MUS112 WEEK 9

SAMPLING AND DIGITAL AUDIO

DR BRIAN BRIDGES

BD.BRIDGES@ULSTER.AC.UK

WWW.BRIANBRIDGES.NET

RECAP

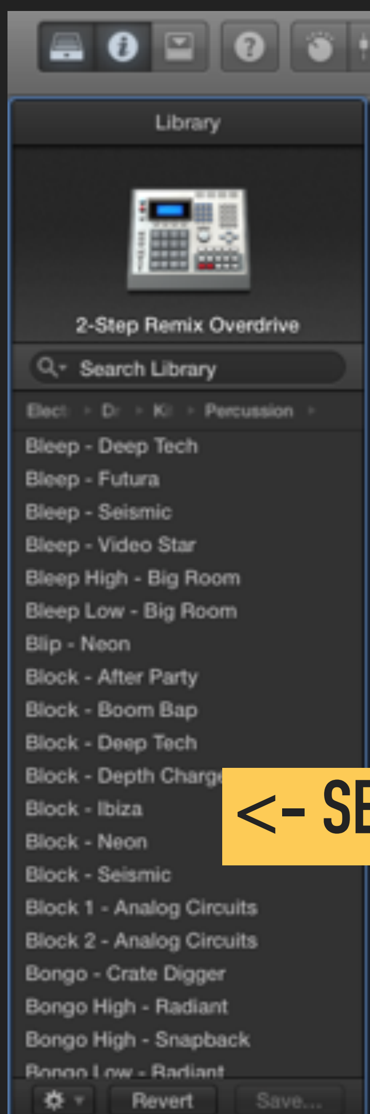
- Last week, we covered MIDI note editing and related topics
- Basics of MIDI messages and differences between MIDI and audio data
- Apple Loops
- Note entry with the keyboard, musical typing function (cmnd+K), piano roll, arpeggiator MIDI plugin
- We also covered drums via Apple loops, Ultrabeat, Drum Machine Designer (front end for Ultrabeat) and Drum Kit Designer

HOLDOVER FROM LAST WEEK: ARP ADDITIONAL FUNCTIONS

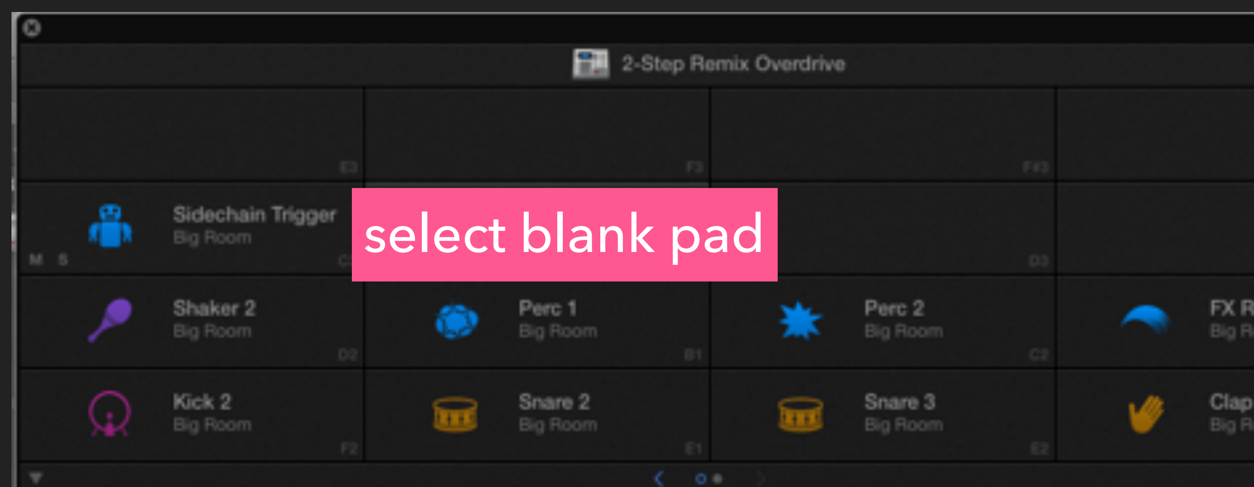
presets available

PLUS DEMO: ARP TO CREATE DRUMS

HOLDOVER FROM LAST WEEK: DRUM MACHINE DESIGNER



SELECT LIBRARY TAB (LEFT-HAND SIDE)



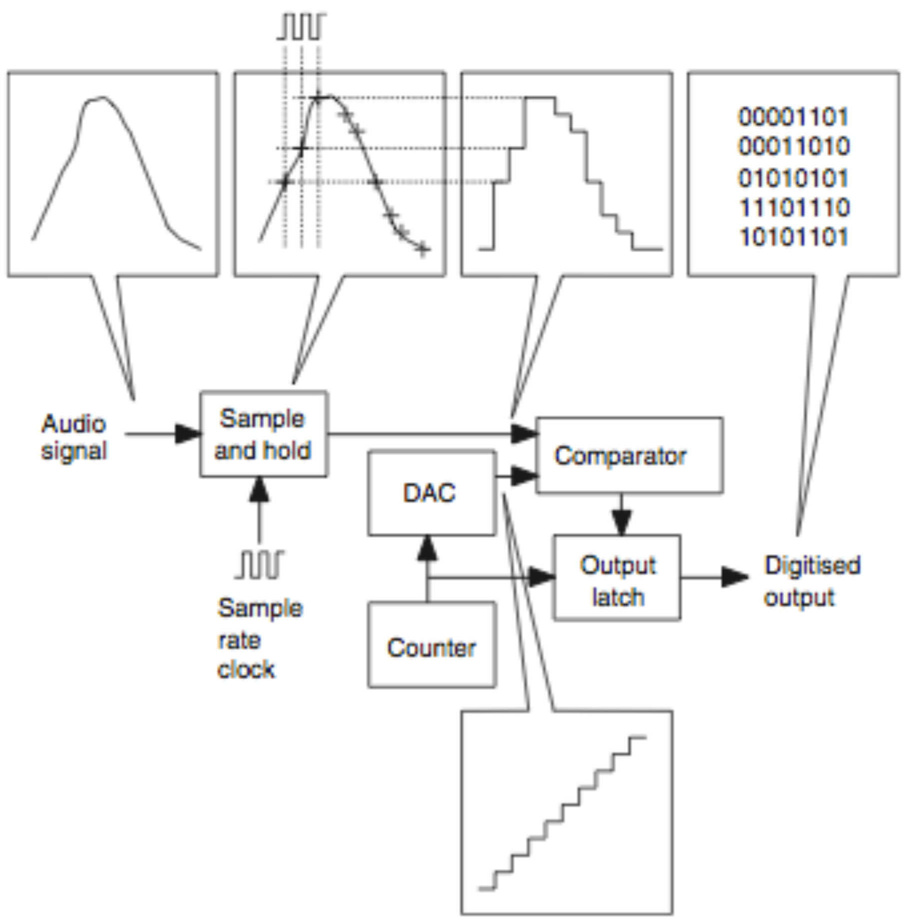
<- SELECT INSTRUMENT FOR TAB

NOTE: IN SPITE OF APPEARANCES, DRUM MACHINE DESIGNER ISN'T AN INSTRUMENT IN ITS OWN RIGHT; INSTEAD, IT'S A 'WRAPPER' AND FRONT END FOR ULTRABEAT (NOTE THAT YOU CAN'T SIMPLY SWITCH A DRUM MACHINE DESIGNER TRACK BACK TO OTHER INSTRUMENTS

RECAP: KEY CONCEPTS: DIGITAL AUDIO

DIGITAL AUDIO: BASICS

- Digital audio is simply 'sound (accurately) described by numbers'
- *Sound is a vibration in the air around us which we receive at our ears (and decode in our brains)*
- Digital audio is simply another means of *encoding* the air pressure vibration that is sound so that we can transmit or alter it
- Digital technology is the key source of the power of modern audio production tools



