Digital Signal Processing

1. Digital signal processing (DSP) is a technique, which uses digital methods to __________.
   a. Create white noise
   b. Display a signal
   c. Market a product
   d. Process a signal

2. In a general DSP system, both the input data and the processed data are stored in the _____________.
   a. ADC
   b. LPF
   c. RAM
   d. ROM

3. The DSP processes the input data using programmed algorithms.
   a. True
   b. False

4. The algorithms used by the DSP are programmed into the ___________.
   a. ADC
   b. LPF
   c. RAM
   d. ROM

5. A DSP will only process ______________ data.
   a. Amplified
   b. Analog
   c. Continuous
   d. Digital

6. A digital filter
   a. Can duplicate the function of an analog filter
   b. Cannot duplicate the functions of an analog filter
   c. Is completely different in function from an analog filter
   d. Is rarely used compared to the analog filter

7. Filter selectivity is determined by
   a. How steep the filter response curve is
   b. The ambient temperature around it
   c. The manufacturer of the filter
   d. The time delay between the input data points

8. Digital filters are better than analog filters because
   a. The roll off rate is slower
   b. They can't distinguish between similar signals
   c. They have better selectivity
   d. They operate at lower frequencies

9. Name the four basic filter types.

10. In analog filters, the ideal response curve is closely matched to the practical response curve.
    a. True
    b. False

11. Calculate the bandwidth of a band pass filter with cutoff frequencies of 3 kHz and 10 kHz.
    a. 3 kHz
    b. 7 kHz
    c. 10 kHz
    d. 14 kHz

12. The notch filter is another name for the
    a. Low pass filter
    b. High pass filter
    c. Band pass filter
    d. None of the above

13. Analog filters can be constructed with
    a. Op amps
    b. RC sections
    c. RL sections
    d. All of the above

14. An active analog filter is one made with
    a. Multiple RL sections
    b. Op amps
    c. Op amps and RL sections
    d. Op amps and RC sections

15. The selectivity of an active filter can be improved by
    a. Adding an RL section
    b. Cascading active filters together
    c. Removing one of the op amps
    d. None of the above

16. By cascading active filters together, the response curve of the filter will match the ideal curve.
    a. True
    b. False

17. The phase response of analog filter can cause ________ to become distorted.
    a. Analog signals
    b. Digital signals
    c. Sine waves
    d. All of the above

18. A phase shift in the harmonics of a digital pulse signal is called ________.
    a. Frequency delay
    b. Group delay
    c. Pulse delay
    d. Signal delay

19. Compared to the analog filters, DSP filters
    a. Have more limitations than the analog filters
    b. Have none of the limitations of analog filters
    c. Have exactly the same limitations as the analog filters
    d. Limitations are still being evaluated

20. The primary function of the averaging filter is to
    a. To amplify the signal
    b. To increase selectivity
    c. To minimize noise
    d. To provide feedback

21. Which filter type uses feedback and can produce superior selectivity?
    a. Averaging filter
    b. Finite impulse response filter
    c. Infinite impulse response filter
    d. None of the above

22. The number of coefficients used in a filter algorithm is also called the number of
    a. Inputs
    b. Operations
    c. Taps
    d. Thumps

23. What is the advantage of a special multiply and accumulate (MAC) instruction?
    a. It eliminates the addition function
    b. It eliminates the sample delay
    c. It helps speed up the operations
    d. It reduces the number of calculations

24. In block diagrams, the ⊗ symbol represents a
    a. Adder
    b. Denominator
    c. Multiplier
    d. Summation

25. Some diagrams use a ____________ to indicate a time delay.
    a. T^{-1}
    b. X^{-1}
    c. Y^{-1}
    d. Z^{-1}

26. The time delay in DSP's are used to represent a time difference between
    a. Data storage
    b. Input and output
    c. Samples
    d. Users

27. An averaging filter is used to minimize ________ noise.
    a. All frequency
    b. High frequency
    c. Low frequency
    d. Single frequency

28. In the averaging filter, the DSP divides by multiplying its fraction because
    a. It doesn't have the capability to divide
    b. It is only an average anyways
    c. It only recognizes decimals
    d. Multiplication is faster

29. The averaging filter is really a _____________ filter
    a. Band pass
    b. High pass
    c. Low pass
    d. Notch

30. If the analog samples are being received from an ADC, the speed of the DSP is determined by the
    a. Design of the DSP
    b. Sampling rate of the ADC
    c. Temperature of the DSP
    d. Time delay of the DSP

31. As the filter receives the third sample, it is multiplied by the following coefficient.
    a. $B_{(0)}$
    b. $B_{(1)}$
    c. $B_{(2)}$
    d. $B_{(3)}$

32. As a new input sample is received, the other samples are multiplied by
    a. A new coefficient
    b. All dropped off
    c. The same coefficient
    d. The sum of the coefficients

33. Each time a new sample is received, a single new output value is created.
    a. True
    b. False

34. If the algorithm has only four taps, the ________ sample will cause the first sample to drop off.
    a. First
    b. Third
    c. Fourth
    d. Fifth

