was done by using bandpass filters to compute

$$s_l[n] = h_l[n] * s[n],$$

and then finding the power of the signal in the time domain,

$$P_l = \frac{1}{N} \sum_{n=0}^{N-1} s_l^2[n]$$

Assuming ideal bandpass filters as in part (a), find A_l in terms of P_l, a, and/or b. State any assumptions that you need to make about N.

2. Suppose that, in the style of the vocoder, we have an unvoiced excitation signal, $e[n]$. Assume that $e[n]$ is zero-mean, unit variance, and uncorrelated, i.e.,

$$E[e[n]] = 0$$
$$E\left[e^2[n]\right] = 1$$
$$E[e[n]e[n-m]] = 0, \quad m \neq 0$$

The last two lines in the equation above can be summarized by saying that the autocorrelation is a delta function, $R_{ee}[m] = \delta[m]$, where the autocorrelation is defined to be

$$R_{ee}[m] = E[e[n]e^*[n-m]],$$

where $e^*[n]$ is the complex conjugate of $e[n]$, which is part of the definition of autocorrelation, but is not relevant for this problem, because this problem uses only real-valued signals. We want to filter $e[n]$ through ten bandpass filters $H_l(\omega)$ $(1 \leq l \leq 10)$, then scale each band by a positive real number A_l, in order to match the energy of each band in the original speech signal $s[n]$. To be more precise, we want it to be the case that, for each l,

$$\int_{-\pi}^{\pi} |H_l(\omega)|^2 R_{ss}(\omega) d\omega = A_l^2 \int_{-\pi}^{\pi} |H_l(\omega)|^2 R_{ee}(\omega) d\omega \tag{2}$$

where $R_{ee}(\omega)$ and $R_{ss}(\omega)$ are the power spectra of $e[n]$ and $s[n]$, respectively.

(a) (1 point) Suppose that the filters are ideal bandpass filters with bandwidth β and center frequency α radians/sample, in other words:

$$H(\omega) = \begin{cases} 1 & \alpha - \frac{\beta}{2} \leq \omega < \alpha + \frac{\beta}{2} \\ 1 & -\alpha - \frac{\beta}{2} < \omega \leq -\alpha + \frac{\beta}{2} \\ 0 & \text{otherwise} \end{cases}$$

Find positive real numbers A_l in terms of $R_{ss}(\omega)$ so that Eq.(2) is satisfied.

(b) (1 point) In 1940, Fourier analyzers were not very cheap. Instead, most spectral analysis was done by using bandpass filters to compute

$$s_l[n] = h_l[n] * s[n],$$

and then finding the power of the signal in the time domain,

$$P_l = \frac{1}{N} \sum_{n=0}^{N-1} s_l^2[n]$$

For stochastic analysis, we need to assume that N is long enough so that

$$P_l \approx E\left[s_l^2[n]\right] = R_{s_l s_l}[0], \tag{3}$$

where $R_{s_l s_l}[m]$ is the autocorrelation of $s_l[n]$. Assuming ideal bandpass filters as in part (a), find A_l in terms of P_l, α, and/or β.

3. One of the advantages of Licklider's duplex theory of pitch perception is its potential for detecting periodic signals in noisy listening conditions. Consider the signal

$$x[n] = v[n] + s[n],$$

where $v[n]$ is zero-mean, unit-variance white noise, and $s[n]$ is a periodic speech signal with the form

$$s[n] = \sum_{k=0}^{N_0-1} S_k e^{j\frac{2\pi kn}{N_0}}$$

In a realistic measurement scenario, the timing of the input signal is unknown, so we can treat n as a uniformly distributed random variable. For example,

$$\begin{aligned} E\left[s[n]\right] &= \sum_{k=0}^{N_0-1} S_k E\left[e^{j\frac{2\pi kn}{N_0}}\right] \\ &= 0, \end{aligned}$$

where the second line follows by taking expectation over the random variable n.

(a) (1 point) Find the autocorrelation of $s[n]$. In this case, even though $s[n]$ is real-valued, the components of the Fourier series are not, so use the formula

$$R_{ss}[m] = E\left[s[n]s^*[n-m]\right],$$

and then use the assumption that n is random.

(b) (1 point) In the preceding section, we saw that the noise power was $R_{vv}[0] = 1$, while the power of the periodic part of the signal is $R_{ss}[0] = R_{ss}[N_0]$. Licklider's model reduces the noise power, without reducing the signal power. Suppose, for example, that we bandpass filter $x[n]$ through ideal bandpass filters with some bandwidth less than the pitch period:

$$\beta = \gamma\frac{2\pi}{N_0}, \quad 0 < \gamma < 1.$$

Now suppose that we compute the autocorrelations in every channel, and then we add together only the channels that have significant energy at a delay of $m = N_0$. What is the resulting noise power, $R_{vv}[0]$?