WEEK 2: SIGNAL FLOW PART 2

- Meeting in the lobby of Berklee at 6:30pm for the Boat trip.
- Auditions are tonight
- David Mash presenting at the scoring stage at 7:00
- Speed dating on Thursday night at 8:00. We are group 2, meeting in the library. By Thurs/Fri you'll know who you are paired with.
- Project booking must be done in person. Wednesday from 2-3, or Thursday 2-3 for Lab Time. Collaboration is allowed.
- Office Hours are Wednesday from 10am-12pm

I. Signal Flow Part 2: Recording Console Basics
   a. Review
      i. Channel Path – Anything from the source to the recorder
      ii. Monitor Math- Anything from the recorder to the listener.
   b. New Terms
      i. BUS – An audio path that "sums" multiple audio paths together.
         1. Types: Mix Bus. Can be in mono, stereo, 5.1, 7.1. They all take multiple channels and sum them into a bus
         2. Auxiliary(aux) Bus- commonly used for effects or headphones. Allows you to send a different mix for live sound
         3. Group Bus – Prior to DAWs you were limited to track count, but it didn't mean you were limited in the amount of mics you can use. You could sum more than one mic to one bus (one track).
      ii. Direct Output – A signal output of the channel path. Typically taken pre or post fader. They take a 1:1 ration. One input of the track goes to 1 output.
      iii. CHANNEL STRIP – A combination of multiple gain stages and audio processes. The rules of consoles is they flow in signal flow order from top to bottom. Input <> Preamp <> Processing <> AUX Bus/Sends <> Mix Bus Pan <> Faders
         1. Memorize this order because every console works this way.
      iv. INSERT – An audio send and return from the channel strip. Used to add additional outboard processing. Typically after the processing but before the fader or after the mic pre but before the processing.
   c. In recording we know we need two audio paths.
      i. Channel Path
         1. Mic Input <> Channel Path Channel Strip <> Direct Outpu or Group Box <> Recorder Input
      ii. Monitor Path

1. Recorder Output <> Line Input <> Monitor Path Channel Strip <> (Aux sends to headphones and effects) <> Mix bus output <> Control Room Playback.
2. Important to always go off the monitor path for headphones. Its what is coming back from the recorder. You should ALWAYS listen back to what you are recording, don't listen from the channel path. Also for overdubs if you are sourcing from the channel path the player will not be able to hear what is being played back.

d. Two types of Recording Consoles
   i. SPLIT CONSOLE – Each channel only contains one path. So you have channel strip faders, and monitor path faders.
   ii. IN LINE CONSOLE – Each channel has 2 audio paths. This is a more modern design. Able to have 2 separate audio paths in one channel. Has two separate inputs that go to two separate places.
      1. Allows for SELECTABLE PROCESSING, so you can either have an EQ recorded to the channel path or just sent to the monitor path. AUX Sends are normally sent to the monitor path but it is also selectable.
      2. Small Faders (channel strip) and Large Faders (monitor Path)

e. What type of console is the System 5?
   i. Always operates as a SPLIT CONSOLE.
      1. Each channel only has one audio path.

f. System 5 Default Config
   i. MONITOR PATH
      1. Pro Tools Outputs 1-48 <> Channels 1-48 (MAIN LAYER) (Inserts 1-48) <> Stereo Mix Bus <> Studio Speakers. Pro Tools 1-48 is normalled from the Pro Tools returs
   ii. CHANNEL PATH
      1. Console Mic Pres 1-48 (via A/D) <> Channels 49-96 (SWAP LAYER) <> Direct Outputs <> Pro Tools Inputs.
      2. Normalled to the Inputs of pro tools.
   iii. System 5 Channel Strip
      1. Rotory Knobs and Switches. (hit EQ and those knobs will display accordingly. Same with compression etc)
      2. Display select. (Pans, EQs etc. are on different pages)
      3. SWAP AND MAIN SWITCH – If the swap layer button is OFF then you know you are on the MAIN LAYER.
      4. An ON BUTTON essentially works as a mute button.

g. Patch Bays
   i. IMPORTANT TERMS

WEEK 3: The Patch Bay & The WorkTape

- The patch bays here are highly succeptible to oxidation. So its smart to exercise the patch point by unpatching and repatching.
- On the Neve 1073 if you go into the Line Inputs it will kill phantom power.
- Beyer M100 was used on Questloves kit on the Booker T record. Everything else was just a 57.

