Audio Studio Primer

by Jason Charney – JHU Digital Media Center – Last Revised: September 14, 2023

The objective of the Audio Studio Primer is to familiarize you with the studio signal setup, get audio flowing using the USB interface and DAW software, studio instruments (routed through the patchbay), microphones, MIDI-controlled software synthesizers, and basic recording/export. **It is not to provide a primer on digital audio theory or comprehensive software/production tutoring.** This can be done through the other Audio Path primers and topic workshops. Check on HopkinsGroups to register for these events, or reach out to Jason Charney, the Multimedia Specialist in Audio at charney@jhu.edu if you need one-on-one help.

*Text in this style indicates notes to staff completing the authorization. Patrons completing the authorization should be **"in the driver's seat"** – that is, they do everything hands-on as directed by staff. Groups of patrons can rotate completing each task so that everyone gets a chance to do something.*

Part 1: First Steps

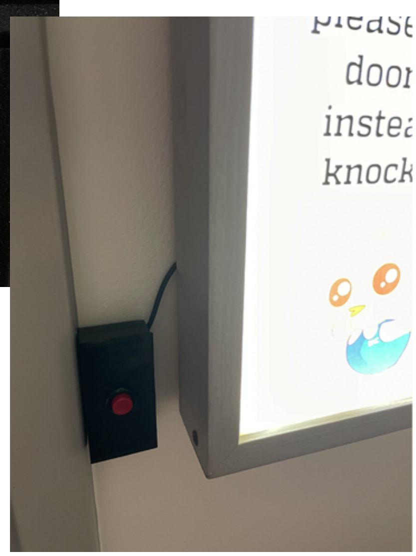
Make sure the Audio Studio is clean, set to its "Default State," and locked before beginning the training. Reserve and check out the Audio Studio in BookIt.

*When patron(s) arrive, make sure they're a member of the DMC. If they're not, they can do the new member sign up after the audio studio training. **Ask the patron if they have any experience** in digital audio, music, recording/editing, etc. and what their interest is in using the audio studio. Every patron must complete every part of the training process even if they're not interested in a particular technique, but this information may help guide the pace and depth of explanation.*

You can book the Audio Studio for as long as you want when the DMC is open, but **be mindful and use it for audio production only**, and not just a quiet place to do homework.

The **front desk staff unlocks the studio when you check it out**. Let the front desk staffer know you're done using it so it can be locked behind you when they leave.

There is a lighted sign outside the studio. Flip the switch by the door to turn it on when you're in the studio. This lets staff know you're in there. If we need to get your attention, staff can push the "doorbell" that flashes a light on the desk.



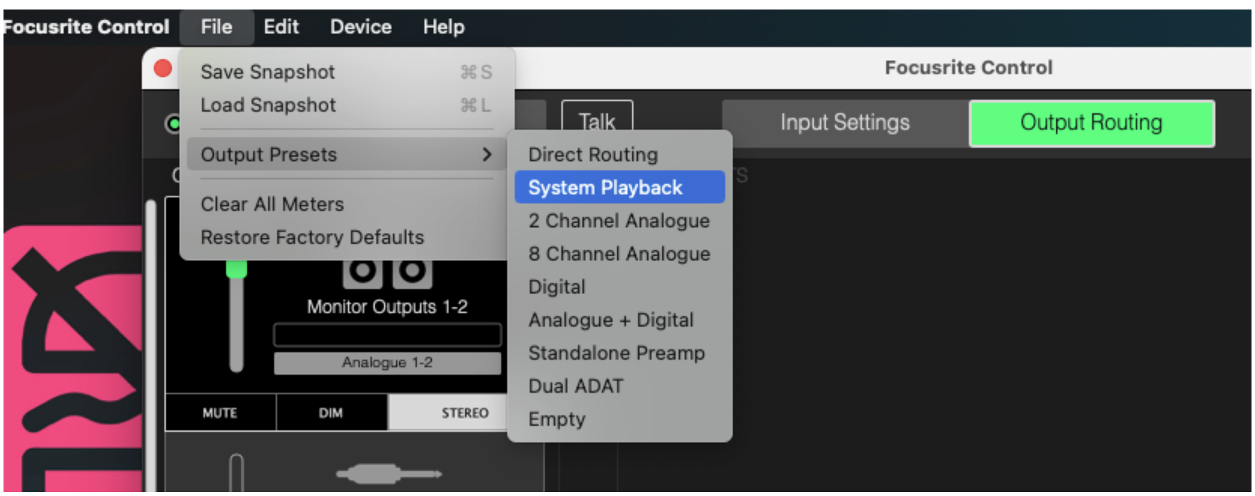
Part 2: Turning Things On

All audio in the studio runs through the equipment rack just to the left of the desk. **Flip the switches on the top power strip from left to right** to power all the gear in the rack. This order is important, as it makes the speakers come on last and prevent loud pops as other gear turns on.

Log into the Audio Studio computer using the **AudioStudio user credentials** that are posted on the monitor. Since all users share this account, it's important to **back up your work** to an external hard drive or cloud storage service since other users may accidentally delete it. You may want to make your own named folder on the desktop to keep your work in one place.

When you log into the AudioStudio account, you'll see the **Focusrite Control** software window open. This software configures the input and output routing for the **Scarlett 18i20 USB interface** located in the equipment rack. All microphones and instrument signals flow into the computer and out to the speakers through this USB device.

Advanced users can reconfigure the signal routing for special purposes in this software. In case a previous user has changed things around, you can reset it to the "default" state for normal use, click **File → Open Snapshot... → System Playback.** However, Focusrite Control does not have to be open while you're working in other software.

