MI 3120 Recording Production II **Name:______________________________**
MIDTERM EXAMINATION **E-Mail:______________________________**

March 26, 2003

Part I: Digital Audio Fundamentals

1. A form of distortion can occur during the conversion of analog signals into digital if the input signal is more than one half the sampling rate. Only portions of the signal will be present when the system samples the waveform and a false image of the waveform, based on the components that were actually sampled, is created. This is commonly referred to as
   a. quantization noise distortion
   b. signal anomalies of the sampling period
   c. aliasing
   d. subsampling distortion
   e. quantization error

2. A system which samples at 48 kHz can correctly process signals of up to 24 kHz. To remove signals above the ____________ Frequency, all A/D converters employ _________________. The 02R employs over-sampling A/D converters to allow the __________________ to be performed in the digital domain. (3 points)

3. The conversion of analog signals into digital signals is performed by the ______________________. The analog signal is sampled every few milliseconds and its level is _________________ into a digital word. The larger the digital word, the more accurate the representation of the analog value. (2 points)

4. The conversion of a digital data stream into analog signals is accomplished by the _______________________. The digital word is buffered and then converted into an analog signal. After conversion, the analog signal is usually processed through a __________________________which removes the step transitions between the digital words. (2 points)

5. _________________ is a mathematical process where a random noise signal is added to the least significant bit of a digital word. With very low level signals, the _________________ error becomes correlated to the signal level. This creates a measurable amount of distortion. By adding ____________, the correlation between the signal level and the _________________ error is cancelled, allowing the digital system to encode amplitudes smaller than the least significant bit. If you change the word size as a signal passes from one digital system to another, being able to add _____________ allows you to maintain a high quality signal. (2 points)

6. Nyquist Sampling Theorem—This theorem defines the process of sampling audio with a digital system. Amongst other things, it states that the sampling frequency of a digital audio system must be at least ________ that of the highest audio frequency, otherwise _____________ will occur. The Nyquist theorem was developed at Bell Labs by C. Shannon and H. Nyquist. (2 points)

7. The scheme for encoding audio data as a series of pulses where each pulse defines a transition from binary one to binary zero is known as
   a. Digital audio encoding
   b. Quantization
   c. Pulse-code modulation
   d. Pulse train encoding
   e. Lossy compression

8. The encoding process when the analog input is approximated to the nearest binary value available (these approximations are not an exact duplication of the analog waveform and are therefore contain errors or noise) is known as
   a. Lossy compression
   b. Binary sub-sampling
   c. Data reduction in the digital domain
   d. Quanitzation
   e. Aliasing

9. The rate at which measurements of an audio signal are taken during A/D and D/A conversion is known as
   a. Sampling period
   b. Sampling rate (or frequency)
   c. Quantization rate
   d. Quatization level
   e. None of the above

