SIGNALS AND SYSTEMS PRACTICE TEST (SAMPLE)

STUDY GUIDE

SAMPLE STUDY GUIDE WITH LIMITED QUESTIONS

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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

1. **In a discrete-time LTI system, which expression corresponds to the eigenvalue for input e^{jωn}?**
   A. H(e^{jω})
   B. H(jω)
   C. H(s)
   D. H(-ω)

2. **Discrete Fourier Transform (DFT) is used to represent a finite-duration, non-periodic signal by sampling its spectrum. Which statement is accurate about this representation?**
   A. The DFT uses a finite sum over N samples to approximate the spectrum.
   B. The DFT yields a continuous spectrum.
   C. The DFT requires the signal to be strictly periodic.
   D. The DFT does not apply to finite-duration signals.

3. **Which formula defines the signal-to-noise ratio (SNR) in decibels?**
   A. 20 log10(P_signal/P_noise)
   B. 10 log10(P_signal/P_noise)
   C. log10(P_signal/P_noise)
   D. 40 log10(P_signal/P_noise)

4. **How is the cutoff frequency defined in terms of H(jω) magnitude?**
   A. The magnitude equals the maximum.
   B. The magnitude equals one-half of the maximum magnitude.
   C. The magnitude equals zero.
   D. The magnitude equals 1/√2 of the maximum magnitude.

5. **Why do multiple copies appear in the spectrum of a sampled signal?**
   A. The spectrum is convolved with a rectangular window in time, producing copies.
   B. The spectrum is unaffected by sampling.
   C. The spectrum is replicated at multiples of Fs due to sampling, with equal amplitude.
   D. The spectrum is inverted in frequency.

6. Which of the following statements is presented as a typical relationship between Fc and Fs in sampling?

A. Fc = Fs

B. Fc = 2*Fs

C. Fc = Fs/2

D. Fs = 10*Fh

7. A time-invariant system has the property that:

A. The input-output relation changes with time.

B. The impulse response depends on time.

C. The output is scaled by time.

D. A time shift in the input results in the same time shift in the output.

8. In the z-transform context, what does the final value theorem relate?

A. The final steady-state value of x[n] as n ∞ to the limit of z X(z)/(z-1) as z 1.

B. The initial value x[0].

C. The sum of all samples.

D. The derivative of the sequence at infinity.

9. For a single-input, single-output time-invariant linear system, the impulse response h(t) is the system's output when the input is which signal?

A. A delta function.

B. A unit step.

C. A sine wave.

D. A ramp.

10. Which statement describes a causal impulse response?

A. h[n] is zero for all n.

B. h[n] is nonzero for negative indices.

C. h[n] is zero for n < 0.

D. h[n] is nonzero for positive indices.

Answers

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

Explanations

1. In a discrete-time LTI system, which expression corresponds to the eigenvalue for input e^{jωn}?

A. H(e^{jω})

B. H(jω)

C. H(s)

D. H(-ω)

*In a discrete-time LTI system, complex exponentials e^{jωn} are eigenfunctions. If you feed the system x[n] = e^{jωn}, the output becomes y[n] = (h * x)[n] = ∑ h[k] e^{jω(n−k)} = e^{jωn} ∑ h[k] e^{−jωk}. The sum ∑ h[k] e^{−jωk} is the discrete-time frequency response evaluated on the unit circle, written as H(e^{jω}). Therefore, the output is simply a scaled copy of the input: y[n] = H(e^{jω}) e^{jωn}. The factor H(e^{jω}) is the eigenvalue corresponding to the input e^{jωn}. This H(e^{jω}) is the system's frequency response for discrete time. The other forms reference continuous-time representations or alternate frequency axes (like H(jω) for continuous-time, H(s) for Laplace-domain, or evaluating at negative frequency H(-ω)), which don't give the correct eigenvalue for a discrete-time input e^{jωn}.*

2. Discrete Fourier Transform (DFT) is used to represent a finite-duration, non-periodic signal by sampling its spectrum. Which statement is accurate about this representation?

