

Signals and Systems Practice Test (Sample)

Study Guide



Everything you need from our exam experts!

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Introduction

Preparing for a certification exam can feel overwhelming, but with the right tools, it becomes an opportunity to build confidence, sharpen your skills, and move one step closer to your goals. At Examzify, we believe that effective exam preparation isn't just about memorization, it's about understanding the material, identifying knowledge gaps, and building the test-taking strategies that lead to success.

This guide was designed to help you do exactly that.

Whether you're preparing for a licensing exam, professional certification, or entry-level qualification, this book offers structured practice to reinforce key concepts. You'll find a wide range of multiple-choice questions, each followed by clear explanations to help you understand not just the right answer, but why it's correct.

The content in this guide is based on real-world exam objectives and aligned with the types of questions and topics commonly found on official tests. It's ideal for learners who want to:

- Practice answering questions under realistic conditions,
- Improve accuracy and speed,
- Review explanations to strengthen weak areas, and
- Approach the exam with greater confidence.

We recommend using this book not as a stand-alone study tool, but alongside other resources like flashcards, textbooks, or hands-on training. For best results, we recommend working through each question, reflecting on the explanation provided, and revisiting the topics that challenge you most.

Remember: successful test preparation isn't about getting every question right the first time, it's about learning from your mistakes and improving over time. Stay focused, trust the process, and know that every page you turn brings you closer to success.

Let's begin.

How to Use This Guide

This guide is designed to help you study more effectively and approach your exam with confidence. Whether you're reviewing for the first time or doing a final refresh, here's how to get the most out of your Examzify study guide:

1. Start with a Diagnostic Review

Skim through the questions to get a sense of what you know and what you need to focus on. Your goal is to identify knowledge gaps early.

2. Study in Short, Focused Sessions

Break your study time into manageable blocks (e.g. 30 - 45 minutes). Review a handful of questions, reflect on the explanations.

3. Learn from the Explanations

After answering a question, always read the explanation, even if you got it right. It reinforces key points, corrects misunderstandings, and teaches subtle distinctions between similar answers.

4. Track Your Progress

Use bookmarks or notes (if reading digitally) to mark difficult questions. Revisit these regularly and track improvements over time.

5. Simulate the Real Exam

Once you're comfortable, try taking a full set of questions without pausing. Set a timer and simulate test-day conditions to build confidence and time management skills.

6. Repeat and Review

Don't just study once, repetition builds retention. Re-attempt questions after a few days and revisit explanations to reinforce learning. Pair this guide with other Examzify tools like flashcards, and digital practice tests to strengthen your preparation across formats.

There's no single right way to study, but consistent, thoughtful effort always wins. Use this guide flexibly, adapt the tips above to fit your pace and learning style. You've got this!

Questions

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- 1. Which statement best describes a discrete-time, discrete-amplitude signal?**
 - A. A The signal is defined at discrete times with continuous amplitude levels**
 - B. B The signal is defined at discrete times with discrete amplitude levels**
 - C. C The signal is defined at continuous times with discrete amplitude levels**
 - D. D The signal is defined at discrete times with discrete amplitude levels**

- 2. What is the effect of the delay operator Z^{-1} on a discrete-time signal $x[n]$?**
 - A. Advances the signal by one sample.**
 - B. Multiplies the signal by z .**
 - C. Delays the signal by one sample in time.**
 - D. Shifts the frequency spectrum.**

- 3. Which operation transforms a circuit's time-domain differential equation into a frequency-domain representation by substituting s with $j\omega$?**
 - A. Take the Fourier transform of the impulse response**
 - B. Use the Z-transform**
 - C. Use the discrete-time Fourier transform**
 - D. Apply the Laplace transform to the differential equation and substitute $s = j\omega$**

- 4. What is line spectra referring to in signal analysis?**
 - A. Line spectra plot amplitudes and phases of frequency components after Fourier representation.**
 - B. Line spectra plots time-domain instantaneous amplitude.**
 - C. Line spectra show only the DC component.**
 - D. Line spectra are derived from non-periodic signals only.**