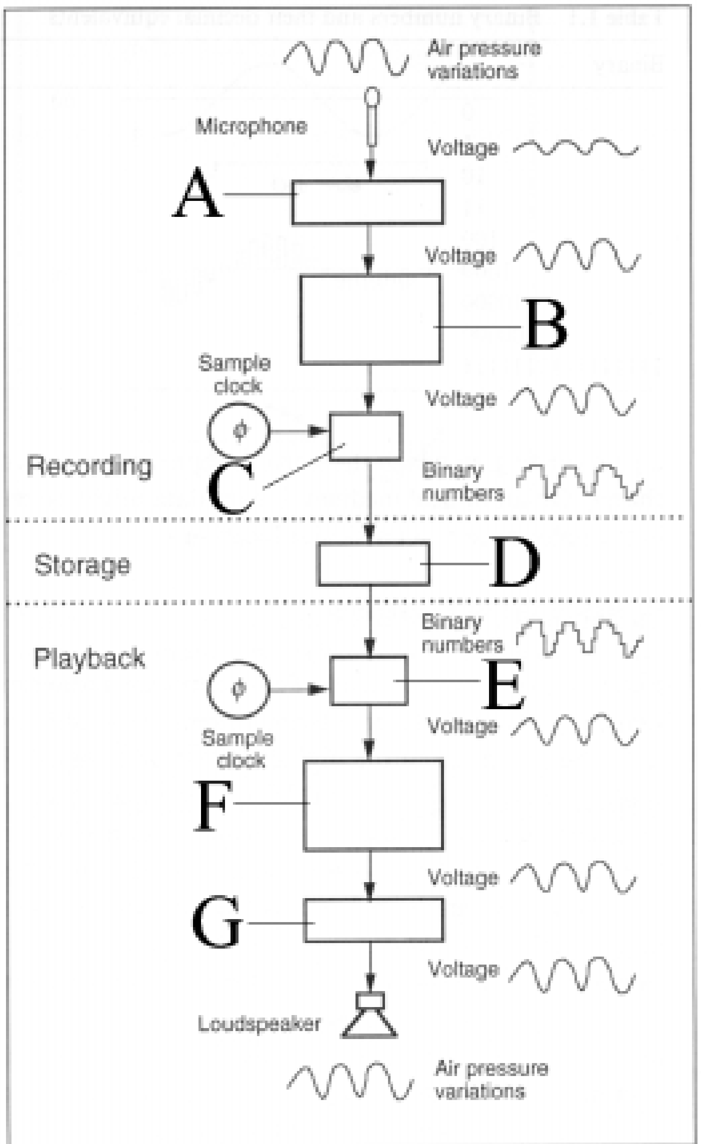
10. T / F The basic editing methods of cut, copy and paste are entirely analogous to those of a word processor and form the foundational level of digital audio editing techniques.

11. DSP means
   a. delinquent salivating puppy
   b. digital saturation protocol
   c. down-sampling pitch
   d. digital signal processing
   e. none of the above

12. Creating fades in digital audio is a DSP function because
    a. Sample amplitude values must be recalculated over some duration
    b. Sample amplitude values must remain constant over the fade points
    c. Nothing much happens to sample values
    d. All sample values go to zero and then positive values
    e. None of the above

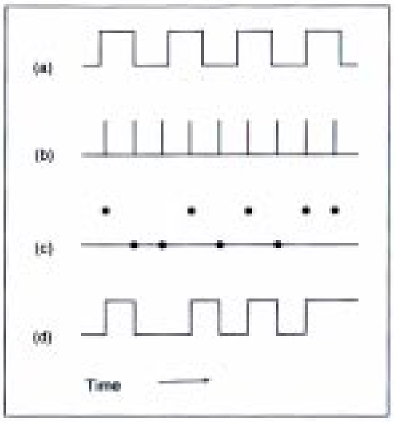
13. T / F To gain change a sample you simply recalculate it either up or down by a user-defined ratio.

14. Normalization is a process of gain change which
    a. makes the sound as loud as it would be normally
    b. makes an abnormal sound appropriate for digital editing
    c. removes D.C. offset
    d. optimizes dynamic range based on the highest sample
    e. reduces the noise floor of recordings by gain adjustment

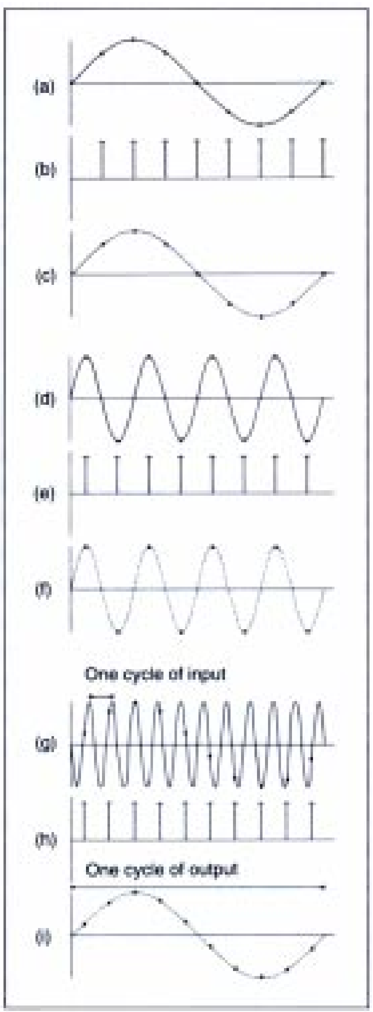


15. The above graphic depicts a simplified electroacoustic chain (ADC/DAC). Label all parts and briefly describe what functions they perform. (7 points)

16.



