Introduction to Digital Music Production

Mac OS X, Logic 9 Express or Pro

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Introduction to Digital Music Production
using Logic Express or Pro 9 on Mac OS X
Course Basics

GOALS By the end of this Introduction, you will:

- Be familiar with the course schedule, student responsibilities and procedures for submitting assignments.
- Have prepared for class by familiarizing yourself with Laulima, the MUS 240 DropBox and Vimeo (refer to Syllabus for web addresses and access instructions)

ASSIGNMENTS To Do (for grade points):

- Read the course syllabus and schedule carefully
- Submit signed “Permission to Use Materials” and “Vimeo Streaming” forms
- Set up a free account on Vimeo and comment on the “MUS 240 Welcome” video
- Log into Laulima to access the MUS 240 Resource folders (everyone), Quizzes (everyone) and Discussion Boards (online students only).
- Submit a blank document to the MUS 240 Dropbox to ensure that you’re able to connect
- Review the study guide for Quiz 1, then take Quiz 1 in Laulima

For online students only:

- Install additional loops into Logic. The Addnl_Logic.zip files are in Laulima’s Resource folder.
- If using a KB37 keyboard, follow the instructions in the Line 6 Home Setup handout.

Having Problems? Go through the AudioMIDI, Signal Check and SoundPrefs handouts

Goals and Assignments will be listed at the start of each lesson.
Both on-campus and online sections of the course will be going through the lessons at the same time. On-campus Students can opt to complete assignments online, and Online Students can opt to attend class for in-person interaction. Whichever you choose to do, it is your responsibility to ensure that you’re fulfilling all course obligations by completing assignments on time. Read carefully through the course syllabus and schedule, paying particular attention to the sections on Student Responsibilities and On Campus Locations for getting your work done.

As noted in the syllabus, all homework and project assignments are to be uploaded to the MUS 240 Dropbox on the WCC Mac server. I’ll retrieve and “viditize” them (turn each into video) and post them on the MUS 240 Vimeo site. This way, all students, on-campus and online, will be able to review and comment on everyone else’s musical masterpieces. Projects will also be posted on the WCC website and be accessible to anyone with an internet connection; you can share the site link with friends and family far and wide!

Agreement Forms

The posting and sharing of student files requires the signing of the Permission to Use Materials and Video Streaming agreements. Online students, send in your agreement via email as instructed in your syllabus. On-campus students, complete and sign the forms; if you are under 18 years of age, please have a parent also sign.

Vimeo

After creating your account on Vimeo, watch and comment on the MUS 240 Welcome video. Send me your Vimeo Username for participation credit.

Laulima

Laulima is the University of Hawai’i system’s online collaboration and learning management system. It’s used by both on-campus and online courses. All documents referenced in this text (the syllabus and schedule, and agreement forms, for example) can be found in the Resource section of Laulima. Folders are organized by Chapter. Log into Laulima to retrieve documents to print out and keep in a binder for reference. You’ll also find larger versions of the videos that appear in the text. You can download these videos to watch on your computer, tablet or smartphone. While in Laulima, be sure to locate the links to Quizzes (all students) and the Discussion Boards (online students only). You’ll be using Laulima often.

MUS 240 Dropbox

As noted in the syllabus, the MUS 240 Dropbox is accessible from any Mac computer connected to the internet, from anywhere. Follow the directions in the Quick Reference section of the syllabus to upload a text document with your name as its title to the Dropbox.
Quiz 1 Study Guide

- Your grade in this class depends on your performance on what types of assignments?
- Where can you find handouts, movies and other course materials that are referenced in the text?
- If you want to do class work on the WCC Campus, in which three places can you find computers with Logic installed?
- To where (or what) do you submit completed work?
- Do you have to write papers for this class?
- What professional DAW software are we using in this class?
- Where will quizzes be taken?
- How much time do you have to complete a quiz?
- Is the MUS 240 Studio in Palanakila 103 soundproofed?
- What are the username and password for the MUS 240 Dropbox server?
- If you lost the key to the MUS 240 Studio what will happen?
- If you do not check your UH Email regularly, what should you do?
- Which video web-streaming service is the class using?
- What are we using the web based video service for?
- Where do you check out the key for the MUS 240 Studio on the WCC Campus?
- Can students move freely between the on-campus and online sections of the course?
- Can students turn assignments in late?
This chapter provides an overview of what’s involved in digital audio production, as well as things to ponder throughout the semester and beyond.

