

21M.380 · MUSIC AND TECHNOLOGY
RECORDING TECHNIQUES & AUDIO PRODUCTION

THIRD QUIZ (QZ3)

DIGITAL AUDIO, MIXING CONSOLES & STRATEGIES, RECORDING SESSIONS, ROOM ACOUSTICS & REVERBERATION

MONDAY, NOVEMBER 21, 2016

30 MINUTES, 10% OF TOTAL GRADE

1 Digital audio

1.1 Sampling theorem (5%)

A piano plays the concert pitch A₄ at 440 Hz. Assume that the piano's spectrum features notable acoustic energy up to about the twentieth harmonic.

What is the minimum sample rate that will be required to faithfully record the above piano note to a digital medium? Quantify the result in kHz and show how you derived your result.

$$f_1 = 440 \text{ Hz}$$

$$f_N = N \cdot f_1$$

$$f_{20} = 20 \cdot 440 \text{ Hz} = 8.8 \text{ kHz} = f_{max}$$

$$f_s \stackrel{!}{>} 2 \cdot f_{max} = 2 \cdot 8.8 \text{ kHz} = 17.6 \text{ kHz}$$

The sample rate must be at least 17.6 kHz.

1.2 Check all statements that are true (10%)

- The *reconstruction* filter in a DAC is a high-pass filter that cuts off all sidebands below the Nyquist frequency.
- One benefit of a higher sample rate is that the reconstruction filter's required slope becomes less steep.
- The term *dither* describes irregularities of the sample clock.
- The process of sampling creates copies of the original spectrum that repeat periodically around multiples of the sample rate.
- Playing back a signal that has been sampled with *jitter* results in distortion.

2 Room acoustics and reverberation

2.1 Room modes (10%)

The frequencies f_n of the expected *modes* in a given room are given as $f_n = \frac{c \cdot n}{2 \cdot d}$ where $c = 340 \text{ m s}^{-1}$ is the speed of sound, $n \in \mathbb{Z} : n \geq 1$ is the mode number, and d is the distance between two parallel walls. Assume that the room you are currently sitting in has a height of 3.4 m and a width of 6.8 m (not a bad estimate) and that the modes between the room's front and back wall are less pronounced due to its acoustic treatment.

1. In the low frequency range up to 100 Hz, which two frequencies would you expect to be particularly problematic in this room? Show how you derived these numbers in the page margin.

- First frequency: 50 Hz
- Second frequency: 100 Hz

2. What makes these frequencies 'problematic' – what does that even mean in this context?

The fact that modes at these frequencies occur along both dimensions, which will raise the likelihood for standing waves at those frequencies. Ideally, modal frequencies are more evenly distributed across the audible spectrum.

3. Which geometrical property of the room (as we have assumed them above) emphasizes this problem, and how could one reduce the likelihood of 'problematic' frequencies?

Room's width and height relate to each other in a simple 2:1 ratio, which naturally supports modes between both pairs of walls. To reduce likelihood, choose non-integer ratio.

2.2 Check all statements that are true (5%)

Why is the *critical distance* d_c of interest to recording engineers?

- Because it informs our decision where on the left-right axis of an ensemble to place a stereo microphone.
- Because it allows us to estimate the ratio of direct sound to diffuse reverberation in a recording.
- Because it defines the recording angle of a stereo microphone.
- Because it informs our decision at which distance from the ensemble to position a stereo microphone.
- Because it tells us where the proximity effect becomes negligible.

n	$h = 3.4 \text{ m}$	$w = 6.8 \text{ m}$
1	$\frac{340 \cdot 1}{2 \cdot 3.4} = 50 \text{ Hz}$	$\frac{340 \cdot 1}{2 \cdot 6.8} = 25 \text{ Hz}$
2	$\frac{340 \cdot 2}{2 \cdot 3.4} = 100 \text{ Hz}$	$\frac{340 \cdot 2}{2 \cdot 6.8} = 50 \text{ Hz}$
3	$\frac{340 \cdot 3}{2 \cdot 3.4} = 150 \text{ Hz}$	$\frac{340 \cdot 3}{2 \cdot 6.8} = 75 \text{ Hz}$
4	$\frac{340 \cdot 4}{2 \cdot 3.4} = 200 \text{ Hz}$	$\frac{340 \cdot 4}{2 \cdot 6.8} = 100 \text{ Hz}$

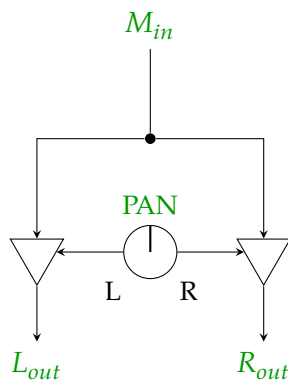
2.3 Room acoustic effects (5%)

You will hear a sound example.

