ENGR 76: Information Science and Engineering Spring 2026 — Lectures 11-12

Digital Communication, Modulation, and Demodulation

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During the first half of this course, we focused on the representation of information. We established the mathematical foundations for converting physical signals into bit sequences: starting with entropy and lossless compression for discrete data, and moving to sampling theory to define the conditions under which a continuous-time signal can be mapped to discrete samples without loss. We explored quantization as the mechanism for mapping real-valued samples to a discrete alphabet introducing an unavoidable source of distortion, and utilized frequency-domain representations, such as the Fourier series, to efficiently approximate signals like natural images by capturing their underlying smoothness with low-frequency sinusoids. Together, these tools provide a comprehensive framework for both the lossy and lossless digitization of the physical world.

The second half of the course shifts from representation to communication. Once information is converted into a bit sequence, it must be transported across space or time. This requires a communication channel: a physical mechanism or medium through which an input "action" at one point creates a measurable "response" at another point. Throughout history, engineers have utilized diverse media to create these channels from copper cables and fiber-optic strands to electromagnetic waves in vacuum/air and acoustic waves in water. Regardless of the specific medium, the key to any channel is a reliable effect-response relationship.

In this new context, our design goals for a communication system are often in tension: we want information transfer to be as fast as possible while remaining perfectly reliable. However, achieving this is difficult because signal transmission is subject to the harsh realities of the physical world: noise, interference, and the fundamental constraints of bandwidth and power. We will study how digital systems navigate these hurdles using three primary strategies:

- **Modulation:** We will examine how to map bits to continuous-time signals (waveforms) specifically designed to survive the characteristics of a given medium.
- **Error-Control Coding:** We will explore how to achieve reliability in the presence of noise by introducing strategic redundancy into the bitstream.
- **Frequency Analysis:** We will see that understanding the frequency representation of our transmit signals plays a key role in both quantifying the bandwidth occupied by our signals and designing systems that allow multiple users to share a medium without mutual interference.

1 Modulation and Demodulation.

In the digital communication pipeline, the first blocks we study are modulation and demodulation. Given a bit sequence $\{b_m\}$, modulation is the process of constructing a continuous-time signal $s(t)$ that encodes these bits in a form suitable for transmission over a physical channel. Demodulation

is the reverse process: given a received waveform $y(t)$, the receiver processes it to produce estimates $\{\hat{b}_m\}$ of the transmitted bits.

1.1 On-Off Keying (OOK) Baseband Signal

We now introduce the simplest possible modulation scheme. The basic idea is extremely simple: during each bit interval, we either transmit a constant signal or transmit nothing at all. A bit 1 is represented by turning the signal "on" at a fixed amplitude, and a bit 0 is represented by turning the signal "off". The waveform therefore consists of rectangular pulses whose presence or absence encodes the bit sequence.

Definition 1 (On-Off Keying: Baseband Signal)**.** *Let $\{b_m\}_{m\geq 1}$ be a bit sequence with $b_m \in \{0,1\}$. Fix amplitude $V > 0$ and bit duration $T_b > 0$. The OOK baseband waveform is*

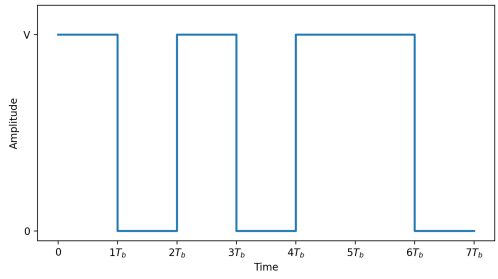
$$x(t) = \begin{cases} V, & t \in [(m-1)T_b, mT_b) \text{ and } b_m = 1, \\ 0, & t \in [(m-1)T_b, mT_b) \text{ and } b_m = 0. \end{cases}$$

The bit duration (or symbol duration) T_b is the amount of time allocated to transmit a single bit and the bit rate R_b is defined as the number of bits transmitted per unit time:

$$R_b = \frac{1}{T_b} \quad \textit{bits per second.}$$

In other words, the time axis is divided into intervals of length T_b, and within each interval the signal takes one of two possible amplitudes: either 0 or V. Information is therefore encoded purely through amplitude levels.