Live sound is infinite in frequency and amplitude (pitch and volume), but audio recording technology is limited to capturing and reproducing just a percentage of that range. Even with continual evolution in technology, our recordings are still missing a lot of “infinity” — frequencies that we may not be able to hear but undeniably feel, and which enhance our perception of a musical event. How does this affect music production?

In audio recording, we’re cramming sounds of infinite frequency and amplitude into a tiny box (audio file). For an analogy, imagine that you’re in a park, having a conversation with a few friends. Throughout the park are groups of people having conversations within their cliques, also numerous trees with their leaves chirping, and a couple of park workers clearing walkways with leaf blowers. The open space and distance allow you to concentrate on your own group’s conversation.
Now take all of those soundmakers and sounds, and put them into a small classroom. Voices from other groups interfere with yours, making it difficult for you to understand what your friends are saying. On top of that, the sound of wind, leaves and birds have been condensed and layered upon each other; sound that what were relaxing outdoors are now intense and irritating. But worse, once the leaf blowers start up, you can’t hear anything else.

If we were recording the park scene, how do we keep individual sounds, including the aural space of the park, distinguishable from one another? How do we faithfully reproduce what we experienced in the park?

Knowing what we’re working with is one place to start. Creating recordings that sound good have a lot to do with skillfully manipulating frequency and amplitude of sounds. The professional digital audio workstation (DAW) is one tool for this job, and is comprised of two main parts: Audio and MIDI.

In music production, we are attempting to recreate, within finite space and technology, audio events of infinite frequency, timbre, amplitude and space.
GOALS
• Understand how sound is generated
• Know how sound is measured
• Know the range of human hearing

TO DO
• Read the lesson and follow links to the websites and videos referenced:
  • Fear of Physics - What Sound Is
  • University of Oregon site
  • Short video “What is Music?”

“A person does not hear sound only through the ears; he hears sound through every pore of his body. It permeates the entire being.”

— Hazrat Inayat Khan
1.1 How Sound is Made

Sound is created when air is made to vibrate, creating fluctuating levels of air pressure. When the fluctuations are in regular, repeating cycles, pitch is generated. Here’s a graphic representation of one cycle of vibration.

Visit the Fear of Physics site for an illustrated explanation of how sound is created and perceived by air pressure. The speed of the pressure waves creates different pitch of sound.

Pitch is measured in cycles per second, or FREQUENCY, in units called HERTZ (abbreviated Hz). The more cycles per second, the higher the pitch; for example, a tone of 100 Hz (100 cycles per second) will be a higher pitch than a 20 Hz tone. In the following figure, the top waveform has the lowest pitch, and the bottom waveform has the highest.

Listen to audio examples of a sine wave at low, mid and high frequencies at this University of Oregon site. (A sine wave is the simplest type of wave, with no overtones. Multiple overtones are what give instruments their characteristic timbre, or sound quality. Since sine waves have no overtones, they are considered “pure.”

The range of human hearing is 20 Hz to 20,000 Hz, or 20 kHz (kilohertz), but frequency of sound doesn’t end there. Theoretically, frequency, and therefore pitch, is infinite in both low and high directions. Experiments have other species detecting frequencies as low as 16 Hz (elephants) and as high as 150 kHz (porpoises). With limitations in technology, how do we ensure that our recordings sound good? Once we understand how sound is recorded and what parameters define a good recording, we’ll be off on the right foot.
The big question you probably have at this point is: “How do these frequencies, sine waves and sounds become music?”

