Studio Exercise 1
Music 192a 2017

In this exercise, we will use the Yamaha Disklavier digital player piano to try several ways of miking a piano. In the process, we’ll also get to know the mixer and computer recording system. Since the Disklavier will play itself, we can concentrate on the selection, placement and hook-up of the microphones. The first order of business is to become familiar with the Yamaha DM-2000 mixer by reading through the mixer documentation section below. (The full manual for the DM-2000 is available as .pdf on the Data partition of the studio computer system hard drive.) We will then use Logic Pro to record and play back our sound files. But first…

Part One: Getting Sound – Meet the Mixer

Digital mixers like the Yamaha DM-2000 are modeled after analog mixers but with the ability to configure internal connections in many different ways depending on the desired use and the particular equipment to which the mixer is attached. [Please refer to Figure 3 for the locations of the mixer controls we are discussing.] The mixer has several layers of inputs that can be routed to and from six interface cards that connect to different external devices like the computer we use for recording. The layers allow the same 24 motorized faders to control 96 total inputs and outputs in four banks plus a master bank. We will focus on the first two layers in this exercise – the first layer (1-24) controls the analog inputs and the second layer (25-48) contains the digital output connections returning from the computer recording system. The layer controls are shown in Figure 3 below. Switching layers changes the targets of the display and controls accordingly. In our case, the analog inputs we use first consist of 16 microphone inputs going from the studio floor box through a snake to the mixer inputs 1-16. There are 8 further analog inputs (17-24) that are used for control room inputs including a Yamaha SY-77 synthesizer and an analog 1/8” TRS connection to user laptops for monitoring.

Since the mixer is a digital type, its internal settings are stored in memories so that it can be configured in many ways depending on its use. The configurations determine how inputs and outputs are connected and this must be selected before the mixer can be used. We have stored the configuration for this exercise as preset #66 “Music 192a St. Ex 1” to connect to the computer and main stereo speakers and this preset should be selected in the “Scene” memory section. The preset can be selected either by using the buttons in the “Scene memory” section or using the cursor in the main display window after selecting “Scene” in the display access section (See Figures 1 & 3 below.) [The scroll wheel moves the cursor to the desired preset in the main display and “Enter” selects it. Be sure the Scene Memory display is on “Recall”.] This preset routes the mixer’s Bus 1 & 2 outputs to the first two inputs in Logic so we can use the mixer controls to create a stereo sound to be recorded. Each preset saves the input and output connections associated with the particular desired use. The outputs from Logic will appear on channels 25-41 (fader bank 2).

Connection to the mixer’s microphone inputs is made through the snake box on the studio floor: the larger box connecting to the Yamaha DM-2000 microphone inputs 1-16 and the smaller box leading to the four outboard API preamps. We will start by connecting microphones to the DM-2000 directly, using inputs 1-16. Be sure to pan the mixer input channels you are using L/R or the microphones will be mixed into mono at the stereo output. (The pan controls in the preset we are using default to center panning.) Start by finding the Shure KSM141 omni-cardioid condenser microphones (use their omnidirectional setting) and Josephson C42 cardioid condenser microphones. The microphones are kept in a file cabinet in the little closet at the side of the landing at the back of the studio. Condenser mics are in the top drawer and dynamics are in the second drawer. Be sure to return them to the proper cabinet drawer when you are done. We will use the microphones in several configurations to compare their application in stereo recording.

The microphones we are using are condenser mics that require phantom power to operate. Phantom power is available on each input channel at the top of the mixing board. Since phantom power applies 48 V DC directly to the mic, it is important to turn the phantom power on and off only when the channel fader and gain
trim knob are turned down. It is best to connect the mics and cables before turning on the phantom power. Once the mics are connected and the phantom power is turned on if necessary, you can turn up the gain trim knob, starting about a quarter of the way up. Then turn up the fader on that channel and be sure the “ON” LED is on and the Insert button is in the OFF position (up). For each input channel you use, the destination should be set to “Stereo” and “Bus 1 & 2” in “Routing” section (see Figure 3 below.) As you change the selected input channel either by touching the fader or by using the “Select” button, the output assignment will change to that set for the selected channel. With the Stereo fader up, “STEREO” selected in the Monitor section, and the speaker volume control up you should hear sound now.

Once you hear sound, you can connect to the computer for recording. Each input channel will send its output to several destinations – we are concerned with two at the moment: the speakers and the computer recording system. Using the preset for this exercise, the stereo output will connect to the speakers and busses 1 & 2 to the first two digital inputs to the computer audio hardware. Once signals are connected to Logic inputs, the Logic outputs will return to the mixer on channels 25 & 26 which should be panned left and right, respectively. At this point, there will be two sets of signals going to the speakers: the “Stereo” outputs from bank 1 and the returns from Logic on bank 2. This results in an undesirable effect because the signals coming back from the computer are slightly delayed relative to the inputs, creating a phasing sound. You can now turn off the “Stereo” buttons on the input channels and monitor through Logic exclusively.