A. The DFT uses a finite sum over N samples to approximate the spectrum.

B. The DFT yields a continuous spectrum.

C. The DFT requires the signal to be strictly periodic.

D. The DFT does not apply to finite-duration signals.

The DFT represents a finite-duration signal by a discrete set of spectral samples, obtained from a finite sum over the available time samples. When you have N time samples x[0] through x[N−1], the DFT computes N complex values X[k] = sum_{n=0}^{N−1} x[n] e^{-j 2π kn / N}, which correspond to N equally spaced frequency bins ω_k = 2π k / N. This means you are sampling the continuous spectrum of the signal (which would exist for all frequencies if you looked at the full Fourier transform) at N discrete points, not producing a continuous spectrum. The input need not be strictly periodic in time; the finite sequence is treated as one period of a periodic extension to justify the discrete spectrum, and the result is a finite, discrete set of spectral components. That's why the statement about using a finite sum over N samples to approximate the spectrum is the accurate description.

3. Which formula defines the signal-to-noise ratio (SNR) in decibels?

A. 20 log10(P_signal/P_noise)

B. 10 log10(P_signal/P_noise)

C. log10(P_signal/P_noise)

D. 40 log10(P_signal/P_noise)

In decibels, the SNR is defined as 10 times the base-10 logarithm of the power ratio because decibels are a logarithmic scale for power. Since SNR compares signal power to noise power, the appropriate form is 10 log10(P_signal / P_noise). If you were instead dealing with voltages or amplitudes, you'd use 20 log10(V_s / V_n) because power is proportional to the square of the voltage (P V^2); the extra factor of 2 inside the log translates to the 20 in front when you convert a voltage ratio to decibels. The other options don't match this power-based definition: 20 log10(P_s / P_n) would misalign with the standard power-to-decibel relation, log10(P_s / P_n) lacks the 10 to place it in decibels, and 40 log10(P_s / P_n) isn't a recognized form for SNR.

4. How is the cutoff frequency defined in terms of $H(j\omega)$ magnitude?

A. The magnitude equals the maximum.

B. The magnitude equals one-half of the maximum magnitude.

C. The magnitude equals zero.

D. The magnitude equals $1/\sqrt{2}$ of the maximum magnitude.

The cutoff frequency is the point where the magnitude of the frequency response has fallen to $1/\sqrt{2}$ of its maximum value. This is the -3 dB point, tied to the fact that power $|H(j\omega)|^2$, so reducing power by half corresponds to reducing the magnitude by $1/\sqrt{2}$. If the passband gain is normalized to 1, then the cutoff occurs at $|H(j\omega_c)| = 1/\sqrt{2}$. The other options don't match this standard definition: maximum magnitude would mean no attenuation, zero would imply complete attenuation, and half of the maximum does not correspond to the half-power point.

5. Why do multiple copies appear in the spectrum of a sampled signal?

A. The spectrum is convolved with a rectangular window in time, producing copies.

B. The spectrum is unaffected by sampling.

C. The spectrum is replicated at multiples of Fs due to sampling, with equal amplitude.

D. The spectrum is inverted in frequency.

Sampling a signal in time causes its spectrum to repeat at intervals of the sampling frequency. This happens because sampling is multiplication of the signal by a periodic impulse train in time, and multiplication in time corresponds to convolution in frequency with that train's spectrum. The spectrum of the impulse train is another train of impulses spaced by Fs, so the original spectrum is copied and shifted to every k·Fs. Each replica has the same shape as the original (and, under typical conventions, the same amplitude), just located at multiples of Fs. This is why you see multiple copies in the spectrum of a sampled signal. If the sampling rate is high enough that these copies don't overlap, you can perfectly reconstruct the original signal; if not, they overlap and cause aliasing.