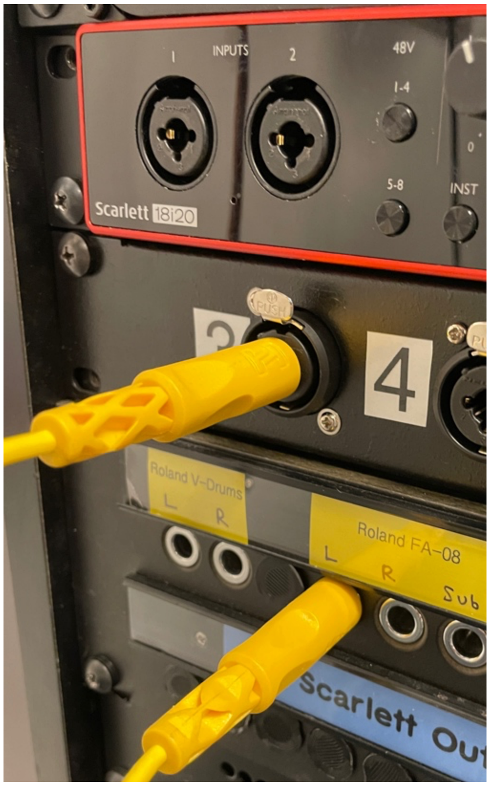


You can **use the Scarlett with your own computer** if you install [Focusrite Control]() on your own Mac or PC and unplug the USB cable labeled "Scarlett" from the studio Mac. If you do this, you **must replace it in the studio Mac's USB port when you're done using it.**

Part 3: Using the patch bay and setting levels

The Audio Studio has **four electronic instruments**: a Roland FA-08 keyboard, Novation Ultranova synthesizer, Roland V-Drums electronic drumset, and a Korg MS-20 synthesizer) are **wired into the patch bay already.** The patch bay is the two rows of ¼" jacks on the lower part of the rack.

Have the patron choose one of the instruments to turn on for the recording demo.



The **"Studio Instrument Output" jacks** (yellow label) correspond to the *outputs* of each of these instruments, labeled above each jack. Take one of the colorful **1/4" patch cables** hanging next to the rack. Plug one end into the appropriate studio output jack and then plug the other end into any input on the Scarlett.

There are 8 **input jacks** on the Scarlett. Two of them are on the front of the Scarlett itself, and 5-8 are in the row below the Scarlett.

← *Roland FA-08 left channel output plugged into Input 3 using a patch cable*

Play a few notes on the instrument (or hit the drums) and watch the **LED level meters** on the Scarlett to show signal flowing.

When you plug a sound source cable into the Scarlett, you'll need to adjust its level. After you've made sure the **output volume knob on the instrument itself** is turned up, the eight **gain control knobs** on the Scarlett can *boost* or *attenuate* the incoming signal for each of the inputs. As you make sound, turn the gain knob so that the **loudest sound you make** *peaks* at 3-4 green LEDs (between -6 and -3 dB). If there's a red LED, it means the signal is *clipping* and they need to turn down the gain. It is always better to be a little bit too quiet than to clip at any point during recording.

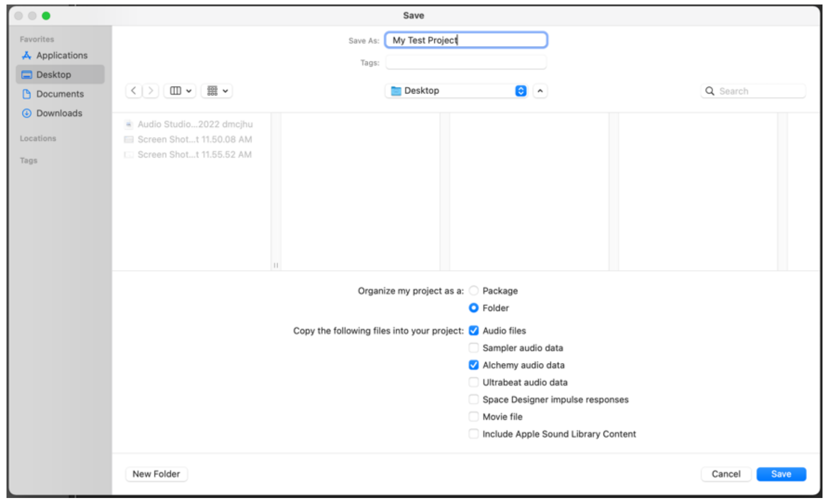
Note: setting the correct level is important for *any* instrument or microphone you plug into the Scarlett. However, the rack patch bay is only used for the four studio instruments. You can plug other instruments or microphones **directly into the Scarlett's eight inputs.**

Part 4: Setting up a project and recording

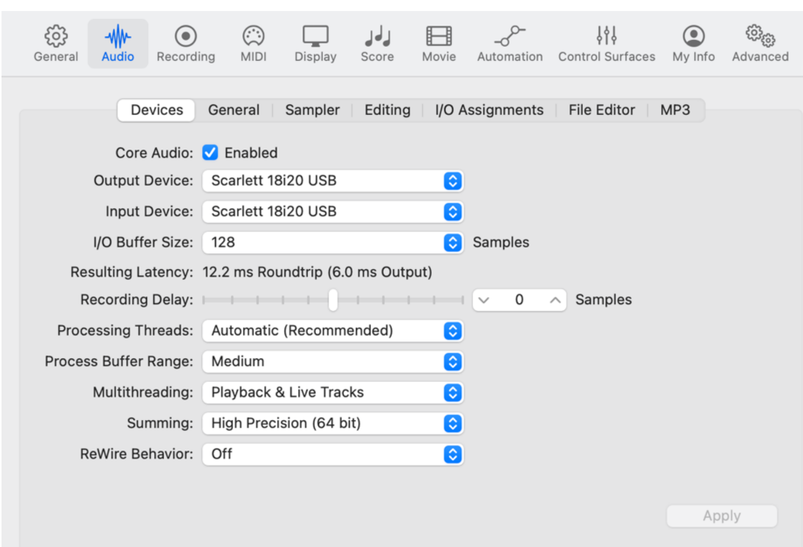
Now, we can see signal flowing into the Scarlett but can't hear anything yet because we **need to have our digital audio workstation (DAW) software open.** This primer will continue using **Logic Pro** as our DAW, but the other DAWs we have installed on the Audio Studio computer — FL Studio, Ableton Live, Studio One, or Reaper — can also function in the same way.

