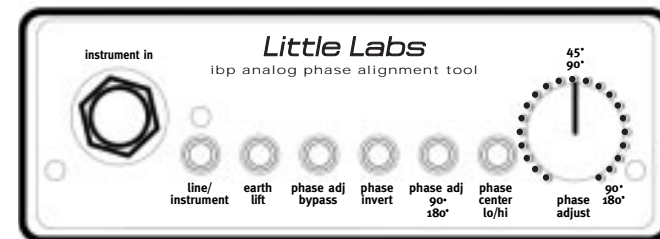


# *Little Labs*

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### IBP ANALOG PHASE ALIGNMENT TOOL



OPERATORS MANUAL

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Dear Audio Professional,

Congratulations on your purchase of the IBP analog phase alignment tool. This tool was developed by me with the help of audio engineers such as yourself, and saw many different forms before it became the mature product in front of you today.

Phase is a very important aspect in sound and has been a fascinating subject to myself ever since 1978 when I wrote a white paper on time and phase alignment of loudspeakers. I wanted to get it published, but instead used it as an in to get a job in a recording facility. Tom Lubin who was the then editor of the now defunct Recording Engineer Producer magazine wouldn't publish it. He said it was to editorial in content for a seventeen year old, oh well it got me a job at Salty Dog studios for \$150 a week (I have included it at the end of the manual, I guess I'm now a published author). Anyway, until now there have been almost no commercial attempts at controlling it past the common console phase (polarity) button. The company Countryman had a battery powered somewhat funky phase adjusting box (I had one at A&M studios) which I've been told the company denies ever making. The famous live touring sound company Clair Brothers I was told by a long time employee had a custom console in the 70s which had a form of phase sweep on each channel. Why Phase has been neglected for so long puzzles me, controlling phase is a powerful thing.

This IBP was designed to put the power of phase control in the palette of the creative audio engineers hand. Audiophile engineers will be quite happy with the sonic characteristics of the IBP and, with or without the Phase adjust circuitry in circuit, the DI built into the unit will hold its own against any of the high ticket DIs (Avalon, Evil twin, Summit, Manley, Demeter).

The IBP was designed using only the finest components including Elna Cerafine capacitors, Roederstein resistors, E-switch light touch silver contact switches, Neutrik XLR connectors, and polystyrene capacitors (in the all pass filters sections). In buying this fine product you are someone who appreciates spending a little extra to support craftsmanship in a world of mass-produced crap. So I thank you and the people who aided me in the development of the product. These people include my physics professor father, and my many supportive recording engineer friends.

Yours truly,

Jonathan Little

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## **READ THIS BEFORE USING THE IBP ANALOG PHASE ALIGNMENT TOOL**

If you never read manuals and prefer to dive in blind please at least read the following before starting:

Before powering up, check your power supply for proper voltage settings. If set incorrectly in North America, you will have severe performance reduction. In Europe, you will blow the power supply up and could possibly damage the IBP itself. Remember 115vac in North America, 230vac in Europe.

When connecting the power supply to the IBP pay attention, the connectors are keyed but an over enthusiastic user could jam the female into the male misaligned and wonder why this phase doo hickey doesn't work.

The re-amp output and line out share the same transformer, they are not intended to be used at the same time. The earth lift switch on the front panel lifts the ground of the line out xlr pin 1 and the sleeve of the re-amp output jack. If you are installing the ibp in a rack and will be extending the re-amp output jack to a remote location use a stereo jack and plug with 3 conductor wiring. This will allow the earth lift switch to function correctly whether using the re-amp or line output.

The IBP sonic performance improves after leaving on for a "burning in" period of twenty-four hours after being off for long periods. On initial power up wait at least three minutes before changing any switch settings. The unit has to stabilize its large reservoir capacitor and zero voltage sources. If you do not wait, loud pops will occur when changing settings. These pops could be damaging to other equipment.

The IBP is a line level balanced (+4dB) in and out, and instrument level unbalanced(-20dB to -10dB) in device. Performance will be reduced if feeding the line level +4 dB xlr balanced in with an unbalanced source. The IBP is not intended to work at mic levels.