Watch this excerpt from the 2009 PBS video “Music Instinct: Science and Song.” The full length video, if you’re interested, is available on Hulu Plus and Netflix.
Lower frequencies have longer wavelengths, and can travel farther than high frequencies. That’s why you can hear a pounding bass from someone blasting their stereo from blocks away.

Frequencies beyond the range of human hearing can be felt and can impact our psyche. Ultra high frequencies can make you feel irritated, as if a bunch of mosquitoes are flying around you. Ultra low frequencies can create an ominous feeling of foreboding – all without being heard.

“Pristine ears,” or perfect hearing, are one of the most prized possessions of musicians and those working in the audio industry. The first range of frequencies we lose through damage or aging are the high frequencies – the very ones which cause listening fatigue. Can you imagine an album mixed by someone with hearing loss? The highs would be accentuated and you’d probably become irritated with its sound after the first few minutes.

So why do so many aspiring professional musicians and recording engineers blast their earbuds or headphones? It’s proven that earbuds, headphones and high volumes can lead to permanent hearing damage, and even worse, once you’ve lost it, it’s gone forever. Read this article and its related stories for more information.
GOALS

• Be able to explain how sound is recorded and reproduced digitally
• Understand what affects the quality of a digital recording
• Be able to name different formats for uncompressed and compressed digital audio files
• Given pictures of audio waveforms, make educated guesses at what sound is represented

TO DO

• Read Lesson 1.2, going through the exercises and examples on the web and in Laulima as instructed
• Complete Assignment 1B to see and hear how sampling rate and bit depth affect audio quality
• Take Quiz 2 after completing the study guide.

These are bagpipes. I understand the inventor of the bagpipes was inspired when he saw a man carrying an indignant, asthmatic pig under his arm. Unfortunately, the man-made sound never equalled the purity of the sound achieved by the pig.

— Alfred Hitchcock
1.2 How Sound is Recorded Digitally

Sound waves are converted to binary digits or “bits” - sequences of ones and zeroes - by an analog to digital converter (ADC). These bits are stored to digital media like hard drives, flash drives, CDs and DVDs. In playback, the recorded data must be converted back to analog sound via a digital to analog converter (DAC) and amplified.

Sampling Rate

An ADC measures, or “samples” the sound at regular intervals. The speed at which the samples are taken is the SAMPLING RATE.

Higher sampling rates result in a more realistic, or “accurate” sounding recording. For reference, the sampling rate for CD quality sound is 44.1 kHz (yes, that’s 44,100 times per second).

Video work often requires a sampling rate of 48 kHz. For example, Blu-Ray DVD audio uses a 48 kHz sampling rate. High end, professional audio systems can befed up computers can record at a sampling rate of 192 kHz.

Exercise: 1A. Sampling Rate

Sampling is the process of taking in information, like looking at something while attempting to capture as much detail as possible.

Imagine you’re standing on the corner of a busy intersection, looking for a break in traffic so that you can cross the street. Now for 30 seconds, blink your eyes at a regular rate, like you’re turning a switch on and off repeatedly. The faster you blink, the more often your eyes open and “sample” of the traffic. The slower you blink, the more likely it is that you miss information, and get an inaccurate picture of the traffic. Result: danger.

In the same way, recording at a higher bit rate will capture more information and create a more accurate recording. Result: more accurate, or realistic sound.

In the figure below, the blue lines represent the result of sampling the same sine wave at a low rate (top line, inaccurate reproduction) and higher rate (lower line, accurate reproduction).

**Bit Depth**

BIT DEPTH is the amount of information taken in per sample. In the Blinking Exercise, bit depth would be everything you can see each time you open your eyes. If you have great vision and are able to focus quickly after each blink, you are more likely to see that taxi cut in front of the white Honda in the far lane. If you don’t have great vision you probably didn’t see the taxi, or even the white Honda, at all.

