ELEC 470 Communication and Information Systems Fall 2014

Review Topics for Exam #2

Please review the "Exam Policies" section of the Exams page at the course web site. Please especially note the following:

1. You will be allowed to use a non-wireless enabled calculator, such as a TI-99.
2. You will be allowed to use two 8.5 × 11-inch two-sided handwritten help sheets. No photocopied material or copied and pasted text or images are allowed. If there is a table or image from the textbook or some other source that you feel would be helpful during the exam, please notify me.
3. All help sheets will be collected at the end of the exam but will be returned to you later.

The following is a list of topics that could appear in one form or another on the exam. Not all of these topics will be covered, and it is possible that an exam problem could cover a detail not specifically listed here. However, this list has been made as comprehensive as possible. You should be familiar with the topics on the previous review sheet in addition to those listed below.

Although every effort has been made to ensure that there are no errors in this review sheet, some might nevertheless appear. The textbook is the final authority in all factual matters, unless errors have been specifically identified there. You are ultimately responsible for obtaining accurate information when preparing for your exam.

Generation of FM signals

- indirect method for NBFM
- Armstrong indirect method for WBFM
- direct method using variable capacitance or inductance in frequency-determining part of oscillator circuit

Demodulation of FM signals

- slope detection: most methods based on differentiating FM signal and then applying band-pass limiter
- can use PLL

Radio/wireless receiver architectures

- tuned radio frequency (TRF)
- homodyne (direct conversion)
    - "homodyne" refers to the local oscillator (LO) being at roughly the same frequency as the RF signal
    - translates RF signal frequency directly to baseband
    - results in double-signal reception unless the baseband image is eliminated via special circuits (like an image rejection mixer)
- superheterodyne
    - "heterodyne" refers to the LO being at a substantially different frequency than the RF signal
    - "super" literally refers to the LO frequency being higher than the incoming desired RF signal frequency. In common usage now, however, a superheterodyne receiver could have an LO either above or below the RF frequency.

- desired signal is downconverted (usually, although upconversion is sometimes used) to one or more intermediate frequencies (IFs)
- highly selective IF filters are practical since they do not have to be tuned
- amplifier gain in the receiver is distributed among RF, IF, and baseband frequencies, which aids in preserving amplifier stability (excessively high gain not required at a single frequency)
- image signal frequency is $2f_{\mathrm{IF1}}$ away from desired signal frequency (where f_{IF1} is the first IF frequency), so image rejection is easy to achieve
- message signal spectrum "flipping" can occur in the frequency translation process
- IF and LO frequencies usually selected to avoid heavily occupied sections of the electromagnetic spectrum
- upconversion architecture can simplify image rejection filtering and ease filter design challenges if coverage of very wide portions of the electromagnetic spectrum is desired

- image rejection mixers
  - uses phase shifters and additional frequency mixers to eliminate image signals (90° hybrids are used at microwave frequencies to provide phase shifts)
  - many architectures will work, but some are more practical to implement and/or are cheaper than others
  - image rejection mixers can be used with receiver front-end image filters to ease the performance requirements of both (i.e., some attenuation comes from filter and some from mixer; one device does not have to provide all of the required attenuation)
  - when analyzing image rejection mixer operation, make sure that the arguments of all sine/cosine functions have physical positive frequencies. For example, if $\omega_a > \omega_b$, change $\cos(\omega_b t - \omega_a t)$ to $\cos(\omega_a t - \omega_b t)$. This works because $\cos(-x) = \cos x$. If there are constant phase shifts in the arguments (like −90° or +180°), don't forget to change their algebraic signs as well.

