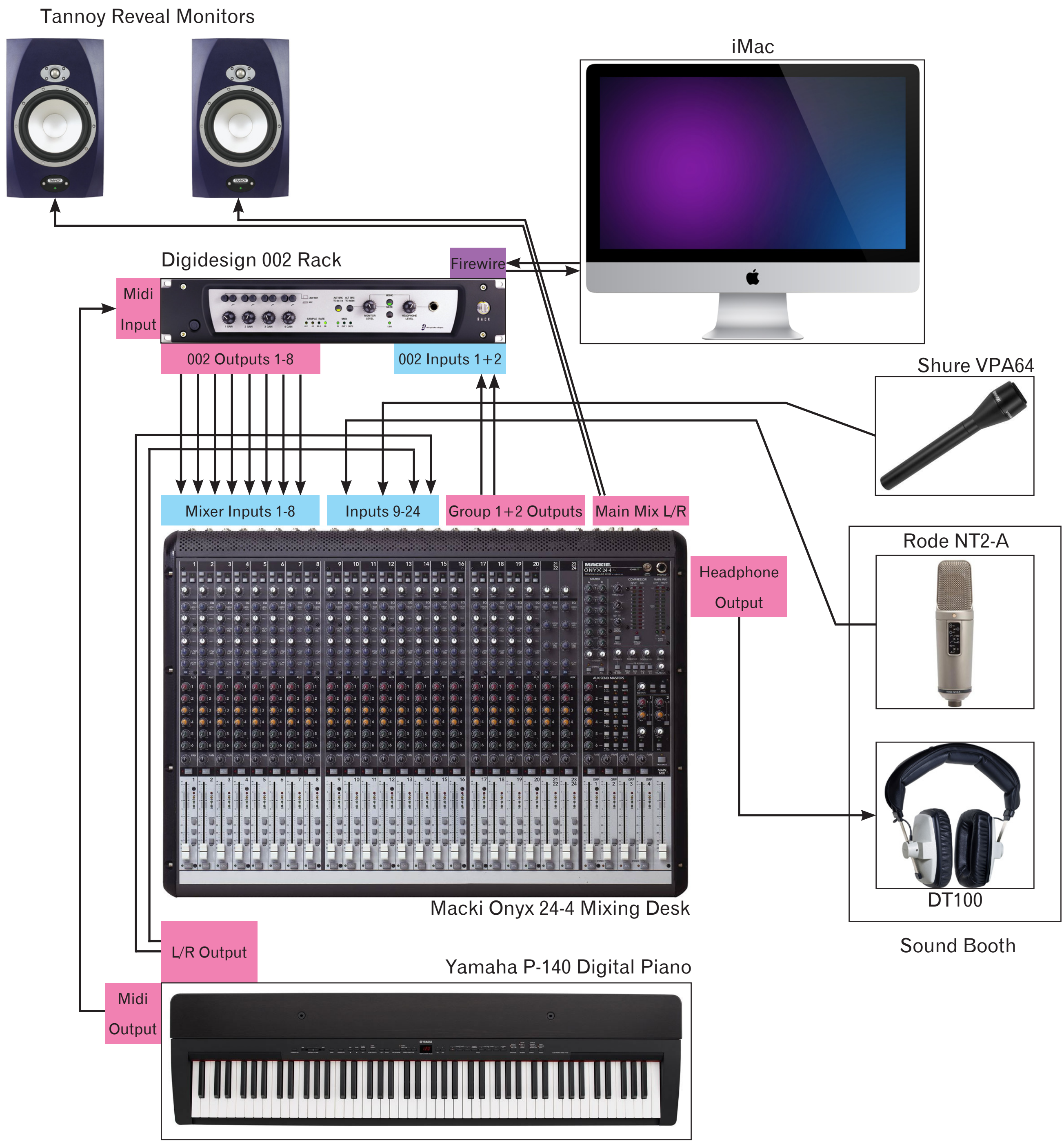
Tracking

<u>Large Sound Studio</u>

Hardware Configuration



Power up/power down procedure

In order to prevent damaging them, the golden rule is:

» ***Turn the monitors on last and off first.***

Everything else can be powered in any order. The monitors have their own power source on the wall.