35. By selecting the correct set of coefficients, the FIR filter can
    a. Completely eliminate all noise
    b. Perform as any of the four basic filter types
    c. Process more than 10 samples at a time
    d. Use the division function in the DSP

36. FIR filters need at least __________ taps to achieve a high selectivity.
    a. 4
    b. 6
    c. 8
    d. 16

37. The higher the number of taps, the
    a. Faster the processing time
    b. Longer the processing time
    c. Lower the selectivity
    d. Simpler the algorithm

38. Many design engineers can now use __________ to determine the coefficients for DSP filters.
    a. Books
    b. Calculators
    c. Estimation
    d. Programs

39. How do IIR filters increase selectivity without increasing the number of taps?
    a. They eliminate the multiplications factor
    b. They only look at previous samples
    c. They use a type of feedback in the calculation
    d. All of the above

40. Which type of filter uses two sets of coefficients?
    a. Averaging filter
    b. FIR filter
    c. IIR filter
    d. All of the above

41. In the IIR filter algorithm, the subtraction function is
    a. Carried out by the DSP
    b. Done by adding negative coefficients
    c. Not needed
    d. Eliminated by increasing the number of taps

42. What are the two basic sections found in the IIR filter algorithm?

43. Which of the following cannot be used for all four basic filter types?
    a. Averaging filter
    b. FIR filter
    c. IIR filter

44. Which of the following is a disadvantage found with the IIR filter?
    a. Can become unstable if not designed right
    b. Have a slow response time
    c. Requires more taps to increase selectivity
    d. All of the above

45. The second most common application for a DSP is
    a. Compression
    b. Filtering
    c. Mixing
    d. Spectral analysis

46. The DFT and FFT algorithms are used to analyze signals in the
    a. Amplitude domain
    b. Frequency domain
    c. Mixed signal domain
    d. Time domain

47. A square wave is made up of a fundamental sine wave and its ______ harmonics.
   a. Even
   b. Odd
   c. 1st ten
   d. Two

48. The statement that any complex signal is made up of sine or cosine waves and its harmonics is defined as the _________ theory.
   a. Discrete
   b. Fourier
   c. Frequency
   d. Transfer

49. _________ is the process of determining the similarities between two signals.
   a. Contemplation
   b. Convolution
   c. Correlation
   d. Illustration

50. In the convolution operation, the values of one group are
   a. All equal
   b. Doubled
   c. Inverted
   d. Reduced by half

51. Another name for the convolution operation is
   a. Bouncing
   b. Flipping
   c. Smashing
   d. Tripping

52. In DFT, the ______________ for both sine and cosine waves are compared to the sampled analog waveform.
   a. Basis functions
   b. Core functions
   c. Frequency
   d. Simple function

53. If the cosine frequency domain plot has an output at K=0, this means the input signals has
   a. A fundamental cosine wave
   b. A fundamental sine wave
   c. An average DC level
   d. Two harmonics

54. The cosine and sine frequency domain plots are combined using the
    a. Convolution theorem
    b. Newton's law
    c. Ohms law
    d. Pythagorean theorem

55. A phase output plot can also be produced from the sine and cosine frequency domain plots.
    a. True
    b. False

56. Which operation is more efficient at higher frequencies?

57. Is it possible to do real time spectrum analysis on high frequency signals?
    a. Yes
    b. No
    c. Unknown

58. A personal computer can be used for DSP as long as
    a. An external hard drive is used
    b. The computer is cost effective
    c. The frequencies and sampling rate are slow enough
    d. The frequency is fast enough

59. One key feature of Microchips dsPIC30F2010 chip is
    a. An extra memory device
    b. An onboard ADC
    c. Its high frequency application
    d. None of the above

60. The primary use of Microchips dsPIC30F2010 chip is
    a. Filters
    b. Memory storage
    c. Mixing
    d. Modulation

61. The DSP program in embedded controller chips is stored in the
    a. CPU
    b. External hard drive
    c. RAM
    d. ROM

62. How does the Harvard architecture differ from the Von Neumann architecture?
    a. Harvard architecture has one memory and two buses
    b. Harvard architecture has three memories and three buses
    c. Harvard architecture has two memories and one bus
    d. Harvard architecture has two memories and two buses

63. Which manufacturer produces over 80% of all DSP chips?
    a. Analog Devices
    b. Freescale Semiconductors
    c. Motorola
    d. Texas Instruments

64. Most microprocessors represent data in
    a. Decimal form
    b. 2's complement binary
    c. Octal binary
    d. 4's complement binary

65. In the 2's complement number, the most significant bit is called the
    a. Alternate bit
    b. Lower bit
    c. Positive bit
    d. Sign bit

66. How can fractional numbers be represented?
    a. By placing a 0 in the MSB
    b. By placing a 1 in the MSB
    c. By putting the binary point to the left of the MSB
    d. By using convolution

67. Floating point numbers allow you to express
    a. All values as negative
    b. Fractional values
    c. Decibel values
    d. Larger binary values