17. The above graphic shows a signal (a) and a sampling rate (b). Describe what concept is depicted and give precise descriptions of (c) and (d). (4 points)



18. Write a comprehensive description of the above graphic. (4 points)

Part II – O2R Console Operations

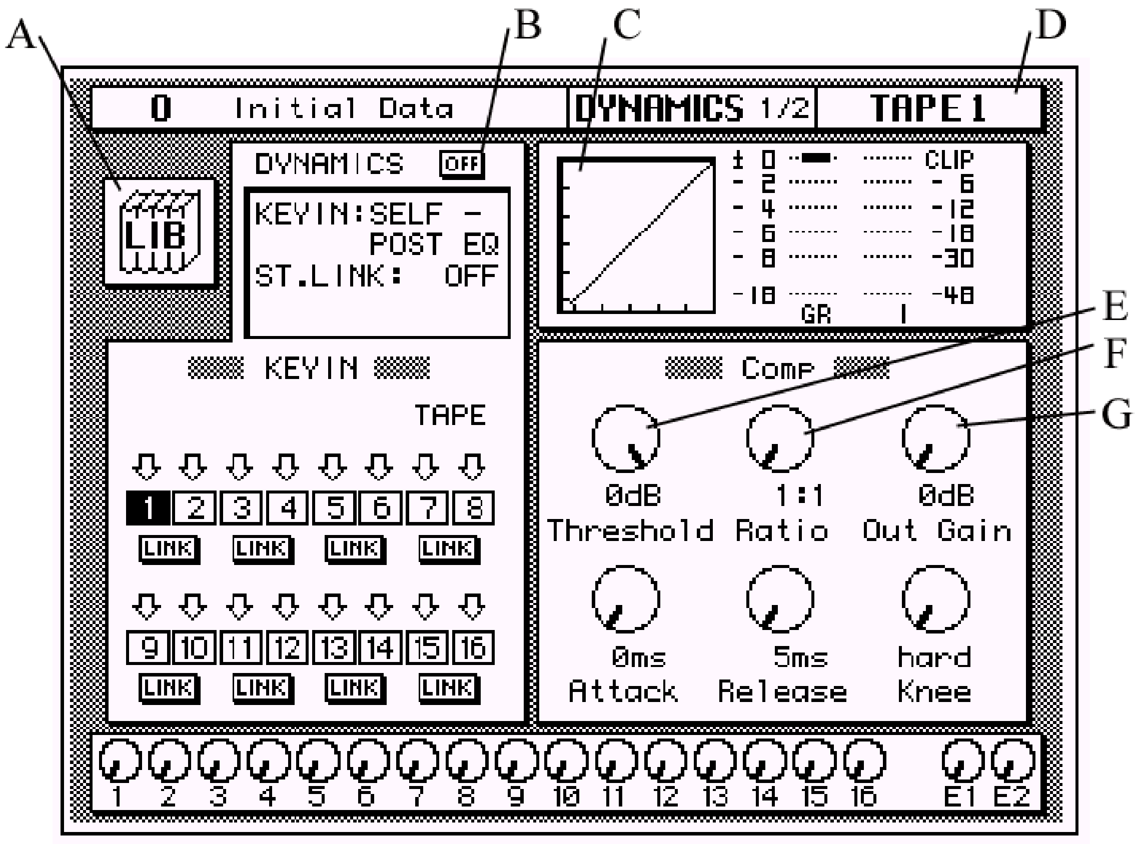
19. T / F It is important to avoid blocking the fader movements when faders move to their AUX send positions by placing objects on the O2R because it is possible to damage the faders.

