21M.361 Composing with Computers I (Electronic Music Composition)
Spring 2008

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[All lab notes are being significantly revised for next term to reflect upgrades in software, different organization, and the incorporation of my own notes on DSP.]

Lab 2.1: Speaker-Mic Feedback; Outboard Processing (Filters, Delay, EQ, Aural Exciter)

Speaker-Mic Feedback and Mults:

1. Always hold onto the mic.
2. Start with it away from the speaker, with Main Mix volume set to zero. When we have done things with Pro Tools in the past, we know that the Main Mix does not alter what is actually happening in Pro Tools, rather, just what we hear. Since we are making a feedback loop, the Main Mix is our main volume and feedback control, thus also affects what is happening in Pro Tools.
3. The two speaker drivers (cones) in each monitor (speaker) produce different feedback sounds; the top one is more varied, but less enjoyable to listen to unprocessed.
4. The region between feedback and no-feedback can produce interesting sounds. The region between the two drivers is also useful.
5. Play this feedback 'instrument' largely by microphone manipulation, careful control of the Main Mix, and attention to the microphone pre-amp (the Pre-Sonus outboard device on the rack—this is a vacuum tube device, and is particular 'warm,' better than the built in mic-pres on the mixer). The patchbay connections to the mic-pres (for now) are irregular: Mic Pre 2 (out) is the rightmost B socket on the lower patchbay; and Mic Pre 1 (out) is above it but its label has fallen off.
6. You can use two mics and/or two monitors if you like. But you might not have enough hands to control levels (unless you are an octopus or squid).
7. Moving the microphone a whole lot can be effective. There are 'sweet spots' around the room where (I think this is what happens) there is constructive interference from wall reflections—anti-nodes of standing waves. Experiment with finding them. Different sounds and pitches can be obtained, especially very deep ones.
8. You can create little trigger sounds, snapping fingers, tapping, whatever. Cardboard tubes are fun (for this lab).
9. The SND filterbank will provide a useful means of finding different pitches. Have it at the beginning of the outboard signal path.
10. You can split a signal using the "mults" on the patchbay, on the upper one to the left. Each mult—there are two—has four sockets. Any of them can act as an input, and all others in the same mult will be outputs, splitting the signal. Don't have more than one input; a mult isn't a mixer. A mixer is. So you can split the signal to go in parallel to the two SND filterbanks, or in series.
11. A nice trick is to have the signal follow the path SND1 -> Delay -> SND2.

Outboard Devices:

1. SND fixed frequency filterbank:
You cannot set Q (quality or resonance, in more simple terms, relative bandwidth) or frequency, just +/- gain, like on a simple graphic equalizer, like on iTunes, etc. The left and right pots are input and output gain, and they both should be near full. If the SND doesn't seem to work, check that its blue light is on. If it is off, then someone has knocked the 'bypass' button to the right of the device. Just press it to make it work again. There is no on/off switch. Play with this toy a lot. It will give you different pitches, and portamenti. And hisses, rumbles, and a lot more.

2. MXR Delay:
First check to see if the red clock is working. If not, switch the device on. The main pot is the larger one in the middle surrounded by blue. It sets the coarse delay time. To the left is the fine tuning for delay time. Speed and width refer to the sinusoidal

fluctuation you can have around the delay time. Mix sets how much of the dry (input) signal you have—fully counterclockwise, it is all dry (thus defeating the purpose of using the delay). Fully clockwise is all wet. It is sometimes useful to change this pot. Regen is internal feedback within the delay unit. Use! It causes there to be more than one echo. The level pot functions in reverse. One of the knobs is missing. Please treat this instrument with care; it is old, and it is mine. (So is everything else on the rack, and the rack frame itself, so please take extra special care!)

3. Aphex EQ:

This is also mine, so treat with extra care. You can set frequency, bandwidth, and +/-gain. Narrowing the bandwidth and accidentally getting a sweet frequency can actually be quite bad for your ears and the speakers. In the middle of the EQ are two buttons; the left one should be pressed in, which means you have access to all 4 bands for your mono input. Alternatively, you can use this device as a stereo EQ, each channel having two bands. In the past, this device has been as useful as the previous two for this assignment. It has a particularly 'warm' sound, emulating a vacuum tube device.

4. Mic-pre:

One for each microphone. Both pots should be set to halfway to start with; the further clockwise the 'drive' pot is clockwise, the richer harmonically your sound will be (or more overdriven—distorted, like old guitar amps).

5. Mixing board EQs:

They are just above the volume faders. Use only if you want to rule the universe: we already have tons of EQing going on.

6. Aural Exciter Type C:

Lately I haven't been able to get this working, though when it is working, it is hard to tune, so sometimes it appears not to be working when it actually is. The inputs and outputs are labeled 'excite.' There are two channels to this toy, three pots per channel. Tune sets a fundamental frequency, drive determines how harmonically rich and 'excited' the signal will become, and mix is similar to mix on the delay, determining dry/wet mix. I think the in/out button just controls the LEDs (but I could be wrong). Perhaps they are bypass buttons, I forget. What does this thing do? It generally improves your life. It puts back in your sound the harmonics it was supposed to have to begin with. Anyway that's what the blurb says. It makes sounds harmonically richer in some way. It was intended to be used for enriching dull vocals. It is hard to use, and you might not hear any effect from it.

Cables

How to patch all these things together? I was kind of hoping you might be able to figure out how to use the patchbay by means of logic rather than mechanical reproduction. But here goes. My suggested signal path for one mic is: mic -> Mic Pre 1-.> SND 1 In...SND 1 Out -> Delay In...Delay Out -> EQ 1 in...EQ 1 Ou -t> [aural exciter if it works] -> Mix In whatever, and SND 2 wherever you like. The mic-speaker feedback is simply mic>pic pre 1>mix in 1 (e.g.)...submix 1...L...main out>studio out. That is all in series. you can use parallel too, using the mults, but I do recommend this serial approach, at least to start with. (For you who don't know, serial means things following one another, and parallel means things happening at the same time.) Be careful with mults: they take only one in (from an out), and up to three outs (to ins). Pretending they are mixers means you are connecting two outs from other devices, potentially damaging them. This is mentioned under Speaker-Mic Feedback and Mults, above. You can use two mics, but we are not octopi, at least not all of us, so it is possible to hold two mics in one hand, for the same speaker, and you can then have one going somewhere, and the other somewhere else, in other words, in parallel. Record everything you do: take the signal from submix 1 (->Mix Out 1 -> 96 In 1, or something like that, etc.)