From Russ, M. 2009. *Sound Synthesis and Sampling*. 3rd ed. Oxford, Focal, p. 61

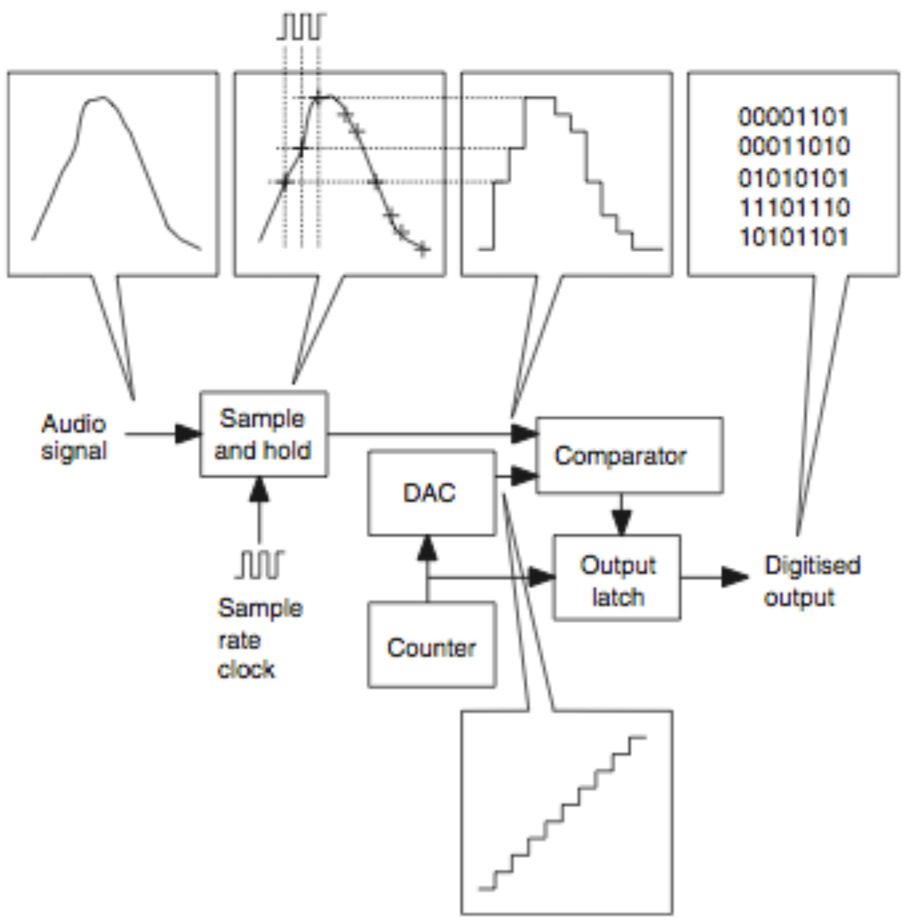
DIGITAL AUDIO: BASICS

- Sampling is the process of changing the continuously-varying (analogue) signal into a discrete set of digital readings (number values for specified time intervals)

- The process here shows an audio signal being sampled at a regular rate: this reads off the size (amplitude) of the air pressure variation for each time interval

- We then have numbers which embody a simplified representation of the original audio (depending on how accurate our sampling system is)

- Accuracy in sampling depends on two things: amplitude resolution (bit depth) and frequency resolution (sample rate)



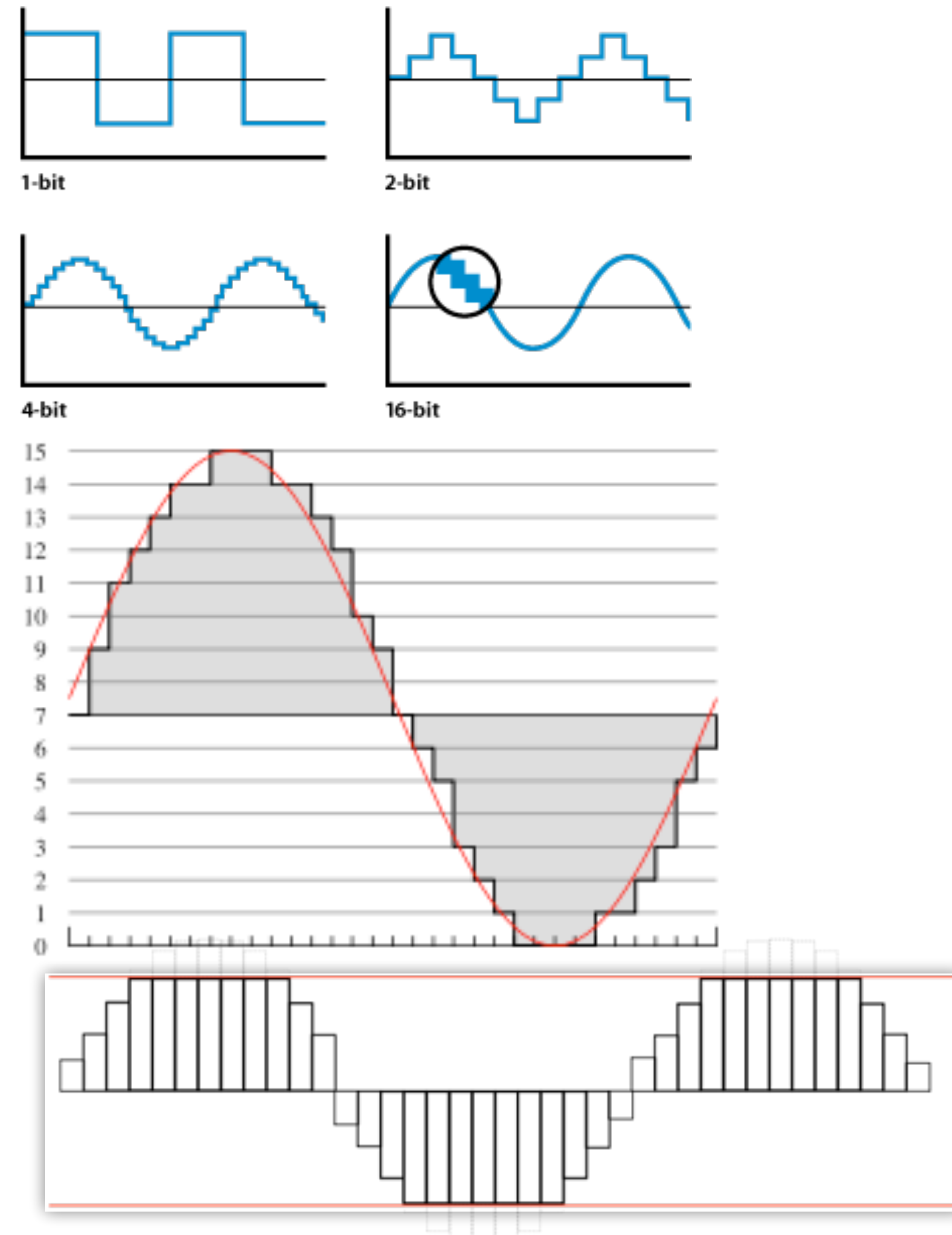
From Russ, M. 2009. *Sound Synthesis and Sampling*. 3rd ed. Oxford, Focal, p. 61

DIGITAL AUDIO: CLIPPING

- Our digital audio sampling system will have minimum and maximum signal levels which it can measure/record reliably based on its bit depth *(1 bit = 2 levels, 2 bits = 4 levels, 4 bits = 16 levels… 16 bits =65,536 levels)*

- If we go over the maximum level, we will not be able to record the signal accurately, we will simply record the maximum level (i.e. number), whatever the signal itself is doing—this is known as clipping

- This results in extreme distortion: rather than the wave being progressively bent out of shape (as is the case with analogue distortion), clipping cuts off the peaks of the wave abruptly (chops the 'heads' off), resulting in sudden and significant audible distortion which is particularly harsh



clipping

DIGITAL AUDIO: SUMMARY

- **Resolution/bit depth gives us the amount of amplitude data and hence, dynamic range for recordings,** between minimum and maximum levels in a system: **expressed in bits**…numbers of on/off binary numbers needed to encode 'normal' decimal numbers **…24 bit=pro recording**=16,777,216 level values, **16 bit=CD audio**=65,536 level values

- **Sampling rate (for frequency resolution):** needs to be twice maximum audible frequency (c. 20 kHz), so rates of **44.1 kHz+** are commonly used

- **Uncompressed and compressed audio:** MP3 files are not full-quality audio, but use a lossy model based on our hearing abilities to reduce file size dramatically (nonetheless, many listeners can hear the difference between MP3 and uncompressed audio such as .wav or .aif/.aiff files)

- **About surround:** *don't* use surround plugins/bounce options unless you are using a multi-speaker surround sound facility (such as studio 2=> you will not be using this facility this year!)