I. The Work Tape
   a. Serves a few purposes
      i. Hearing their work for the first time
      ii. First time to work together
      iii. First tracking session in the scoring stage
      iv. Limitations
         1. No more than 4 mics maximum
         2. No more than two musicians
         3. No headphones
         4. No click
         5. No overdubs
         6. No EQ, Compression or Processing
   b. What to turn in
      i. Session Setup Sheet
      ii. Session Floor Plan
      iii. A Photo of the session
      iv. An MP3 bounce of the session
   c. Tips for the session
      i. Use the console Pres
      ii. Refer to picture taken on how to get maximum rejection on each mic for a singer/songwriter acoustic guitar setup.

1. MJ - Billie Jean. No two elements conflict at the same time.
2. Britney Spears - Toxic

B. Tempo Mapping in Pro Tools

- Pro Tools as a default changes the grid to fit the audio when tempo mapping, where Ableton warps the audio to fit the grid. Both can do the same if the default settings are changed.

1.Setting up the session

1. Go to the I/O Window
    1. Input Page - The physical input into Pro Tools. You can create paths (1/2) or sub paths (mono tracks). Always create a main stereo path and then mono sub paths to give you the most flexibility.
    2. Output Page - The Physical outputs leaving Pro Tools getting to the D/A converter. There are NO subpaths on the outputs page. Its setting each output as a stereo output, the reason is so that we can create busses.
    3. Bus Page - A bus is an internal audio path in Pro Tools. Without reassigning all outputs in Pro Tools you can route the bus page to do that for you if you want. Its must easier when moving studios this way, so you don't have to change your I/O setup.
    4. Clearing I/O Setups
        1. The Default button - If you hold the Option Key down while you're importing settings and then hit default, it imports the Input, Output and Bus pages all at once.
        2. Export IO settings - At the scoring stage, hold Option and then hit Import, then click Scoring Stage Default. The MADI inputs and outputs won't be there anymore, and it will just show 1-2 etc. ALWAYS delete the IO setup from before because the Bus window is additive. Which means that you will import new busses in addition to the ones you already have.
    5. Import the audio - CMMD I
        1. Select the Stems. Add All only adds them to the session, it does NOT copy them to the session. ALWAYS use Copy All when importing audio into a session. Don't use drag and drop because it does not move those audio files to the audio files folder, it just adds them. You can Save Copy In in

order to physically collect all audio files that were only added instead of copied.

2. You can also use the audition window to listen to files to check out tracks. NEVER use it to listen to mixes, it is a completely different audio driver.
3. When you click Done, it will default save your audio clips into the audio files folder.
4. First issue with the song is that its not set to a grid. Need two important tools on to do this:
    1. Tab to Transient. When you hit the tab button, it will automatically tab over to the next transient. Work with this on ALWAYS. Its very useful.
    2. Link Timeline and Edit Selection. Alt+cmmd+tab will turn that on.
    3. Timeline Follows Insertion. Should always be on as well for this to work.
5. What to map to.
    1. The kick is pretty laid back and on 2+4, so it would be better to use the congas as the template. At some point the congas will drop out so you have to map to the drums. Eventually you can move back to the congas and map.
    2. Use Cmmd+I, and type in bar 1 at the start of the downbeat. Now the grid will begin to change.
    3. set the grid to Quarter notes.
    4. Find the point in the song where the next 1 comes. Click Apple+I again and set the next bar. Use tab to transient to do this on every bar on the track.
6. Now create a click track.
    1. you need to listen to the click with the music in order to hear how the tempo is being mapped.
    2. Make sure you have the Conductor track enabled in the MIDI window. This allows you to make tempo changes.
    3. Option+CMMD{} will change the view size of the

range.

3.421 - Sounds similar to 57 but would sound better distorted

4.On Axis vs Off Axis 421

1. On Axis, sounds great clean. Very present.
2. Off Axis, sounds much softer. Would use more for distorted guitar sounds