20. T / F Compression can be considered a kind of automated gain control.

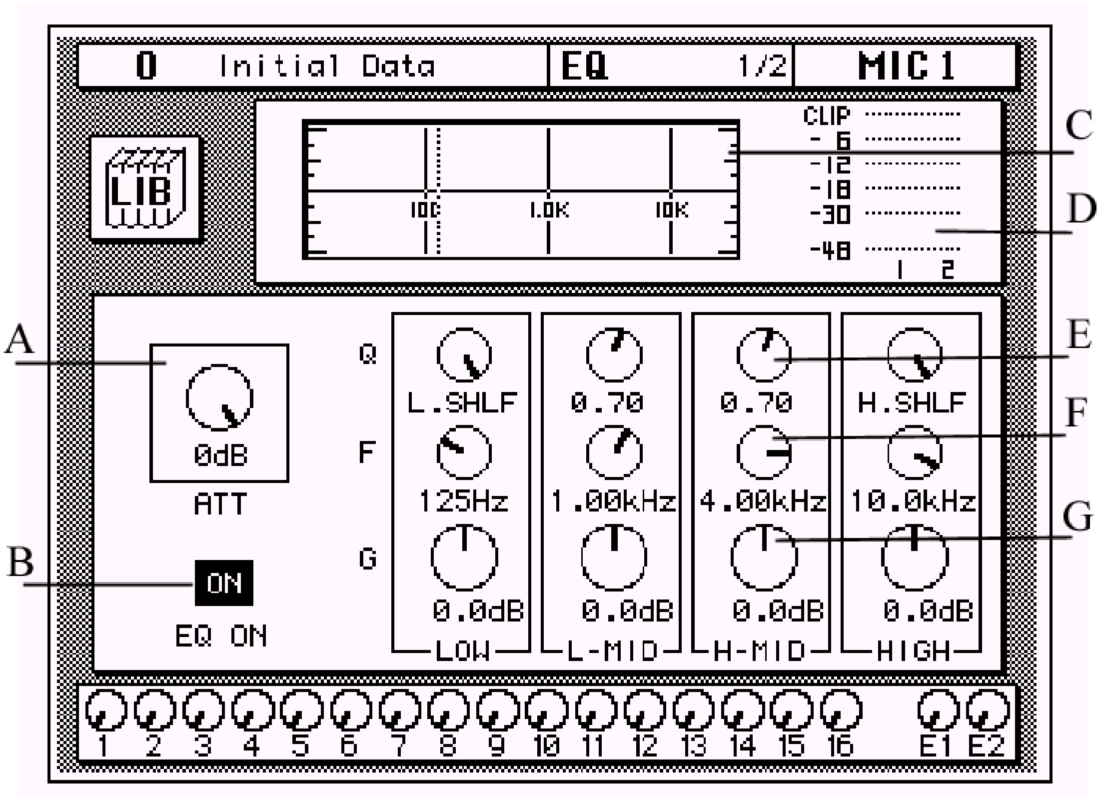
21. Gating is best described as
    a. A process for changing the mix level of a signal
    b. A destructive form of editing
    c. A kind of automated muting technique
    d. A process to avoid in professional audio work
    e. None of the above

22. It is possible to reset the EQ curves on the OR2 by
    a. turning the power off and then on again
    b. using the ProjectManager software
    c. going to the EQ screen and pressing "reset"
    d. press and hold LOW/HPF and then HI/LPF
    e. press and hold LOW/LPF and then HI/HPF