DIGITAL AUDIO PROBLEMS: CLIPPING

This happens when we record or mix

at too high a level: ‘chopping the heads off’

red ‘overs’ = pretty serious distortion

solution: turn problematic channels down before tracks are bounced!

https://www.youtube.com/watch?v=9uEtworGLrU

EX. WITHOUT CLIPPING

Inside digital audio workstations, we can get away with this to some extent due to smart digital signal handling, but it IS crucially important when you come to mixdown and play your file elsewhere

http://modernmixing.com/blog/2014/06/07/why-you-shouldnt-care-digital-clipping/

DIGITAL AUDIO PROBLEMS: CLIPPING

This happens when we record or mix

at too high a level: ‘chopping the heads off’

red ‘overs’ = pretty serious distortion

solution: turn problematic channels down

before tracks are bounced!

https://www.youtube.com/watch?v=9uEtworGLrU

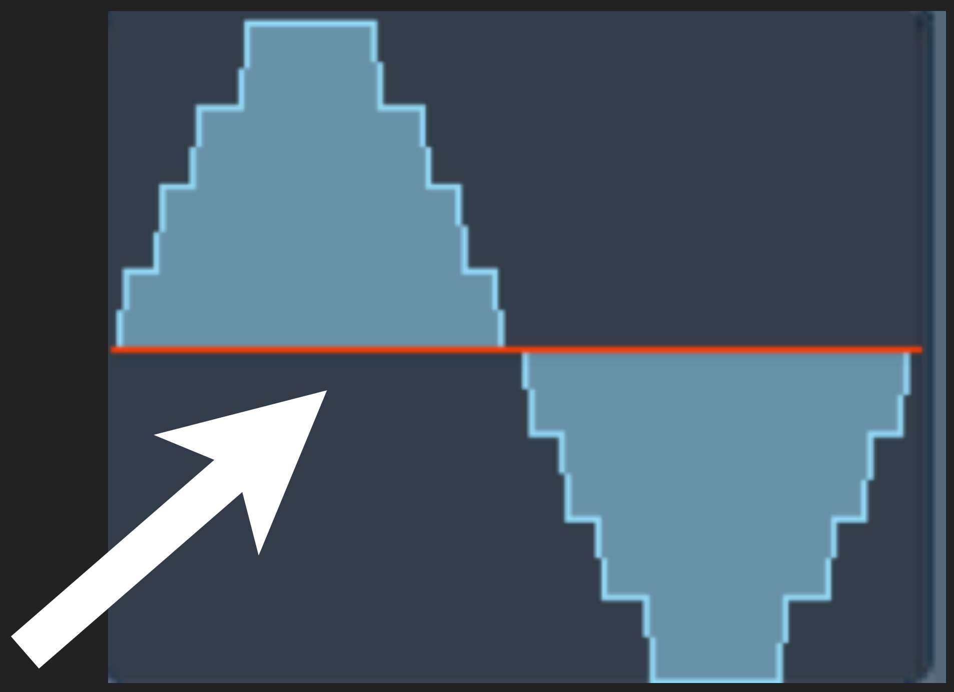
Inside digital audio workstations, we can get away with this to some extent due to smart digital signal handling, but it IS crucially important when you come to mixdown and play your file elsewhere

http://modernmixing.com/blog/2014/06/07/why-you-shouldnt-care-digital-clipping/

DIGITAL AUDIO PROBLEMS: LOW BIT DEPTH (AUDIO RECORD LEVEL TOO LOW)

If we record at too low a level, we may have problems at the other end: low-level/quiet sounds (e.g. the *decay phase* of a note or the *tail* of a reverb may become distorted: quantisation noise

This effect is exacerbated by lower sampling rates (16 bit, 12 bit…8 bit if you're feeling retro!)



sound wave becomes

'pixellated'

ASIDE & TRIVIA: 8 BIT OR CHIPTUNE?

- Some people like to work with really low-fi digital recordings or synthesis
- So-called *8-bit* (AKA *chiptune*) music is often not actually 8–bit: it’s 4-bit (for even crunchier effects)
- The name 8-bit comes from the main processor of old computers and games consoles, not their sound chip
- Chiptune is therefore a more accurate name!
- An example of the genre, here: www.youtube.com/watch?v=A5TqDx7iWvQ

DIGITAL AUDIO PROBLEMS: LOW SAMPLING RATE

- Sampling rate is how frequently a computer records or generates a number for changing level data in an audio file
- All sound is vibration: digital sound is a set of numbers storing or generating that vibration
- **We can hear vibrations in a range from 20 per second (20 Hertz) to an upper limit of 16,000 to 20,000 per second (16–20 kiloHertz)**
- Samping rates need to be **double** the highest frequency in a piece of audio: frequencies higher than this limit are 'reflected down' into lower ranges, 'messing up' the frequency spectrum (and making the sound 'harsh', 'edgy', or 'metallic'…the technical term is **inharmonic**)
- Therefore, we use sampling rates of **44.1 kHz** plus to accurately encode audible sound

DIGITAL AUDIO PROBLEMS: LOW BIT-RATE MP3S

- MP3 (MPEG 1, Layer 3) is an audio standard for **data compression** of an audio file so that it may be more easily transmitted over networks or stored on low-capacity devices
- Typically, it offers savings of 6:1 or greater on **uncompressed file sizes**
- However, there is a trade-off: it's a **lossy** compression scheme, removing audio data based on a model of the behaviour of our inner ear structures
- In some cases, **due to the structure of our ears, sounds in a similar frequency range may 'block' (mask) other sounds…**
- **In short, MP3 is not a suitable archival format**; use uncompressed .wav, .aif/.aiff etc.

http://productionadvice.co.uk/why-mp3-sounds-bad/

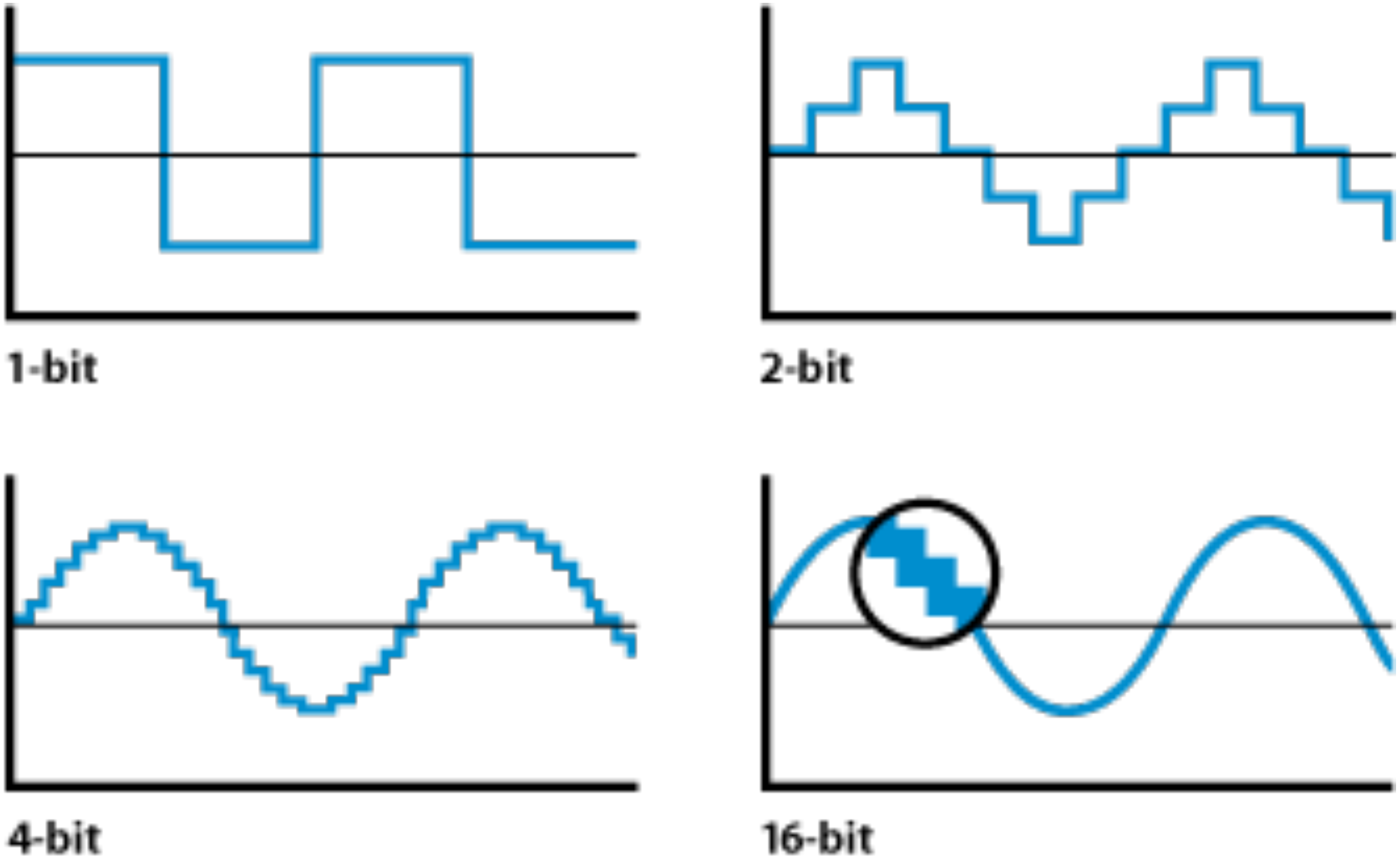
http://www.soundonsound.com/techniques/what-data-compression-does-your-music