Example 1. *Consider the bit sequence $1, 0, 1, 0, 1, 1, 0$. Then the resulting OOK baseband waveform with bit duration T_b and amplitude V is as follows:*



1.2 Demodulation

We now turn to the receiver. The transmitter maps bits $\{b_m\}$ to a waveform $x(t)$. The channel introduces noise and distortion, and the receiver observes a waveform $y(t)$. The goal of demodulation is to process $y(t)$ and produce estimates $\{\hat{b}_m\}$ of the transmitted bits. In other words, we must decide, for each bit interval $[(m-1)T_b, mT_b)$, whether a 0 or a 1 was transmitted.

1.2.1 Method I: Midpoint Sampling

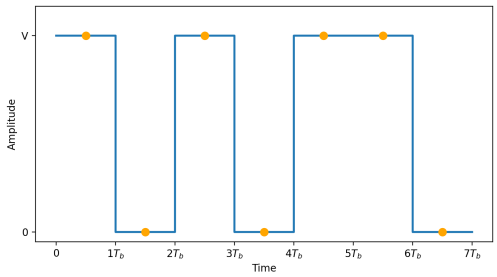
The simplest idea is to sample the received signal once per bit interval. Since $x(t)$ is constant within each interval in OOK, a natural choice is to sample at the midpoint:

$$t_m = (m-1)T_b + \frac{T_b}{2}.$$

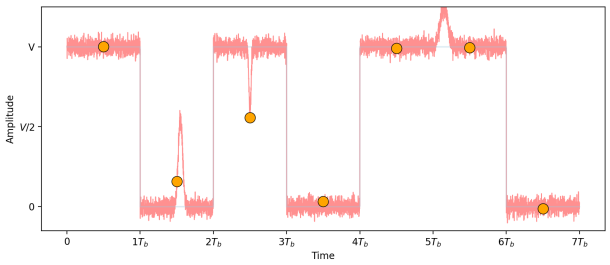
We then declare

$$\hat{b}_m = \begin{cases} 1, & y(t_m) > \alpha, \\ 0, & y(t_m) \le \alpha, \end{cases}$$

where α is a threshold, typically chosen to be $V/2$. If the channel is noiseless, so that $y(t) = x(t)$, this method works perfectly. Indeed, when $b_m = 1$, we have $y(t_m) = V$, and when $b_m = 0$, we have $y(t_m) = 0$. Choosing any threshold α in the interval $(0, V)$ guarantees correct decoding.



(a) Sampling in a noiseless scenario



(b) Sampling in presence of noise

However, this strategy is very sensitive to noise. Since the decision depends on the value of the signal at a single time instant, a brief noise spike near the sampling time can flip the decision even if the rest of the interval is relatively clean. In the figure above, most of the third bit interval contains a signal close to V, but a noise spike near the sampling time causes $y(t_m)$ to fall below the threshold, leading to an incorrect decision. This illustrates a fundamental weakness of single-sample detection: it does not use all the information available in the interval.

