REMARK: A 10% percent “per day” deduction will be applied to each report handed in after the due date and time.

Modeling a Double Sideband Suppressed Carrier Communications System for Underwater Acoustic Communications

The main objective of this laboratory is to model a double sideband suppressed carrier (DSB-SC) communications system for the transmission and reception of speech signals in an underwater medium. The maximum frequency content of a speech signal is assumed to be $F_V = 4000\text{Hz}$ and the “one-sided” channel bandwidth is said to be $B = 16000\text{Hz}$. A receiver must be designed to recover a signal $x_r(t) \approx s(t)$ (see Fig. 3).

The channel itself is modeled as a composition of two basic systems, an ideal, linear phase, low-pass filter $T_C$, with cut-off frequency $F_L = 16000\text{Hz}$, and a signal summing system, $T_N$, where the channel underwater noise signal, $n(t)$, is added to the output of the linear phase filter $T_C$. This DSB-SC communications channel is given the name $T_{CN} = T_{AM}$. Thus, it is given by cascading $T_C$ and $T_N$ (see Fig. 2).

The overall sampling frequency of the DSB-SC communications system must be set to a value greater than twice the maximum frequency (Nyquist-Shannon sampling theorem) content of the output of the demodulator at the receiver side. That is, $F_S > 2(f_c + F_V)$, where $f_c$ is the carrier frequency of the modulator (demodulator) and must be $8000 < f_c < 12000$. The input signal $x_m(t)$ to the modulator is the sum of a “wanted” speech signal $s(t)$ and an “unwanted” interference signal $g(t)$. After the demodulator, an ideal, linear phase, low-pass filter, $T_L$ is used, with cut-off frequency $F_M = 4000\text{Hz}$, to recover the “wanted” speech signal $s(t)$ (see Fig. 1).
PROJECT TASKS:
Each student group must perform the following tasks. Each task is worth **20 points** for a total of **100 points**.

1. **Design** an ideal, linear phase, band-pass filter to eliminate the input interference signal $g(t)$.

2. **Design** an ideal, linear phase, low-pass filter to model the channel filter $T_C$.

3. **Modify** the amplitude of the noise signal $n(t)$ and describe observed changes at the receiver output.

4. **Design** an ideal, linear phase, band-pass filter $T_R$ to reduce receiver noise from the input signal $y_{co}(t)$.

5. **Design** a linear phase, low-pass filter, with cut-off frequency $F_V = 4000\,Hz$ for the receiver’s filter $T_L$.

Hand in a formal report in **.pdf** format, with input and output plots, in time and frequency (magnitude only for the frequency plots) of every subsystem of the system.