EE4000 **Semester 1**
Signal Processing 3 **O P KENNY**

<u>Assignment 2016</u>
<u>MATLAB Cochlear Implant Simulation Program</u>

Background

One speech processing strategy used in cochlear implant processors is the Spectral Maxima Sound Processor (SMSP). In this scheme, the speech waveform is analyzed using a bank of 16 band-pass filters and a spectral maxima detector. A block diagram for the operation of this strategy is shown in Figure 1.

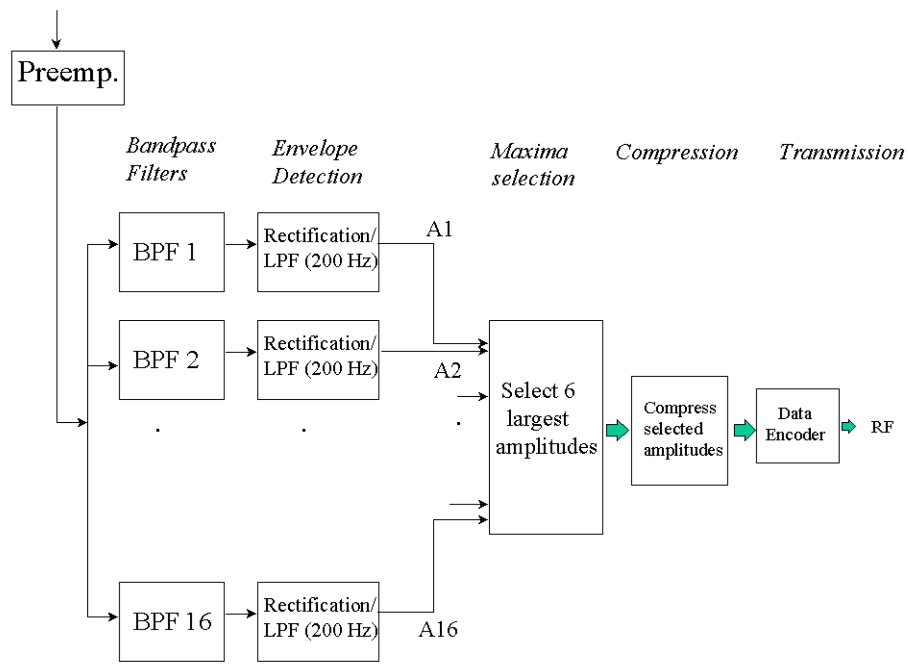

Figure 1 Block diagram for the SMSP Cochlear Implant Strategy

The signal form the microphone is first preprocessed by an amplifier to get a suitable dynamic range for the analogue-to-digital converter. The signal is then sent through a bank of 16 band-pass filters with center frequencies ranging from 250 to 5.4 KHz. The output of the filter is rectified and low-pass filtered with a cutoff frequency of 200 Hz. After computing all 16 filter outputs, the SMSP processor selects, at 4msec intervals, the six largest filter outputs. The six amplitudes are compressed logarithmically with the result transmitted to the electrode array through a radio link. The electrode array is shown in Figure 2, and the output of each of the filters is allocated to each electrode. The apical electrode is allocated to the filter with the lowest center frequency, while the basal electrode is allocated to the filter with the highest center frequency. Only the 16 most-apical electrodes are activated, while the remaining basal electrodes in the 22-electrode implant are left inactive.

Figure 2 Cochlear implant electrode array

After the amplitude selection stage, amplitudes have to be mapped for individual patients for all the filter outputs. The patients map ranges form a threshold level to a comfort level. These to levels are commonly referred to T and C levels. The Threshold is the minimum level that a patient can hear and the C level is the maximum level that a patient can hear without discomfort. The mapping function for each channel is given by,

$$A = (C - T)x + T \ ,$$

where A is the mapped amplitude and x is the input amplitude that ranges from 0 to 1. For this experiment, assume T=.1 and C=.9.

Electrodes are stimulated one at a time. The order the six chosen electrodes stimulate the nerve ending is from the biggest to smallest amplitude. Hence every 4msecs six electrodes are stimulated. A useful tool to analysis the cochlear implant strategy is the electrod-ogram, which show each electrode is stimulated over the evolution of time. A typical electrodogram is shown in Figure 3.