1.2.2 Method II: Average Value Detection

A more robust strategy is to use the entire bit interval rather than a single sample. Since OOK encodes information through amplitude, it is natural to measure the average amplitude of the received signal in each interval:

$$V_m = \frac{1}{T_b} \int_{(m-1)T_b}^{mT_b} y(t)\, dt.$$

This quantity captures the average amplitude of the signal during the bit interval. The decision rule becomes

$$\hat{b}_m = \begin{cases} 1, & V_m > V_{\text{thresh}}, \\ 0, & V_m \le V_{\text{thresh}}. \end{cases}$$

Let us first understand what happens in the noiseless case, where $y(t) = x(t)$. If $b_m = 0$, then $y(t) = 0$ throughout the interval and hence

$$p_m = \frac{1}{T_b} \int_{(m-1)T_b}^{mT_b} 0\, dt = 0.$$

If $b_m = 1$, then $y(t) = V$ throughout the interval and therefore

$$V_m = \frac{1}{T_b} \int_{(m-1)T_b}^{mT_b} V \, dt = \frac{1}{T_b}(VT_b) = V.$$

Consequently, in the ideal case without noise, the average amplitude matches the intended symbol: 0 for a bit "0" and V for a bit "1". In this noiseless scenario, any decision threshold chosen strictly between 0 and V (typically $V/2$) guarantees perfect, error-free decoding.

The fundamental advantage of using the average amplitude, effectively integrating the signal over the entire bit interval, is its inherent robustness. By observing the signal across the full duration T_b rather than sampling a single point, we exploit the statistical nature of noise. Most noise processes are zero-mean, meaning that over a sufficient time interval, positive and negative fluctuations tend to cancel each other out. While a single "midpoint" sample is highly vulnerable to being flipped by a short burst or a stray impulse of noise, the average value is much harder to displace. This "averaging out" process makes the receiver significantly more resilient to noise. In practice, extreme noise events can still push the average across the threshold, meaning errors cannot be entirely eliminated; however, by utilizing the signal's value across the entire interval, we maximize our use of the available information and substantially reduce the probability of error.

1.3 Frequency-Domain Properties of OOK

We now analyze OOK in the frequency domain. Since modulation produces a continuous-time waveform, its frequency content determines how much bandwidth is required for transmission. Recall that a continuous-time signal of duration L seconds can be written as

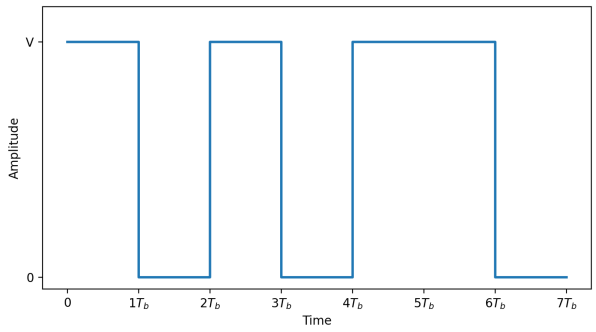
$$x(t) = b_0 + \sum_{j=1}^{\infty} a_j \sin\left(2\pi \frac{j}{L} t\right) + b_j \cos\left(2\pi \frac{j}{L} t\right).$$

Its spectrum is visualized by plotting the magnitudes

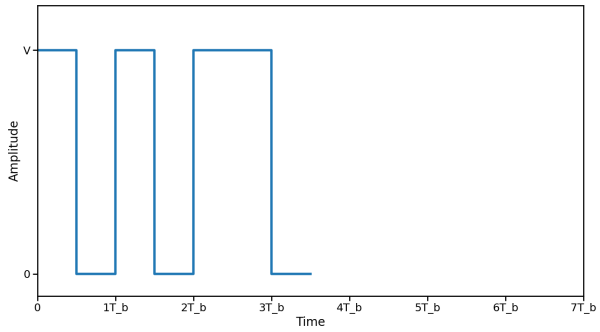
$$A_j = \sqrt{a_j^2 + b_j^2}$$

as stems at frequencies j/L.

Effect of Increasing the Bit Rate. Suppose we double the bit rate. Since $R_b = 1/T_b$, doubling the rate means replacing T_b by $T_b/2$. In time, this squeezes the waveform by a factor of 2. If the original waveform is $x(t)$, the faster waveform is approximately $x(2t)$.