DIGITAL AUDIO PROBLEMS: LOW BIT-RATE MP3S

- Think of the effect of trying to listen to music in a car with a loud engine! You miss some of the lower frequencies because the car 'blocks' them
- MP3 works on this principle: certain sounds/parts of sounds in a mix will **mask** others, therefore you don't need to encode them
- MP3 **encoders** can even try to filter out sounds that you might here to reduce the file size further, but at a cost to audio quality
- Guideline: low bit-rate MP3s (e.g. 128 kbit or less) are likely to interfere with the quality of your audio (sometimes in unpredictable ways)
- Solution: use 148/196 kbit or higher for casual listening, but used **uncompressed file formats** (.aif, .wav) **for archiving**

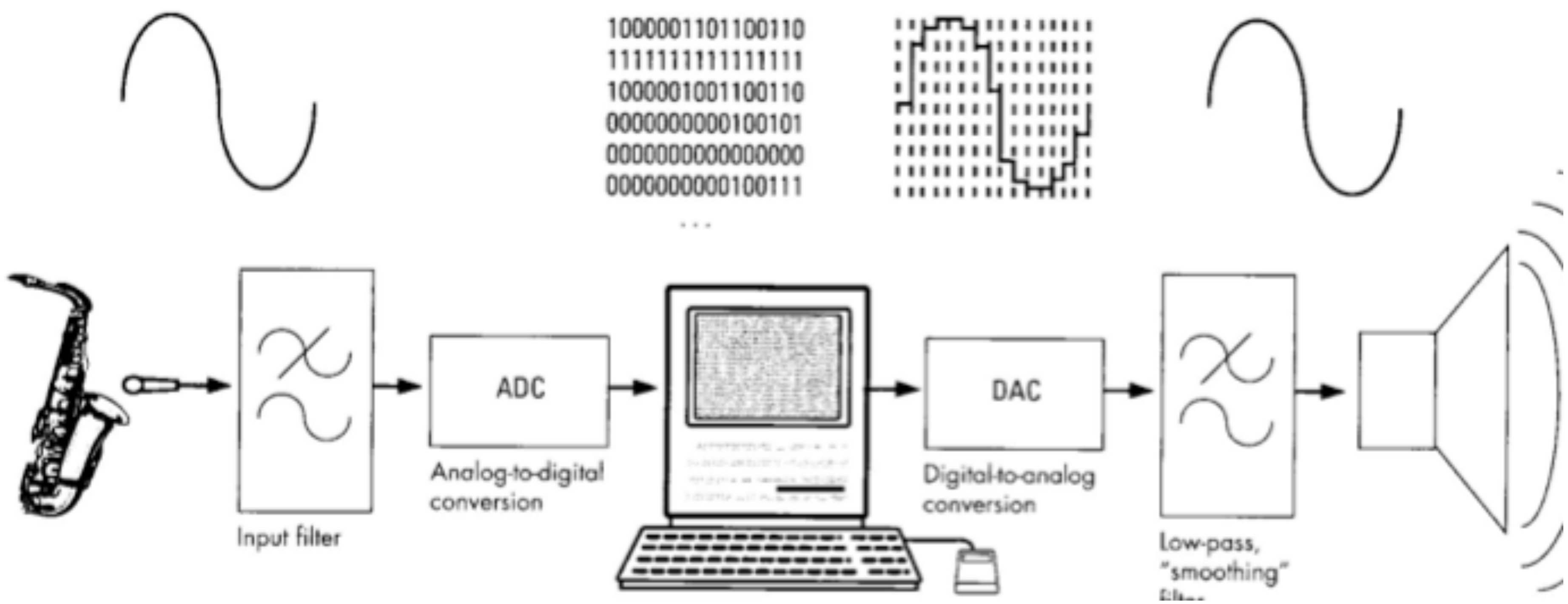
DIGITAL AUDIO PROBLEMS: 'MORALS'

- Take care of levels when recording and/or mixing to prevent clipping
- Use at least 16 bit audio settings when recording but preferably, 24 bit to prevent quantisation noise and allow a reasonable amount of dynamic range
- Use a sampling rate of 44.1 kHz or higher (opinion is still divided on the relative merits of higher rates, but that's a more advanced issue)
- **NB: Don't use compressed audio files (e.g. MP3s) as an archival format! Use uncompressed (PCM: .aif or .wav)**



INTRO TO DIGITAL SAMPLING

SAMPLING AS INSTRUMENT



- Digital sampling is the use of digitally recorded or encoded audio data for playback of instrument sounds or other sound sources on demand
- As discussed previously, it is not the same as creating sounds 'from scratch' using synthesis methods and processes
- We don't always have to create our own samples; modern digital audio workstations (DAWs) such as Logic have extensive instrument sets based on samples

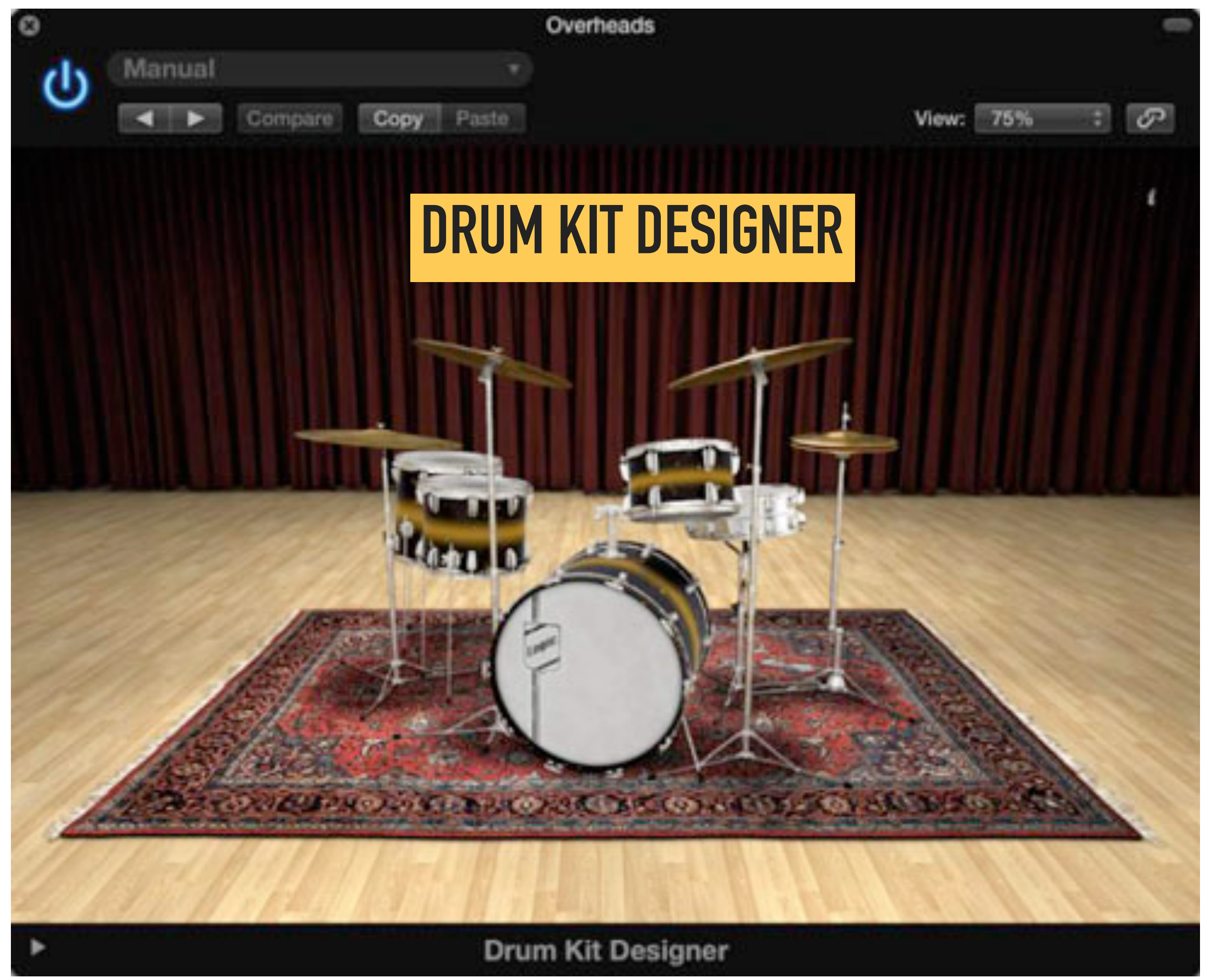
INSTRUMENTS IN LOGIC BASED ON SAMPLING TECH.