5. Similarly, what happens if a signal is very broad in the time domain regarding its Fourier transform?
- A. It becomes even broader in frequency.
 - B. It becomes very narrow in frequency.
 - C. It becomes zero for all frequencies.
 - D. It becomes periodic.
6. Which statement about energy signals is true?
- A. It has zero energy.
 - B. It has finite energy and zero average power.
 - C. It has finite energy and nonzero average power.
 - D. It has infinite energy and nonzero average power.
7. If F_c equals $F_s/2$, what does this imply about the filter's passband with respect to the sampling rate?
- A. It allows up to half the sampling frequency in the baseband.
 - B. It allows frequencies above the sampling frequency.
 - C. It stops all frequencies.
 - D. It has no relation to the sampling frequency.
8. In an ideal analog-to-digital conversion chain, which order of blocks yields a digital output?
- A. A Sampler, Quantizer, Encoder, yielding digital output
 - B. B Encoder, Sampler, Quantizer, yielding digital output
 - C. C Quantizer, Encoder, Sampler, yielding digital output
 - D. D Sampler, Quantizer, Encoder, yielding digital output
9. The term bandwidth in this context refers to which portion of the spectrum?
- A. A All frequencies including negative
 - B. B Positive frequencies only
 - C. C Zero frequency only
 - D. D Frequencies above the Nyquist frequency

10. In a linear time-invariant system with a single input and output, the relation between input $x(t)$ and output $y(t)$ is given by which expression?

- A. $y(t)$ equals $x(t)$ convolved with $h(t)$.**
- B. $y(t)$ equals $x(t)$ times $h(t)$.**
- C. $y(t)$ equals the derivative of $x(t)$.**
- D. $y(t)$ equals the integral of $x(t)$.**

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Answers

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1. B
2. C
3. D
4. A
5. D
6. D
7. A
8. D
9. B
10. A

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Explanations

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1. Which statement best describes a discrete-time, discrete-amplitude signal?
- A. A The signal is defined at discrete times with continuous amplitude levels
 - B. B The signal is defined at discrete times with discrete amplitude levels**
 - C. C The signal is defined at continuous times with discrete amplitude levels
 - D. D The signal is defined at discrete times with discrete amplitude levels

The statement tests the combination of how time is represented and how amplitude is represented. A discrete-time signal is defined only at specific time instants (like $t = nT$), not for every moment in time. A discrete-amplitude signal means the signal can take only a finite set of values due to quantization. When both properties apply—values exist only at discrete times and the amplitudes are drawn from a discrete set—you have a discrete-time, discrete-amplitude signal. That's exactly what the description says: defined at discrete times with discrete amplitude levels. The other descriptions mix one discrete aspect with a continuous one, so they don't capture both features at once. For example, sampling with continuous amplitude would still allow any real value, and a signal defined for all time with discrete amplitudes isn't truly discrete-time. An everyday example is a digital audio sample: it's taken at distinct times and each sample is quantized to a finite number of levels.

2. What is the effect of the delay operator Z^{-1} on a discrete-time signal $x[n]$?
- A. Advances the signal by one sample.
 - B. Multiplies the signal by z .
 - C. Delays the signal by one sample in time.**
 - D. Shifts the frequency spectrum.

Delaying by one sample is what the delay operator does in discrete time. When you apply Z^{-1} to $x[n]$, you get $y[n] = x[n-1]$. In other words, every sample is taken from one step earlier, so the signal is shifted to the right by one sample (often with $x[n]$ replaced by zero for $n < 0$ if the signal is assumed causal). Advancing by one sample would come from multiplying by z , giving $y[n] = x[n+1]$, which is the opposite shift. A pure time delay does not move the spectrum itself; it adds a phase factor that varies with frequency, leaving the magnitude spectrum intact.

3. Which operation transforms a circuit's time-domain differential equation into a frequency-domain representation by substituting s with $j\omega$?