Open up Logic Pro and create a new **Empty Project**. A window will prompt you to **create a new track.** Go ahead and click "OK" without configuring any settings.

Select "**File → Save"** from the menu bar. In the prompt, select "Organize as a Folder" option and give the file a unique name. A folder is created for the project file itself (with extension .logicx), and the **audio files you record and import are saved in subfolders in this project.** It's important that these subfolders always stay together with your project file. The Logic project file itself is quite small; it just points to specific audio files on the hard disk. When you play back audio, the software is reading the files from disk from the appropriate time point in the file, processing them through effects, and being played back in real time through your speakers.



If you import audio files (e.g. downloading sounds from the Internet), put them in the project folder *before* adding them to the Project. When you backup their work to open on another computer, **copy the WHOLE folder** and not just the .logicx project file. This principle applies to any other DAW and its project files as well, not just Logic Pro!

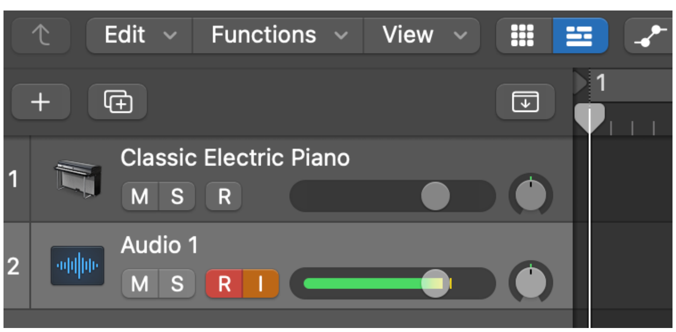


Next, select "**Logic Pro → Settings → Audio...**" in the menu bar. Make sure that "Scarlett 18i20 USB" is both the Output Device and Input Device.

Now we can actually start recording audio. Click **"Track → New Audio Track..."** in the Logic menu bar.

The **Inspector** on the left side of the Main Window shows information about whichever track is highlighted in the **Main Window** (the track sequencer). Press ⌘I if the inspector is not visible.

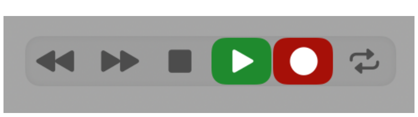
Select the input channel for where you plugged in the patch cable in the Inspector using the drop-down **Input menu** – Channels 1-8 are the 8 input jacks on the Scarlett and the panel below it.



Next, **record-enable** the track (red R button on the track) and turn on **input monitoring** (orange I button on the track) and play some notes on the electronic instrument. Finally, some sound! Careful, it may be loud.

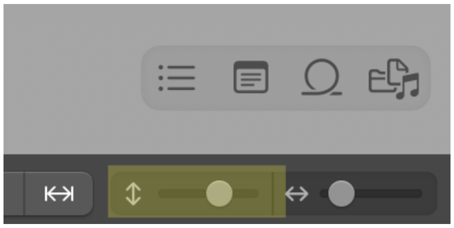
The **Monitor volume knob** on the Scarlett controls the output volume of the speakers but that they should make sure that the software gain meter is still in the green! Don't confuse the **input gain/level** for the output **monitor volume.**

Now it's time record a few seconds of audio.

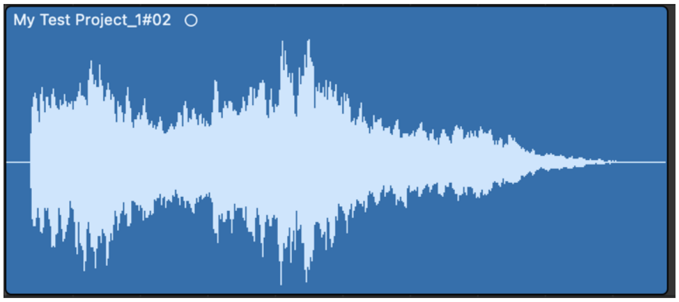


Press the **Record** button in the **Transport** (the playback controls at the top of the Main Window).

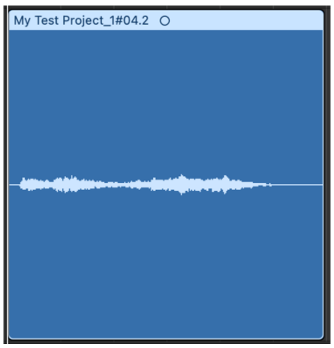
If the Metronome is annoying, you can press **K** to toggle it on/off ☺



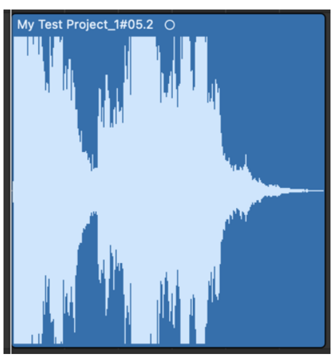
After recording, press Spacebar or stop button in the Transport. Now take a look at the *waveform* you recorded on the track — this is a visual representation of the recorded sound. To get a better look at the waveform, you can increase the height of each track using the **Track Zoom controls** in the upper right of the main window. Pull the vertical arrow slider to make each track "thicker" on the Main Window screen.



Here's a healthy waveform example. There is a big range of difference between silence (represented by the thin line in the middle of the waveform at the beginning and end) and the ***peak amplitude***, about halfway through.