- Which one of the following effects are you hearing? Select only one answer.
 - Artificial reverb
 - Comb filter
 - Flutter echo
- How can you explain the characteristic manner in which the sound in this example changes over time? How was this achieved? Select only one answer.
 - Automation of a pre-delay control
 - Adjustment of a digital oscillator's fundamental frequency
 - Change of distance between microphone and sound source

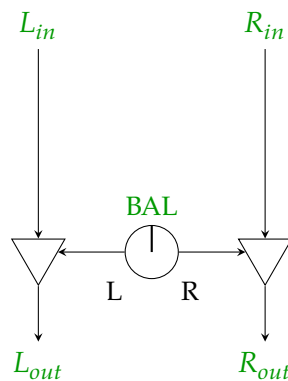
3 Mixing consoles

3.1 Panpot or balance control? (5%)



Left image:

- Panpot
- Balance control

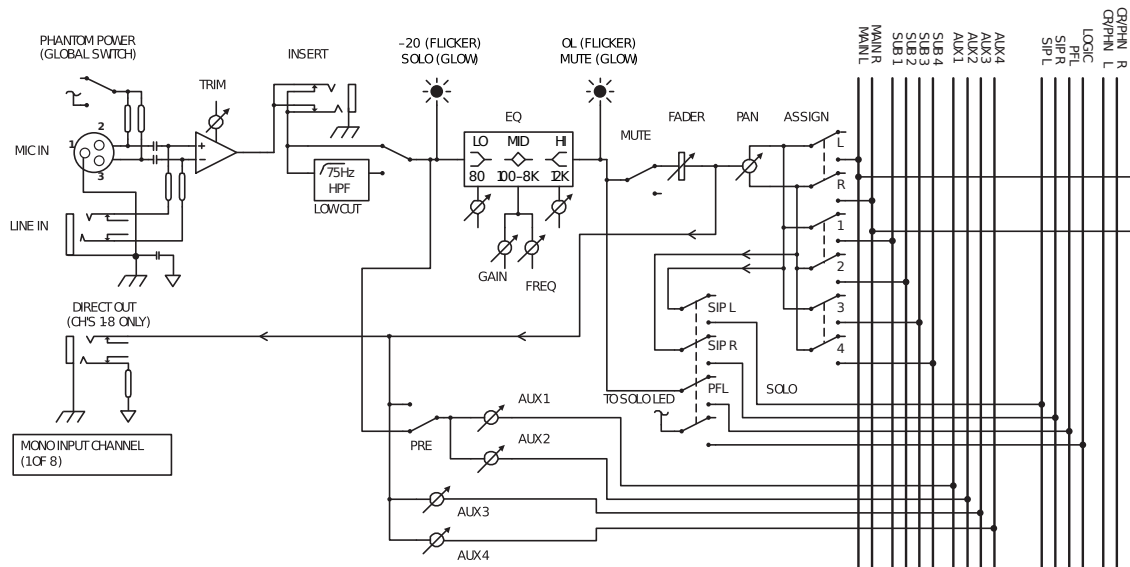


Right image:

- Panpot
- Balance control

3.2 Signal flow (15%)

Consider an analog mixing console's input channel.



1. Does the channel fader affect the level at which the signal appears at the Direct Out(put)? Yes No

2. You are engineering a live show and want to fine-tune the EQ on the guitar by soloing the guitar to listen to it pre-fader on your monitoring headphones. Will you be able to hear the effect of your EQ on the headphones? Yes No

Argue in favor of your answer: The PFL (pre-fader listening) solo bus is post-EQ.

3. Would the guitarist her-/himself likely hear the effect of the same EQ on her-/his monitoring loudspeaker? Yes No

Argue in favor of your answer: Monitor speakers use pre-fader aux sends, but pre-fader auxes 1+2 are also pre-EQ on this mixer

4. You use this console to mix a live show. The band has brought their favorite 19" mix-in effects: an external reverb unit, a chorus effect, and the world's biggest flanger. You explain that even though this mixer has 4 auxiliary sends, it will be tricky to accommodate all three effects if the band requires even just a single monitoring loudspeaker. Why?

Aux 3+4 are post-fader, aux 1+2 can be switched together between pre- & post-fader. Two possible configs: 4 post-fader, or 2 pre- & 2 post-fader aux. But we'd need 3 post-fader (effects) and 1 pre-fader aux (monitor).