EXS24 SAMPLER: ALLOWS EXTENSIVE EDITING

ULTRABEAT: DRUM SAMPLES WITH SYNTHESIS

DRUM MACHINE DESIGNER: IMPORT SAMPLES

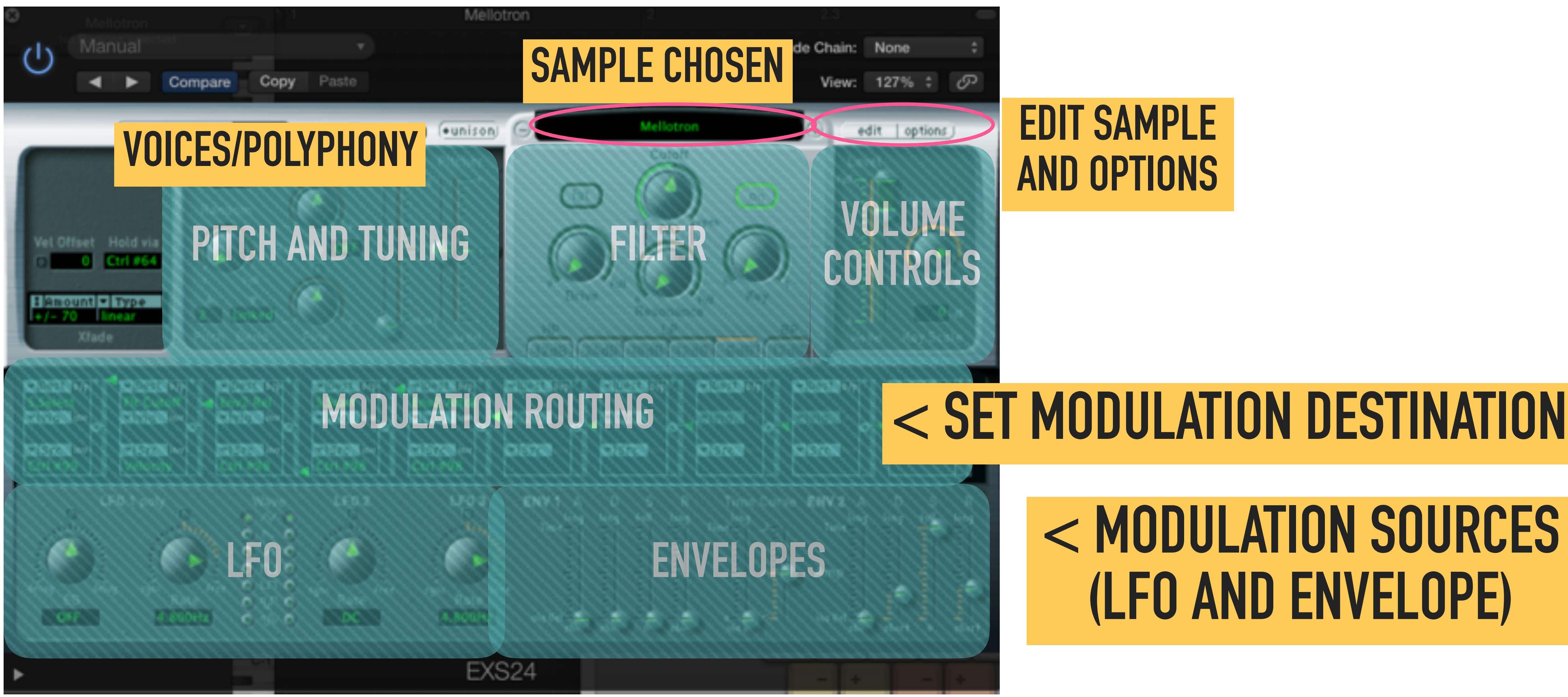
- DON'T FORGET LOGIC'S
- APPLE LOOPS FUNCTIONALITY



DRUM KIT DESIGNER

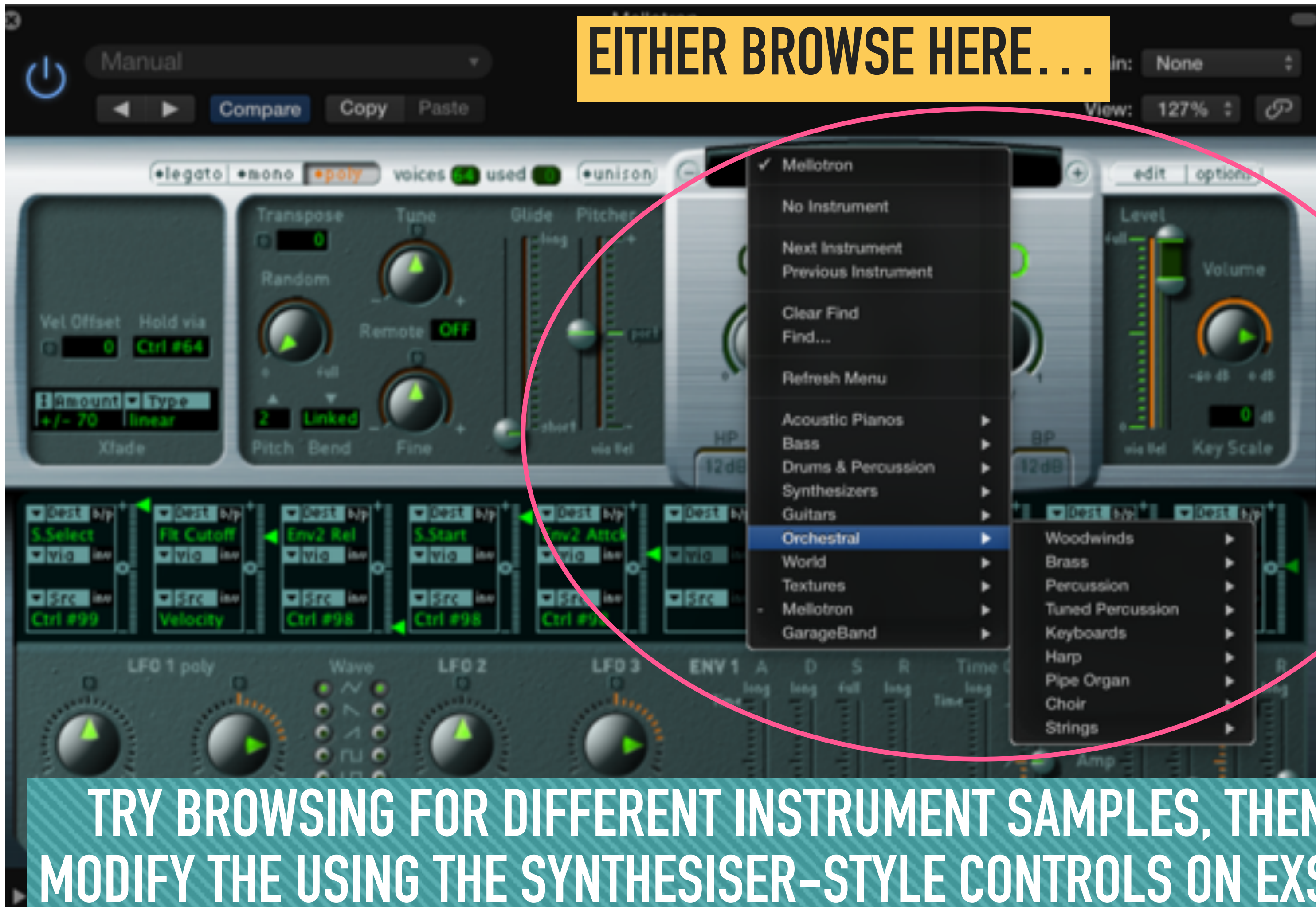
EXS24 AS INSTRUMENT

- *EXS24 mkII is a software sampler. It plays back audio files, called samples, that you load into it. These samples are combined into tuned, organized collections called sampler instruments. As sampler instruments are based on audio recordings, they are ideally suited to emulating real instruments such as guitars, pianos, and drums.* (https://documentation.apple.com/en/logicstudio/instruments/index.html#chapter=12%26section=0)