(a) $x(t)$: Bit rate $1/T_b$



(b) $x(2t)$: Bit rate $2/T_b$

Now consider what happens to the frequency components. If $x(t)$ contains a term $\cos(2\pi ft)$, then $x(2t)$ contains $\cos(2\pi f2t) = \cos(2\pi(2f)t)$. Every frequency component is doubled. Thus, squeezing a signal in time stretches it in frequency. For example, if $z(t) = 1 + 3\cos(2\pi 5t)$, then its band is $[0, 5]$. Replacing t by $2t$ gives $z(2t) = 1 + 3\cos(2\pi 10t)$, whose band is $[0, 10]$. Squeezing the signal in time doubles its bandwidth. Since reducing T_b squeezes the OOK waveform in time, increasing the bit rate increases its bandwidth. We therefore expect the bandwidth to scale proportionally to $1/T_b$.

The Frequency Spectrum of OOK Consider an OOK waveform with bit duration T_b and total length L, where typically $L \gg T_b$. The waveform consists of rectangular pulses of width T_b placed at multiples of T_b, with amplitudes either 0 or V. When we compute its Fourier series representation over the interval of length L, we obtain discrete coefficient magnitudes $A_j = \sqrt{a_j^2 + b_j^2}$ at frequencies j/L.

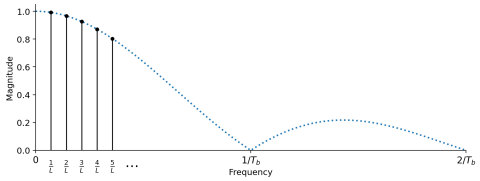
The magnitudes A_j follow a sinc-shaped envelope. Specifically, the overall profile of the spectrum is proportional to the absolute value of the normalized sinc function, which is defined as:

$$\text{sinc}(x) = \frac{\sin(\pi x)}{\pi x}.$$

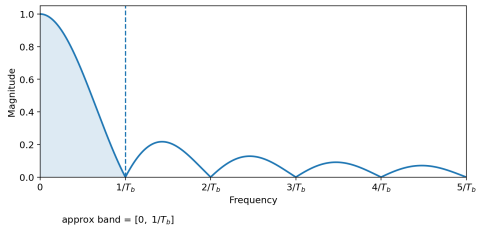
For our OOK signal, the frequency components are bounded by the following envelope:

$$|\text{sinc}(fT_b)| = \left|\frac{\sin(\pi fT_b)}{\pi fT_b}\right|$$

This envelope reaches its maximum at frequency 0 and has zero-crossings whenever the argument fT_b is a non-zero integer. In other words, these "spectral nulls" occur at every integer multiple of the bit rate $(1/T_b, 2/T_b, 3/T_b, \dots)$. While the exact height of the individual frequency stems (A_j) will fluctuate depending on the specific sequence of random bits transmitted, their collective decay pattern, the spectral nulls and bandwidth footprint are governed by the shape of this sinc envelope depicted below.



(a) Discrete spectrum with sinc envelope



(b) Continuous view of sinc-shaped decay

Although the spectrum extends to arbitrarily large frequencies, the magnitudes decrease rapidly beyond the first zero at $1/T_b$. Most of the signal energy lies in the main lobe between 0 and $1/T_b$, and the side lobes beyond this frequency are significantly smaller. This gives us the following approximate bandlimited property of the OOK signal.

Fact 1. *For OOK modulation with bit duration T_b (equivalently, bit rate $1/T_b$), the baseband signal is approximately bandlimited to the frequency range $[0, 1/T_b]$.*

Remark 1. *This bandlimiting is only approximate. The spectrum of OOK has a sinc-shaped decay and extends to arbitrarily high frequencies, but most of the energy lies in the main lobe up to* $1/T_b$*. There exist other modulation schemes that produce exactly bandlimited signals but these are beyond the scope of our course.*

1.4 Parameter Choice and the Role of Frequencies

There are two primary design parameters in OOK modulation: the amplitude V and the bit duration T_b.

