The 'PhaseBuster'

An Adventure into Audio Phase Manipulation

The directionality, localization and ambiance aspects of a stereo audio performance are defined not only by amplitude differences between the left and the right channels, but perhaps even more importantly by time and phase differences between program components that are common to both channels. We can do little to change the relative intensities of sounds within an existing stereo soundstage, but some very interesting and possibly useful effects can be had by manipulating phase between the two channels.

PUBLICATION CREDITS

The bulk of this write-up first appeared as a build-it-yourself article in the November, 2015 issue of *audioXpress* magazine.

BACKGROUND

Inspiration for this project sprang from two incidents separated by more than 20 years. The first involved one feature of an audio diagnostics product that the author's company developed back in the 1990s.

To determine and accurately meter the dynamic range of radio broadcasts, we combined the left and right channels to monaural with simple L+R summing. But the absolute energy of the sum did not quite agree with what was contained in the individual channels when measured separately. Another project, proceeding in parallel at the time, was an unrelated collaborative effort with the CBS Technology Center, formerly known as CBS Labs. The late Emil Torick represented CBS in this venture, and kindly volunteered his wisdom on the other matter that had us stymied. He suggested offsetting the phase of one stereo channel by 90 degrees to yield a more consistent stereo program energy profile for mono that is derived from combining stereo program channels. In fact, Mr. Torick whipped-out a mathematical validation for this idea, which we christened "QDM," for Quadrature-Derived Monaural.

This quadrature trick worked quite well, giving a more honest representation of bass and vocal track energy in pop music recordings. Pop music is typically recorded as multiple individual tracks with bass and vocals commonly panned to the center of the soundstage. This results in a certain degree of energy buildup when left and right channels are summed to mono.

The more recent inspiration for this project came from my own frustration in mixing-down certain orchestral recordings to monaural. A longtime friend employs the historic "Decca Tree" microphone technique (check *Wikipedia* for this) in making very lifelike binaural recordings of live symphonic performances. I, on the other hand, take great delight in cutting lacquer discs with a vintage lathe that predates stereo. In mixing my friend's .wav files to mono, I found that a lot of the music suddenly went missing. Certain instruments, along with a good deal of the venue ambiance, simply disappeared! While this phenomenon can also be noted to a lesser extent in multi-tracked, artificial-stereo pop music, my friend's live recordings, with his mics arranged for true binaural/3D imaging, just refused to mix-down to satisfactory mono.

Thus one goal of this PhaseBuster project was to reduce a stereo program to more realistic mono by imparting the 90-degree phase offset between channels as suggested all those years ago. And as an adjunct to this idea, why not use this technique to correct stereo material such that it would inherently mix-down to better mono when the two channels were combined with simple L+R addition? Although we don't normally reduce stereo to mono intentionally, today's FM receivers incorporate blend-to-mono circuitry to minimize background noise as the signal fades or as multipath effects distort the FM baseband. Although absolute directionality of a stereo program with an intentional 90-degree offset between the two channels might well suffer at the expense of maintaining better mono compatibility, it could be worth a try.

SYNTHESIZING STEREO

When we first tried that 90-degree offset idea for mono mix-down years ago, just for the heck of it we tried the reverse as well; that is, to create a pseudo-stereo program from a monaural source.

Various methods have been proposed over the years to synthesize stereo from mono material. In the early days of stereo discs, record companies used crude frequency reassignment to convert their existing monaural catalog to the new format. “Re-channeled for stereo” was often as simple as scooting the bass to the left and the treble to the right, although this was quickly refined with a greater number of frequency divisions and alternating the bands between the left and right channels. Bass frequencies were kept in the mono domain, however, and pop music producers would later ensure that everything below about 200Hz was recorded as mono. After all, bass is pretty much omnidirectional by nature and more efficiently reproduced by two loudspeakers driven in phase. Also, avoiding low frequency vertical groove modulation ensures better tonearm tracking and reduced playback distortion of stereo records.