## **?? PHASE ??**

Is phase the bastard child of time? Stay tuned... Well sort of, they are related but I'd like to point out what might seem obvious to some, two sounds can be in phase but not necessarily in time.

The IBP is not a delay line and with modern digital audio workstations you can move tracks around and put them in time. You can also use a digital delay line and align a track in time. When in one of the more common situations that finds a use with the IBP, the classic combination of direct bass and miked bass cabinet, you have a time delay between the direct sound and the sound traveling from the speaker to the mic diaphragm. You don't really expect the two sources, mic and direct to be in time, you just want them in phase. Now when you move them in time by physically moving the mic, or shifting the miked track back in time in your DAW (digital audio workstation), or delaying the DI track with a digital delay, are you in phase? Not entirely, you will not be compensating for the phase response of the speaker cabinet or the phase response of the microphone, and there is more, did you EQ the track (all analog and most digital EQ adds phase shift)? Should you care? No!! Mainly your concern is how it sounds. Due to the long wavelengths of lower frequencies, when two sources containing low frequency content are combined, you'll find if they are not in phase you will notice more than anything a loss of low end, resulting in a hollow thin sounding track. So you move the mic, use the IBP to sweep the phase, use a digital delay to delay the Di, or shift the track in your DAW until it sounds full and punchy.

The IBP has the advantage over the delay line, in that you can put the tracks in phase with one another with a simple, continuously variable twist of a knob, avoiding the D to A / A to D process, and in doing so, keeping the track out of the digital domain (if you care). Also with time delay, no compensation is possible for phase response error due to EQ or cabinet and mic phase response.

The IBP has the advantage over the DAW, in that you can put the tracks in phase with one another in real time, and avoid the D to A / A to D process and in doing so, keeping the track out of the digital domain (if you care). Also it is always a

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good idea to have what is printed to your DAW or tape right to begin with, rather than fix it in the mix (although you can fix it in the mix with the ibp if you want to). Also the DAW as with time delay, no compensation is possible for phase response error due to EQ or cabinet and mic phase response.

The advantage of using the IBP over moving the microphone is, after getting used to using the ibp, the most powerful argument for using the ibp or multiple ibps, on every tracking session. The ibp allows you to leave the mics where they individually sound the best. Moving a mic to compensate for the phase cancellation with another mic will compromise the individual mics sound. In other words, leave the mics where they sound the best, use the ibp or ibps when adding mics when phase problems occur.

For more information on the power of the ibp, check out page 7 of this manual. The first piece is an excerpt from a post in rec audio pro by the infamous Fletcher, it is informative as well as entertaining.

## **?? HOW DOES THE IBP WORK ??**

The IBP phase shift circuitry works by running through two passive symmetrical all pass filters. It works entirely in the analog domain. All pass filters are like high pass or low pass filters as in your EQ or speaker crossover network, but only the phase is affected not amplitude. The IBP has relatively flat amplitude response well past 96kHz. In most but not all cases at least two all pass filters are needed to recreate what is necessary to correct a phase problem or emulate phase shift in nature. One filter is used for the bottom of the frequency spectrum. The other is used in series for the top, creating a somewhat linear phase vs. frequency response. This is because you need to create a phase curve that is approximately twice the phase shift when you double the frequency. So for 45 degrees of phase shift at 40 Hz you need approximately 90 degrees of Phase shift at 80 Hz for it to sound natural. I say approximately because every situation is different with sound bouncing around, EQ added (with corresponding phase shift), phase response of speakers, microphones etc. Phase curves in nature have many variables that determine the amount of phase shift vs. frequency.

The switch Phase adjust 90 degrees /180 degrees selects between one or two series all pass filters in circuit. Phase center lo/hi selects between a higher bandwidth phase adjust utilizing one all pass filter for lo and the second for hi frequencies or a lower bandwidth phase adjust which uses both all pass filters for the lo frequencies.

