Laboratory for week 7:

Digital Signals & Systems

Introduction

In order to process an acoustic signal on a computer the **continuous analogue waveform** has to be converted into a **discrete digital waveform**. 'Analogue' means that at any point in time, a value of the wave is available; 'discrete' means that only a limited number of values are available over time. In other words: a digital signal consists of a limited number of samples per second taken from the analogue signal. This number of samples is called the **sampling rate**. The standard sampling rate for CD quality is 44100 samples/second.

But a digital signal has to be discrete not only on the time axis, but also on the amplitude axis. Only a limited number of amplitude values is possible in a digital system. This discreteness is referred to as the **quantisation** level. The current quantisation standard is 16 bit, which means that the whole range of the amplitude is divided into 65536 possible values ($2^{16} = 65536$). Lower quantisation levels cause again changes to the signal shape and thus introduce harmonic distortion to the signal.

Scientific Objectives

- To get acquainted with the effects of changes in quantisation level to the signal.
- To see how systems can be simulated digitally.

Learning Objectives

- To learn using the software Praat
- To learn the effects of varying the number of quantisation level.
- To gain familiarity with processing signals on a computer.
- To gain familiarity with creating systems on a computer.

Apparatus

You will use the software Praat on the computers in the lab. It is downloadable for free from the internet under www.praat.org.

(You will work with two options in Praat that have especially been created for this lab ('Create Sine wave' and 'Re quantise') and are therefore not included in the Praat version you can download from the internet. If you are interested in having these two extra options for your home use then please talk to Volker.)

1.) Quantisation of a sine wave

Using 'Create Sine wave...': For the following table create a sinusoid of 200 Hz, and 0° phase with a sampling rate of 22050 samp./sec. in any case and vary the number of bits according to the values in the table. Please do this for the amplitude values of 1mV and ½ mV. Then:

- Count the number of quantistion levels (For 8 bit and higher it will be difficult, actually impossible, to count the quantisation levels but maybe you can calculate them!)
- Measure the intensity of the fundamental frequency in the spectrum in the 'Praat objects' window (not in the 'Praat picture' window!). By clicking on the respective peak values in the spectrum you will be able to obtain their intensity in dB (in red) to the left of the spectrum.
- Measure the amount of harmonic distortion in dB.

	amplitude = 1 mV	
number of bits	quantisation levels	harmonic distortion (in dB)
1		
2		
4		
8		
12		
16		

Questions:

1.) How does harmonic distortion develop as a function of quantisation levels? Why?

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2.) Quantisation of real speech

So far so good: We have seen that the level of quantisation can actually change the shape of a sine wave significantly and thus introduce strong harmonic distortion, highest in case of the 1 bit example. But, as you may already know, sine waves are a rather artificial concept and they almost never occur in the real world.

With complex waveforms, low quantisation levels introduce another source of error: since many samples lying on the border between quantisation levels randomly fall into the higher or lower quantisation level, a low amplitude noise is produced that is called the **quantisation noise**.

In the next section, you will re-quantise a real speech example by changing the number of bits in its representation. You do that by the use of the option 'Re quantise…' under 'New' in the 'Praat objects' window. Apart from the number of bits you do not need to change any of the settings in this window. Create a version of the test file for each quantisation level possible and answer the following questions after you have listened to your examples:

- At what quantisation level does speech become non-intelligible? Would you have expected the result you came up with?

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- At what levels of quantisation are you able to perceive quantisation noise in the signal?

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3.) Digital Systems

So far we, we have only dealt with digital signals. If we modify the values of a discrete signal by some function or formula or even by hand, we can change the signal shape and thus act as a kind of a system. With this technique we can create digital imitations of real systems, for example, filters. And even more: since we have control over every single sample of the signal we can control our digital systems much better than any analogue system. This is one of the main reasons why digital techniques are nowadays so commonly used in signal processing.

In the following section, you will work with the ESystems software that you have already met before. Start the software from the same directory from which you started Praat. In the ESystems menu, you will find the 'Tutorial' under 'Help'. Please open it and work through section 5. 'Introduction to digital systems'. Only the brave will attempt **section e** on convolution!