The early years of stereo TV sound marked a comeback for stereo synthesis. Bob Orban devised one of the better analog stereo synthesizers (US Patent 3,670,106), which saw widespread use by television broadcasters in the 1980s. RCA even proposed building a stereo synthesizer into the TV set itself (US Patent 4,239,939). Both strategies split the audio spectrum into filtered frequency bands alternating between the left and right loudspeakers to impart a certain degree of ambience to the performance.

These cited stereo synthesis methods are static in their operation. They do not, themselves, generate dynamic motion within the soundfield. But another clever synthesizer does just this, and is the subject of a DIY article by Ethan Winer. It first appeared in the June, 1979 issue of *Recording Engineer/Producer* magazine and may still be available on Mr. Winer’s Website: www.ethanwiner.com/St-Synth.html. Mr. Winer’s design employs a similar band-splitting approach, but with an interesting twist: the band crossover frequencies are constantly moved up and down at a subsonic rate to give a pronounced ambience sensation.

The broadcast industry is largely responsible for the “loudness wars” that began with AM radio in the 1960s. Despite claims to the contrary, not only are loudness wars still with us, but they have metastasized from top-40 programming to nearly all AM/FM radio formats, as well as to television commercials and even to CD and downloadable music releases. I’ll avoid a rant on loudness here as it’s outside the scope of this article. But at some point someone simply must spearhead a crusade back toward sonic sanity.

Compressing the dynamic range of music to just a few dBs is certainly bad enough. But once the effort to make radio programming louder reached a point of diminishing return, sure enough, some radio Program Director asked his engineer, “Well, if we can’t be louder, can we be wider?”

Most broadcast audio processors (those mysterious boxes ahead of the transmitter) now offer a means of expanding the stereo image, or making the stereo wider, perhaps even wider than a perimeter defined by the placement of the speakers themselves… a first-order impression of ‘surround sound’ from just a pair of stereo speakers! Popular techniques employ dynamic expansion of the L-R stereo difference signal, frequently making use of phase manipulation as well. The result can certainly make a performance seem more spacious, sometimes too spacious, with an obvious ‘hole in the middle’ and a sense that the performance is somewhat disjointed. Sadly, just like the loudness situation, this practice has migrated into music production techniques as well.

Nevertheless, there are instances where a bit of stereo image enhancement can sound pleasant. Recordings from an earlier, more conservative era are a case in point. Consequently, a ‘spatializing’ utility was incorporated into the PhaseBuster project.

PROJECT GOALS

1. To create a monaural mix from stereo without losing audible components that end-up out of phase, either accidentally or by design.
2. To process a stereo program for better mono compatibility; that is, a program that will sum to L+R mono without program component loss.
3. To create a pseudo-stereo program from a monaural source.
4. To ‘spatialize’ a stereo source, increasing the apparent soundstage width.

THE BUILDING BLOCKS

Figure 1 shows a typical first-order all-pass filter section. This single op-amp stage exhibits non-inverting, unity gain for DC and low audio frequencies. At very high frequencies, the capacitor essentially grounds the non-inverting input and the stage flips the signal by 180 degrees, still with unity gain. Between these extremes the output will exhibit a frequency-dependent phase offset from the input signal, but will maintain a flat amplitude response. Figure 2 graphs the amplitude and phase response of this circuit, which has values for a 90-degree phase lag at 1kHz.

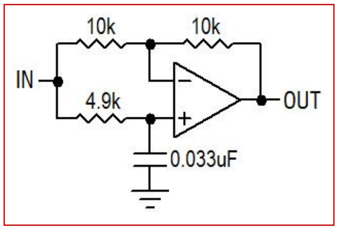


Figure 1

The PhaseBuster utilizes a number of these stages. The frequency-dependent phase offset shown in Figure 2 is a simple matter, but our need for a 'quadrature' phase offset (90 degrees at *all* frequencies) is a bit more complex.