This waveform is ok, but it's *quiet*, with **low dynamic range.** There's not much difference between silence and its loudest point. This can be fixed by turning up the **output volume** of the electronic instrument and/or turning up the gain trim knob on the Scarlett interface (or getting closer to the microphone, if you're recording something quiet with a mic).



This waveform is **clipping**, meaning its peak amplitude levels are not being accurately captured (notice the flat tops and bottoms?) This means the gain is too high and it will sound distorted when played back. Clipping corresponds to hitting the "red" portion of the level meter.

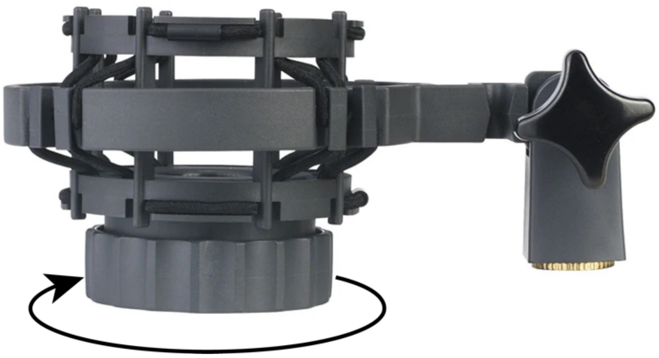
Note that it's **always better for a recording to be a little too quiet than for it to be clipping:** you can make a recorded audio file louder, but you can't easily fix the distortion in an audio file that has clipping.

Awesome! You've made your first recording! Now click "**File → Save...**" in the Logic menu to save your project and continue.

Part 5: Recording with a microphone

If there are multiple patrons in the training session, have one person be the "performer," adjusting the microphone position and singing/speaking into it, and the other be the "recording engineer," setting the levels and pressing record.

There is a matched pair of AKG C214 microphones that live in a box on top of the rack. These microphones are great for recording voice and acoustic instruments with detail and clarity. You can always check out other microphones, cables, and stands from the front desk or bring your own into the studio.



Open the box and screw a **shockmount** onto the mic stand. Place the microphone in the cradle, and tighten the ring around the base.

The front of a microphone is where the *logo* emblem is. Face it to the sound source when you record.

Take an **XLR cable** from the wall and plug the **female end into the mic,** and the **male end into an input** on the Scarlett. Note: it should "click" into the microphone but won't click when you plug it into the Scarlett.

Adjust the mic stand so it's at a comfortable position to record your voice. Typical microphone placement should have the sound source (i.e., your mouth) 3-4 inches away from the front of the mic.

With multiple patrons, the "performer" can stand behind the "engineer." For a single patron, have them adjust the microphone height and position so they can both operate the computer and record their voice.

Make a test sound close to the mic like a snap or clap and watch the LED meters on the front of the Scarlett like you did with the studio instrument in the previous part. However, you shouldn't see any of the signal LEDs light up – what's going on!?

The AKG C214s are *condenser* microphones. This kind of microphone requires **phantom power**, a 48V constant charge that is supplied by the *preamplifier* (the circuitry connected to each jack on the Scarlett). Turn phantom power on using two buttons on the front of the Scarlett, between the Channel 2 input jack and Channel 1 gain knob. You can toggle phantom on and off for inputs 1-4 and 5-8 – it won't hurt anything else that's plugged in. If they are not getting signal when a microphone is plugged in and the gain knob is high, this is the first thing to check!

Once phantom power is turned on, setting levels for the mic is just like setting them for the studio instruments: **make the loudest sound they're going to make and never hit the red LED** at the top of the meter.

Start speaking/singing in front of the microphone and adjust the gain accordingly.

Now in Logic Pro, create a new Audio Track and select the correct input for the mic you just plugged in. Arm it for recording so you can see the signal coming in, but **don't turn on input monitoring** (orange "I" button) **yet!**

If you monitor the microphone through the speakers, the microphone will pick up its own output and could create unpleasant and dangerously loud **feedback.** This is why having headphones in the studio is helpful: we can mute the speakers and listen to the computer output without it being captured by the microphone.

Put on the studio headphones. This pair hangs on a hook on the wall, but you're welcome to bring your own or check out a different pair from the front desk.

There's a speaker **mute** button on the front of the Scarlett. Engage the button to mute the studio monitor speakers.

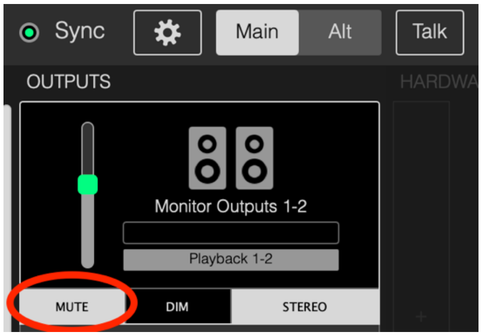
Play back the track you already recorded and **adjust volume** of headphones to comfortable level.

Now, you are safe to turn on input monitoring to hear your own voice through the headphones as you record.

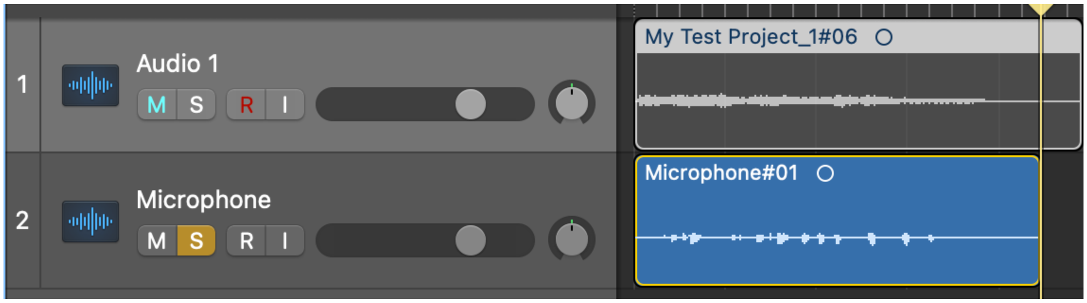
Record a short snippet of voice over the existing instrument recording. As before, look at the waveform after done recording to make sure it looks healthy!

