Work for Digital Signal Processing Supervision II

Please attempt all the questions below and submit your work by 6pm on the day before the supervision. Regardless of whether you submit the main body of the work in PDF or hand-written form, please include in it any code samples and their output, and also send email them to me as separate files so that I can run/unit test them. Please also include all your working and any additional thoughts with your work.

Questions

1. Examples sheet exercises 10–13, 16 and 17.

   For any spectrogram you plot, adjust the `window`, `noverlap` and `nfft` parameters such that the "pixels" in the plot appear square (see `> doc spectrogram` for a description of the parameters). Adjust the limits of the x- and y-axes so that the plot displays the relevant information most clearly. It is conventional to display time on the x-axis of audio spectrograms, and on the y-axis of IQ spectrograms. For spectrograms of IQ data, make sure that the lowest frequency appears on the far left of the plot.

2. Examination question: 2013 Paper 9 Question 5 (attached at the end of this document).

3. (a) What is modulation?

   (b) For continuous signals, explain amplitude modulation and frequency modulation. Explain how frequency modulation can be viewed as a special case of phase modulation.

   (c) What is the difference between coherent and non-coherent demodulation? When might it be important to use coherent demodulation (exhibit an example)? When may it be less important?

   (d) What is IQ modulation?

   (e) In your own words, explain the operation of ASK, QPSK, 8PSK, and FSK.

4. Research and explain the operation of a Tayloe (Quadrature Product) Detector. Structure your explanation by describing the high-level relationship of the output signals to the input signal, and then by describing at a low-level how the desired output is obtained. Why must the commutator be driven at four times the centre frequency? What are the advantages of the Tayloe Detector over other methods of performing the same function? What are its disadvantages? Practically, what determines the highest and lowest frequencies at which a Tayloe Detector is useful?

5 Digital Signal Processing (MGK)

(*a*) Consider a digital filter with impulse response

$$h_i = 2\alpha \cdot \frac{\sin[2\pi(i-n/2)\alpha]}{2\pi(i-n/2)\alpha} \cdot w_i \quad \text{where} \quad w_i = \begin{cases} 1, & 0 \le i \le n \\ 0, & \text{otherwise} \end{cases}.$$

(*i*) What type of filter is this? [4 marks]

(*ii*) How are the sampling rate f_s at which this filter is operated and its −6 dB cut-off frequency f_c related to parameter α? [2 marks]

(*b*) In an open-source audio-effect library, you find a C routine for processing a recorded voice to sound like it came over an analog phone line:

```
#include <math.h>
#define N 512
#define PI 3.14159265358979323846
void phone_effect(double *x, double *y, int m)
{
  double w, p, f, g, h[N+1];
  int i, k;
  for (i = 0; i <= N; i++) {
    w = 0.54 - 0.46 * cos(2*PI*i/N);
    p = 2 * PI * (i-N/2) / 10;
    f = w * ((p == 0) ? 1 : sin(p)/p) / 5;
    p = 2 * PI * (i-N/2) / 100;
    g = w * ((p == 0) ? 1 : sin(p)/p) / 50;
    h[i] = f - g;
  }
  for (i = 0; i < m; i++) {
    y[i] = 0;
    for (k = 0; k <= N && k <= i; k++)
      y[i] += x[i - k] * h[k];
  }
}
```

The input array `x` and the output array `y` each hold `m` samples of an audio recording (mono) at sampling frequency $f_s = 32$ kHz.

(*i*) Explain in detail what operation is implemented here (e.g., type of filter, order, cut-off frequency) and how it has been constructed. [8 marks]

(*ii*) You want to use this algorithm on audio recordings with a sampling rate of 48 kHz. What do you have to change in the source code to ensure that the audible effect remains the same? [6 marks]