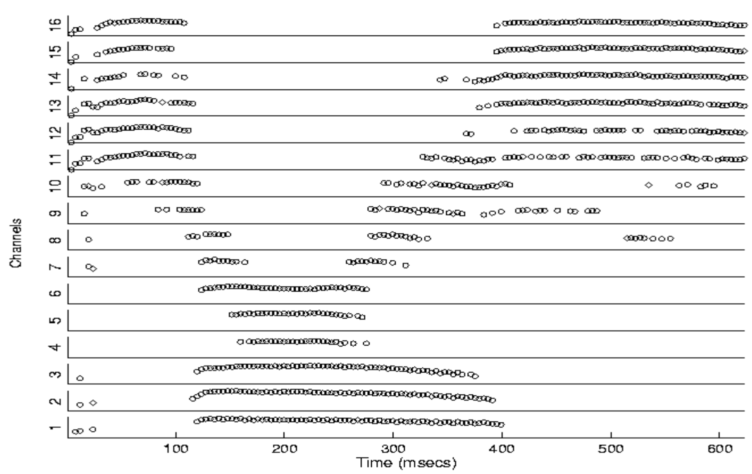

Figure 3 Typical Electrodogram Display

Task 1

The task is to write a matlab simulation for the SMSP strategy and display an electrodogram after the mapping function, but before the compression stage. The filter requirements are is linear phase with the center frequencies for the 16 band-pass filters being 250 Hz, 375 Hz, 500 Hz, 625 Hz, 750 Hz, 937 Hz, 1187 Hz , 1437 Hz, 1687 Hz , 2000 Hz, 2375 Hz, 2812 Hz, 3312 Hz, 3875 Hz, 4563 Hz, 5375 Hz.

The corresponding bandwidths for the 16 filters are:- 124 Hz, 126 Hz, 124 Hz,126 Hz, 124 Hz, 250 Hz, 250 Hz, 250 Hz, 250 Hz, 376 Hz , 374 Hz, 500 Hz, 500 Hz, 626 Hz, 750 Hz, 874 Hz .

Note: The transition band for the filters is not defined.

A test signal is provided, mbet.wav (male saying the word “bet”) that can be copied and read into MATLAB using the audioread function. The signal is than resampled at 16 KHz, which can be done using the resample function. The matlab commands to read in the .wav file is as follows

```
[y, FS]=audioread(fname);

y1=resample(y,16000,Fs);
```

In addition to producing the electrodogram, indicate the region in the electrodogram where the vowel ‘e’ is located highlighting the formant frequency structure of this vowel.

Assignment Task 2:

With normal hearing the volume of the sound drops with respect to frequency. For this reason, in a hearing test results are measured in dB (HL). This is a dB scale, but measured with reference to a normal hearing level. Hence dB (HL) corresponds to the gain needed to restore hearing to the normal hearing level. Ideally, the dB (HL) for a normal hearing person would be around 0dB (HL) across the frequencies. In this task, we will take the dB (HL) to be dB (HL) = $20\log_{10}(G)$, where G is the gain required.

Design a filter bank consisting of 15 filters for a hearing aid. Each filter has to be weighted to produce 0dB HL. The output of the weighted filters are added together to produce the required amplified sound.

The results from a hearing test are as follows:

The frequencies were used in the testing:- 250 Hz, 500 Hz, 750 Hz, 1000 Hz, 1250 Hz, 1500 Hz, 1750 Hz, 2000 Hz, 2250 Hz, 2500 Hz, 2750 Hz, 3000 Hz, 3250 Hz, 3500 Hz, and 3750 Hz

The corresponding dB (HL) were evaluated to be:- 9.0dB, 14.0dB, 37dB, 53dB, 57dB 41dB, 34dB, 38dB, 40dB, 43dB, 45dB, 46dB, 48dB, 49dB, 50dB, and 51dB];

As it can be seen this person has hearing loss around 1kHz.

The file hearingloss.mat, contains a test signal, Y, which simulates what the hearing loss might sound like. You can use this signal to evaluate the performance of your hearing aid. The corresponding normal hearing level signal is the signal, X. The sampling frequency used was 8 kHz.

Demonstrate that your hearing aid works by passing test tone signals at frequencies corresponding to those used in testing. Calculate the gain in dB and compare to the dB (HL) levels in the testing.

Marks distribution:
This assignment is worth 15% of your final mark for EE4000. The marks will be distributed according to the following schedule:

Level of achievement- 8%
Report and Software documentation-7%

Assessment Criteria:

1. Your report shall be assessed on its technical content, and justifications of your method of solving the given tasks.

The assessment is designed to allow the student to demonstrate their level of competency in the area of digital signal processing.

Your submission:
You are to submit the following items for assessment:

1. A report which details the design procedure used for the filters.
2. Your MATLAB software- both printed and source file. The simulation software should be a single m-file.