Choice of Amplitude V. The amplitude V determines the signal strength. Increasing V increases the separation between the two hypotheses (bit 0 and bit 1). A larger separation makes the system more robust to noise and reduces the probability of bit error. However, V cannot be increased arbitrarily. Transmitters are subject to power constraints due to hardware limitations, battery life, regulatory limits, and interference considerations. Thus, while increasing V improves reliability, it is fundamentally constrained by physical and regulatory considerations.

Choice of Bit Duration T_b. The bit duration determines the bit rate: $R_b = \frac{1}{T_b}$. To transmit information faster, we want a larger bit rate, which means making T_b smaller.

Making T_b smaller, however, reduces the amount of time available to average noise within each bit interval. In average value detection, the integral

$$\frac{1}{T_b}\int_{(m-1)T_b}^{mT_b} y(t)\,dt$$

is computed over a shorter interval, so the averaging effect is weaker. As a result, the probability of bit error typically increases as T_b decreases. Nevertheless, this is not the most fundamental limitation. Error probability can be reduced by increasing V (within limits) or by using channel coding to introduce redundancy. The more fundamental constraint arises from bandwidth.

Why Frequency Matters: The Physical Reality of Bandwidth As we have seen, the OOK waveform is effectively bandlimited to the frequency range $[0, 1/T_b]$. This creates a fundamental physical trade-off: to increase the bit rate, we must reduce the symbol duration T_b, which proportionally "stretches" the signal's footprint in the frequency domain. This is the precise technical reason why the terms *high-speed* and *high-bandwidth* are often used interchangeably. Faster communication literally occupies more "room" in the frequency domain.

At this point, one might ask: why do we care about the frequency representation at all? Is not the Fourier series simply a mathematical trick for representing functions in a different way?

On the contrary, the frequency representation corresponds to something very physical and real: the **physical waves** we launch into the environment. When we transmit a signal, the Fourier series tells us exactly which waves are present in the medium. For example, for acoustic channels every frequency component in our math represents a literal pressure wave. If the Fourier series shows energy at 5 kHz, there is a physical wave vibrating the air 5,000 times per second. In wireless systems, a frequency of f Hz represents an electromagnetic wave oscillating at that rate as it propagates through space.

Because these waves are physically present, the frequency representation describes the **spectral footprint** of our transmission in the environment. This is how diverse communications co-exist in the air or on the shared fiber-optic cables that serve entire neighborhoods. If we know the exact "lane" of frequencies a signal occupies, we can design other systems to operate in different "lanes" simultaneously without mutual interference. Just as two cars can travel the same highway if they stay in different lanes, two communication systems can share the same medium if they occupy different frequency bands.

In the United States, the electromagnetic spectrum is treated as a limited public resource managed by the Federal Communications Commission (FCC). The FCC allocates distinct frequency bands to specific services to prevent chaos. For example, the FM broadcast band spans 88–108 MHz and is divided into non-overlapping 200 kHz channels, each assigned to a different station. As another example, the standard 2.4 GHz Wi-Fi operates between 2400 and 2483.5 MHz. This band is divided into multiple channels (such as Channels 1, 6, and 11), allowing different routers in the same apartment building to operate without "clashing" as long as they stay in their assigned lanes.

The full complexity of this management can be seen in the United States Frequency Allocation Chart. Consequently, a communication system must operate within its assigned band. This available bandwidth directly limits how small T_b can be, effectively capping the achievable bit rate. Increasing a system's rate often requires licensing more bandwidth, a process that can involve significant financial investment.

Furthermore, we must be mindful of frequency because physical channels are frequency-selective. Different frequencies experience different levels of attenuation or distortion. For example, a standard audio speaker or microphone has a limited frequency response; it may be unable to move fast enough to reproduce high-pitched tones or lack the physical size to move enough air for deep bass. Similarly, walls might block 5 GHz Wi-Fi signals more effectively than 2.4 GHz signals. To ensure reliable transmission, we must locate our signal's spectrum within the favorable portion of the channel's frequency response.