Now, *disarm* the track for recording by toggling off the R and I buttons.

Now it's safe to unmute the speakers. Another way of muting/unmuting the speakers is using the **Focusrite Control software** instead of the button on the interface.



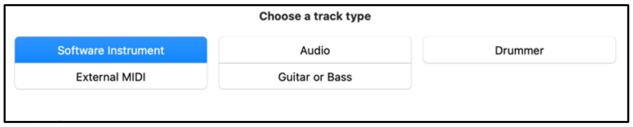
When listening back or recording, you may want to hear tracks independently from one another. You can Mute individual tracks to silence them, or Solo them to mute all other tracks. Press the yellow S button on the vocal track you just recorded to Solo it and listen to it play by itself through the speakers.

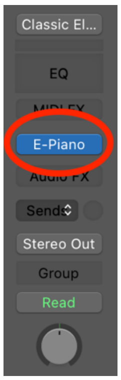


Part 6: Software Instruments

In the final part of the primer, you're going to learn about **software instruments,** which are ***plugins*** – apps that run within a ***host*** **DAW** like Logic. Logic comes with a lot of great free instruments, but the DMC has extra installed software instrument plugins available across all DAWs.

Select **Track → New Audio Track** from the menu bar and create a new **Software Instrument** track (*not* Audio track like we did before).



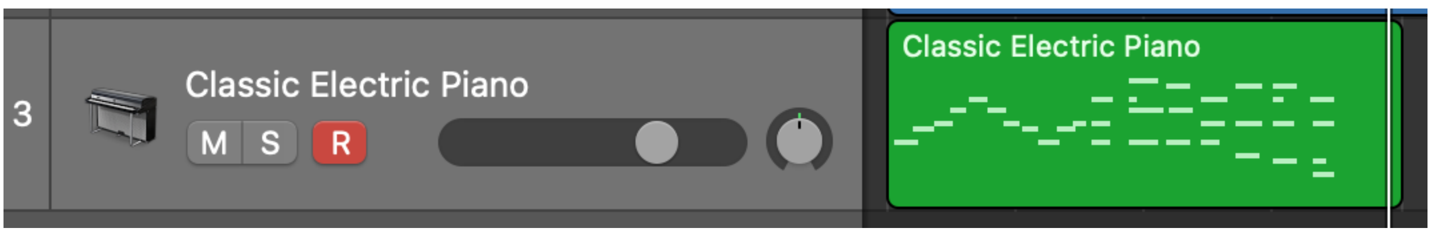


The "input" to this track is the virtual instrument plugin itself, instead of a physical jack on the Scarlett interface.

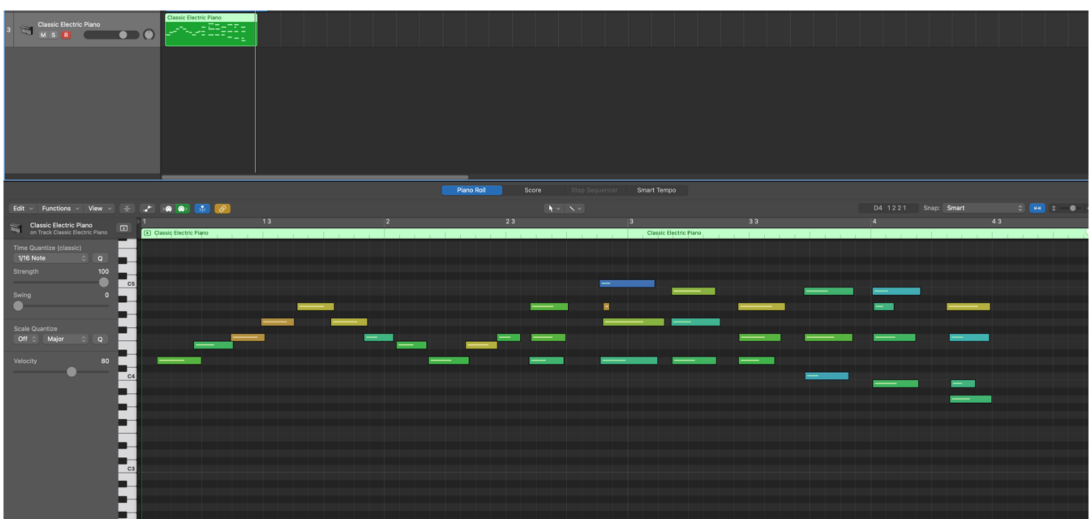
You can play the instrument using **MIDI**, a messaging protocol for controlling both software and hardware synthesizers. Any of the electronic instruments in the Audio Studio can send MIDI to the computer – you can even play software instruments using the electronic drums!

Make sure the track is record enabled (red R on the track). Play a few notes on the keyboard of the studio instrument and listen to the sound that comes out.

Now, hit Record. Play and record a little bit, and then stop recording. You'll see that instead of a waveform being recorded, we're getting a bunch of parallel horizontal lines in the **clip region**.



Double-click on the clip you just recorded. This opens the clip's ***piano roll*** viewer.



Each **individual note you recorded in the piano roll** contains a *pitch* (*which* note on the keyboard you played: bottom to top on the piano roll = left to right on the keyboard = low to high) and a *velocity* (how fast/hard you pressed the note. Most synthesizers will scale the volume and timbre of the resulting sound if you play softer, just like a real piano).

You can **edit the notes in your performance,** adjusting each note's pitch and duration individually in this view, and change their velocity (represented by the color in the note bar).