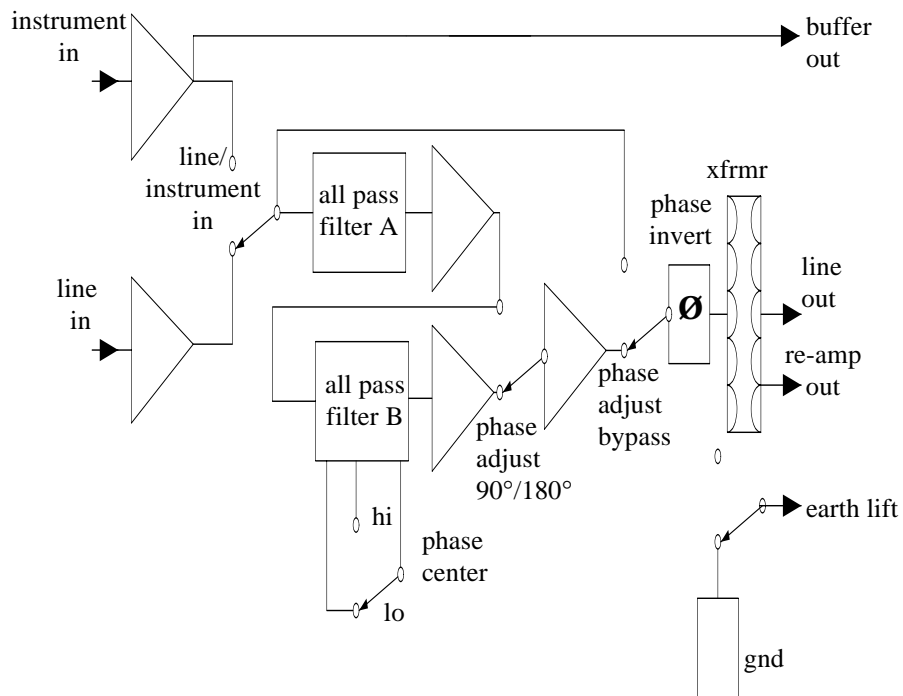
The IBPs electronics are for buffering and gain make up only. The IBP is entirely balanced and symmetrical from input to output. The output transformer in the ibp allows you to plug freely into balanced or unbalanced circuitry without any interface problems and also allows you to lift the ground.

The DI section puts a very high impedance input circuit specifically designed for instrument pickups ahead of the phase adjust circuitry. This hi impedance presents no load on your pickups allowing you to hear the full tonality of your instrument. The gain of the DI section can be varied from 12 to 26 dB of gain. The buffered out put is tapped after the instrument input circuit, before the phase adjust circuit, and can be used independently, even when the IBP is used in line level mode.

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The re-amp output will bring the +4dB db output to guitar level (about -14dB) and is tapped after the phase adjust circuitry and after the transformer. It is the same output as the line output but at a lower level and higher impedance. The re-amp output and line out share the same transformer, they are not intended to be used at the same time. The earth lift switch on the front panel lifts the ground of the line out xlr pin 1 and the sleeve of the re-amp output jack. If you are installing the ibp in a rack and will be extending the re-amp output jack to a remote location use a stereo jack and plug with 3 conductor wiring. This will allow the earth lift switch to function correctly whether using the re-amp or line output.

Check out the following functional block diagram.



## GAIN & NOISE AND DIRECT BOXES

Proper gain between stages in an audio system is critical for low noise and low distortion. A passive type Direct Box using a transformer has to lose voltage in order to have its input (primary) impedance high and its output (secondary) impedance low. Typical voltage loss for this type of passive device is 20dB. This is the only reason Direct Box outputs have always been run into the microphone inputs of a console.