2 Passband Signals

So far, we have focused on a **baseband** signal $x(t)$ whose spectrum lies approximately in the range $[0, B]$, where $B \approx 1/T_b$. In this scheme, most of the signal's energy is concentrated near frequency 0. However, as we have discussed, modern communication systems do not have the luxury of using the entire spectrum at will; they are assigned specific frequency bands $[f_{\min}, f_{\max}]$ with $f_{\min} > 0$.

If every transmitter utilized frequencies near zero, their spectra would overlap completely, resulting in mutual interference. There would be no physical mechanism to separate one user's signal from another's. Beyond this regulatory necessity, baseband communication is often physically impractical for wireless transmission because there is a fundamental relationship between the frequency of an electromagnetic wave and the size of the antenna required to transmit it. To radiate energy efficiently, an antenna's length must be a significant fraction of the signal's wavelength ($\lambda = c/f$). At baseband frequencies (near 0 Hz), the wavelength is enormous—stretching for hundreds of miles. Transmitting a baseband signal wirelessly would require an antenna kilometers long. By shifting the signal to higher frequencies (where wavelengths are centimeters or millimeters), we can use the small, portable antennas found in smartphones and laptops.

To overcome these hurdles, we need a mechanism to translate the spectrum of $x(t)$ from near-zero frequencies to a higher allocated band $[f_{\min}, f_{\max}]$. This process is known as passband modulation.

The core objective is to shift the baseband waveform so that its spectrum is centered around a carrier frequency f_c. The key idea is to multiply the baseband signal by a high-frequency sinusoid. Before describing the general mathematical scheme, let us first look at a simple example to build our intuition.

Example 2. *Let* $z_1(t) = 1 + \cos(2\pi 100t)$*, whose frequency band is* $[0, 100]$*. Choose* $f_c = 1000$ *and consider*

$$z_2(t) = z_1(t)\sin(2\pi 1000t).$$

Using the identity

$$\sin(a)\cos(b) = \frac{1}{2}\big(\sin(a+b) + \sin(a-b)\big),$$

we obtain

$$z_2(t) = \sin(2\pi 1000t) + \frac{1}{2}\sin(2\pi 1100t) + \frac{1}{2}\sin(2\pi 900t).$$

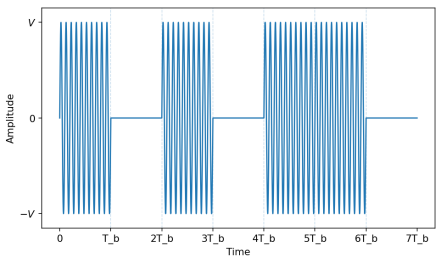
The original frequency band $[0, 100]$ *has been shifted to* $[900, 1100]$*, which equals* $[f_c - 100, f_c + 100]$*.*

This example illustrates the general phenomenon: multiplying by $\sin(2\pi f_c t)$ creates new frequency components at $f_c + f$ and $f_c - f$ for every baseband frequency f. Now consider a general signal $x(t)$ that contains a frequency component $\cos(2\pi ft)$ with $0 \le f \le B$. Multiplying by $\sin(2\pi f_c t)$ produces components at $f_c + f$ and $f_c - f$. Thus, each baseband frequency is translated symmetrically around f_c. If $x(t)$ occupies the band $[0, B]$, then after multiplication the signal occupies $[f_c - B, f_c + B]$. The width of the band becomes $2B$ and its location changes.