Using an ADC with the capacity for a higher bit depth will result in a more accurate sounding recording. For reference, *the bit depth for CD quality sound is 16-bit.*

**Higher Rates**

Technology is always improving, and consumer, home studio audio interfaces can have specs of 192 kHz and 32-bit. CD quality, however, is 44.1kHz/16-bit. Why aren’t we recording and creating CDs at a sampling rate of 192 kHz and a bit depth of 32?

The main reason is resources. Higher sampling rates and bit depths demand greater computer processing power, more storage space, and faster hard drives with higher data transfer rates. For example, recording a single track for one minute at 192 kHz/32-bit versus 44.1 kHz/16-bit requires *six times more* storage space and data transfer speed. Multiply that load by 10 tracks and 5 minutes . . . most consumer desktops and laptops will choke at handling that much data.

A second reason is that there is still a lot of debate about how high is “too high” -- humans can only hear so much, machines can only reproduce so much, etc. If you’re interested in the debate, do an internet search on “recording at 192 kHz” and read some of the articles that come up.

If we have better equipment and unlimited hard drive storage, can we record at 192 kHz/32-bit? Of course you can. Note, however, that *files of different sampling rates cannot be mixed* within a single project. And, once you get to producing the end product, you will have to convert your 192 kHz sampling rate to 44.1, and DITHER your 32-bit bit depth down to 16. A discussion of sample conversion is beyond this course, but it is important that you understand what dithering is.

**Dithering**

How do we get recordings at higher bit rates down to “fit” into 16-bit CD specifications?

Think of bit depth as word length on a Scrabble board. You have a 20-letter word “supercalifragilistic” that you need to fit into 12 spaces, analogous to fitting 20 bits into 12 bits. The simplest thing to do would be to truncate the word: cut off a chunk of 8 letters from the beginning, the end, or the middle. Results would be “supercilfra,” “ifragilistic” or “superlisc.” None of these result in an accurate picture of the original. You could, instead, intelligently abbreviate the word by cutting out vowels,
and end up with a pretty good abbreviation that’s closer to the original: sprclfrglstc.
In digital audio, this process of intelligent abbreviation is called DITHERING. This is a simplistic definition, but for the purpose of this class, “dithering” will refer to intelligent conversion of bit depth.

Unlike differing sample rates, files of differing bit-depths can be mixed within a single project. You can choose to use lower bit rates to create special effects.

1.3 Audio File Formats

Files intended for CD production and forms of high quality audio playback are normally saved as PCM files in their “raw” uncompressed state, at 44.1/16. On PC based platforms these are WAV files; on Mac platforms, these are AIFF (or AIF) files. There is no difference between a .wav and .aif file other than the platform it was created on; they are both in PCM raw format.

An uncompressed raw file can be large, as it contains all of the original recorded data. Files intended for web streaming and download are therefore COMPRESSED in order to facilitate streaming and download speed, as well as to save host server storage space. The easiest way to understand compression of an audio file is to refer back to how dithering works. The encoder (compression software) in a DAW intelligently removes data that will be least missed in audio playback, resulting in a file of smaller size.

There are many different compression algorithms (formulas) resulting in many different types of compressed files. The most common are MP3, AAC, FLAC and MPEG-4.

MP3 Files

In this class, we’ll be bouncing (condensing multiple tracks to stereo) and saving our projects as MP3 files at a 44.1 kHz sampling rate and 256 kbps variable bit-rate.

Bit-rate has to do with download or transmission speed. KBPS, or kilobits per second, refers to the amount of data, measured in kilobits, that is processed per second. Important: this is not the
same as bit-depth, which refers to the amount of data recorded per sample.