With an active Direct Box voltage loss is not a problem. In fact the active stages typically sound better with gain (something more than unity). Most passive instruments need very little or no gain to run into a microphone preamplifier. Most active instruments have more than enough gain to bypass the microphone preamplifier stage altogether, and run directly into a line input. It is important to note this because if you find yourself heavily padding an instrument in order to run it through the high gain input section of your console (the microphone preamplifier), you will get a dramatic increase of NOISE. The proper way to run a Direct Box in this situation is to feed the Direct Box output into your console line inputs rather than the console microphone inputs. This way, no gain is unnecessarily added or padded, and a happy gain structure (low noise and low distortion) is the result.

The Little Labs IBP analog phase alignment tool is capable of putting out +32dB on the balanced output with phase adjust out and +19dB to +26dB with phase adjust in. The actual gain of the Direct Box section is set by the trim on the back of the DI from 12 to 26dB of gain.

Knowing the gain structure of each electronic device in the signal path of whatever you are recording or reinforcing is becoming increasingly critical. Digital recording is now the norm. Tape hiss and tape compression masking effects are no longer available to cover up engineer's oversights. Hiss, hums, and hard clipping are inexcusable to an increasingly critical audience.

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## USER TIPS AND RANTS

1. This first piece is an excerpt from a post on rec audio pro written by Fletcher of Mercenary Audio:

"From first hand experience, lemme tell you, I've been craving a box like this for a couple of decades now. I first encountered a device of this nature in the early 80's, and was unable to procure one for my own purposes. The fact that Jonathan took the time and trouble to put one on the market is more of a 'god send' than a commercial endeavor from my perspective, then again our store sells them (actually, the store offers them for sale, the only one that was actually purchased was returned as the purchaser said he couldn't tell the difference).

"The last stuff I mixed, I brought up the "Front Of the Kit" mic (which had a fair amount of bass bleed in it as I had recorded the band in one room), then the bass DI through the IBP. I pushed a couple of buttons, turned the knob a little and eventually a nice clear (golly the bass sound I was looking to achieve) came out of the speakers. I hit 'bypass' on the IBP, the bass sound 'washed out'; engaged the IBP, bass sound came back again. Then I added the bass mic signal to the 'FOK' and the DI signal... same deal, pushed some buttons, turned some knobs, eventually - bingo - the tone for which I searched. Hit the 'bypass' buttons on each a couple more times, dicked with the knobs a bit more, bypass, knobs, bypass, knobs, until I ended up with this bass track that coupled with the drum track in a manner that I had *never* before achieved! Added the rest of the drum kit and went on mixing. The unit wasn't available when I tracked the material but it's going to be a mainstay item for my tracking sessions from here on out.

Interesting little box. I've considered doing the same thing in a DAW, nudging one signal up or down by the proper amount; it doesn't work the same. Not even close. I've tried it in a DAW, I've tried it with the IBP, the results aren't even marginally close. With a DAW you can move the entire track up or back a frame (or sub-frame) at a time. It'll get you close, but not there. This unit seems to change the character of the phase characteristics of the signal much as a transformer will slow down treble frequencies while allowing bass frequencies to pass first, or piss poor circuit design will slow down bass frequencies

allowing the upper frequencies to pass first. By moving things in a DAW you're moving them all at the same time. It appears to me (and I sure as shit could be wrong about this, but it's my impression of what's occurring) that the IBP is altering the relationship between the high and low frequencies allowing for a different way to line up the sounds.

"Unfortunately, the typed word lacks the ability to adequately describe the difference, but if you hear it, you'll have heard it (or like the brother that bought one and returned it, you may not hear it. Or it may not make enough difference to you to warrant the expense, which is OK too). About the only way I can describe the difference is that it's kinda like the difference between a hand job and a blow job. They both achieve the same eventual goal, but one achieves that goal in a so much more pleasant manner.

"Eventually I have a feeling that I'm going to get into both shifting the time with a DAW (OK, I don't use a DAW, but I can do similar shit with my hard disk recorder), as well as fucking with the phase via the IBP box. The things I heard when using the IBP just blew my mind (and made the stuff I was mixing soooooo much easier to mix I was floored)! I did six mixes on that project. It wasn't until around mix four that I really started to get a handle on what the hell this box was all about. Unfortunately, there is no budget remaining to go back and remix the first two songs, but if there were, I would do it in a heartbeat!