6. Which of the following statements is presented as a typical relationship between Fc and Fs in sampling?

A. Fc = Fs

B. Fc = 2*Fs

C. Fc = Fs/2

D. Fs = 10*Fh

The key idea is how often you should sample a signal relative to its highest frequency content. According to the Nyquist criterion, you must sample at least twice as fast as the highest frequency component to avoid aliasing. In practice, though, real systems add margins for non-ideal filters and reconstruction, so engineers often choose a sampling rate well above the bare minimum—commonly around ten times the highest frequency present in the signal. That conservative rule-of-thumb is why the relationship Fs = 10·Fh is considered typical: it communicates a substantial safety margin between the signal's highest frequency and the sampling rate, reducing distortion and making reconstruction easier. The exact requirement is still Fs ≥ 2·Fmax, but 10× is a common design choice to ensure clean sampling and reconstruction.

7. A time-invariant system has the property that:

A. The input-output relation changes with time.

B. The impulse response depends on time.

C. The output is scaled by time.

D. A time shift in the input results in the same time shift in the output.

Time invariance means that delaying the input in time yields an identical delay in the output, with the same waveform. For a linear time-invariant system, the output is $y(t) = \int x(\xi)\, h(t-\xi)\, d\xi$. If you feed a time-shifted input $x_d(t) = x(t-\tau)$, the resulting output becomes $y_d(t) = \int x(t-\tau-\xi)\, h(t-\xi)\, d\xi$. With a change of variables, this simplifies to $y_d(t) = y(t-\tau)$. So delaying the input by τ delays the output by the same amount, preserving the shape. This is the defining behavior, whereas changing with time, having an impulse response that depends on absolute time, or output scaling with time would break this property.

8. In the z-transform context, what does the final value theorem relate?

A. The final steady-state value of $x[n]$ as $n \ \infty$ to the limit of $z X(z)/(z-1)$ as $z \ 1$.

B. The initial value $x[0]$.

C. The sum of all samples.

D. The derivative of the sequence at infinity.

The final value theorem shows how the long-run (steady-state) value of a discrete-time sequence $x[n]$ can be read directly from its Z-transform $X(z)$ by looking at a limit as z approaches 1. When $x[n]$ converges to a constant L and the ROC is appropriate (the system is stable and no problematic poles block the unit circle), the constant L is obtained from the limit of $z X(z)/(z - 1)$ as $z \quad 1$. The idea is that near $z = 1$, $X(z)$ has a behavior dominated by the tail $x[n] \quad L$, and the factor $z/(z - 1)$ isolates that tail value as z approaches 1. This is the standard way to extract the final steady-state value from the transform, provided the usual conditions hold.

9. For a single-input, single-output time-invariant linear system, the impulse response h(t) is the system's output when the input is which signal?

A. A delta function.

B. A unit step.

C. A sine wave.

D. A ramp.

In an LTI (linear time-invariant) system, the impulse response $h(t)$ is defined as the output when the input is a Dirac delta signal $\delta(t)$. This is because, for any input $x(t)$, the output is the convolution $x(t) * h(t)$. The Dirac delta acts as the identity for convolution, so feeding $\delta(t)$ directly reveals $h(t)$ itself: $y(t) = \delta(t) * h(t) = h(t)$. The other inputs produce different responses: a unit step gives the step response (the integral of $h(t)$ if the system is causal), a sine wave produces a steady-state response related to the system's frequency response, and a ramp yields a response tied to the system's integration properties. None of these equal the impulse response itself.

10. Which statement describes a causal impulse response?

A. h[n] is zero for all n.

B. h[n] is nonzero for negative indices.

C. h[n] is zero for n < 0.

D. h[n] is nonzero for positive indices.

For a causal discrete-time system, the output cannot respond to inputs that haven't happened yet. The impulse response h[n] represents how the system reacts to a unit impulse applied at n = 0. If the system is causal, there can be no reaction before the impulse, so there is no response at negative times. That is why h[n] must be zero for all negative indices. This makes the statement the best fit for causality: the impulse response is zero for negative n. In contrast, saying h[n] is zero for all n would describe a degenerate system with no response at all, which isn't generally the case. Saying h[n] is nonzero for negative indices would imply the system anticipates the impulse, violating causality. Saying h[n] is nonzero for positive indices isn't the defining property of causality—the system could have nonzero or zero values at positive times; what matters is the absence of any response before the impulse.

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:

https://signalssystems.examzify.com

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