MP3 encoders can have a bit-rate range from 8 kbps to 320 kbps. Our selected bit-rate is a balance between audio quality and effective streaming and download speeds. While a raw file may arguably sound better, no one would enjoy waiting for large song file to download or start playing, especially when previewing multiple songs. Many iTunes and Amazon MP3s that are sold online are encoded at 128 kbps, likely because these services were established when dial-up internet was the norm. Today’s high speed internet connections, however, can handle higher bit rates, and the 256 kbps that we’re using is very common.

Most recently, DAWs are able to use variable rate MP3 encoding (VBR). In sections of the song file where there is less activity (and therefore, less information), the encoder might compress the file down to 64 kpbs, then raise the bit-rate up to 320 kpbs in sections where there is a lot of activity and information. This results in high quality MP3 files.

NOTE: By using the 44.1/256 VBR setting to encode our MP3 files, we are setting 256 kbps as an average bit-rate.

Here’s a link to a chart that summarizes sampling rates, bit-depths and bit-rates, their application, and a subjective judgment of the resulting audio quality.
Analog recording accurately replicates original sound waves, while the accuracy of digital recording depends on sampling rates and bit depths, which are at the mercy of technology. While digital technology allows portability (iPods) and affordable home recording, audio purists continue to argue the supremacy of analog. Do a web search on “Analog vs Digital audio” and read a couple of articles to get an idea of the pros and cons of both.

Digital audio is just one step along the digital revolution. Without data being converted to digital bits, there would be no personal computers or tablets, no internet, no cell phones, no CDs and DVDs, no HDTV, and no digital photography. It wasn’t until 1993 that the worldwide web (internet) was made available to the public. Now, just 30 years later, it’s difficult to imagine a world without internet based companies such as Google, Facebook and Amazon.
Quiz 2 Study Guide

• How is sound created?
• What is the measurement unit of pitch?
• What is the range of human hearing?
• What converts sound to digital form?
• What are the two most important settings that affect the quality of a digital sound recording?
• What is sampling rate?
• What is bit depth?
• What are the sampling rate and bit depth settings of CD quality recording?
• What is the main reason why home recordists don’t record at the highest sampling rate and bit depth possible?
• What are the recommended settings for recording in this class?
• What is dithering?
• What is a “raw” file?
• What is the difference between a WAV and AIFF file?
• What are some of the commonly found formats for compressed files?
• What format are we using for class?
• What is bit rate?
• What is the bit rate we’re using for our class files?
LESSONS 1.4 AND 1.5

MIDI, Audio Cables

GOALS

• Know what M-I-D-I stands for
• Be able to give a simple description of how MIDI works
• Be able to explain what the benefits of MIDI are
• Identify Middle C for reference
• Be able to identify parameters that can be edited in MIDI
• Be able to identify the most common types of cables used in digital music production

TO DO

• Read Lessons 1.4 and 1.5, going through any exercises and examples as instructed.
• Watch Movie 1.4 to see and hear how MIDI works
• Become familiar with common types of cables used in music production
• Take Quiz 3 after working through the study guide

I did my first live shows in the mid 80’s and in those days I had a boatload of equipment that always seemed to be going wrong.

– Thomas Dolby, synthesizer impresario and one of the first musicians to use MIDI in live concerts
1.4 MIDI

MIDI is the acronym for Musical Instrument Digital Interface, a communications protocol which allows the joining of two or more digital instruments. One instrument acts as a master, and the others as slaves. This allows one digital instrument to trigger playback of multiple digital instruments; this one master is known as the CONTROLLER. The instruments can be any brand, size or instrument type; as long as they have a MIDI connection, they can be used.

When MIDI was first invented in the mid-1980’s, workstation set ups looked like this. There were no software instruments, so musicians had to own a physical keyboard or sound module for each type of synthesizer or drum machine that they wanted. They also had to purchase separate hardware units for audio processing: reverb, delay, chorus and compression units. This was all extremely expensive -- and crowded!

All you need to work in MIDI now is a controller and a computer with a MIDI recorder, or sequencer, installed. All professional DAWs are comprised of an audio recording