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MAS.160 / MAS.510 / MAS.511 Signals, Systems and Information for Media Technology
Fall 2007

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MAS 160/510 Problem Set One

Due in class Wednesday September 12, 2007

1. Periodicity

Determine the fundamental period of each of the following signals:

- (a) $x_a(t) = \cos(4\pi t)$
- (b) $x_a(t) = 10 \cos(3t + \frac{\pi}{16})$
- (c) $x[n] = \cos(0.04\pi n)$
- (d) $x[n] = \cos(5\pi n) + \cos(\frac{4}{5}\pi n)$

2. Sampling

- (a) Let T_s , the sampling period for a continuous time signal, be $3/2$ and let $x(t) = \cos(2\pi t)$. Sketch $x[n] = x(nT_s)$ for $n = 0, 1, \dots, 8$.
- (b) Let $T_s = 1/3$ and let $x(t) = \cos(2\pi t)$. Sketch $x[n] = x(nT_s)$ for $n = 0, 1, \dots, 8$.

3. Review of Complex Numbers(DSP First problems at the end of appendix A)

- (a) A.3(b)
- (b) A.3(c)
- (c) A.4(a) to (e)
- (d) A.8

4. Systems and Block Diagrams

The video and audio components of the DVD are stored in a compressed format (MPEG-2 for video and Dolby Digital for audio). These compression processes can only take in digital signals as inputs. Assume that all our inputs (e.g. film, audio tapes, etc.) and outputs (e.g. TV, speakers, etc.) are analog signals. Draw the block diagrams of the audio and video systems necessary for DVD creation and playback.

You are given the following components:

- (a) Lots of A-to-D converters
- (b) Lots of D-to-A converters
- (c) an MPEG-2 encoder
- (d) an MPEG-2 decoder
- (e) a Dolby Digital encoder
- (f) a Dolby Digital decoder
- (g) an Optical Disc Writer
- (h) an Optical Disc Reader

5. **Matlab: Introduction** For those of you using MATLAB the first time here are some instructions on how to start:

(a) Find Matlab

- Go to any Athena Cluster.
- At the Athena prompt, type **add matlab** to setup MATLAB locker in your Athena environment (you only need to do this the first time you use MATLAB).

OR

- Login to any media lab unix machine

(b) Start Matlab

- type `matlab` at the prompt

this should initiate the Matlab shell, and display a prompt like below:

>>

(c) Use Matlab

The following steps will guide you through the Matlab exercises (Note the % indicates a comment and need not be entered.) At the Matlab prompt, type

```
x = [1:1:140]; % sets up the domain
yCarrier = cos(pi*x/8); % create a sinusoid to act as a carrier
plot(x,yCarrier); % and plot it
ySignal = rand(size(yCarrier)); % create a signal to be transmitted
plot(x,ySignal); % and plot that too
ySum = yCarrier + ySignal; % create an average signal
yDiff = yCarrier - ySignal; % create a difference signal
```

(d) Using only linear combinations of the `ySum` and `yDiff` signals, recover the `ySignal` and plot it.

The following MATLAB exercises (found in Appendix C of the *DSP First* text) should be treated as walkthrough tutorials to important concepts. You can ignore any references to *instructor verification* or a *lab report*. We will specify which items we would like turned in as part of the homework. However, it would be difficult to do only the parts that are to be turned in (i.e. it would be unwise to skip steps in the lab).

6. *DSP First* Lab 1

Items to be turned in:

- The `expand` function from C.1.2.6.
- The `replacez` function from C.1.2.7.
- Plots specified in C.1.3.1.

7. *DSP First* Lab 2

Items to be turned in:

- Your `sumcos` function and resulting plots from C.2.2.2.

- (b) Plots specified in C.2.3.2.
- (c) Plots and answers to questions specified in C.2.4.

8. Additional problem (for MAS.510): Listening to Sinusoids

We have seen many examples of sinusoids where amplitude, frequency, and phase remain constant. However, these parameters can be varied, often with interesting and surprising results. In this problem, you will use MATLAB to explore these possibilities.

Start with a basic sinusoid of the following form:

$$x(t) = A(t) \sin(f * 2\pi t + g(t)) \tag{1}$$

- (a) In MATLAB, use the equation above to create a constant amplitude ($A(t) = 1$) constant phase ($g(t) = 0$) 300 Hz tone lasting two seconds. Then alter its frequency so that it rises linearly from 300 Hz to 3000 Hz. Listen to the resulting signal using the MATLAB function `sound`. Plot enough of the signal so that the change in frequency is visible. This signal is called a *chirp*. (Note: use the `sound` command at the Matlab prompt. And when you are selecting a sampling rate try to use 8000 Hz, and see if your results sound better.)
- (b) Start again with the 300 Hz tone. This time, vary the *amplitude* using the following equation:

$$A(t) = \sin(N2\pi t) \tag{2}$$

Keep the phase constant (i.e. $g(t) = 0$). Vary N with a few values from 1 to 300 and listen to the result. How does the sound change with different values of N ? Plot enough of the signal so that the change to the signal is apparent.

- (c) Return to the constant amplitude signal ($A(t) = 1$). This time change the *phase* to be the following:

$$g(t) = \cos(M2\pi t) \tag{3}$$

Try several values of M between 5 and 500 and listen to the result. How does the sound change with different values of M ? Plot enough of the signal so that the change in the signal is apparent.

- (d) Oftentimes, the modifications themselves contain the signal of interest, while the original sinusoid becomes what is called the *carrier signal*, with frequency f . The changes applied in parts (b) and (c) are called *Amplitude Modulation* (AM) and *Frequency Modulation* (FM), respectively. Where else have you heard these terms? Give some examples of your favorite values of f .