**Plugging Microphones Through Patch Bays**

Plugging microphones through patch bays is useful in a studio setting, however, a few things should be considered so that no equipment gets damaged, for example, if phantom power is used it could damage the console.

As the plug is inserted into the patch bay, its tip will briefly connect with the ring contact in the socket, while the ring of the plug touches the sleeve contact. This short-circuits the phantom supply, bridging the +48V line straight to the main return and, though the phantom supply should manage, the resulting voltage increase can cause permanent damage to the input circuitry of the mixer channel. The mic input stage can be destroyed completely but it is more usual to find a steady decrease in performance as different circuit components worsen.

Also, unless the labeling of the patch bay is clear, it’s very easy to plug a line source into a mic input accidentally. The result will be a loud and dreadfully distorted signal, but it is possible for the phantom power on the mic sockets to destroy the output stage of the line source you are trying to connect.

One simple way to resolve these issues is to make sure that the phantom power is turned off before plugging in the mic to the bay. Some mixers have the phantom power button on the back, making it harder to see or make sure it’s off. Some more newer consoles have them on the main board, but it’s one key that you should always make sure of when connecting a mic through a patch bay.

If you are concerned about this kind of accident, a reasonably practical alternative (and one I personally recommend) is to patch the mics on a dedicated XLR patchbay -- Canford Audio, for example, can supply suitable panels. XLR sockets are designed so that the earth connector (pin 1) contacts first, with the two signal pins connecting fractionally later, thus avoiding any chance of the phantom supply being shorted. The contacts are also much better suited to delicate microphone signals, and you cannot accidentally plug a line-source jack plug into an XLR socket! In most broadcast studios the usual practice is to supply phantom power direct to the studio wall points, which is an alternative method of avoiding having phantom power across the jackfield.

**Signal Flow Chain**

At the beginning of the signal flow chain there has to be a microphone line. This line is a direct transfer of the audible sound to the console. Microphones work as transducers and convert the audio into an electrical current. Speakers are also transducers as they convert the electrical signal to an audible sound. Microphone lines give no effect to the sound; they provide the most basic and clean audio.

The auxiliary send provides a space for plug-ins to be activated. Plug-ins allow the engineer to insert a special effect on the audio signal. Many engineers use reverb or delay to produce a unique effect on a singer’s voice or even add in a loud distortion on the lead guitarist’s riff, for example. The auxiliary send is another part of mixing that enhances the audio’s character, but it’s not something that’s requierd.

It is important to know at which volume level (dB) the signals are set at before recording. If a signal is too loud, it can blow the speakers and damage more recording equipment.

It’s often that when listening to music, it’s possible to point out where the instrument that’s being played is coming from. The pan knob allows the engineer to “place” the instrument in either the more left or right side of the recording. This adds in a great feel to the music.

After all these steps have been completed, another and final step to consider would be synchronization. For example, if a drum kit is being recorded, it would usually need at minimum 6 mics. Instead of having to adjust the volume for 6 lines, the mics could be synchronized into a sub group and have them adjusted with one knob rather than 6.

 Once all the audio has been set and adjusted through these steps, a master mix is in place.

**PATCH BAYS PROS AND CONS**

**CONS**

* Having to control cables by getting more cables
* More cables increases the odds of one cable going bad, ruining tracks, or just degrading the signal
* The shortest analog signal path is always recommended, and by adding a patch bay and cables , it becomes lengthened
* If you already have more inputs and outputs than you need and you have easy access to them, there’s no real point in having a patch bay

**PROS**

* You have more gear than you have inputs on your console or audio interface. The fewer inputs you have, the more a bay is useful
* The back of your audio interface or soundcard is in an isolated, out-of-the-way place
* You have an analog mixer with inputs, busses, sends, returns and a lot of outboard gear, like effects units, compressors, pedals, samplers, synths and standalone audio recorders. Total flexibility is needed in order to connect them in a unique fashion for each project that’s being done
* You need a way to divide the audio signal to go to different processors while not disturbing the original signal path
* If there’s a modular multi track recorder with separate channel ins and outs and want flexibility patching channels to an analog mixer, for example, may require a few more patch bays