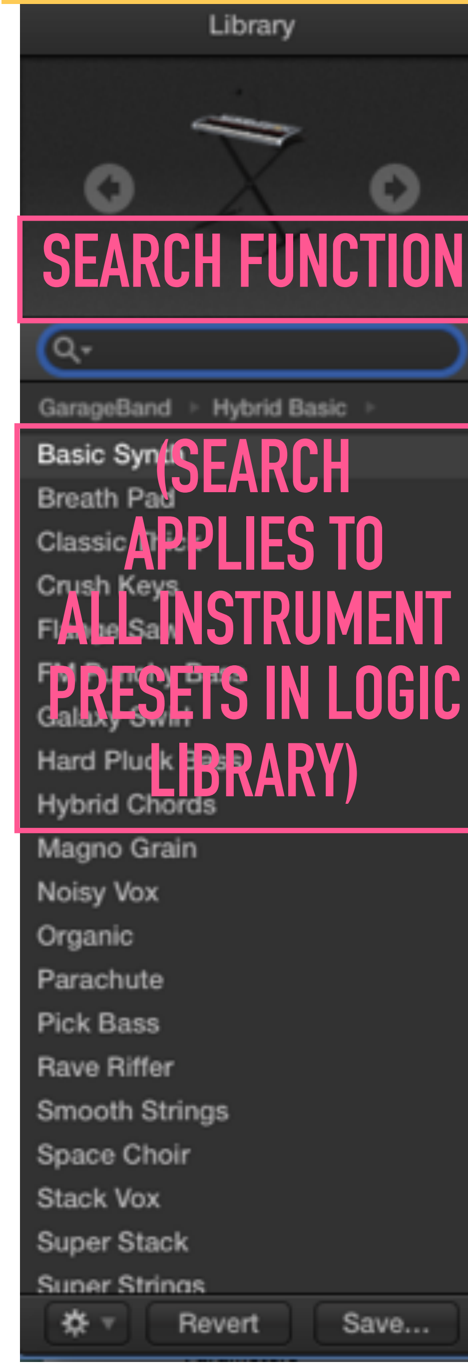
SELECTING EXS24 SAMPLED INSTRUMENTS

EITHER BROWSE HERE…



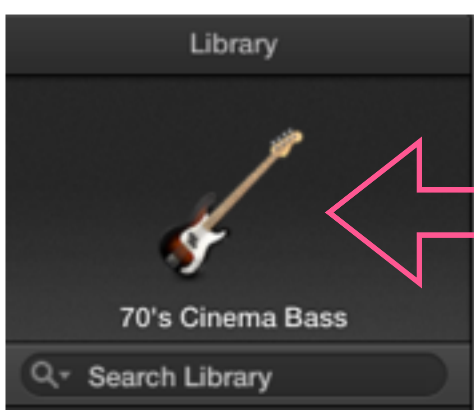
TRY BROWSING FOR DIFFERENT INSTRUMENT SAMPLES, THEN MODIFY THE USING THE SYNTHESISER-STYLE CONTROLS ON EXS24

…OR USE LIBRARY PANE IN ARRANGE



SEARCH FUNCTION

(SEARCH APPLIES TO ALL INSTRUMENT PRESETS IN LOGIC LIBRARY)



YOU CAN EVEN BROWSE FOR KEYWORDS IN SYNTH/SAMPLER INSTRUMENT PRESETS WITHOUT HAVING EXS24 ALREADY SELECTED

USING YOUR OWN SAMPLES: 1 (SINGLE PERCUSSION HITS)

USING DRUM MACHINE DESIGNER

DRAG AND DROP AUDIO FILE FROM OS X FINDER/DESKTOP TO THE DESIRED ‘SLOT’ ->

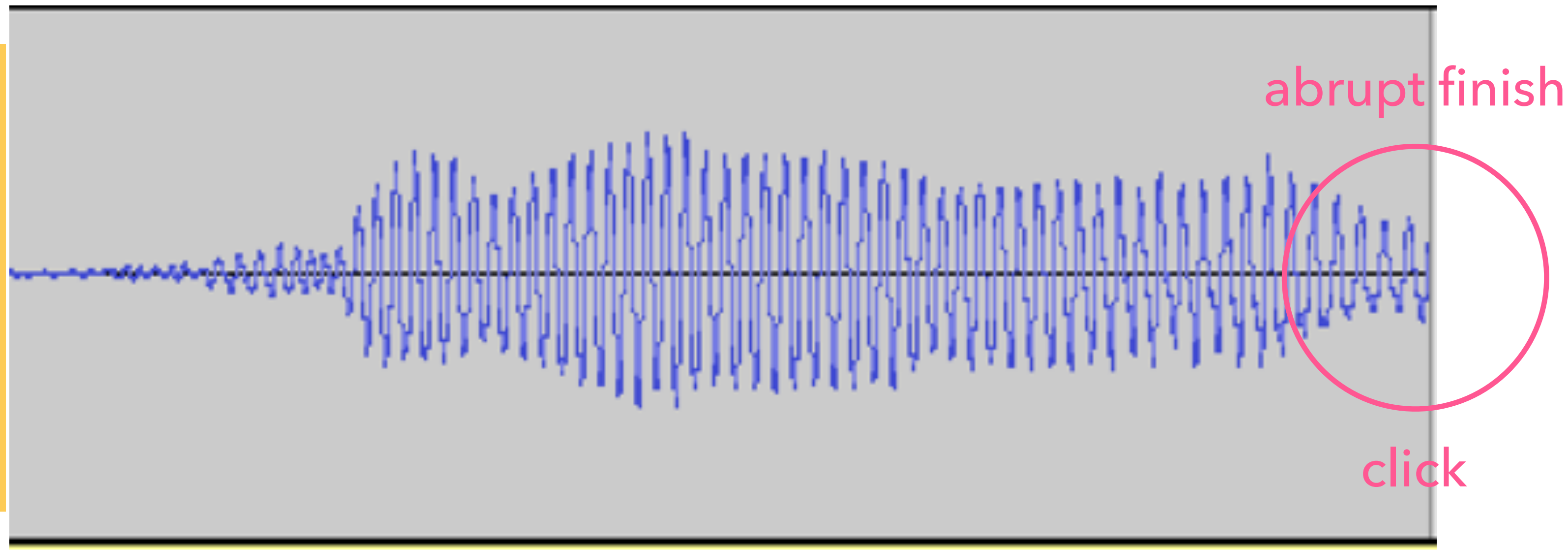
ADJUST PITCH, VOLUME, PAN AND ENVELOPE

FILTER (TONE CONTROL) AND EFFECTS
NB: NAMING CONVENTION OF FILTERS IS ‘CUT’ RATHER THAN ‘PASS’
...HIGH CUT=LOW PASS

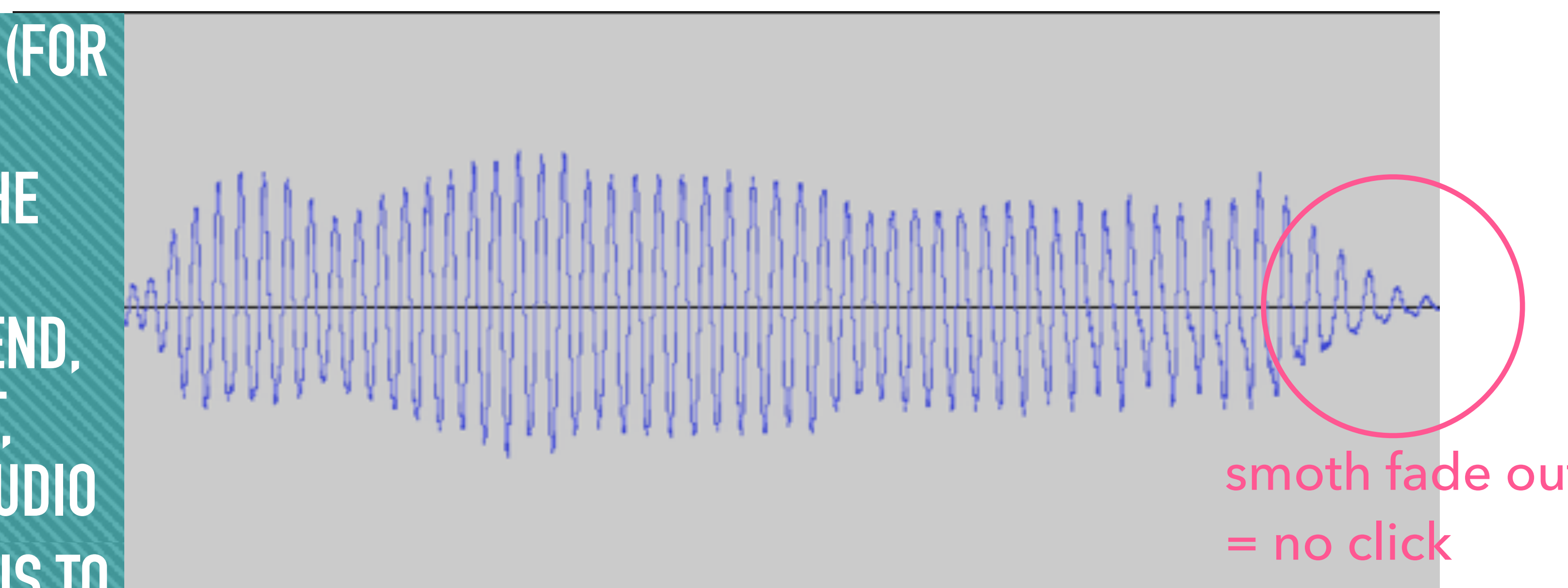
USING YOUR OWN SAMPLES: 1 (SINGLE PERCUSSION HITS)