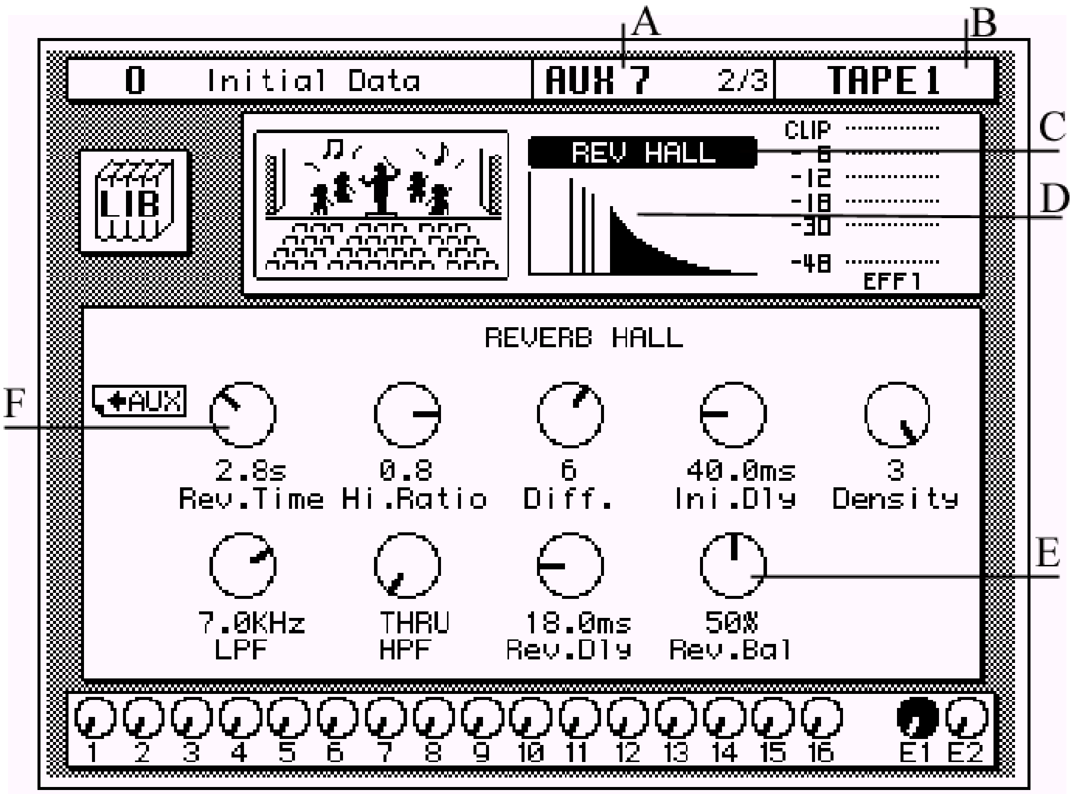
Please continue to next page….



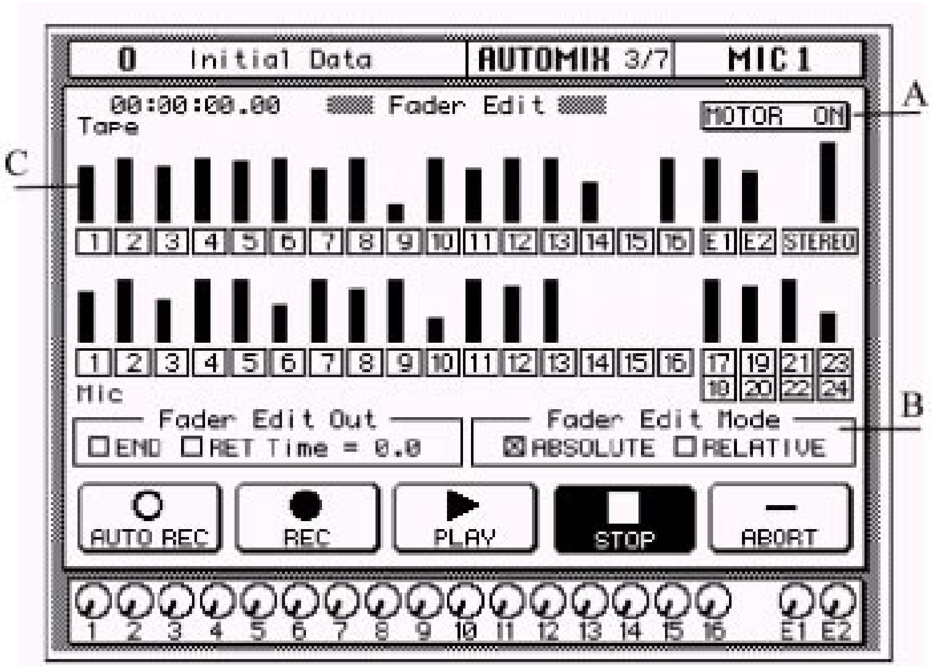
23. What does this O2R screen do? Provide labels and descriptions for all marked items. (8 points)