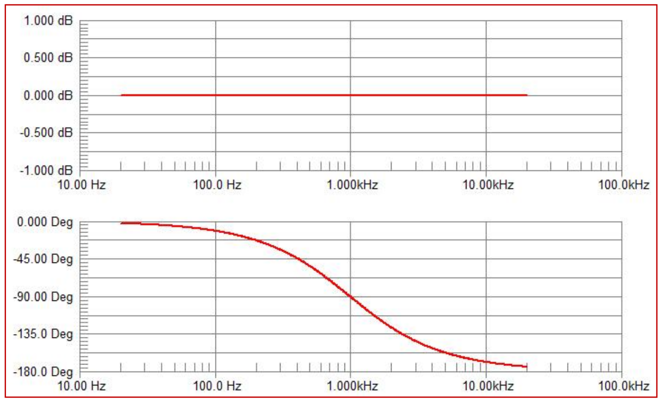


Figure 2

Figure 3 is the complete PhaseBuster schematic. At the center-left of the diagram, note the arrays of cascaded all-pass filter stages. There are three banks of cascaded filters, and the top two, utilizing op-amps IC3, IC4, IC5 and IC6, are of initial interest to us here.

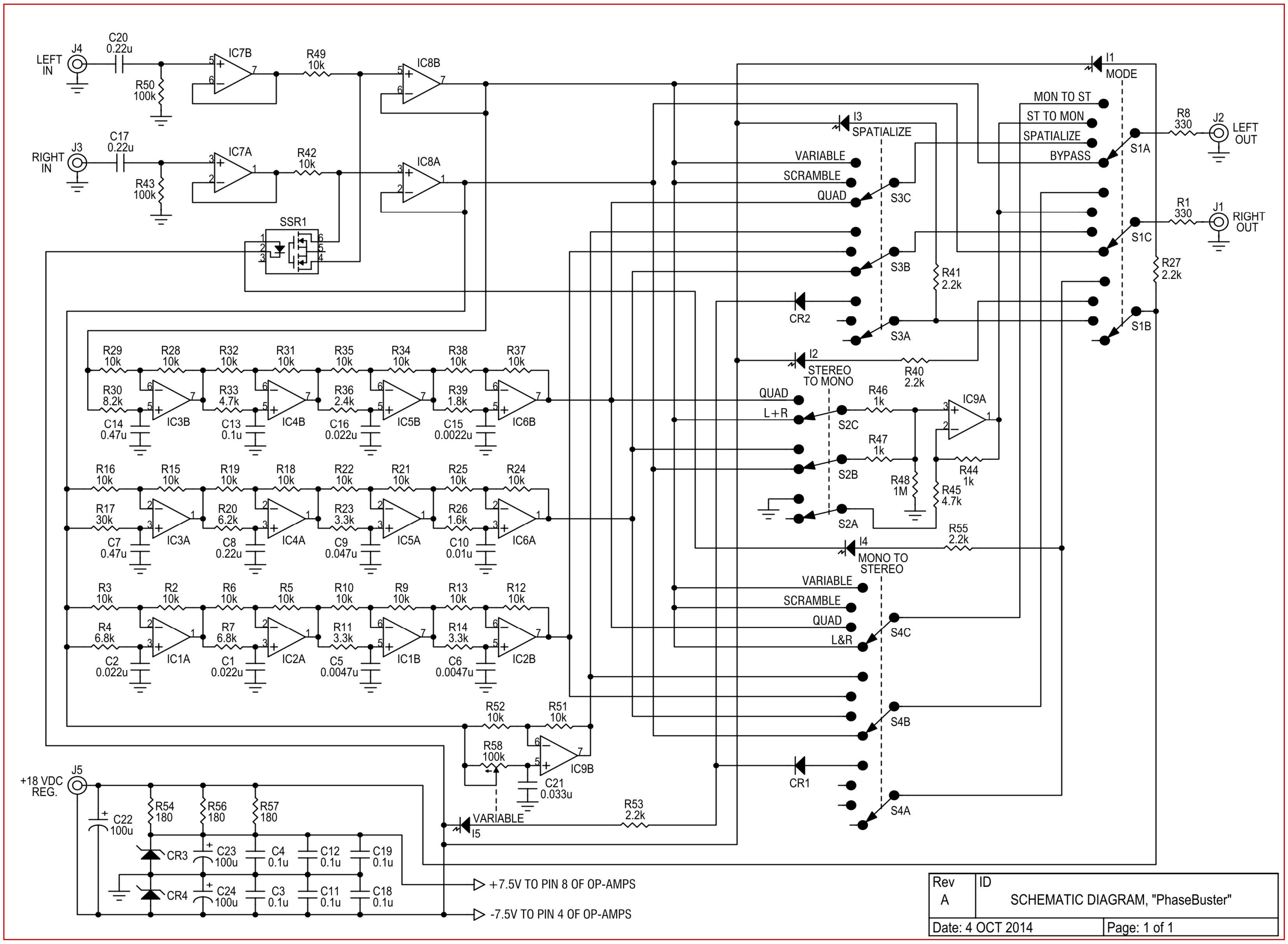


Figure 3

THE BUILDING BLOCKS

The amplitude and phase characteristics of these top two filter banks are graphed together in Figure 4.

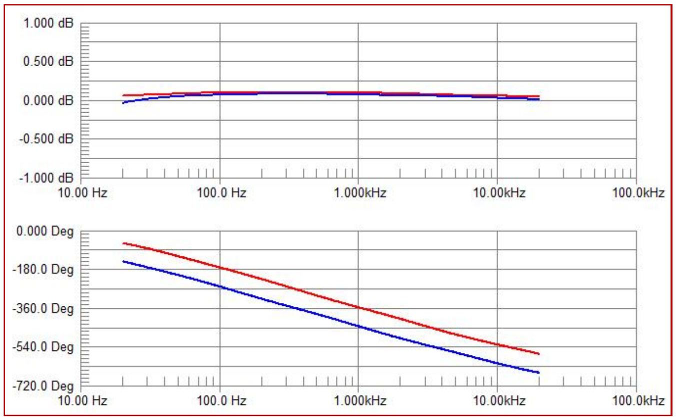


Figure 4

With the specified component values, four cascaded all-pass sections will yield a frequency-dependent phase offset that follows a relatively straight line from 20Hz to 20kHz, a phase-lag slope of about 180-degrees per decade. But note that the phase difference between the outputs of the two banks is a constant 90 degrees at any and all frequencies in the audible range: a true quadrature relationship.