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3.3 Auxiliaries (10%)

You are mixing a live show and connect a bass player, a guitarist, a saxophone and a vocalist to the first four channels of your mixer (in that order). You assign the first auxiliary send to an external reverb unit. You assign auxiliaries 2–4 to feed the stage monitor loudspeakers of the vocalist, the saxophone player, and the monitor box which the bassist and guitar player share (in that order). During the sound check, the following issues arise. Which auxiliary sends do you need to adjust on which input channels?

1. You decide that one can never have too much reverb, especially on vocals.
 - Input channel number: 4 (vocals)
 - Auxiliary send number: 1 (reverb send)
 - Auxiliary send should be set to...
 - pre-fader
 - post-fader
2. The saxophone player requests to hear more of the bass on her monitor.
 - Input channel number: 1 (bass)
 - Auxiliary send number: 3 (sax monitor send)
 - Auxiliary send should be set to...
 - pre-fader
 - post-fader
3. The guitar player requests to hear less of the bass on her monitor.
 - Input channel number: 1 (bass)
 - Auxiliary send number: 4 (bass/gtr monitor send)
 - Auxiliary send should be set to...
 - pre-fader
 - post-fader

After you have completed these edits as requested, which member of the band is most likely going to complain and why? The bassist, who shares a monitor loudspeaker with the guitar player and now hears less of her-/himself on that loudspeaker.


4 Recording sessions

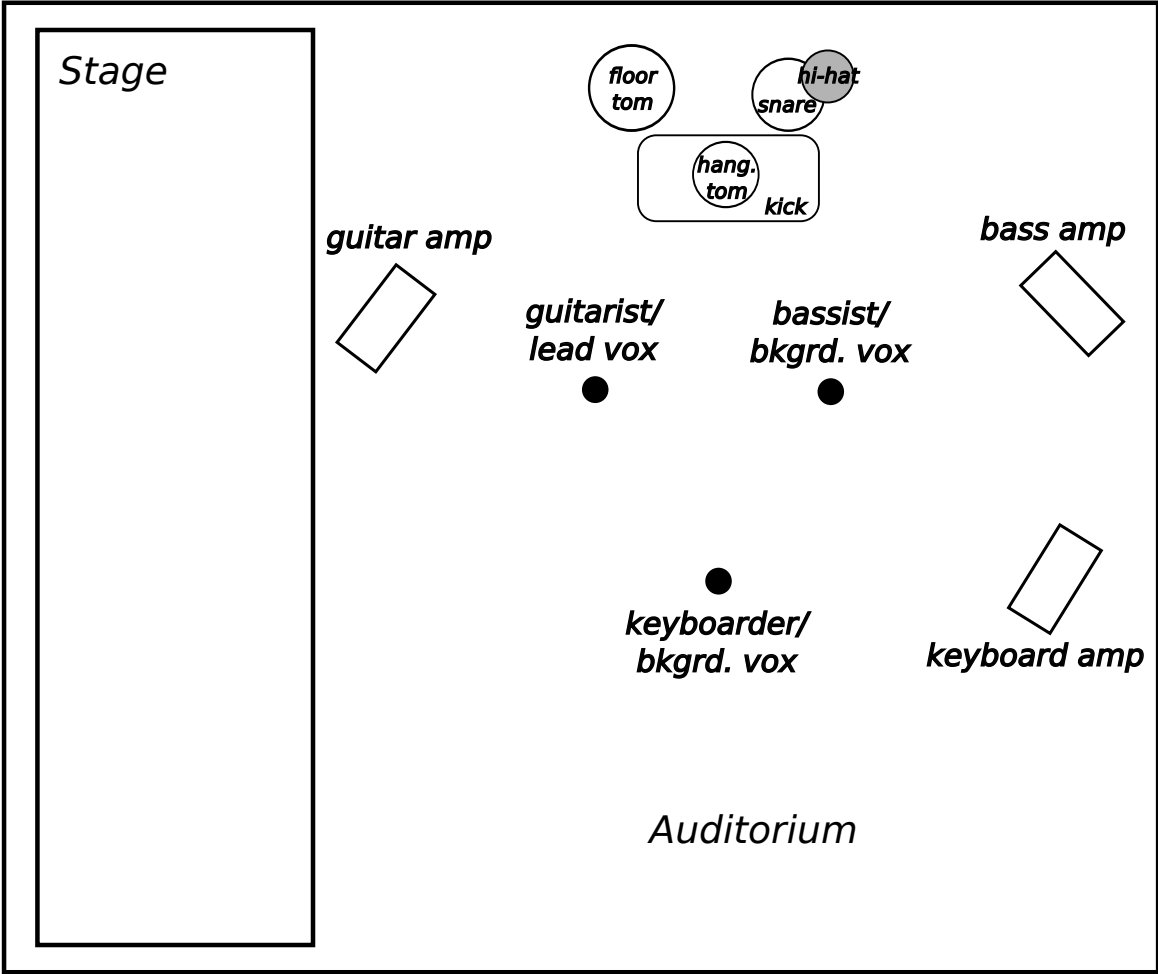
4.1 Fictional recording session (20%)

