Subject: Audio Geeks: Mid-side mojo and questions
Posted by duncan on Mon, 29 Jan 2007 05:59:50 GMT
View Forum Message <> Reply to Message

OK, not an exclusively "Paris" type question, but not entirely OT either: I'm mixing some tracks here that were recorded to ProTools by another guy in a nice Neve room in LA. Tracks are nice and fat... Trio jazz piano/vocal type stuff... But this piano: tracks are all set up "Mid - Side"...

So -- I'm an all in the box guy here on my humble Paris system. My approach for this is to copy the "side" track to another channel, flip phase, pan the two copies left and right, group those faders and push them up to taste in combination with the "mid" channel, which is doing most of the work and is panned -- doh -- center... However, the session engineer -- an old-school English guy with about 300 times more experience than I, says he has a better way to decode this. His way: send the "side" channel to an aux (post fader, pre pan), return it to another channel in the mixer (set on "input"), leave them both center panned, the first channel at "0" (nominal gain), then flip phase and push up the return's fader until it nulls -- which it won't, actually... (here I say "huh?" and he says "Yeah, it's a ProTools thing") But anyway, push it up until it "thins out" as far as it's going to... Then, leave that fader right there forever, pan the channels left and right, and control levels for both "sides" with the one fader (of the channel which is sending...). Which works actually very well -- and *does* sound different and better than the "copy channels and flip phase" method... But, in a word -- Why...?!

By the way, in order to do this his way on my rig, I'm sending to an external output -- adat lightpipe looped back to itself -- which is then patched to another mixer channel which is set to active input... Probably introducing some itty-bitty latency here which is why the channels don't null... I'm also going to test this (for kicks) with analog in and out -- but I doubt things will get any better, as there would also be a couple more conversions taking place, and who needs that? One other potential issue: these are 24 bit files at 44.1 -- and I'm wondering if my lightpipe patch for the aux is truncating the signal to 20 bits? (I know this has been discussed in the past, but I'm just dumping all the questions in here now in case anybody wants to actually clarify *everything* for me in one go...)

Thanks, I realize this is rather long-ish... And I'd almost put a smiley-bob here -- if I actually ever did that sort of thing.

-- chas (thinking about buying all of DJ's stuff, bailing it up, and sending it into earth orbit as the first truly vintage digital audio satellite array)