**University of Nottingham**

**Faculty of Engineering**

**Department of Electrical and Electronic Engineering**

**H64DSP – Digital Signal Processing for Telecommunications, Multimedia and Instrumentation**

**Coursework- Filtering of a Voice Signal**

**1. Introduction**

The coursework component for H64DSP is worth 30% of the module assessment. The coursework makes use of MATLAB to process sound signals which have been corrupted by noise and interference.

There are Workshops that take place for one hour each week in the Tower Computer Room (3rd Floor), where you have access to computers running MATLAB but you can of course work on this project outside this slot. MATLAB is available for download for all registered students. In addition, MATLAB is also run on computers in both the Tower and other communal computer rooms in the University.

The main educational aim of this project is to reinforce the theoretical material introduced in the lectures. You will have the opportunity to apply a wide variety of filters and other signal processing techniques and you should, by the end of the project, have a better understanding of these techniques.

**Please note that this project is worth just 3 credits, so you should spend no more than 2 to 3 hours a week on it.** If you find yourself having to work longer than this, or have any other problems, then please feel free to contact me.

**2. Background to the Project**

**2.1 Data Files**

You are provided with the following MATLAB files containing a voice signal:

*H64DSP1718.m4a -* Clean signal

*corrupt1718.m4a -* corrupted version of the clean signal.

Information about reading and writing sound waves in MATLAB can be found at:

<http://uk.mathworks.com/help/matlab/import_export/read-and-get-information-about-audio-files.html> (last accessed 18/9/17).

For example, if you want to read an audio file, then you use the command **audioread**:

e.g. [y, fs] = audioread(‘corrupt.m4a’)

The array *y* will contain the data values. *fs* will give you the sampling frequency (Hz).

If you want to hear the signal, then you should use the **sound** command:

sound(*y*,*fs*)

Note that the default value for fs is 8 kHz whilst the recorded signal has been sampled at 48 kHz, so, if you miss out the *fs* value then the recording will sound sinister!

Hearing the signal after processing may give you an indication of how close to the clean signal is the processed signal. However, if loudspeakers are not available, comparing the plots of the variations with time of the clean and processed signals is just as good a procedure for comparison.

After having processed the data, if you want to write it to a file with \*m4a format then you should use the **audiowrite** command:

e.g. audiowrite(‘processed.m4a’,y,fs)

Note: in this example the chosen filename is processed.m4a, but any name can be chosen as long as the suffix .m4a is used.

Note that if you read the audio file *H64DSP1718.m4a* using the command:

[y fs] = audioread(‘H64DSP1718.m4a’)

then array *y* will consist of two columns. For this recording, both columns of *y* are identical, so you need only choose one column (say the first) to compare with the corrupted signal.

**2.2 Aim of the Project**

The aim of the project is to process the data in *corrupt1718.m4a* in order to recover as much as possible the original signal contained in *H64DSP1718.m4a*. **Please note:** *H64DSP1718.m4a* is provided for your reference – information from this file must not be used to process the signal in *corrupt1718.m4a*.

You can use a variety of methods to process the signal. Digital filters are introduced in Lectures 5 and 6 but you are welcome to read ahead and use some of these filters earlier if you want to.

**3. Deliverables**

You should produce a report, approximately 10-15 pages in length, which should be word processed. See below for more details of the format of the report.

Submission of the coursework is **online** using the **Moodle** page for **H64DSP/H64DS2**.

You should submit your **report** and **MATLAB program files** in a single \*zip file. Please use the Windows \*zip facility and not any other compression program (such as \*rar) as I will not be able to open these files.

The report should be saved as a **PDF file**. **Please do not submit any data files as these will be very large** – I should be able to reproduce the signals you have obtained by running your programs.

The submission deadline is:

**Monday, December 11th 2017 before 3 pm**

**4. Report**

The report is to be structured as follows:

*(1) Introduction*

What are the aims and objectives of the project?

Which methods were used in your project?