The third filter bank (IC1/IC2) is similar to the others, but has a steeper phase-shift characteristic that turns-over (begins) at a higher frequency, as graphed in Figure 5.

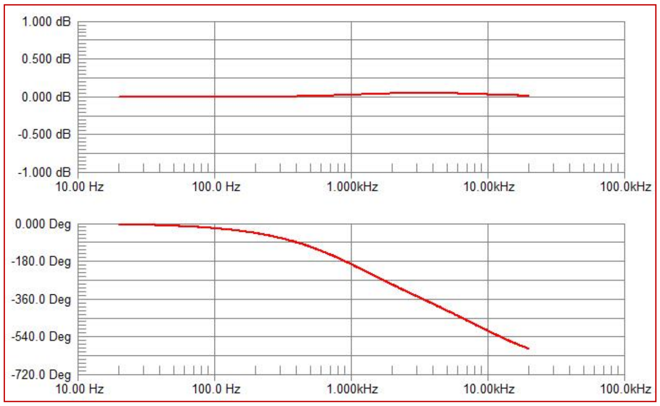


Figure 5

This filter bank performs phase 'scrambling,' a term we'll touch on later. For now, just note that this filter has little effect on bass frequencies, but subjects the midrange and top-end to a radical phase-lag slope.

The remaining all-pass section is the single stage involving op-amp IC9B. This is essentially the circuit of Figure 1, except that resistor R58, which determines the phase-shift turnover frequency, is a variable control. This allows setting the 90-degree phase shift frequency point over the near-6-octave range of 50Hz to 3kHz.