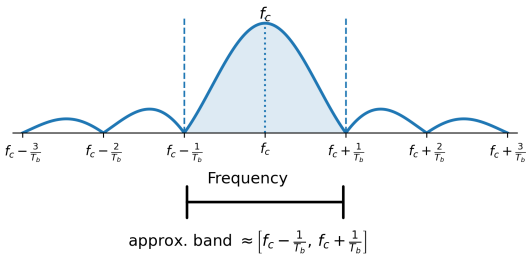
We now apply this frequency-translation idea directly to OOK modulation. Let $x(t)$ be the baseband OOK signal constructed using amplitude V and bit duration T_b. To shift its spectrum from $[0, 1/T_b]$ to a band centered around f_c, we multiply by a high-frequency sinusoid and define

$$s(t) = x(t)\sin(2\pi f_c t).$$

The sinusoid $\sin(2\pi f_c t)$ is called the **carrier wave**, and f_c is the **carrier frequency**. To distinguish between $x(t)$ and $s(t)$, we will refer to $x(t)$ as the **baseband OOK signal** and $s(t)$ as the **passband OOK signal**.



(a) Passband signal $s(t)$



(b) Spectrum of $s(t)$

This passband signal corresponds to a very natural communication strategy: we pick a specific "tone" for transmission and transmit that tone for a fixed duration to indicate that a bit is 1, while remaining silent to indicate a 0.

Mathematically, when $b_m = 1$, the transmitted signal over the interval $[(m-1)T_b, mT_b)$ is $V\sin(2\pi f_c t)$, and when $b_m = 0$, the transmitted signal is 0. In this scheme, the high-frequency

carrier is turned on and off in accordance with the bit sequence. In this relationship, the baseband waveform $x(t)$ acts as a slowly varying envelope that controls the amplitude of the rapidly oscillating carrier.

Fact 2 (Bandwidth of OOK Modulation)**.** *For OOK modulation with bit duration T_b (equivalently, bit rate $1/T_b$), the baseband signal $x(t)$ is approximately bandlimited to the frequency range $[0,\ 1/T_b]$. The passband signal constructed using $s(t) = x(t)\sin(2\pi f_c t)$, where $f_c > 1/T_b$, is approximately bandlimited to the frequency range $[f_c - 1/T_b,\ f_c + 1/T_b]$.*

The bit rate remains $R_b = 1/T_b$. For baseband OOK, the approximate band is $[0, 1/T_b]$, so the bandwidth is $1/T_b$. For passband OOK, the approximate band is $[f_c - 1/T_b, f_c + 1/T_b]$, so the bandwidth is $2/T_b$. Thus, for passband transmission, the bandwidth equals $2R_b$. Increasing the bit rate necessarily increases the required bandwidth.

Choosing Carrier Frequency and Bit Duration Based on an Allocated Band Suppose we are allocated a frequency band $[f_{\min}, f_{\max}]$ for transmission. We would like the passband OOK spectrum to lie entirely inside this band. Since passband OOK occupies approximately $[f_c - 1/T_b, f_c + 1/T_b]$, we require

$$f_c - \frac{1}{T_b} \geq f_{\min} \quad \text{and} \quad f_c + \frac{1}{T_b} \leq f_{\max}.$$

To utilize the band efficiently, we center the signal within the allocated range by choosing the carrier frequency as the midpoint,

$$f_c = \frac{f_{\min} + f_{\max}}{2}.$$

Let $B = f_{\max} - f_{\min}$ denote the bandwidth of the allocated channel. With the midpoint choice of f_c, the conditions above reduce to $2/T_b \leq B$. Thus, the maximum achievable bit rate satisfies

$$R_b = \frac{1}{T_b} = \frac{B}{2}.$$

This shows that the bit rate depends only on the width of the allocated band, not on its location. The carrier frequency determines where we transmit, while the available bandwidth determines how fast we can transmit.

Note that with this strategy, we are technically fitting the "main lobe" of the signal into the band. Because the sinc-shaped envelope of the spectrum extends infinitely, a small amount of energy will inevitably leak outside our allocated band. Regulators such as the FCC generally permit a controlled amount of such leakage, provided the out-of-band power remains within strictly defined emission masks.

