Using effects with DAWs

Processing Sounds

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Recording and mixing audio almost always involves sound-shaping processors and various special effects, for several reasons:

- Imperfections of sound sources (instruments, voices, samples, and sound effects)
- Microphone, DI, preamp, and A/D-D/A converter limitations
- The need to reshape the frequencies and dynamic behavior of multiple sound sources so they can be successfully blended together
- The artistic element — using special effects to "color" sounds so they can stand out

Using DAWs and compatible DSP plug-ins (digital signal processing software) allows users to experiment with sound-shaping tools that were previously not available. Hundreds of different plug-ins from dozens of manufacturers are available for your DAW of choice, and many of them are of the highest professional grade.

This article examines some of the more common types of processors and basic usage methods, as well as interfacing with external analog and digital sound processors.

TYPES OF PROCESSORS

Equalizers: When auditioning sounds for recording or mixing, we often use terminology not traditionally related to sound production: "thin," "fat," "muddy," "crispy," "cloudy," "harsh," "honky," "hollow," "bright," and "dark" are some of the expressions that describe a sound's timbral quality. EQs are the most suitable processors for shaping a sound's spectral response and tone.

Dynamics Processors: The difference between loud and soft sections in a performance determines a sound's dynamic range. Too wide of a dynamic range can exceed that of the capture or playback medium, and the way to restrict the range is with dynamics control processors such as a compressor or peak limiter.

Increasing the input control in either a compressor or limiter will allow a sound's softer moments to come up in level, while not allowing the louder parts to get even louder, thus narrowing the sound's dynamic range. Gates (which mute a channel in the absence of a signal) and expanders (which expand, rather than compress, dynamic range) are also dynamics processors. Frequency-dependent compressors, such as de-essers, narrow the dynamic range of selected frequencies only.

Space and Ambience Simulators: Every sound we hear is a combination of the sound source itself and the way it interacts with the acoustic environment in which it is being recorded or monitored. Some sounds lack this interaction, such as artificial sounds generated by synthesizers and sounds recorded using DI (direct interfacing or direct injection) techniques. Others are recorded in rooms that are too small and baffled to produce any reflections, thus sounding "dry." Reverb processors simulate the reflections and echoes that occur from sounds interacting with an acoustic environment. Reverberated sounds will be a bit more natural and live ("wet"), while processing multiple sounds with the same reverb (unifying their ambience) will give them a more cohesive sound when mixed together.

Special Effects: This final family of processors "colors" sounds in a unique way, rather than correcting or solving problems they may have. The most commonly used effects are:

ANALOG TAPE AS REALTIME PROCESSOR

Although not as convenient to work with as DAWs, there are still some distinct qualities associated with the sound of analog tape and tape machines. Even if a project was produced entirely using a DAW, it's still possible to process some or all of a session's tracks with an analog tape during mixdown. This doesn't involve transferring the audio to an analog tape reel for mixing, but rather uses the tape machine as a real-time effect processor.

Place a reel of tape on any analog machine (the more tracks it has, the more processing you can do at once), set up the tape speed (slower will make the tape last longer, faster may improve sound quality), and arm all tracks. Route the outputs from your DAW to the tape machine, and the machine's outputs back to the inputs of your DAW. It's now possible to send some or all of a DAW session's tracks through the tape machine. Switch the machine to "repro" mode, which places both the record and playback heads on the tape at the same time, and start recording. Sounds are recorded onto the tape, and, after a few dozen milliseconds, play back from the tape via the repro (playback) head (tape is rolling free in record the whole time). The reason for this delay is the physical distance between the record and playback heads on the analog machine. (The record head is located before the playback head.) The tape machine's output can then be sent back into the DAW for mixing or to be recorded to new tracks.

Once you've figured out the delay time between the record and playback heads, shift each track you're sending to tape to start earlier by as many milliseconds as the delay time, making them play back in time with other DAW tracks not affected by the analog tape process.

As the tape is functioning only as a real-time sound processor, you can stop and start the DAW playback whenever you want, loop sections, and keep editing as you go. When the tape runs out, don't rewind it — just flip it over and start rolling again. This method works beautifully, and is an elegant way of incorporating some real analog qualities into digital audio, without losing any of the benefits offered by a DAW.