OPERATING MODES

The heart of the PhaseBuster comprises the four phase-shift networks described above. PhaseBuster left and right inputs are hard-wired to these networks, and network outputs are selected with front-panel rotary switches for the different operating modes.

Front-panel controls consist of four rotary switches and one potentiometer. Spare switch contacts light front-panel LEDs beside each knob to indicate which controls are active in any particular mode. The following mode discussions include a description of the signal path, an explanation of what the circuit is doing in each case, notes on what you might expect to hear and ideas for further experimentation.

BYPASS

BYPASS simply routes the PhaseBuster input directly to the output. Power does need to be applied, however, as there are a couple of buffer stages in the signal path. The PhaseBuster is designed to work at average, 'zero-VU' signal levels in the –10dBv (–8dBu) range common to consumer audio gear. The clipping level is about +15dBu.

SPATIALIZE

This is the stereo-enhancement mode for widening the stereo image, or soundstage. There are three SPATIALIZE sub-modes.

1. QUADRATURE routes the left and right inputs through the all-pass filter banks that impart a 90-degree phase offset between the two channels. This will have the effect of spreading the virtual center-channel (L=R) program components between the two loudspeakers. In pop music, vocals and bass tracks will no longer be centered midway between the two speakers. Music may exhibit rather nebulous directionality, but images will be stable. You may not notice much of a difference unless you are sitting in the listening sweet spot. Someone in the next room probably won't know that there has been a change at all.

   QUADRATURE is also the means of altering a stereo program so that it will make a better mono mix downstream. This concept is covered in more detail in the STEREO TO MONO discussion later on.

2. SCRAMBLED passes the left channel unchanged, but puts the right channel through the all-pass filter bank that has the steeper phase slope shown in Figure 5. Because this filter bank doesn't do much below 200Hz, this mode will have negligible effect on bass frequencies. But program components in higher registers will tend to move within the soundstage, depending on their pitch. For example, a piccolo going up and down the scale might move from left to right and back again because this filter bank can put those frequencies through a complete 360-degree phase rotation. Vocal sibilants may likewise seem to detach from the singer.

3. VARIABLE also passes the left channel directly, and routes the right channel through the adjustable all-pass stage. Now the VARIABLE control knob will be active to change the 90-degree phase lag frequency. Turned all the way down, the all-pass section is essentially out of the circuit and you should not hear a difference. As you turn the knob clockwise, however, the right channel will maintain a more normal (not inverted) phase below the indicated frequency and a more out-of-phase condition above it (see Figure 2).

Unless you set a particular music passage into a short repeat loop, you may not hear much difference between VARIABLE and SCRAMBLED as you switch back and forth. The VARIABLE position may lend better stability to the soundstage, as the phase of high frequencies never lags by more than 180 degrees.

Here is the one important consideration concerning both the SCRAMBLED and the VARIABLE modes. The resultant 'enhanced' stereo is valid only in a listening environment. Audio processed this way cannot be electrically mixed-down to monaural without creating gaping holes in the mono frequency response. These modes are for playback/listening only.

STEREO TO MONO

This second PhaseBuster mode is helpful in creating a more compatible monaural mixdown from the two stereo channels. The mono mix in this mode is routed to both the left and the right PhaseBuster outputs.

L+R combines the stereo input channels in traditional summing style as the name implies. QUADRATURE SUM, on the other hand, inserts the 90-degree phase offset between the stereo channels prior to summing them, and introduces a 2dB gain in the output path as well. Because program components common to both channels lean toward a more out-of-phase than an in-phase condition in quadrature summation, this bit of gain helps match the apparent loudness of most program material between the two switch positions.

I found that the QUADRATURE SUM position restored at least some of the musical content that went missing when the two stereo channels were combined in simple L+R fashion. Listening to the resulting mono, QUADRATURE SUM returned ambiance to the performance, sounding much the way mono LPs used to in simpler times.