Angle modulation and nonlinearities

- FM and PM signals can be amplified by efficient class C amplifiers (and other nonlinear "switching" amplifiers, like class D, E, F, … and Doherty amplifiers) without distorting the message signal. Amplitude information is unimportant in FM/PM.
- only necessary to preserve the number and spacing of zero crossings of the FM/PM waveform
- FM/PM also immune to effects of fading due to use of band-pass limiting

Angle modulation and immunity to interference (capture effect)

- if A = FM signal amplitude and $I \cos \omega t$ = interference signal at input of receiver, and $A \gg I$, signal strength of interference at output of *FM* detector is

$$y_d(t) = \frac{I\omega}{A}\sin \omega t$$

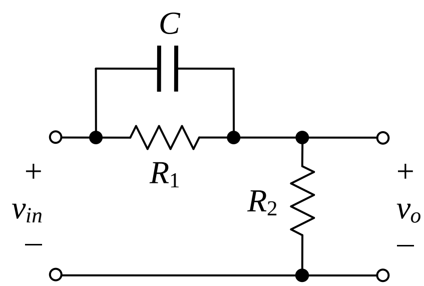
- if A = PM signal amplitude and $I \cos \omega t$ = interference signal, signal strength of interference at output of *PM* detector is

$$y_d(t) = \frac{I}{A}\sin \omega t$$

- signal strength of weak interference is suppressed in an FM/PM detector; thus, the desired signal "captures" the receiver output

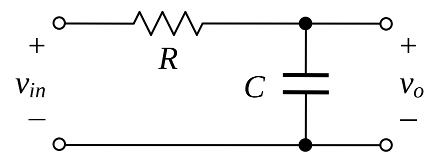
- noise can be treated as a type of interference, so in the case of white noise at input to receiver (see Fig. 5.14 on p. 286 of textbook),
    - noise spectral density at the output of an *FM* detector is directly proportional to frequency
    - noise spectral density at the output of a *PM* detector is uniform with frequency
- pre-emphasis in FM
    - amplifies high-frequency components of message signal in transmitter (before white noise is added to signal at the receiver's input port)
    - signal gain is proportional to frequency, just like noise response of FM demodulator
    - common pre-emphasis circuit, with Bode plot zero frequency f_1 and pole frequency f_2 defined by

$$f_1 = \frac{1}{2\pi R_1 C} \quad \text{and} \quad f_2 = \frac{1}{2\pi\left(R_1 \| R_2\right)C}$$



- de-emphasis in FM
    - attenuates output of FM demodulator in receiver as $1/f$ above first cut-off frequency f_1 (same f_1 as in pre-emphasis filter)
    - signal frequency response is flattened; high-frequency noise (which entered at the input of the receiver) is attenuated
    - counterpart to pre-emphasis in transmitter
    - common de-emphasis circuit, with pole frequency f_1 defined by

$$f_1 = \frac{1}{2\pi RC}$$



Direct digital synthesis (DDS)

- major parts of DDS system: tuning word register, phase accumulator, ROM containing sine look-up table (also called phase-to-amplitude converter), digital-to-analog converter (DAC), low-pass filter
- output frequency f_o is given by

$$f_o = M\,\frac{f_c}{2^n}$$

  where f_c = clock frequency, M = binary tuning word (ROM address increment), and n = no. of bits in ROM address
- in theory, output frequency can be as high as $f_c/2$ (Nyquist limit), but in practice the upper limit is much lower than $f_c/2$ because ideal low-pass filters are not available to apply to output of DAC

- binary sine wave amplitude representation in ROM look-up table is sometimes truncated to N bits, where $N < n$
- a primary disadvantage of DDS is the generation of quantization noise, which is inversely proportional to the number of quantization levels squared L^2 (where $L = 2^N$); however, quantization noise can be made arbitrarily small by making N (and L) large enough.
- hybrid frequency synthesizers can combine DDS and PLL (e.g., DDS can supply a tunable reference frequency signal to PLL)

<u>Relevant course material:</u>

Homework:	#3, #4
Mini-Projects:	#1
Textbook:	Sections 5.3-5.7
Supplements:	“Mixer Circuits and Image Frequencies”
	“Receiver Design,” from D. Pozar, *Microwave and Wireless Systems*
	“Fundamentals of Direct Digital Synthesis (DDS),” by Analog Devices, Inc.
Web Links:	(none)
Matlab:	(none)