• Many signals are present in the frequency range $0 - X$ Hz. For instance, when human speech is converted to an electrical signal with a microphone, the signals is mostly present in the $0 - 3.5$ KHz range. An analog video signal (following the NTSC standard) is present in the $0 - 4.3$ MHz range. These signals that extend all the way from 0 Hz (or close to 0 Hz) to $X$ Hz are called “baseband signals”. Clearly, baseband signals can overlap, and cannot be transmitted over the same channel as they are. From linearity of the Fourier Transform, the overlapping frequencies will add up, making it difficult to separate. Hence, we need a way to change the frequency of a signal. One such process is “modulation”, where one signal is used to change the characteristics of another (Section 4.1).

• If we choose to change the characteristics of orthogonal signals, for example, we can add up these signals and then separate them out without trouble. For instance, we could use sinusoidal signals of different frequencies and separate them using filters.

• We can ”modulate” a signal in pretty much two ways: change the amplitude, or change the frequency. Equivalently, we can think of changing the frequency as changing the phase, because the two are equivalent. We will first concentrate on Amplitude Modulation (AM).

• The data or message to be transmitted is usually called the “message signal”. When the message signal is multiplied by a cosine signal of frequency $f_c$, in the frequency domain, this is equivalent to shifting the message signal by $f_c$ (from the modulation property of the Fourier Transform). This cosine signal is usually called a ”carrier”, denoted by $c(t)$.

• Section 4.2 explains the math behind multiplying a signal with a carrier, and depicts what happens in the frequency domain. If the message in the frequency domain, $M(f)$, occupies frequencies in the range $0 - B$ Hz, it will occupy a bandwidth of $2B$ around $f_c$, when it is multiplied with $c(t)$. The left and right side of the spectrum about $f_c$ is called lower and upper sideband, respectively. Because we get these two upper and lower sidebands, this form of modulation is called “Double Sideband Suppressed Carrier” AM (DSB-SC AM).

• We can take a modulated signal $m(t)c(t)$, and then multiply it with $c(t)$ again. This will yield $m(t)c^2(t) = m(t)/2 + m(t)\cos(4\pi f_c t)/2$. The $m(t)/2$ term is basically the original message signal scaled by a constant, and the $m(t)\cos(4\pi f_c t)/2$ term is the message signal shifted to the frequency $2f_c$. If we use a low pass filter to remove frequencies beyond $B$, we can recover $m(t)$. This process is called ”demodulation”. Clearly, we need $f_c > B$ for this to work, or else the signal term would overlap in frequency with the $m(t)\cos(4\pi f_c t)/2$ term. In practice, $f_c >> B$ is preferred. Since we use the same signal we used as a carrier for modulation in demodulation also, this is called “synchronous” or ”coherent” demodulation.

• Both the above modulation and demodulation systems can be build with multipliers, which can be thought of as variable gain amplifiers. Using advances in semiconductor technology, we can build amplifiers that switch at very high speeds, which makes high speed communication possible.