MONO TO STEREO

This mode is the reverse of the STEREO TO MONO mode, and has four sub-modes. In all cases the PhaseBuster left and right inputs are first combined to make sure that the material is monaural to start with, even if the source is stereo. This L+R addition is performed after the input buffering stages, rather than simply paralleling the inputs. (Stiff, low impedance voltage-source outputs ought never to be fed into one another.) If you do feed just one input with your mono source, expect a 6dB loss through the system. Apply a single mono source to both inputs for overall unity gain.

1. L=R does not provide any stereo synthesis, it merely routes the monaural signal to both the left and the right PhaseBuster outputs. Mono played this way ought to sound as if it's coming from a point right between your speakers… no 'soundstage' at all.

2. In the QUADRATURE mode, the mono source drives the two all-pass filter banks with characteristics plotted in Figure 4. The two banks are routed to the left and right PhaseBuster outputs, yielding two sources that have a 90-degree phase shift between them at all frequencies.

   As stated in the L=R discussion above, a single channel of audio fed to two loudspeakers should seem to emanate from a point midway between. If you then reverse the wiring to one speaker, so that the two are driven out of phase, it should instead appear that sound is coming from each of the speakers separately, the classic hole-in-the-middle effect. But if you drive your two speakers with signals that are in quadrature (90-degrees apart at all frequencies) the effect is a veritable 'wall of sound' between the speakers. This is hardly pseudo-stereo, however, as there is no directionality or localization at all; in fact, just the opposite is true. Nonetheless, this effect may have some utility. For example, a simple PA system with speakers on each side at the front of a hall might give an illusion of better coverage… or maybe not. It is an interesting effect worth further experimentation.

3. SCRAMBLED routes the mono source directly to the PhaseBuster left-channel output, and through the steep all-pass filter bank (Figure 5) to the right-channel output. Unlike the SPATIALIZE/SCRAMBLED mode, which added an enhancement effect to a program that was already stereo, this mode synthesizes a pseudo-stereo effect from a mono source. But unlike the traditional stereo synthesizers discussed earlier, all audio frequencies are present in both output channels; there is no frequency band division.

   Instead, the apparent location of a sound in this artificial soundfield will still depend somewhat on its pitch, not because of an amplitude difference between the channels, but because of the phase relationship between them at any given frequency. The localization effects are being generated and perceived in the listening environment, they are not inherent in the signals that feed the two loudspeakers. Both speakers are being driven with full-range audio, not interleaved bands of frequencies.

   "Busy" might best describe the sonic effect of this mode, the same description for SPACIALIZE/SCRAMBLED operation. Vocals and instruments alike shift about within the soundstage, strings and vocal sibilants in particular. This soundfield can be impressive, but perhaps confusing and annoying to music purists.

4. The VARIABLE mode of stereo synthesis is perhaps the most interesting, especially as the circuitry involved is so exceedingly simple. Once again, the left PhaseBuster input is routed directly to the left output, but this time the single op-amp adjustable phase-shift stage is inserted in the right channel signal path.

   Assume the VARIABLE knob is set at 1kHz, giving the phase response graphed in Figure 2. Low audio frequencies will remain in phase between the two output channels, which will feed two stereo speakers in tandem for efficient bass reproduction. As program components reach the 90-degree phase shift frequency, the two channels will be in quadrature (and remain reasonably enough in quadrature over a modest range), so that mid-range sounds will appear to fill the soundstage between the two speakers. High frequency material will approach, and perhaps even reach, an antiphase relationship (180 degrees apart) and appear to come from the two speakers independently. The overall effect is a wide sound with a pleasant degree of soundstage dynamics and without appearing too confusing.

   The best pseudo-stereo performance setting seems be in the 1kHz to 3kHz range of the VARIABLE control. At too low a frequency setting the bass is weak and much of the audio is completely out of phase. Set too high, there is little stereo effect at all.

