Solution of ECE 316 Final Examination Su09

1. The communication system technique that allows multiple transmitters to operate at the same time and also allows a receiver to select and listen to only one transmisison is called

   Frequency multiplexing

2. Why is DSBTC modulation used by commercial AM radio stations instead of DSBSC?

   DSBTC allows the use of an envelope detector for demodulation which is generally much simpler and cheaper to design than the synchronous demodulation methods required for DSBSC.

3. What advantage does SSBSC modulation have compared with DSBSC modulation?

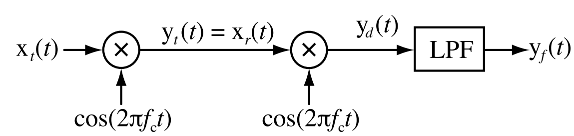
   SSBSC uses half the bandwidth of DSBSC for the transmitted signal.

4. What are the two most common types of angle modulation?

   Phase Modulation

   Frequency Modulation

5. In the system below $\mathrm{x}_t(t) = \cos(100\pi t)$, $f_c = 10\ \text{kHz}$ and the cutoff frequency of the unity-gain, ideal lowpass filter is 100 Hz. What are the signal powers of $\mathrm{y}_t(t) = \mathrm{x}_r(t)$, $\mathrm{y}_d(t)$ and $\mathrm{y}_f(t)$?



$$\mathrm{y}_t(t) = \mathrm{x}_r(t) = \cos(100\pi t)\cos(20{,}000\pi t) = (1/2)\left[\cos(19{,}900\pi t) + \cos(20{,}100\pi t)\right]$$

$$P_{\mathrm{y}_t(t)=\mathrm{x}_r(t)} = \frac{(1/2)^2}{2} + \frac{(1/2)^2}{2} = 1/4$$

$$\mathrm{y}_d(t) = (1/2)\left[\cos(19{,}900\pi t) + \cos(20{,}100\pi t)\right]\cos(20{,}000\pi t)$$

$$\mathrm{y}_d(t) = (1/4)\left[\cos(100\pi t) + \cos(39{,}900\pi t) + \cos(100\pi t) + \cos(40{,}100\pi t)\right]$$

$$\mathrm{y}_d(t) = (1/4)\left[2\cos(100\pi t) + \cos(39{,}900\pi t) + \cos(40{,}100\pi t)\right]$$

$$P_{\mathrm{y}_d(t)} = \frac{(1/2)^2}{2} + \frac{(1/4)^2}{2} + \frac{(1/4)^2}{2} = 1/8 + 1/32 + 1/32 = 3/16$$

$$\mathrm{y}_f(t) = (1/2)\cos(100\pi t) \Rightarrow P_{\mathrm{y}_f(t)} = \frac{(1/2)^2}{2} = 1/8$$

6. List the magnitudes and angles of the s-plane pole locations for a normalized 5th-order lowpass analog Butterworth filter.

All pole magnitudes are 1. The angle between poles is $\pi / 5$ radians. One pole is on the negative σ axis and the others occur in complex-conjugate pairs. Therefore the magnitudes and angles are

Magnitudes	Angles
1	π
1	$4\pi / 5$
1	$-4\pi / 5$
1	$3\pi / 5$
1	$-3\pi / 5$

7. A lowpass filter has a transfer function $\mathrm{H}(s)$. A new filter is designed by replacing s by ω_c / s. What kind of filter is the new filter?

The new filter is highpass.

8. A lowpass digital filter is designed first using the impulse invariant technique and then using the step invariant technique. Which one is guaranteed to have the correct response magnitude at zero frequency and why?

The step invariant technique will have the correct gain at zero frequency because it reproduces samples of the step response and the final value is correct.

9. A digital filter is designed using direct substitution to approximate an analog filter with poles at $s = -10$ and $s = -5 \pm j8$ and a zero at $s = 0$. If the sampling rate is 20 Hz, where are the digital filter's poles and zeros in the z plane? (Numerical locations.)

$$s = -10 \Rightarrow z = e^{-10/20} = 0.6065$$
$$s = -5 \pm j8 \Rightarrow z = e^{(-5 \pm j8)/20} = 0.7788\angle \pm 0.4$$
$$s = 0 \Rightarrow z = e^{0} = 1$$

10. A first-order lowpass analog Butterworth filter with a corner frequency of 200 Hz is approximated by a digital filter using the bilinear technique. The sampling rate is 2000 samples/second. Where are the poles and zeros of the digital filter in the z plane?