You are asked to record a progressive metal band in Killian Hall. The band has been moved off stage onto the carpet to minimize undesirable reflections. Its members play:

- Electric guitar & lead vocals
- Keyboard & background vocals
- Electric bass & background vocals
- Drumkit (for which the drummer forgot crash and ride cymbals)

You have access to the following moss equipment:

- 2 pcs. Audio-Technica AT4041
 - 2 pcs. Earthworks TC 20mp
 - 2 pcs. AKG C414
 - 1 pc. Royer R-101
 - 1 pc. Mojave MA-200
 - 2 pc. Shure Beta 58A
 - 1 pc. Shure SM 57
 - 1 pc. Sennheiser e604 (clip-on)
 - 2 pcs. Sennheiser MD421
 - 1 pc. Audix D6
 - 1 pc. Audio-Technica ATM250DE (dual capsule)
 - 2 pcs. Radial JDI passive mono DI box
1. In the stage plan on page 7, indicate how you will pick up the band. You do not have to use all available equipment, but you should also not make the setup too minimal. Feel free to include ambience mics and/or a main stereo mic (indicating the stereo technique that you have chosen) as you see fit.
 2. Mark every microphone that requires phantom power with 48V. E.g.,  48V indicates: "AKG C414, requires phantom power"



5 Mixing strategies

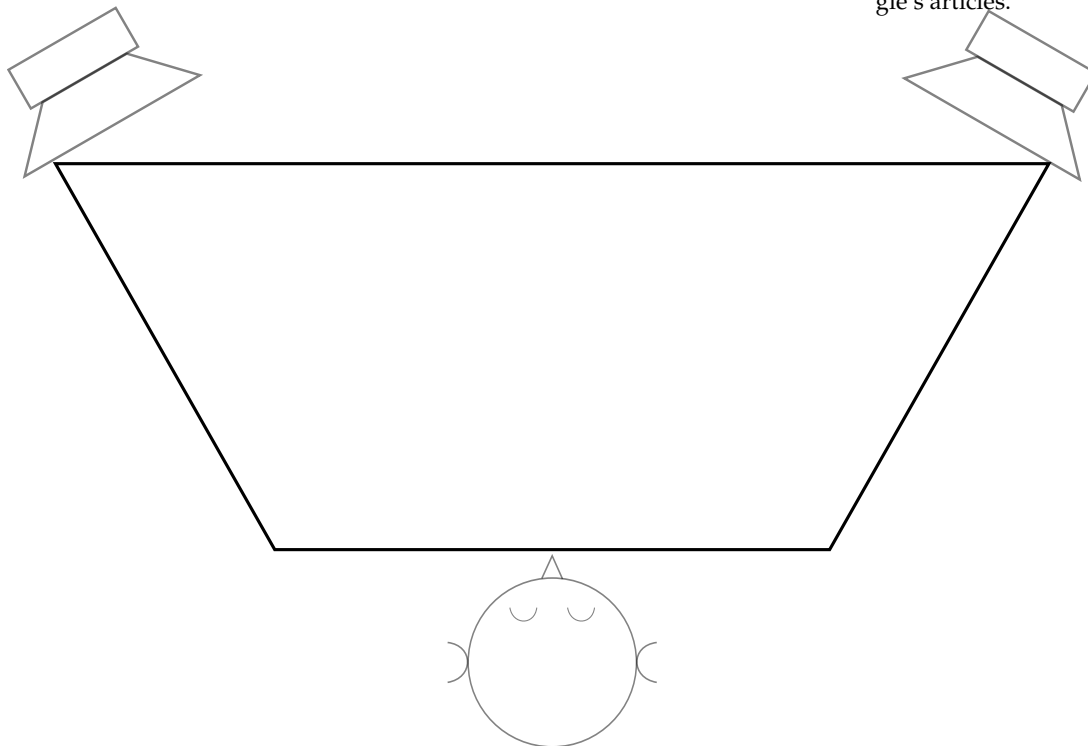
5.1 Fictional mixdown (15%)

You are asked to mix a jazz recording that features the following instruments:

- Female lead vocalist
- Male lead vocalist
- Rhythm guitar
- Saxophone
- Double bass
- Drumkit

Draw a plot of the stereo stage below that illustrates how you would approach this mix in terms of spatial balance. Indicate position, width and depth of each instrument.¹

¹ Note that the diagram is upside down by comparison to John Earle's articles.



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