This single-stage stereo simulation circuit proved so effective that it was incorporated in an experimental wire recorder project, bringing pseudo-stereo to a medium that is capable of surprisingly good fidelity, but with no means of accommodating a second audio channel.

OTHER CIRCUITRY

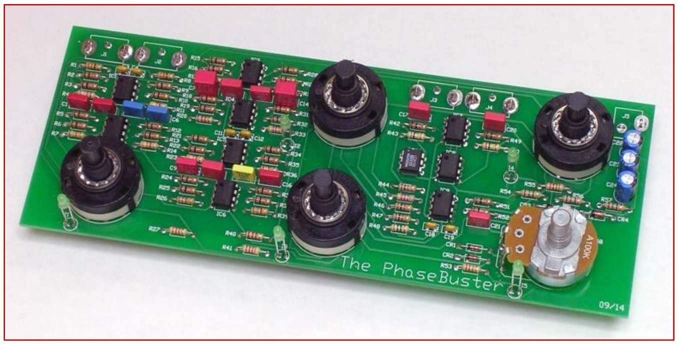
There is really not much to the PhaseBuster beyond the all-pass filter sections. The left and right channel inputs are buffered by IC7A and IC7B, which provide unity gain with high input impedance for minimal loading of whatever feeds the unit.

SSR1 is a solid state relay that sums the inputs to monaural for the MONO TO STEREO mode. Although not specifically intended for audio switching, these clever little chips are capable of handling high-level AC signals without introducing significant distortion. Their only drawback is that they are a bit slow to turn on and off (milliseconds), not a concern here.

The PhaseBuster is powered by a regulated 18-volt "wall-wart" switching-type power supply and further regulated by a pair of zener diodes to create a bipolar 7.5-volt supply for the op-amps. The 18-volt input must float for this to work, meaning that the PhaseBuster can't share ground-referenced power from some convenient existing source.

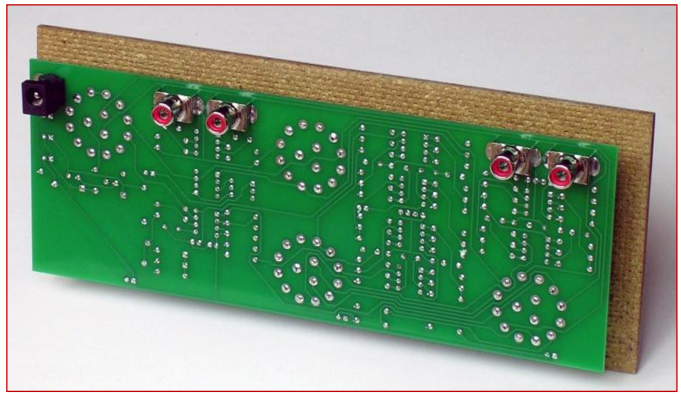
CONSTRUCTION

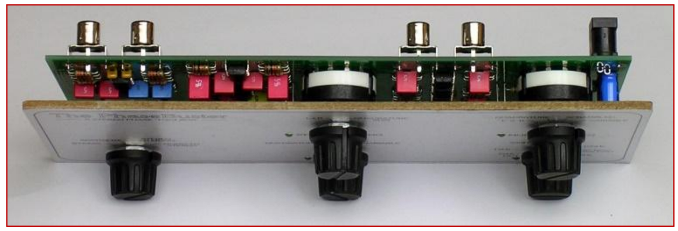
PhaseBuster circuitry fits easily on a single 3-by-8-inch circuit board as shown below.



The only hand wiring consists of three short, bare-wire jumpers between the VARIABLE pot and the board. Almost all the components, including the rotary switches, are on one side of the board; the input, output and DC power connectors are mounted on the rear.

A front panel was fabricated from 1/8-inch hardboard in this example, and when drilled for the switches and pot, the entire assembly forms a 'sandwich.'





Additional construction details and a full component parts list may be found in the original magazine article. A PDF of the article, circuit board Gerber files and a panel artwork file will be sent for the asking.