Gain

On a technical level, making good sounding and clean recordings is largely about managing recording levels between different devices, and between processing sections within a single device. This is called gain structure.

Analogue Gain Structure

Essentially, gain is amplification of the signal, measured in decibels (dB). Gain is a factor in most audio processes.

At every stage of recording, from the microphone to the final output, gain must be optimised to prevent both noise (caused by insufficient signal level relative to the noise floor) and distortion (cause by excessive signal level leading to signal clipping).

The first gain stage is the preamp on a mixer, stand-alone channel strip or soundcard. The small voltage produced by a microphone is amplified to the operating level of the unit in question.

Mackie Onyx Metering Section (with PFL meter)

On an analogue mixer this level is normally 0dB. Turn the input gain pot until the signal hovers around 0dB as indicated by the PFL (pre-fader listen) meter when the PFL button is pressed for that channel. This level gives the best SNR (signal-to-noise ratio) while leaving plenty of headroom for transient peaks.

A gain of 0dB represents an unchanged signal level: output = input. This is sometimes called unity gain. Thus when the input gain pot is placed at 0dBu (or 'U' on the Mackie mixer for unity) no gain is being applied.

Mackie Onyx Channel Strip

Logic X Channel Strip

Similarly, when the input fader is at 0dB, the fader is not affecting the level of the signal. The same goes for output faders. Therefore, the simplest and cleanest recording chain is a microphone with gain applied at only one stage (the preamp, controlled by the input gain pot), with all subsequent faders set to 0dB.

Remember that most audio processes involve gain in some form. Equalisers (eqs), for example, are in essence frequency selective gain units. If a lot of eq boost or cut is used, the amount of gain applied by the preamp may need to be lowered or raised accordingly to keep the overall signal at the ideal level.

Simply, the idea to keep in mind is that whatever you are doing in each process you should aim to keep the signal at the ideal level for the piece of equipment you are using. For most professional audio equipment in the analogue realm, this is 0dB.

Digital Gain Structure

The dB scale is logarithmic relative to some known reference at 0dB, so there are different flavours depending on the context. For example, on a professional analogue mixing desk, gain controls the voltage of the audio signal, the reference is a fixed unloaded voltage of 0.7746V, and the scale is named dBu. Therefore an audio signal of 0dBu has a voltage of 0.7746V.

In the digital realm, the scale used is dBFS (dB Full Scale), and the reference for 0dBFS is the maximum peak signal possible as represented in binary by an audio sample. Consequently, there are no positive dBFS values.

In the analogue realm, we try to operate with signals at or around 0dBu as discussed. The clipping point (the highest level possible without undesirable distortion) for pro analogue mixers will be around +20 to +30 dB above this depending on quality (the Mackie mixer is +21dB). This allows undistorted representation of transient audio peaks, and is called headroom.

In the digital realm however, since 0dBFS is the maximum signal, there is no inbuilt headroom. There is also no standardised conversion between analogue and digital decibel scales. However, in practice, all that it's necessary to realise is these decibel variants represent audio signals in the same relative fashion: a change in signal level of *x*dB will sound the same in either case. Therefore, digital recordings should be made with headroom created artificially by the operator, by simply recording with levels metering in a DAW (Digital Audio Workstation) such as Logic at around -20dBFS. To ensure the greatest SNR, always record 24bit files. This way, unpredictable audio sources can be recorded at an average level much lower than -20dBFS with no increase in noise.

In summary, to record an audio source through the Mackie into Logic with good gain structure:

Mackie Onyx Master Section

1. Power up mixer and iMac first, then the monitors
2. Load an audio template in Logic and arm a track to record
3. Start with all faders down and all channels muted
4. Plug a microphone into a free mixer channel, activate +48V if required
5. Press the PFL button at the bottom of the channel strip
6. Audition the performer and apply input gain so the PFL meter hovers around 0dB
7. Apply channel EQ and engage HPF (High-Pass Filter) as desired
8. Adjust input gain to compensate for EQ if necessary
9. Route channel to Group 1 and/or 2
10. Raise Group fader to 0dB and unmute
11. Unmute and raise channel fader so that meter in Logic hovers around -20dB (you should end up with the mixer fader at around 0dB)
12. Record 24bit files in Logic (this should be the case by default)