Give recommended method with reasons for this recommendation

*(2) Methods and Results*

This is the main section of the report and should describe the following:

* The signal processing methods you have used and how you have implemented them using MATLAB.
* The reason why you have used each particular method. Justification for using specific choice of input parameters.
* Results shown in graphical form
* Quantitative comparison of processed and clean data

*(3) Discussion of Results*

* If a particular combination of signal processing methods worked well, then why is that the case?
* If a particular combination of signal processing methods worked badly, then why is that the case?

*(4) Conclusions*

Recommendation for the best combination of signal processing methods to be used and the reasons you are recommending this combination.

**5. Assessment**

The assessment of the report will be as follows:

***(1) Introduction – 10%***

First Class (M Eng)/Distinction (M Sc)

Aims and detailed objectives given. List of methods that have been used. Recommended method and brief reasons why this method worked better than the others.

2(i) (M Eng)/Merit (M Sc)

Aims and objectives given. List of methods that have been used. Recommended method given but no reasons why this method worked better than others.

2(ii) (M Eng)/Pass (M Sc)

Aims and objectives given. List of methods used not given or recommended method not given.

Third Class (M Eng)/Soft Fail (M Sc)

Aims and objectives given. List of methods used not given and recommended method not given.

Fail (M Eng)/Hard Fail (M Sc)

Just a few sentences lacking any detail required for higher assessments.

***(2) Methods and Results – 40%***

First Class (M Eng)/Distinction (M Sc)

Comprehensive set of signal processing methods applied, going beyond what was covered in class.

Detailed explanation of why you have selected each signal processing method and why this method should be good at reducing the effects of noise and interference.

Details given of input parameters for each method and detailed justification for the choice of these parameters.

Clearly presented results that show to the reader the relative merits of each method that you have used.

Excellent choice of figures of merit to compare processed data with clean data.

2(i) (M Eng)/Merit (M Sc)

Several signal processing methods applied, a few methods beyond what was covered in class.

Fairly detailed explanation of why you have selected each signal processing method and why this method should be good at reducing the effects of noise and interference.

Details sometimes given of input parameters for each method and some justification for the choice of these parameters.

For the most part: clearly presented results that show to the reader the relative merits of each method that you have used.

Good choice of figures of merit to compare processed data with clean data.

2(ii) (M Eng)/Pass (M Sc)

Reasonable set of signal processing methods used, but not going beyond what was covered in class.

Some justification of the choice of signal processing methods used.

Input parameters missing

Results presented in graphical form but may not be clear to the reader in some cases.

Fair choice of Figures of Merit to compare processed with clean data.

Third Class (M Eng)/Soft Fail (M Sc)

A few methods chosen to process data based on what was covered in class.

No justification for choosing the methods used.

Input parameters missing

Results sometimes presented clearly

Just visual comparison used between processed and clean data

Fail (M Eng)/Hard Fail (M Sc)

Just one or two methods chosen to process data

No justification for choosing the methods.

Input parameters missing

Results presentation not clear

No comparison between processed and clean data

***(3) Discussion of Results – 40%***

First Class (M Eng)/Distinction (M Sc)

Detailed reasons given as to why each particular method or combination of methods works well or works badly.

2(i) (M Eng)/Merit (M Sc)

Some detailed reasons given as to why each particular method or combination of methods works well or works badly.

2(ii) (M Eng)/Pass (M Sc)

Some reasons given, with limited depth, as to why each particular method or combination of methods works well or works badly.

Third Class (M Eng)/Soft Fail (M Sc)

A few reasons given as to why a particular method or combination of methods works well or works badly.

Fail (M Eng)/Hard Fail (M Sc)

Very few if any reasons given as to why a particular method works well or does not work. Just a summary of the results given.

***(4) Conclusions – 10%***

First Class (M Eng)/Distinction (M Sc)

Recommendation for the best combination of signal processing methods to be used and detailed justification why you are recommending this combination. Optimal input parameters for the recommended method are given.

2(i) (M Eng)/Merit (M Sc)

Recommendation for the best combination of signal processing methods to be used and some justification for recommending this combination. Optimal input parameters for the recommended method are given.