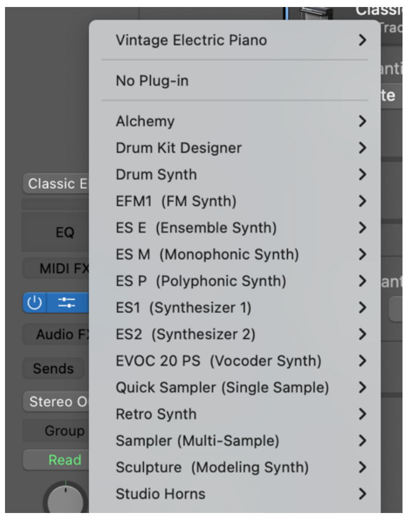
You don't even need to record using the keyboard first if you prefer to just draw in notes directly to the Piano Roll! Hold the Command key (⌘) to switch to the secondary Pencil tool, then click in the piano roll to add notes.

Unlike the signals you recorded as audio waveforms, the software instrument "performs" in real time when you press play. Logic passes the MIDI note/velocity information into the plugin and renders it into audio. Why is this useful? It means we can change the software instrument and **play the same performance using different sounds** at any time.

Double-click in the middle of the instrument name in the Inspector channel strip to open the plugin's interface.

Here is the default *patch* (synthesizer parameter settings that make a particular sound) for the sound. You can adjust the parameters of the software instrument in this window, or choose from many included presets in the dropdown menu. Every Logic instrument includes several preset combinations of parameters.

You can choose a new instrument using the input select menu in the Inspector – hover over the right side of the blue plugin name and click the up and down arrows to bring up the software instrument list.



Select another one such as the "Retro Synth" and you'll see the plugin window interface change.

Press play on the Transport to hear your performance through the new instrument!

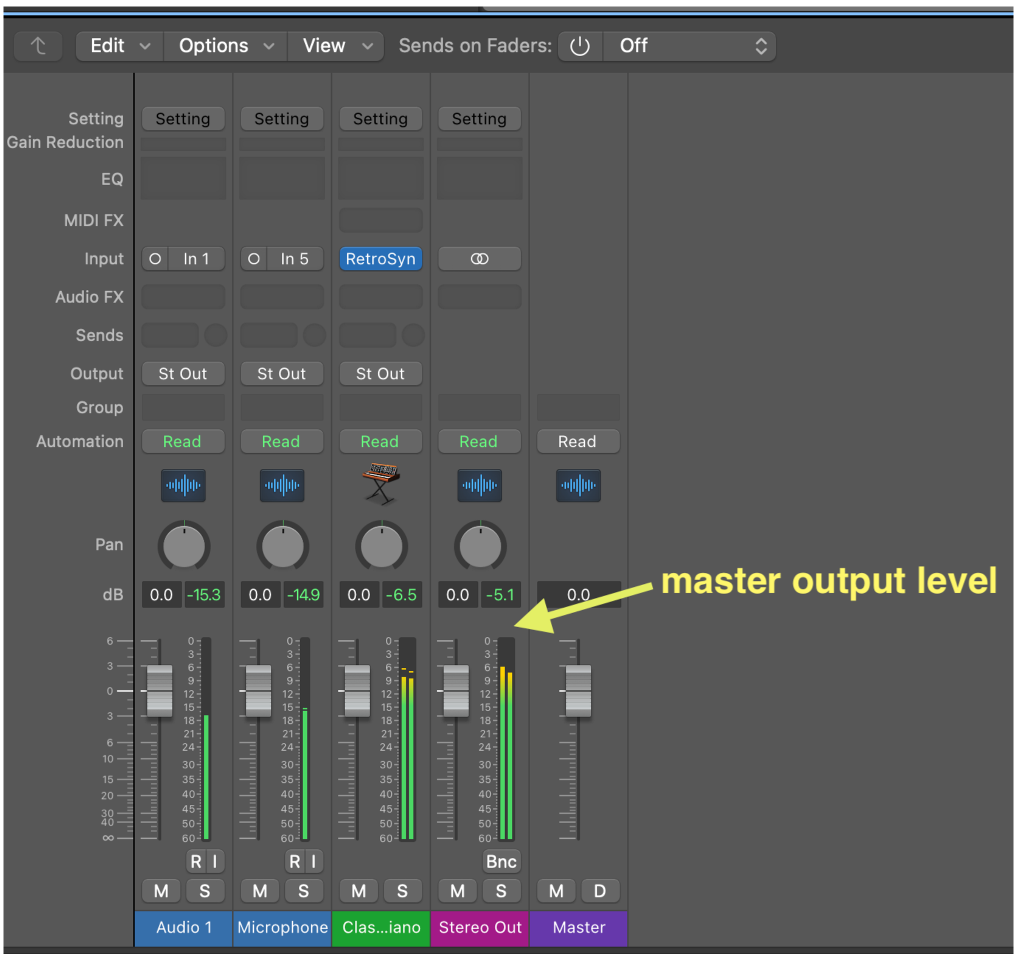
Part 7: Exporting

Now it's time to render your whole mix as a single audio file: recorded instrument, vocal track, and software instrument.