- A. Take the Fourier transform of the impulse response**
- B. Use the Z-transform**
- C. Use the discrete-time Fourier transform**
- D. Apply the Laplace transform to the differential equation and substitute $s = j\omega$**

The essential idea is to use the Laplace transform to convert the time-domain differential equations that describe a circuit into algebraic equations in the complex frequency variable s . Derivatives become multiplications by s , so the entire relationship between voltages and currents becomes an equation in s . By then evaluating that s -domain expression on the imaginary axis, substituting $s = j\omega$, you obtain the frequency-domain representation $H(j\omega)$ that characterizes how the system responds across frequencies. This is the standard route from time-domain dynamics to a frequency-domain transfer function. The other options don't fit this exact process. Taking the Fourier transform of the impulse response gives the frequency response directly but doesn't describe converting the governing differential equation itself. The Z-transform and the discrete-time Fourier transform apply to discrete-time signals or systems, not to continuous-time differential equations.

4. What is line spectra referring to in signal analysis?

- A. Line spectra plot amplitudes and phases of frequency components after Fourier representation.**
- B. Line spectra plots time-domain instantaneous amplitude.**
- C. Line spectra show only the DC component.**
- D. Line spectra are derived from non-periodic signals only.**

Line spectra represent the frequency-domain content of a signal as discrete lines at specific frequencies. Each line corresponds to a sinusoidal component revealed by a Fourier representation, with its height showing the component's amplitude and its angle giving the component's phase. For periodic signals, these lines appear at harmonic frequencies of the fundamental; non-periodic signals tend to produce a continuous spectrum rather than discrete lines. The DC component is just one line at zero frequency, but line spectra overall describe the distribution of energy across all frequencies, not just a time-domain amplitude or a single value.

5. Similarly, what happens if a signal is very broad in the time domain regarding its Fourier transform?

- A. It becomes even broader in frequency.
- B. It becomes very narrow in frequency.
- C. It becomes zero for all frequencies.
- D. It becomes periodic.**

When a signal lasts a long time, its variations are spread out over a large time interval, so it changes slowly. In the Fourier domain, slow variation translates to energy concentrated at low frequencies. So the Fourier transform becomes very narrow in frequency, concentrating near zero frequency. In the limit of an ideal constant signal (infinite duration), the spectrum collapses to a single line at DC. Conversely, a signal that is very brief in time exhibits a broad spectrum. If a signal were perfectly periodic, its spectrum would consist of discrete lines rather than a continuous narrow band.

6. Which statement about energy signals is true?

- A. It has zero energy.
- B. It has finite energy and zero average power.
- C. It has finite energy and nonzero average power.
- D. It has infinite energy and nonzero average power.**

Energy signals are those whose total energy is finite, defined by $E = \int_{-\infty}^{\infty} |x(t)|^2 dt$. For such signals, the average power is zero, because $P = \lim_{T \rightarrow \infty} (1/2T) \int_{-T}^T |x(t)|^2 dt$, and the integral in the numerator is bounded by the finite energy E while the dividing interval $2T$ grows without bound. So the true statement is that an energy signal has finite energy and zero average power. The idea is that all the signal's energy is packed into a finite amount of time, so when you average it over increasingly large time windows, the average power tends to zero. Zero energy would mean the signal is zero almost everywhere, which isn't an energy signal. Finite energy with nonzero average power cannot happen because the average power of an energy signal must vanish in the infinite-time average. Infinite energy with nonzero average power is outside the definition of an energy signal.

7. If F_c equals $F_s/2$, what does this imply about the filter's passband with respect to the sampling rate?

- A. It allows up to half the sampling frequency in the baseband.**
- B. It allows frequencies above the sampling frequency.
- C. It stops all frequencies.
- D. It has no relation to the sampling frequency.