C. Signal re-amping

- Guitar Amp Trick!
  - Start with taking the guitar (Instrument level) to the DI.
  - Come out of the DI at Mic Level.
  - This goes out to the Wall Panel via XLR to 177 or 178 in the Control room.
  - Patch from mic line 177 into a mic line in the live room (129)
  - Leave 129 via XLR to a second DI at mic level
  - Leave the DI at Instrument Level back into the Guitar Amp in another room.
  - Now mic the amp, and the guitar player is in another room
  - Convenient because you have a ground lift on both ends which can completely isolate the player from the amp.
  - Does NOT work with an active DI. Only use a Passive DI. The only reason it works is because it works through a transformer.
  - You also must go through mic/tie lines.
  - Possibly loosing a bit of high frequency, and or level.
  - Should never really be a problem if using a good DI like a Radial or a Jensen.
- RE-Amping a Guitar Signal.
  - Before Re-Amp boxes, the method described above was the standard way to re-amp a guitar.
  - The problem with Re-amping a guitar with a DI box is that you are using the wrong impedance to send signal to the amp.
  - Insert sends 1-48 are normalled to the Pro Tools outputs.
  - Patch from say Pro Tools input 14 into 129
  - Turn the level down
  - out of 129 into a DI box and into the amp.
  - NOW that there are actual Re-Amp boxes there is a different way to do it.
  - Pro Tools output of the DI signal (in this case it is 14.)
  - There are no “Pro Tools Outputs” on the patch bay. They are all digital. SO you have to patch from the insert send of the corresponding Pro Tools input which acts as a discreet Pro Tools output.

17. Filter out anything that is fighting low end space.
18. Almost always run reverb post fader.

WEEK 9: Melodyne Vocal Tuning, Vocal shootouts.

- Two things to buy while at Berklee that will get you work: Melodyne, and Izotope RX.

I. Melodyne and Vocal Tuning
- An incredible tool, and completely necessary as a modern music producer.
- On the market right now, Melodyne is the best option.
- Its important to distinguish between using vocal tuning as a creative effect (EDM, T Pain), and using it as an effective tool to make something still sound natural.
- Do not leave this class as a bad vocal tuner. Being able to use this as another tool in your arsenal as a producer is essential as a modern producer.

1.Melodyne
1. Realize that when you use Melodyne, you are running another DAW on top of Pro Tools
1. Transfer means record. Print the entire vocal at one time, because the program doesnt do well with punch ins.
2. Always make sure your vocal is comped and edited before you start using it.
3. Hit stop, and then Melodyne will analyse the vocal along a type of midi grid.
4. You can adjust both the modulation and the pitch of the singers voice independently, which gives you the ability to achieve a more natural vocal.
5. ALWAYS give Melodyne the cleanest signal possible, meaning transfer the dry signal without EQ, Effects, or Compression.
6. Melodyne will play any time you press play in Pro Tools, but what is really happening is your audio is playing out of the Melodyne plug in.
7. Right click allows you to open the tools window.
1. The Zoom tool gives you a lot of functionality
2. The hand key allows you to move around
3. The PITCH TOOL - is your most important, and useful tool.

1. DSP – Using the Processing the Pro Tools card. You are using HDX for example. Which means there is little to no use of your computer. This gives you very little latency
2. Native – The opposite. Running off of your computers CPU.
3. It is still helpful to share your DSP on another card, but the new macbook pros (this computer) makes this basically an old system
4. Pro Tools also has AudioSuite, which makes it so you can print an effect to an audio file. This does not happen in real time. Audiosuite uses Native. You would utilize Audiosuite for plug ins like RX Suite.

e. Using Izotope RX in Pro Tools
    i. Uses:
        1. Broadband Noise Reduction – Remove a sound or frequency under an entire track. For instance if you want to remove Air Conditioning noise or guitar hum. Helps best if you have the noise by itself. This lets it "learn" the noise, and applies a broadband compressor to remove that specific noise from the track.
            a. For removing Guitar Hum for example: highlight a section of the noise by itself. Go up to AudioSuite, find De-Noiser, and select RENDER. This will create a new audio file and place it directly on top of the highlighted Audio. Obviously a good idea to do in this situation in to start a playlist.
            b. Whole File Handle Space – this allows you in audiosuite to have a number of seconds on either side of your selection to fade in or out, instead of just the selected audio.
            c. Under RX De Noiser, go into Simple. Advanced gives you much more parameters. Select "Learn" once you highlight an audio file. Now that RX has learned the noise, you can now select the whole file and reduce noise.
            d. The NOISE REDUCTION bar can be

flourishes.
    c. Vocals, bass and drums are in the center (the most important things in the mix).
9. Use your ears, and use balance and panning.

ii. EQ
1. Parametric EQ – Usually the only EQ you'll use in mixing. Going to have 3 parameters that are going to be present:
    a. Gain – Allows us positive or negative gain. The center is the frequency you are choosing.
    b. Q or Bell – A higher Q is a narrower bandwidth, a lower Q is a wider bandwidth. Some EQs are not parametric and have a set Q or bandwidth.
    c. Shelves – Boosts everything either above or below a certain frequency range. Low or high shelves. Great for reducing frequencies in bass/kick, or boosting high frequencies. NOT to be used as a filter. This will sound very phasey and strange.
    d. Filters – Cuts out everything above or below a certain frequency. The Q of the filter just adjusts the steepness of the cut.
    e. George Massenberg created the best parametric analog EQ of all time. His plug in is also very great.