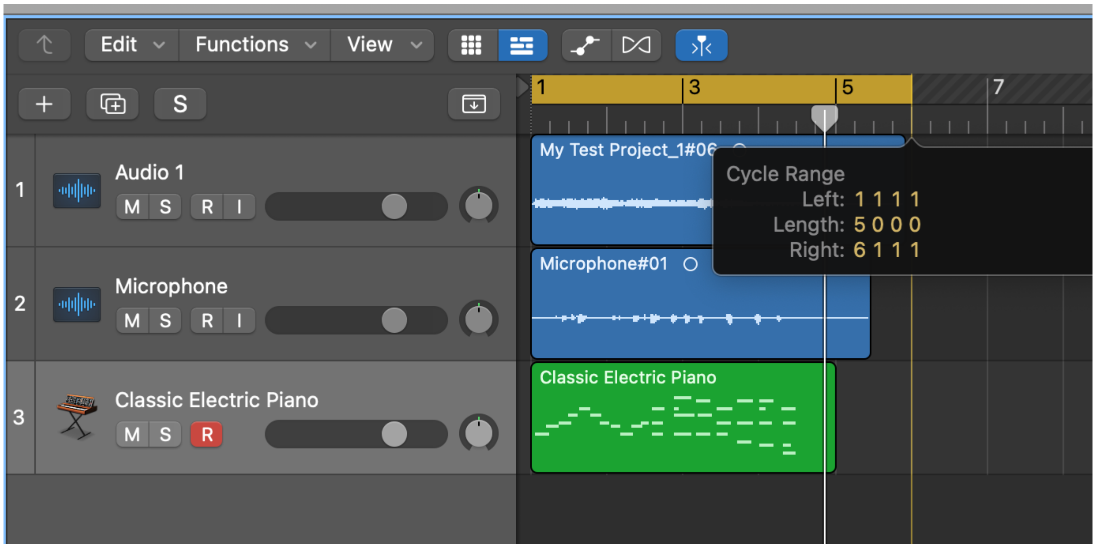
You were careful to make sure that each signal you recorded was not clipping on input, but it's important to make sure the **sum** of all your signals **does not clip on output:** adding together to be too "hot" and leading to distortion and unwanted artifacts.

Press X to bring up the Mixer subwindow. We can see each track's levels side by side in this view, and the overall **Master output.**

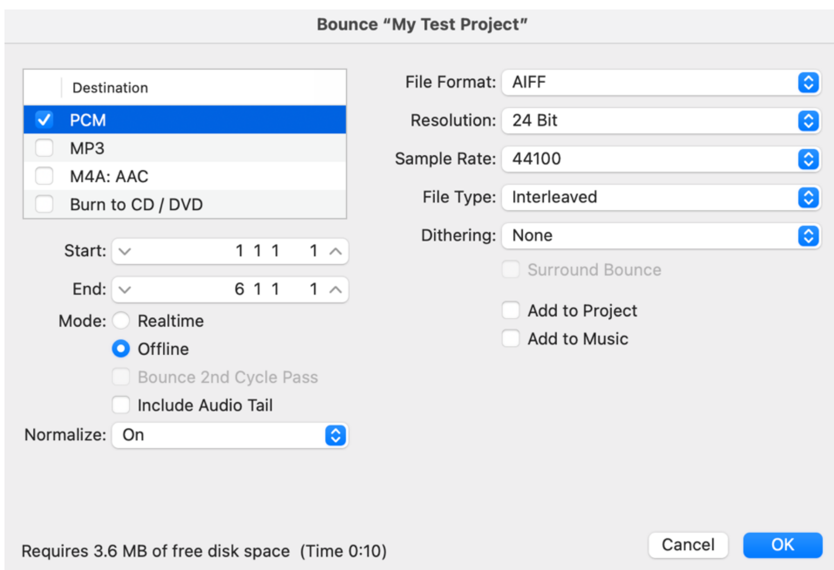
When you press "play," watch the meters on the Master output. They should never hit the red! You can mix the individual track levels to balance them and especially make sure that your **overall Master output never clips** (hits the red zone at the top of the meter).



Now, let's export our whole mix — in Logic, this is called **Bouncing** the mix ("export" is specifically for rendering individual clips). Select the time range you want to export by dragging your mouse within the time ruler at the top of the Main Window. An orange region ("Cycle Range") will set the time boundaries of your track. **Make sure that you don't have any tracks (unintentionally) soloed or muted!** The bounced file will reflect these track states.



Next, click **File → Bounce → Project or Section…** in the menu bar.

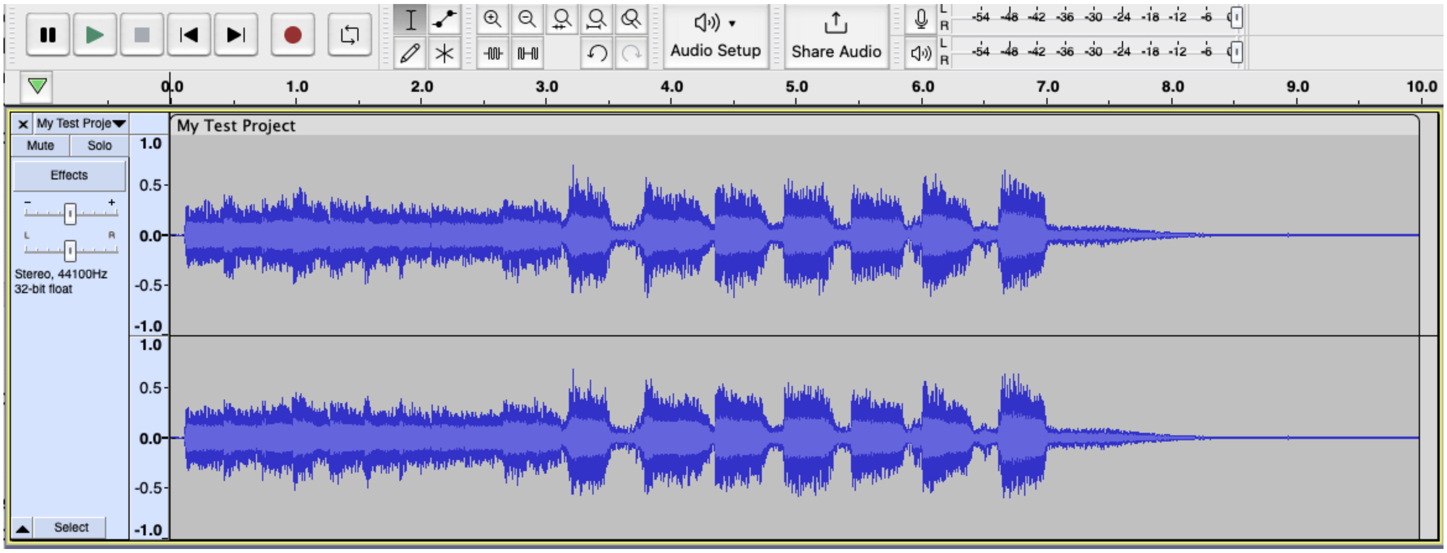


You can configure several things about your bounce here.

