Midterm Examination $\#2$ Nov. 21, 2007

MIDTERM #2 SOLUTIONS

1. (a) With digital modulation, the transmitted signal $w_z(t)$ obeys

$$
w_z(t) = \sum_n s[n]p(t - nT).
$$

The values $[s[0], s[1], s[2], s[3] = [3, 1, -1, 3]$, yield $w_z(t)$ illustrated below for $T_p =$ $T/2$ (left) and $T_p = T$ (right).

(b) Recall that convolving two symmetric rectangles $p(t)$ of width T_p yields a symmetric triangle $c(t)$ of width $2T_p$ and height $\int_{-\infty}^{\infty} |p(t)|^2 dt$. Here, $\int_{-\infty}^{\infty} |p(t)|^2 dt =$ $\int_{-T_p/2}^{T_p/2} \left| \frac{1}{\sqrt{2}} \right|$ $\frac{1}{T_p}$ $^{2}dt = 1$. Thus, we get the following:

$$
c(t) = \begin{cases} \frac{t+T_p}{T_p} & t \in [-T_p, 0) \\ \frac{T_p-t}{T_p} & t \in [0, T_p) \\ 0 & \text{else} \end{cases} \qquad \qquad \begin{cases} c(t) & s[n] \rightarrow [c(t)] \end{cases} \qquad y(t) \rightarrow y[k] \\ \text{else} \qquad \qquad \begin{cases} \frac{s[n] \rightarrow [c(t)] \end{cases} \qquad y(t) \rightarrow y[k]}{\text{if } \text{tr}(t) \text{ or } \text{tr}(t) \text{ is } T} \end{cases}
$$

(c) With a noiseless trivial channel, the received signal obeys

$$
y(t) = \sum_{n} s[n]c(t - nT).
$$

The values $[s[0], s[1], s[2], s[3] = [3, 1, -1, 3]$, yield $y(t)$ illustrated below for $T_p =$ $T/2$ (left) and $T_p = T$ (right). The individual contributions of each symbol are depicted using dashed lines.

(d) The Nyquist criterion can be stated as $c(mT) = \begin{cases} 1 & m = 0 \\ 0 & k \end{cases}$ $0 \quad m \neq 0$ for integer m. As evident from the figure below, the Nyquist criterion is satisfied for any $T_p \in (0, T]$:

2. (a) The SRRC frequency response $P_{\text{src}}(f)$ is sketched below for various α .

(b) The RC frequency response $C_{\text{rc}}(f)$ is sketched below for various α .

(c) From the homework, we know $S_y(f) = |P_{\text{src}}(f)|^2 S_n(f)$. Since $P_{\text{src}}(f) \in \mathbb{R}$, we have

$$
S_y(f) = P_{\text{src}}(f)^2 N_0
$$

=
$$
C_{\text{rc}}(f) N_0.
$$

3. (a) Because $2e^{-j2\pi (f_c+f_0)t} = 2e^{-j2\pi f_c t} \cdot e^{-j2\pi f_0 t}$, we just need to put $\boxed{e^{-j2\pi f_0 t}}$ in the box.

(b) The diagram on the left implies

$$
x(t) = [u(t)e^{-j2\pi f_0 t}] * f(t)
$$

\n
$$
= \int_{-\infty}^{\infty} u(\tau)e^{-j2\pi f_0 \tau} f(t-\tau)d\tau
$$

\n
$$
= \int_{-\infty}^{\infty} u(\tau)f(t-\tau)e^{j2\pi f_0(t-\tau)}d\tau \cdot e^{-j2\pi f_0 t}
$$

\n
$$
= [u(t) * [f(t)e^{j2\pi f_0 t}]]e^{-j2\pi f_0 t}
$$

which can be interpreted as " $u(t)$ gets convolved with $\overline{f(t)e^{j2\pi f_0 t}}$, and the result is modulated by $e^{-j2\pi f_0 t}$." Thus, the impulse response $|f(t)e^{j2\pi f_0 t}|$ goes in the box.

(c) Since $\mathcal{F}{f(t)e^{j2\pi f_0 t}} = F(f - f_0)$, the modulation acts to shift the passband of the LPF right by f_0 Hz:

The effect of this modulation on up/down-conversion can be understood by recalling the two key roles of the LPF: (i) suppressing the undesired signal component that exists at $f \in [-2f_c-B, -2f_c+B]$ Hz and (ii) preserving the desired signal component that exists at $f \in [-B, B]$ Hz.

- i. To suppress the unwanted component, we need $-2f_c + B \leq -B_s + f_0$, or equivalently $f_0 \geq B_s + B - 2f_c$.
- ii. To preserve the desired component, we need $-B_p + f_0 \leq -B$ and $B_p + f_0 \geq B$, or equivalently $-B_p + B \le f_0 \le B_p - B$ or equivalently $||f_0|| \le B_p - B$.
- 4. (a) $h(t)$ can be replaced with $h_p(t)$ because these filters are identical at all frequencies occupied by the input signal (i.e., $f \in [-f_c - B, -f_c + B] \cup [f_c - B, f_c + B]$).
	- (b) As in problem 3(a), we can write $2e^{-j2\pi (f_c+f_0)t} = 2e^{-j2\pi f_c t} \cdot e^{-j2\pi f_0 t}$.
	- (c) Problem 3(b) confirms that this is the correct way to reverse the order of filtering and modulation.
	- (d) We can always swap the ordering of linear time-invariant operations.
	- (e) Problem $3(c)$ shows that up/down-conversion is transparent even when the LPF is modulated by f_0 Hz, provided that $|f_0| \leq B_p - B$ and $f_0 \geq B_s + B - 2f_c$.
	- (f) $h_p(t)e^{-j2\pi f_c t}$ can be replaced with $h_z(t)$ because these filters are identical at all frequencies occupied by the input signal (i.e., $f \in [-B, B]$).