$$\mathrm{H}(s) = \left[\frac{1}{s+1}\right]_{s \to s/400\pi} = \frac{1}{s/400\pi + 1} = \frac{400\pi}{s + 400\pi}$$

$$\mathrm{H}(z) = \left[\mathrm{H}(s)\right]_{s \to \frac{2}{T_s}\frac{z-1}{z+1}} = \left[\frac{400\pi}{s + 400\pi}\right]_{s \to \frac{2}{T_s}\frac{z-1}{z+1}} = \frac{400\pi}{\frac{2}{T_s}\frac{z-1}{z+1} + 400\pi}$$

$$\mathrm{H}(z) = \frac{400\pi(z+1)/2000}{2z - 2 + 400\pi(z+1)/2000} = \frac{0.6283(z+1)}{2.6283z - 1.372} = 0.2391\frac{z+1}{z - 0.522}$$

Pole at $z = 0.522$ and zero at $z = -1$.

11. In designing a digital filter to approximate an analog lowpass filter, which design technique guarantees that the digital filter's magnitude response is zero at $\Omega = \pm\pi$?

The bilinear technique.

12. Which type of digital filter can be designed to have a linear phase shift in its passband?

    FIR

13. Which type of digital filter is guaranteed stable regardless of the sampling rate or the coefficient values?

    FIR

14. Generally, in order to design a digital filter with a certain passband gain, transition bandwidth and stop-band attenuation, which type of digital filter accomplishes the design goal with fewer multiplications and additions?

    IIR

15. What is a disadvantage of finite-difference digital filter design using forward differences?

    It can design an unstable digital filter as an approximation to a stable analog filter.

16. Of the three window types, Rectangular, von Hann and Blackman, if they are all used to design the same type of digital FIR filter with the same number of samples in the impulse response

    (a) Which one yields the narrowest transition from passband to stopband?

        Rectangular

    (b) Which one yields the greatest stopband attenuation?

        Blackman

17. An ideal operational amplifier is connected in the inverting amplifier configuration. The impedance $Z_f(s)$ in the feedback path consists of a resistor and capacitor in series and the input impedance $Z_i(s)$ is a single resistor. The transfer function is the ratio of the output voltage to the input voltage.

    (a) What is the numerical slope of a magnitude Bode diagram of the frequency response at very low frequencies (approaching zero)?

        +20 dB/decade

    (b) What is the numerical slope of a magnitude Bode diagram of the frequency response at very high frequencies (approaching infinity)?

        0 dB/decade

18. If a filter design requires a monotonic magnitude frequency response in the passband which type or types of filter should be used?

    Butterworth and Chebyshev Type 2

19. If the most important aspect of a filter design is that the width of the transition between passband and stopband be as small as possible which type or types of filter should be used?

    Elliptic

20. The CTFT of a continuous-time signal has impulses at ± 3 kHz , ± 7 kHz and ± 8 kHz . It is sampled at 10 kHz to form a discrete-time signal $\mathrm{x}[n]$. The DTFT of $\mathrm{x}[n]$ is $\mathrm{X}(e^{j\Omega})$. At what numerical values of Ω in the range $0 \le \Omega \le \pi$ is $\mathrm{X}(e^{j\Omega})$ non-zero?

In the DTFT $F = f / f_s$ and $\Omega = 2\pi F$. Therefore $\Omega = 2\pi f / f_s = \omega / f_s$. Also $\mathrm{X}(e^{j\Omega})$ repeats periodically in Ω with period 2π . So the direct non-zero locations for $\mathrm{X}(e^{j\Omega})$ are $\pm 0.6\pi$, $\pm 1.4\pi$ and $\pm 1.6\pi$ and all of these repeat with period 2π . Therefore the impulses in $\mathrm{X}(e^{j\Omega})$ are at

$\pm 0.6\pi$, $\mp 1.4\pi$, $\pm 2.6\pi$, $\mp 3.4\pi$, $\pm 4.6\pi$ $\cdots$
$\pm 1.4\pi$, $\mp 0.6\pi$, $\pm 3.4\pi$, $\mp 2.6\pi$, $\pm 5.4\pi$ $\cdots$
$\pm 1.6\pi$, $\mp 0.4\pi$, $\pm 3.6\pi$, $\mp 2.4\pi$, $\pm 5.6\pi$ $\cdots$

Of these, these ones that fall in the range $0 \le \Omega \le \pi$ are 0.6π and 0.4π .