2.1 Demodulating Passband Signals: From Amplitude to Power

In the first half of this course, we saw that for baseband signals (where a bit "1" is a constant voltage V), we could decode the signal by computing the **average amplitude** over the bit interval. However, once we move to passband communication, this simple strategy fails.

The Problem with Average Amplitude Recall that a passband bit "1" is represented by a high-frequency tone: $y(t) = V\sin(2\pi f_c t)$. If we attempt to decode this by taking the average amplitude over the bit interval T_b, we compute:

$$\frac{1}{T_b}\int_{(m-1)T_b}^{mT_b} V\sin(2\pi f_c t)dt$$

Because the carrier frequency f_c is typically much larger than the bit rate, the positive and negative half-cycles of the sinusoid cancel each other out. The result of this integral is effectively zero, regardless of whether a "1" or a "0" was transmitted.

The Solution: Decoding Based on Average Power To distinguish a high-frequency tone from silence, we must look at the **average power** of the signal. The standard strategy is to square the received signal before integrating. Squaring the signal ensures that all values become non-negative, preventing the cancellation effect. For each bit interval m, the receiver computes the average power:

$$p_m = \frac{1}{T_b}\int_{(m-1)T_b}^{mT_b} y^2(t)dt$$

In the noiseless case:

- **When $b_m = 0$:** The signal $y(t) = 0$, so $p_m = 0$.
- **When $b_m = 1$:** The signal $y(t) = V\sin(2\pi f_c t)$. Using the identity $\sin^2(\theta) = \frac{1}{2}(1-\cos(2\theta))$, the integral yields:

$$p_m = \frac{V^2}{2}$$

By looking at the average power, we have created a clear distinction between our two states. The receiver now chooses a threshold (typically $V^2/4$) to decide between bit 1 and bit 0.

2.2 Synchronization

So far, we have assumed that the receiver knows exactly where each bit interval begins. In other words, we have assumed that the receiver and transmitter are perfectly synchronized—meaning they have agreed on the same origin $t = 0$ and are using the same "clock" to define the intervals $[(m-1)T_b, mT_b)$.

In practice, this assumption is unrealistic. Because the transmitter and receiver are physically separated, the signal travels through the medium with an unknown delay. The receiver has no inherent way of knowing when the first bit was launched. Synchronization is the process of aligning the receiver's local clock with the transmitter's, effectively ensuring that both entities place the origin $t = 0$ at the same physical moment in the signal.

If the bit boundaries are misaligned, the receiver's mathematical process fails. Without a shared $t = 0$, the integral

$$p_m = \frac{1}{T_b}\int_{(m-1)T_b}^{mT_b} y^2(t)\,dt$$

will no longer capture a single, discrete bit. Instead, it will average over portions of two adjacent bits. Even in a perfectly noiseless case, this "mixing" of bits produces intermediate power values—falling somewhere between 0 and $V^2/2$—that do not clearly correspond to either state. This leads to ambiguous results and incorrect bit decisions.

A simple synchronization scheme is as follows: we add a **dummy 1** at the beginning of every transmission to serve as a "start bit." The receiver continuously monitors the incoming signal and detects the first time the received power exceeds a set threshold. This specific moment is taken as the receiver's origin, $t = 0$. Once this starting point is determined, the receiver knows that the dummy bit occupies the first interval, and all subsequent bit intervals are spaced at integer multiples of the bit duration T_b.

This approach works reasonably well when the signal-to-noise ratio is high and the channel is stable. However, it is particularly vulnerable to bursty noise. A short, high-energy burst of noise can cross the threshold prematurely, causing the receiver to "start its clock" at the wrong time. Because synchronization defines the timing grid for all subsequent bits, a single false detection at the beginning can shift the entire alignment, potentially corrupting an entire packet of data. In practice, more sophisticated synchronization schemes are used to improve reliability and robustness.