Figure 1: Output and control sections

Figure 2: Faders and pan pots
Figure 3: Mixer overview
Part 2: Using Logic X

In order to make recordings, we need to select a software platform for the computer. We will use Logic Pro X to start as it’s closely integrated with the Macintosh computer and operating system. We will need to learn how the program deals with recording the inputs. The Mac Pro computer we use has a system hard drive for the software and separate audio disks for the audio files. Users log in using their CCRMA network accounts. This means user desktops reside on the CCRMA server and not on the local system disk. **Sound files cannot reliably be recorded over the network so we must be sure we are recording to one of the local audio drives when we set up the recording session.**

We will be using Apple’s Logic Pro to begin our recording project. The mixer connects through an AES/EBU multi-channel digital interface (MOTU 112D) that also generates the master clock signal. This keeps the digital systems synchronized at the same sample rate. **The sample rate is set by Logic and routed through the Big Ben clock to the mixer and other devices.** The minimum preferred sample rate is now 48 kHz since CDs (44.1 kHz) are no longer the driving force in music distribution and 48 kHz sounds better and is the standard sample rate for sound accompanying video.

When you first run Logic X, you will need to open Logic Preferences and enable the advanced tools – click on “Advanced”. Check “Show Advanced Tools” (otherwise it’s basically Garage Band). Audio must be selected and the other options may also be useful:

![Logic Preferences](image)

Next, create a new empty project. Once the project opens, you should go to “File/Project Settings/Recording…” and choose a folder on one of the Audio drives. Create a folder with your name and store all your projects inside that folder. See figure below. From the same window you can configure the input and output hardware settings.
Then select “Audio” to set the sample rate:

Next, select “Logic Pro/Preferences/Audio” and set Input and Output Device to 112D. This connects the MOTU 112D digital interface to the AES/EBU inputs and outputs of the DM-2000 and provides 16 channels in and out of Logic from the mixer.
The session sample rate should be 48kHz. Create one new stereo track using inputs 1 and 2. Logic inputs 1 and 2 should be selected for recording – they will be fed from busses 1 and 2 on the DM-2000.

The figure above shows the main Logic window than controls the recording and playback process. Familiarize yourself with the way the window handles the various functions. Many of these displays can be customized and eventually you will want to do so but at first the minimal functions will be sufficient.
Part 3: Recording the piano

To start the piano playing, insert the Disklavier Sample disk and let it load. You can then just push “play” or change the program with the Song Select buttons. (The controls are on the small box at the lower right below the keyboard: the on/off button is on the left of the control box.) You can also change the tempo and volume if you wish. Now it’s time to experiment with different ways of miking a piano. (See Chapter 6 in the Huber Microphone book.) The main approaches to stereo miking are the spaced pair and the coincident pair. Somewhere between these are the near-coincident techniques like ORTF. An additional technique is mid-side (MS) in which a forward-facing directional microphone is combined with a side-facing figure eight microphone to produce a synthesized stereo pair by matrixing. Each of these techniques involves the localization cues to different degrees. We will compare their sounds to determine which produce the most convincing stereo image and best sound for the piano.
The spaced pair technique uses two matched microphones spaced apart to capture the sound, placed in front of the sound source at a distance approximately 1/3 to 1/2 of the distance from the center to the outer edge of the source. Spaced microphones will capture both time-of-arrival and intensity spatial cues. Spacing the microphones too far apart will cause a hole in the center of the stereo field while too close together will make the edges of the sound source seem indistinct. Spaced microphones are usually omni-directional although directional types like cardioids may also be used. The directional characteristics of the mics will determine the optimum distance apart and from the sound source.

Coincident pairs rely entirely on amplitude (intensity) differences to create a sense of spatial placement, since both microphones are in the same place there is no time-of-arrival difference. This is usually done with a pair of cardioids (or other directional mics) placed next to or above/below each other and angled from 90-120 degrees apart, depending on the size of the sound source and the distance from the mics. This technique tends to have problems with the outer edges of the sound source if the mics are too close to the source or too narrowly angled. The angle between the mics will determine how wide the stereo field seems in the recording.

Somewhere in between these extremes are the near-coincident placement techniques like ORTF. ORTF is a tradeoff between spaced and coincident systems and clearly violates the 3:1 rule, since the mics are much closer together than the rule would suggest. These techniques make use of both intensity and time of arrival information, but they risk causing comb-filtering effects if the tracks are combined into mono. For any combination of more than one microphone, you should always check what happens when you combine the channels into mono, as this will show you any comb-filtering problems that may arise.

For both coincident and near coincident mic mountings, use the stereo bar. It holds two microphones and attaches to the stand while allowing manipulation of the angle and spacing between the mics. We use quick-release adapters on all the microphone stand adapters, both to speed up mounting and to avoid cross-threaded stands and adapters. Please use these quick-release mounts and don’t remove them.

If you can’t make up your mind or wish to be able to alter the stereo sound field after recording, you can employ the M-S system. This system uses a directional mic facing the source (M) and a very close figure-eight mic with its null facing the source (S). These may be recorded to separate tracks and added together on the mixer to produce a phantom coincident pair with an adjustable angle between them. This is accomplished by panning the M to the center and assigning the side mic to two separate channels on the mixer, one of which is set out-of-phase with the rest. The side mic channels are panned hard right and left. By varying the balance of mid to side mics in the mix you can widen or narrow the stereo image. Another advantage of this technique is that if you collapse the stereo to mono there is no chance of phase problems since the side mic channels cancel, leaving just the mid mic. The DM-2000 allows adjacent channels (1,2; 3,4; etc.) to be paired and decoded as M-S. See page 103 of the DM-2000 manual to see how to select the M-S pairing and choose which input is set to M.
Shure KSM-141

AKG C414XLS

Two of the available multi-pattern condenser microphones are shown above. Both require phantom power; the C414 function switching only works when phantom power is on.