2(ii) (M Eng)/Pass (M Sc)

Recommendation for the best combination of signal processing methods to be used with a few reasons given for recommending this combination. Optimal input parameters for the method may or may not be given.

Third Class (M Eng)/Soft Fail (M Sc)

Optimal method for processing data given with no reasons or no input parameters given.

Fail (M Eng)/Hard Fail (M Sc)

A few sentences given but with no recommendations given and no input parameters for these methods stated.

*Note on assessment:*

You will not be penalised for trying out a method that does not work – this happens all the time with research! You will be given credit for explaining why a method does not work (or works well). You will not receive much credit if you present a lot of results with no explanation. Try using a variety of combinations of processing methods comparing the advantages and disadvantages of each combination.

**6. Connection with the Lectures**

Many of the signal processing methods introduced in the lectures will be relevant to the coursework. This project should help you in the understanding of these methods.

The following is a suggested schedule for the project, for you to be able to apply the methods discussed in the lectures:

*Weeks 1 and 2*

Using MATLAB compare the clean and corrupted signals. Can you identify the various types of corruption that have been caused to the original signal?

You should use the first two sessions to get used to MATLAB and writing simple programs to display the signals – use the **plot** command to plot the corrupted and clean signals as a function of time. What features do you note in the noisy signal that are not present in the clean signal? How could you get rid of them?

*Week 3*

In the third lecture, we look at the applications of the Discrete Fourier Transform (DFT) and its efficient implementation, the Fast Fourier Transform (FFT). How does the FFT amplitude spectrum of the corrupted signal compare with that of the clean signal? Are there any features there that can be attributed to noise and interference?

If you would like to start processing the data, you may consider first preprocessing the data using the *median filter* before applying any other digital signal processing methods.

Further details of the median filter can be found at:

<http://fourier.eng.hmc.edu/e161/lectures/smooth_sharpen/node2.html>

and

<https://uk.mathworks.com/help/signal/ref/medfilt1.html>

(both last accessed September 17th 2017)

Note, any median filters must be applied before any other method used; the reason for this will be explained in Lecture 4.

*Week 4*

In the lectures that week, we look at techniques such as median filtering and Wiener filtering. Could either, or a combination, of these techniques be applied to processing the corrupted signal?

*Weeks 5 and 6*

For these lectures we discuss the various types of digital filter (FIR and IIR) and describe various ways to design these filters. The appropriate MATLAB commands are given in the lecture notes. Can these be used to reduce the effects of noise and any interference?

*Weeks 7 and 8*

During these two weeks, we discussed adaptive filters, which are digital filters with time varying coefficients. Can any of the interferences be reduced with adaptive filters?

*Weeks 9 and 10*

During these weeks you should finish off your report for submission by the deadline.

**7. Other useful MATLAB Commands**

During the course of the lectures, various MATLAB commands will be noted in the lectures – try reading up on these in the Mathworks web site to see if they could be useful (**sptool** could be particularly useful):

**Spectral Analysis**

Demonstrations: **fftdemo**

• DFT: **fft ifft**

• Windows: **bartlett triang hanning hamming blackman chebwin kaiser**

• JTFA: **spectrogram**

• PSD (Welch) : **pwelch** (used to be called **psd**)

**Digital Filtering**

• **fir1** truncation +weighting for common filters (LP etc)

• **fir2** truncation + weighting for a sampled magnitude response

• **firls** least square design

• **firpm** chebyshev design

• Filter properties **grpdelay freqz impz zplane**

• **conv filter fftfilt** (to implement filtering)

• **sptool** (versatile tool for design, filtering, analysis)

**8. Help with project**

The methods you use are entirely up to you. Please feel free to discuss any ideas you may have with me or if you require more help. I would be interested in listening to your ideas. You can come to my office if you like or you may wish to check my availability by email first.

Remember – there are a large number of algorithms that you could come up with each with their own advantages and disadvantages.

*Malcolm Woolfson, September 18th 2017*