When a signal is sampled at rate F_s , the useful spectrum that you want to keep lies from 0 up to $F_s/2$, known as the Nyquist limit or baseband. A reconstruction low-pass filter with cutoff equal to $F_s/2$ will pass all frequencies in that baseband, preserving the original content up to the Nyquist frequency, while attenuating the spectral replicas that occur every F_s due to sampling. So the passband extending to $F_s/2$ means you retain frequencies up to half the sampling rate in the baseband, which is exactly what you need for faithful reconstruction. The other possibilities don't fit because passing frequencies above the sampling rate would let in the image spectra created by sampling, which would distort reconstruction; stopping all frequencies would remove usable content; and claiming there's no relation to the sampling rate ignores the fundamental Nyquist constraint that governs what can be faithfully represented after sampling.

8. In an ideal analog-to-digital conversion chain, which order of blocks yields a digital output?

- A. A Sampler, Quantizer, Encoder, yielding digital output
- B. B Encoder, Sampler, Quantizer, yielding digital output
- C. C Quantizer, Encoder, Sampler, yielding digital output
- D. D Sampler, Quantizer, Encoder, yielding digital output**

The process to turn an analog signal into digital always goes in three steps: sampling in time, quantizing the sampled values, and then encoding those quantized levels into a binary digital stream. Sampling converts the continuous-time signal into discrete-time samples. Quantization then maps each sample's amplitude to the nearest level in a finite set, making the signal discrete in amplitude. Finally, encoding takes those quantized levels and produces a binary code stream. This order is essential because encoding needs discrete, quantized values to produce a digital output; you can't generate binary data from a continuously valued signal without first sampling and quantizing. Swapping steps would mean encoding an imperfect or continuous signal or sampling after encoding, which wouldn't yield a proper digital representation. Therefore, the correct sequence is Sampler, Quantizer, Encoder, yielding digital output.

9. The term bandwidth in this context refers to which portion of the spectrum?

- A. A All frequencies including negative
- B. B Positive frequencies only**
- C. C Zero frequency only
- D. D Frequencies above the Nyquist frequency

Bandwidth describes the range of frequencies that carry signal energy. For real signals, the spectrum is symmetric about zero, so the information about how wide the signal is comes from the positive-frequency side. The bandwidth is essentially the highest positive frequency present (the span from 0 up to that point). Counting negative frequencies would just duplicate what the positive side already shows. Frequencies above the Nyquist frequency aren't part of the signal's true spectrum—they're aliases caused by sampling. So the positive-frequency portion best captures the bandwidth.

10. In a linear time-invariant system with a single input and output, the relation between input $x(t)$ and output $y(t)$ is given by which expression?

- A. $y(t)$ equals $x(t)$ convolved with $h(t)$.**
- B. $y(t)$ equals $x(t)$ times $h(t)$.
- C. $y(t)$ equals the derivative of $x(t)$.
- D. $y(t)$ equals the integral of $x(t)$.

For a linear time-invariant system, the output is obtained by convolving the input with the system's impulse response. The relationship is $y(t) = \int x(\tau) h(t - \tau) d\tau$. This means every value of the input acts like a scaled copy of the impulse response shifted in time, and the overall output is the sum of all those shifted responses. This convolution reflects both linearity (superposition of scaled responses) and time invariance (shifts in time of the input produce corresponding shifts in the response). In short, the output is not just a simple product or a derivative or an integral of the input in general; it's the convolution with the impulse response, which generalizes all linear, time-invariant behavior. (Special cases exist: a differentiator corresponds to $h(t) = \delta'(t)$, an integrator to $h(t) = u(t)$, etc., but the general relation is the convolution.)

Next Steps

Congratulations on reaching the final section of this guide. You've taken a meaningful step toward passing your certification exam and advancing your career.

As you continue preparing, remember that consistent practice, review, and self-reflection are key to success. Make time to revisit difficult topics, simulate exam conditions, and track your progress along the way.

If you need help, have suggestions, or want to share feedback, we'd love to hear from you. Reach out to our team at hello@examzify.com.

Or visit your dedicated course page for more study tools and resources:

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We wish you the very best on your exam journey. You've got this!

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