USING DRUM MACHINE DESIGNER

SOMETIMES SAMPLES END ABRUPTLY (THE SOUND DIDN'T FULLY FADE OUT BEFORE YOUR EDIT, LEAVING A SUDDEN DROP TO ZERO AND AN AUDIBLE CLICK)



THE EASIEST WAY TO FIX THIS (FOR NOW) IS TO
OPEN THE AUDIO FILE IN THE AUDACITY APPLICATION,
SELECT THE REGION AT THE END,
EFFECT (MENU): FADE OUT,
THEN FILE (MENU): EXPORT AUDIO

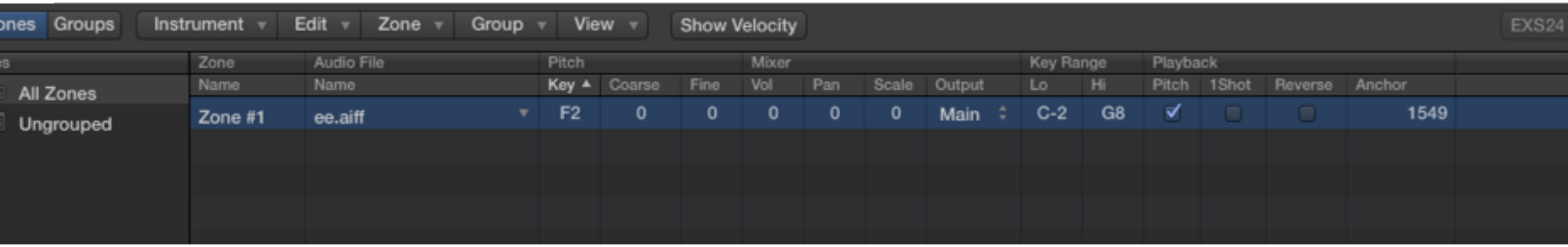


THE EASIEST WAY TO FIX THIS IS TO
OPEN THE AUDIO FILE IN THE AUDACITY APPLICATION,
SELECT THE REGION AT THE END,
EFFECT (MENU): FADE OUT,
THEN FILE (MENU): EXPORT AUDIO

THEN REIMPORT THE FILE INTO LOGIC'S DRUM MACHINE DESIGNER (DRAG AND DROP NEW VERSION)

USING YOUR OWN SAMPLES (2): EXS24

USING YOUR OWN SAMPLES (2): EXS24



key: set root key (note) of sample (if it is pitched) so that the keys are pressed, they make the note you expect!

coarse and fine tuning of sample

volume (+12/ - 96) and pan (0=centre, R= + 100, L = -100)

scale: scales level of sample across key range; root key is loudest, followed by fade-out

output: only useful for surround/multichannel: output numberings beyond stereo

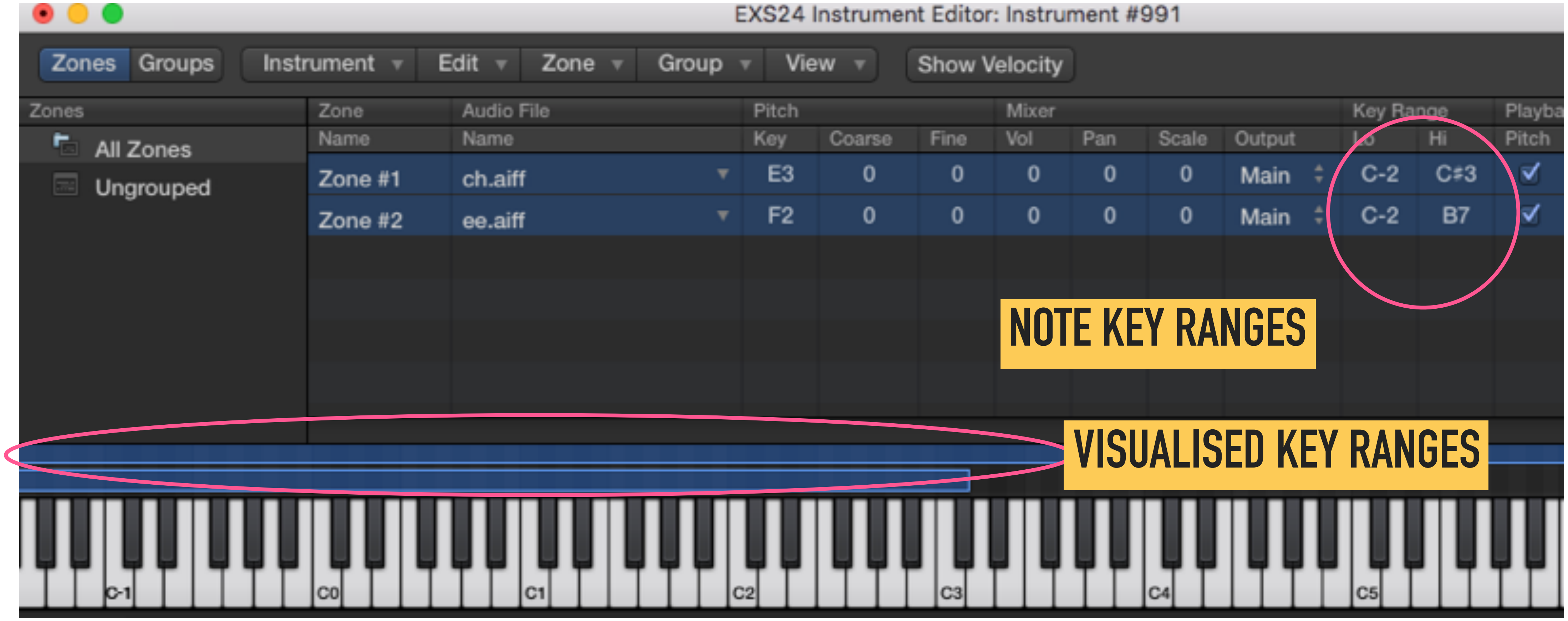
key range: which notes on the keyboard play the sample

playback: pitch (off for percussion) 1 shot (play entire sample, regardless of env), reverse

anchor: set start point of sample

https://documentation.apple.com/en/logicexpress/instruments/index.html#chapter=9%26section=14%26tasks=true

USING YOUR OWN SAMPLES (2): EXS24 MULTISAMPLES



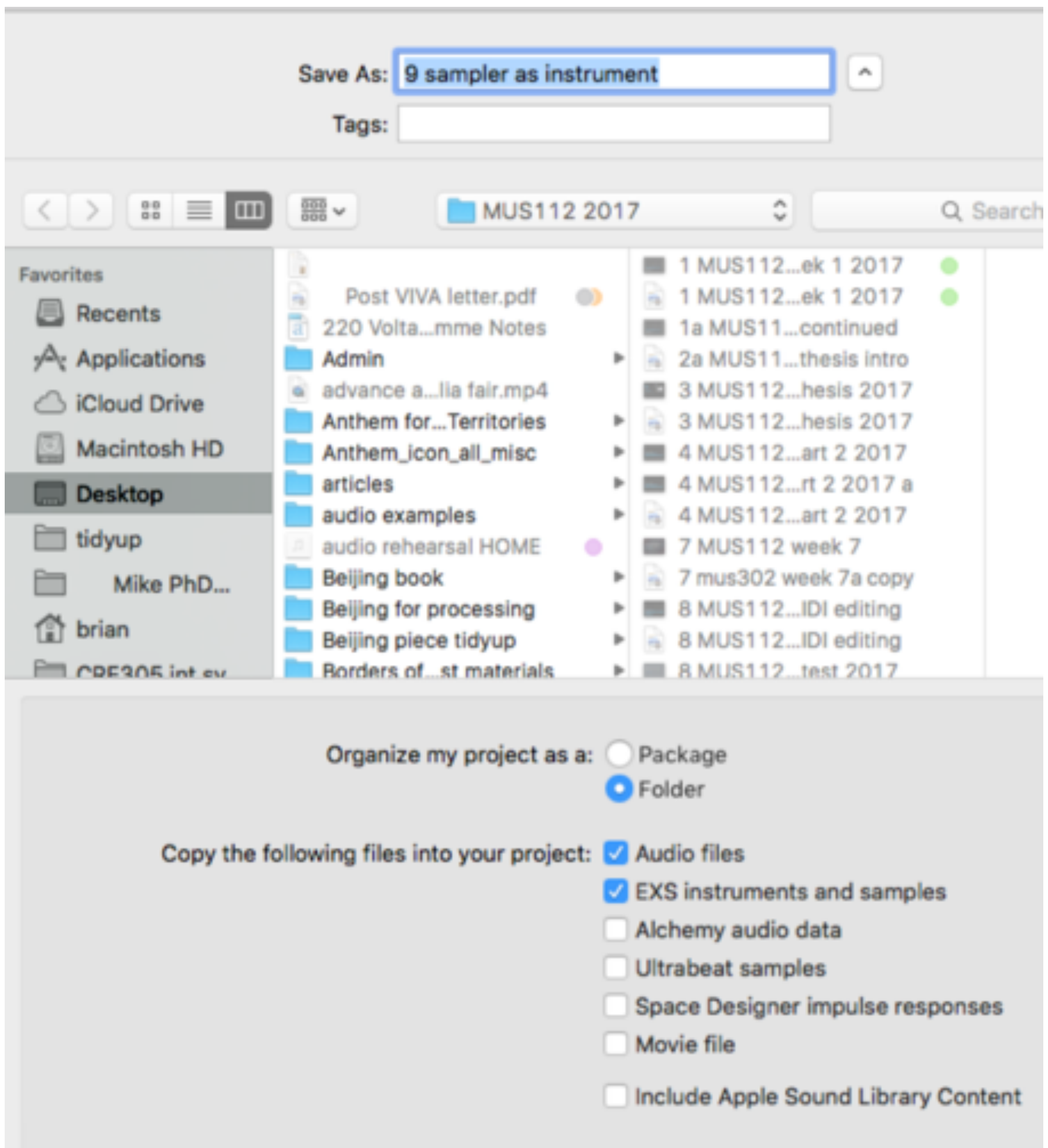
MULTISAMPLING IS WHEN YOU APPLY MULTIPLE SAMPLES TO A SINGLE SAMPLER INSTRUMENT
IT IS OFTEN USED EITHER TO SAMPLE DIFFERENT PITCH RANGES OF A MUSICAL INSTRUMENT
OR TO SAMPLE DIFFERENT DYNAMICS OF A MUSICAL INSTRUMENT