24. What does this O2R screen do? Label all marked elements and provide a description of the functions. (8 points)



25. What does this O2R screen do? Label all marked elements and provide a functional description. (7 points)



26. Provide a description of the use of this O2R screen. Label and describe the functions of the marked elements. (4 points)

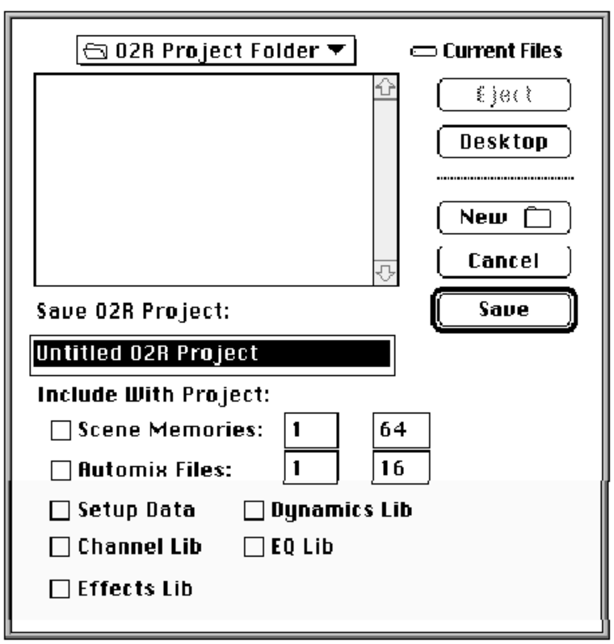
-

Part III: ProjectManager

27. What happens when you answer YES to this prompt? (2 points)

28. Why is it important to use ProjectManager for O2R sessions in CARA? (2 points)

29. The Re-Synchronize command in ProjectManager
    a. Runs a check on SMPTE and MIDI timecode
    b. Sets the internal buffer of the program to NULL
    c. Resets the serial connection and uploads all O2R data
    d. Resets the serial connection and blanks the O2R data
    e. All of the above



30. Explain the usage of this screen in detail. (5 points)

MI 3120 MIDTERM – Continued
ESSAY QUESTIONS

Use Blue Books. Put your name and email on the cover of the book. Please write legibly and neatly.

1. Answer one (1) of the following two (2) questions: (15 points) **–** Thompson: Digital Audio Technology – A Primer Parts I and II (found on your course web pages)

A: *Describe the ADC process touching on the following terms and concepts a) anti-aliasing filter, b) quantization, c) Nyquist Theorem, d) quantization error and e) sampling rate. Draw a basic diagram of the ADC which supports your response.*

B: *Describe the DAC process touching on the following terms and concepts a) smoothing filter, b) reconstruction of the analog signal, c) signal to noise ratio, d) dither and e) sampling rate. Draw a basic diagram of the ADC which supports your response.*

2. Read the article "boom…BOOM!" in February, 2003 Issue of Mix Magazine

Provide a brief synopsis of the salient points of this article. (10 points)

3. Read the article "Recording Vocals From A to D" in the February, 2003 Issue of Mix Magazine

Provide a brief synopsis of the salient points of this article. (10 points)

4. Read the articles (3) comprising the section "Surrounded" in the March, 2003 Issue of Mix Magazine

Provide a brief synopsis of the salient points of this articles. (10 points)

5. *Dr. Robert Willey in his webpage HOW TO STUDY, http://www.flowwwing.com/willeyrk/creative/papers/LEARN/LEARNING.HTML discusses some of his perspectives for effective learning. Outline his major points and suggestions and describe how they might apply to the life-long process of learning for the professional audio technologist. (10 points)*

EXTRA CREDIT – 25 points will be awarded for an excellent response to this question. No credit is given for a partial or incomplete response.

Two factors in digital audio are all important - frequency range, in terms of total bandwidth, and signal-to-noise ratio. Discuss these two features of the technology by comparing them to the analog domain, human auditory perception and acoustic principles. Importantly, describe what changes are coming in the near future which will influence both frequency range and signal to noise ratio.