- **File Format:** make sure that it says "AIFF" or "WAV." These are **lossless** formats that are the highest quality audio (vs. mp3 or other compressed formats that degrade the quality of the audio in order to save space).
- **Normalize:** With this selected, the final file will be adjusted so that the loudest point of the rendered file hits 0 dB (the highest possible amplitude value to store in a digital audio file). Recommend turning this OFF as you can always normalize the file later if you need to.
- **Include audio tail:** Toggle this "on" especially if you have effects like reverb on any of your tracks. It will allow the rendered file to go a bit beyond the time selection and allow sounds to fade out naturally rather than sharply cut them off.

- **Check the Time** in the bottom left of this window to make sure it's the length of the track you intend to export!

Now, click OK and save your file! It's recommended to then **open and listen to the file** to make sure it bounced correctly, isn't too quiet or too loud, and doesn't have extra unintended silence at the beginning or end. You can drag your bounced file onto a new track in your existing Logic project and Solo it, or use another program like Audacity (a free audio editor) to open it and look at the waveform. Here's the bounced track in Audacity.



Notice that it's **stereo** (it has two channels: left and right). It also **does not have any clipping** and has relatively healthy levels overall. There's some silence at the beginning and end that could be trimmed off, but nothing too long.

Great job! Send an email to dmcstaff@jhu.edu with your exported WAV or AIFF attached.

Confirm the patron has completed the training and add them to the authorized users in Active Directory. If they're done with the studio, make sure that they shut it down and return it to the default state. Lock the door and check out of the space in BookIt.

Glossary

Amplitude – the "strength" of a signal, how much a signal deviates from silence or equilibrium (and therefore how energetic the signal is). In audio, a higher amplitude corresponds to a louder sound. Measured in decibels (or dB). In digital audio, you'll see meters have a maximum of 0 dB (with operable ranges in negative dB). 0 dB represents the highest amplitude possible in a digital audio system.

Clipping – When a signal's amplitude is too high to be accurately captured and causes distortion. You can tell it's happening if the tops of the waveform in any of your recorded regions is tall and flat (aka "clipped"). Whenever you see a meter hit "the red" at the top, it means clipping is occurring.

DAW –Digital Audio Workstation. Any audio editing and mixing software such as Logic Pro, Ableton Live, FL Studio, Reaper, Adobe Audition, Pro Tools, etc.

Dynamic Range – the difference between the lowest amplitude and highest amplitude in an audio file, experienced as the "quietest" and "loudest" points. Context-dependent: could refer to a portion of a single instrument track in a song's mix, the whole song, or even songs across a whole album.

Gain – the amount an audio signal is increased by an amplifier. Adding gain (such as using the knobs on the front of the Scarlett interface) essentially makes the incoming signal louder before it is digitized and stored in the computer. You can also add gain to a file after you've recorded it.

Master Output – the single stereo track through which all other tracks are summed and output to your speakers and headphones. When you bounce or export a mix from a DAW, this is the track that is rendered.

MIDI – Musical Instrument Digital Interface. A messaging protocol that allows software and hardware synthesizers and samplers to communicate, either over 5-pin MIDI cables or USB. The most common MIDI messages often combine a note (the key played on a keyboard controller) and velocity (how hard a key was pressed) to capture performances.

Mono – an audio format that has only one *channel* of audio. When you play it through speakers or headphones (which have two channels, left and right), it plays at equal volumes through both of them.

Normalize – A process in which all amplitude values of a recorded audio file are analyzed, and then boosted proportionally so that the highest amplitude value reaches 0 dB.

Patch Bay – a studio tool consisting of several audio jacks that expose all input and output points in one place, even though they may be connected to instruments around the studio. In the DMC's Audio Studio, each Studio Instrument's output can be easily connected via short ¼" cables to the inputs of the USB audio interface.

Phantom Power – a 48V charge that can be supplied through an XLR jack to a condenser microphone, which uniquely require it in order to pick up sound.

Plugins – software applications that can run inside a "host" DAW and process digital audio. They can be used for effects processing of recorded audio, or may be software instrument sound sources themselves. The Audio Studio's computer has several special plugins available to all DAW software, and each DAW has its own suite of built-in plugins.

Stereo – an audio format with two channels of audio, frequently experienced as "Left" and "Right." To record something in stereo, you need to record with two microphones simultaneously.

Transport – the area in a DAW that has the play, stop, rewind, record, etc. buttons. May also display timing information about the project.

Volume – the level of gain applied to the *output* of a signal without regard to what's actually recorded in your software. You could turn up a quiet recording very loud, or listen to a loud recording very quiet.

XLR Cable – The three-pin cable usually used for microphones. The "female" end plugs in the microphone and the "male" end plugs into the mixer or interface.

¼" TS cable – stands for "Tip/Sleeve," a cable that carries a mono signal. The ¼" diameter format is used for guitar and other electronic instruments, and the patch bay in the DMC's Audio Studio.

Resources for Further Learning

DMC workshops – register on HopkinsGroups

LinkedIn Learning courses

Learning Synths – Ableton

[youtube people]

Audio Studio on DMC Discord