Equalizers and Phase Shift

In the beginning all equalizers were analog electronic circuits using capacitors and inductors. These components shift the phase of AC signals passing through them. If you combine a signal with a phase shifted version of itself (after passing through the capacitor or inductor), the frequency response is altered. As one cycle of the wave is rising, the shifted version is falling, or perhaps it hasn't yet risen as high. So when the two are combined they partially cancel *at some frequencies only* thus creating a non-flat frequency response. Therefore analog equalizers work by intentionally shifting phase, and then combining the original signal with the shifted version. In fact, without phase shift they would not work at all!

Most digital equalizers mimic the behavior of analog equalizers, but with a completely different circuit design. Instead of using capacitors and inductors to shift phase, they use taps on a digital delay line. A digital delay line is a series of memory locations that the numbers representing digitized audio pass through. The first number that arrives is stored in Address 0. Then, at the next clock cycle (44,100 times per second for a 44.1 KHz. sample rate) the number in Address 0 is shifted into Address 1, and the next incoming sample is stored at Address 0. As more numbers enter the input they are shifted through each memory location in turn, until they eventually arrive at the output. This is the basis for a digital delay, and you can alter the delay time by changing the total number of addresses each number passes through or the sample rate or both. (A series of memory addresses used for this purpose is sometimes called a *shift register* because of the way the numbers are shifted through them.)

To create an equalizer from a digital delay line you tap into one of the intermediary memory addresses and feed a varying amount back to the input. Just like the feedback control on a tape recorder-based delay like an old EchoPlex. Except without all the wow and flutter. You can also reverse the polarity of the tapped signal before sending it back to the input to get either cut or boost. The bottom line is the delayed sound combines with the input - just like a flanger effect - to create peaks and dips in the frequency response. By controlling which addresses along the delay route you tap into, and how much of the tapped signal is fed back into the input and with which polarity, you create an equalizer.

With an analog EQ the delays (phase shift) are created with capacitors and inductors. In a digital EQ the delays are created with a tapped shift register. But the key point is that *all* EQ shifts phase, unless it uses special trickery.

Some people confuse the sound of Phaser and Flanger effects with the phase shift that's inherent in all equalizers. Phaser and Flanger effects use phase shift internally in the same way an EQ does, but what they really do is alter the frequency response. *That* is what you hear when you run a track through a Flanger. It's the comb filtering - a series of many peaks and dips in the frequency response - that drastically changes the sound, not the audibility of the phase shift used to create that effect. The "problems" caused by phase shift have been repeated so many times by magazine writers and audio salespeople that it's now commonly accepted, even though there's not a shred of truth to it.

Some people claim they can hear phase shift in equalizers because when they boost the treble they hear a "phasey" sound. So they wrongly assume what they hear is the damaging phase shift everyone talks about. In truth, what they are hearing is comb filtering that was already present, but subdued. When a microphone is near a room boundary like a wall or ceiling, or when placed near the open lid of a grand piano, the delay between the direct and reflected sound creates a comb filter acoustically in the air. When the treble is boosted by EQ the comb filtering becomes more apparent. But the EQ did not add the phasey sound, it merely brought it out.

The same thing can happen when mixing tracks recorded with multiple microphones in a room. For example, a mike near the snare drum picks up the snare sound as well as sound from the nearby kick drum. So when the snare and kick mikes are mixed it's possible for the low end to be reduced - or boosted - because of acoustic interference caused by the arrival time difference between mikes. So while phase shift is indeed the cause of the response change, it's the response change that you hear, not the phase shift itself.

For even more compelling proof that phase shift alone is inaudible, see this gem I recently discovered:

Some Experiments With Time

ADDED JULY 9, 2009: If you'd like to hear for yourself whether phase shift is audible, download this freeware VST plug-in:

http://www.lesliesanford.com/VSTEffects/SanfordPhaser.shtml

Set Feedback and Modulation to 0, Left and Right Frequency the same and near the middle, and Wet/Dry to full Wet (slider all the way to the right). Then disable Sync and Quad, set Mod Source to None, and select 4 or 6 stages which gives far more phase shift than you'd get from any gear in a normal situation. Now play a track without the plug-in, stop playback, and play it again with the plug-in engaged.