HERE'S A SIMPLE EXAMPLE WITH A 'CH' SOUND AND AN 'EE' SOUND LAYERED
TOGETHER OVER PART OF THE KEYBOARD

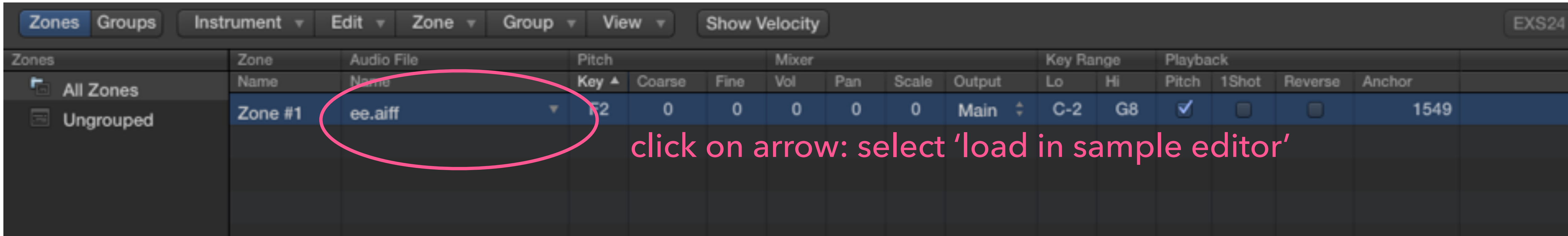
THIS IS A MORE ADVANCED TOPIC; WE'VE ONLY SCRATCHED THE SURFACE OF SAMPLING (SIMPLE
SAMPLING OF SINGLE SOUNDS FOR A KEYBOARD INSTRUMENT OR DRUM MACHINE)...
BUT HOPEFULLY YOU CAN APPRECIATE ITS POWER

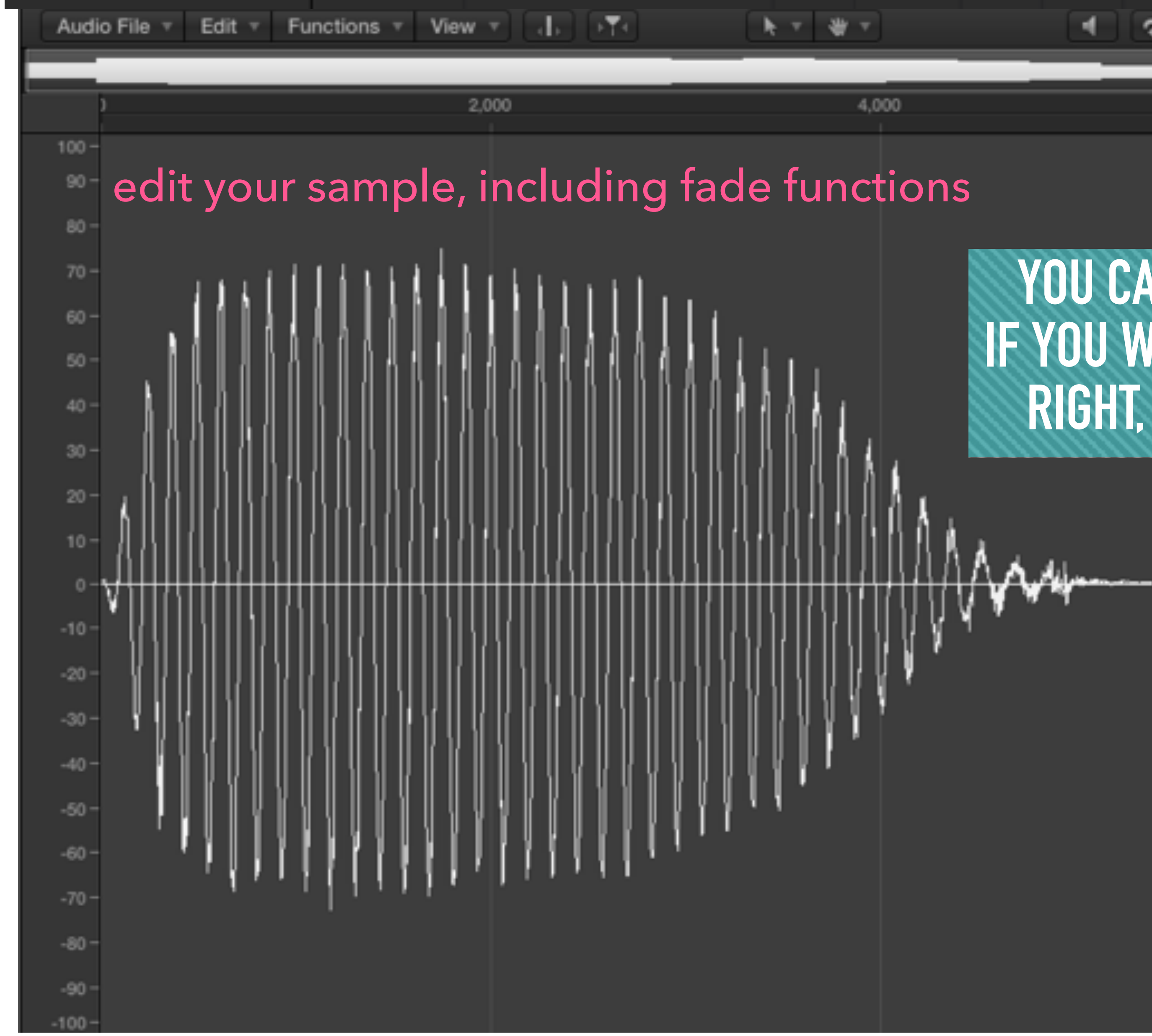
FILE MANAGEMENT

- Remember, if using external sampled audio, save your audio files and sampler instruments and samples within the project folder: choose File: 'Save As' for these options



OTHER SAMPLING-RELATED FEATURES: SAMPLE EDITOR





YOU CAN ALSO SET LOOP POINTS ON SAMPLES IF YOU WANT THEM TO SUSTAIN (DIFFICULT TO GET RIGHT, BUT WORTH A TRY ON SOME SOURCES)

FURTHER READING/REFERNCE

- Russ, M. 1996. *Sound Synthesis and Sampling*. Oxford: Focal.
- d'Escrivan, J. 2012. *Cambridge Introduction to Music Technology*. Cambridge: Cambridge UP. [in library as e-book - log in via portal to view/download temporary copy]
- http://www.soundonsound.com/techniques/lost-art-sampling-part-1
- http://www.musicradar.com/tuition/tech/a-brief-history-of-sampling-604868
- A Brief History of Sampling from Eclectic Method on Vimeo.
- Listening to accompany last week's lecture: Spotify arpeggiator playlist
- Listening to accompany this week's lecture: Spotify sampling playlist