2. Using EQ in frequency reduction.
    a. Start by high pass filtering the bass.
    b. The kick should live in the 30hz-80hz range, and the bass should live on top of that. This is HUGE for mixing. This allows you to avoid the need of compression, EQ, or changing levels on either the bass or kick.
        i. Find the head resonance on the

reproduced. These were often recorded in bathrooms, giant stairwells etc.

2. Springs – A sound is played which excites a spring, this is then captured added back into the dry sound. These were highly popular because reverb was finally portable.
3. Plates – A sound is played which a large metal sheet (plate), this is then captured and added back into the dry sound. These are incredibly heavy, and are usually in a “plate room” very far away from where the music is being made because they are highly succeptible to outside noise.

iii. Tape Delay and Digital Reverbs

1. Tape machines need three heads to function. An Erase Head, a Record Head and a Play Head. ALWAYS in the order of ERP (erase, record, play). Les Paul invented a technique called Sound on Sound, which is where the erase head is removed so that he could just record on top of all the previous tracks. Essentially inventing multitrack recording.
2. Issues with using a tape machine for delay – You are limited in delay times to the speeds the tape machines motor will create. 3.5ips, 7.5ips,15ips, 30ips etc. The tape runs out eventually, so you’ll have to keep refreshing the reel.
3. Echo Plex has basically a tiny tape machine inside of it.
4. Next came analog delay chips. Solid state chups that could create variable delay. Once digital audio arrives, delay becomes very easy. The Lexicon PCM 42 is probably the most famous digital delay and allowed for a lot of variability
5. Algorithmic Reverb – By combining many digital delays together and utilizing modulation and filters, algorithmic digital reverb is born. The EMT 250 and EMT 251 were the first pieces of gear that did this and are still highly sought after.
6. Golden Age of Digital Reverb – By the 1980s as digital circuitry became more affordable digital

a. Create an Audio Track and select a specific compressor, de esser, EQ chain that you would typically use on Vocals for example. Do the same for Bass, Guitar, Organ, etc.

8. All of the busses should stay at zero, and your automation should be happening on the audio tracks.
9. This Routing described in these steps is the most effective gain stage possible for a Pro Tools session.
10. Update your Mix Template as you move from project to project, or create a new one for each album.

iv. NOW to Import this into an existing session.

1. First thing when importing your Template into an existing session is to clear out the I/O Window Outputs
2. Now you just have to assign your audio tracks to the proper busses.

II. Introduction to the System 5 Console

a. Introduction to MADI

i. MADI stands for Multichannel Audio Digital Interface
ii. MADI allows you to send up to 64 channels on one cable. If you double the sample rate, 48-96k, you will half the amount of channels (from 64 to 32). The old MADI cable ran 56 or 28 channels.
iii. MADI is carried through a 75ohm Coaxial Cable, a BNC connector.

1. Very durable, high strength and low cost. But you cant really send across a far distance. Still can go 100 meters, but you couldn't get across a football field.
2. The System 5 is running entirely on Coaxial Cable.

iv. OR it can be transferred through a Fiber Optic Cable

1. Can go Extremely long distances, but is fragile and very costly.

b. Dante

i. Very similar to MADI, but it runs multiple channels of digital audio over one network cable much like the

Pro Tools, Ableton or Logic.

4. Euphonix Converters – about 10 of them. Each one maxes out at 28 channels. This is because if you are running at 96K you will max out at 28 Channels. Blue snake goes to the control surface.
5. Comms Control – Shitty. Controls headphones, monitros, talkback etc over serial controls. The serial connection fails all the time and is not designed particularly well.
6. Remote Mic Pres – This is where the mic pres on the small channel fader are.
7. Master Clock and Distribution Amp – Word Clock and the distribution is for all of the converters.
8. Pro Tools HDX System – Total of 128 channels of Pro Tools in AKSS.
9. Aviom System – A16D Pro
10. Monitor Processing and Amplifiers.