"Since those mixes I had an opportunity to record some guitar tracks. I rarely if ever put more than one mic on a guitar cabinet, but seeing as we had some time to dick around, I figured, 'what the hell,' and put 3 mics on a Marshall 4x 12" cab. Mic 1 was a Royer R-122; mic 2 was a Shure SM-57; mic 3 was a Sennheiser MD-421. The Royer was about a half inch to an inch off the grill (just because the logistics of getting a side address mic pushed into the grill), the 57 and the 421 were both bending the cloth. I got a tone I quite liked from the 122, but for shits and giggles brought up the 57. It made things sound a bit comb filtered. I muted the 122, moved the 57 around until I liked the tone of it alone, then brought the 122 back in. Again a bit of that kinda hollow comb filtered shit. Enter the IBP. A couple button pushes, some knob twiddling,

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bypass, twiddle, bypass, twiddle. All of a sudden, great big motherfucking guitar tone that sat in the track like a good little schoolboy all prepared with his assignment! Repeated the same process with the 421, got a balance on the three mics, and proceeded to get a guitar track like I'd never quite gotten before. Changed the guitar, changed the amp top, changed the balance of the 3 mics, cut a second rhythm guitar track, and the shit was sounding seriously good (and the vocals sat in the track like a king on his throne)!"

2. This second piece are excerpts from three posts on gearslutz.com written by Eric Greedy aka Recorderman:

Entry 1 "I hate to sound redundant or like some religious zealot, but my way of mic'ing (using cans for critical placement by ear of almost all mic set-ups) has gotten me results that are super phase accurate. Shit just punches. Still, there are times (such as when using two mics on an a-gtr) where I must compromise one mic or the other so that the phase of the sum of the two doesn't hamper the sound. Now, I can place each mic on the a-gtr where I want that mic to be (relative to it's specific contribution, eq wise, ect.); then I can adjust the phase with the IBP to have the best of both worlds. IT ROCKS."

Entry 2 "Tracked drums, bass, gtr's and vocal the other day: used one IBP on the bottom snare mic, and another on bass DI (along with the amp also mic'd). The snare (57 top, 414 bottom w/IBP) was fatter than a complete 180 flip on the bottom would have provided; and the Bass was more forward than usual with the DI & Amp mixed evenly (I had the IBP on the DI side). Worked very well on both. Later today on a two mic'd a-gtr.

"This device (in my opinion) is best for use on one leg of a pair that are being used to capture a common source. I.E. the bottom mic of a top & bottom mic'd snare, tom, etc.

"My devils advocate tells me to shut up because it's my new secret weapon (along with Transient Designers, Dolby Stretches, and my twisted way of mic'ing). I wish I could afford a rack of twenty. At the least I'm getting another one tomorrow before my next session. Now if I could only have someone build me an AATP (add another ten's place) for my bank account."

Entry 3 "Found another good use for this box. Recording a singer and his acoustic guitar. Singer got an SM7 and the a-gtr got one of those Beyer 201 (I think that's the model. Kinda of a smoother 421 with a little more clarity).

Placed the mics where I liked for the sounds of their source (i.e. voc & a-gtr, respectively) to be optimized unto themselves. In other words, I didn't give a hoot as to the mic's working together per say, like you normally have to do, because of relative phase issues between the two. No, this time I stuck the a-gtr mic where it made the gtr sound right, and the same with the vocal.

"Now in the control room, I stuck the IBP on the vocal, after the pre/before the DAW (I was looking to maybe keep the a-gtr and later recut the vocal). While monitoring in one speaker mono, I tweaked the IBP till all sounded good - and it did! Great little box. Now I've just got to get those matched pair of 121's, a pair of Vipers and a Transient Designer. Then my gearsluttery for the year shall be sated."

3. This third piece is an excerpt from a post on gearslutz.com written by Michael Wagener in response to someone asking what the difference was between moving the tracks in time and shifting phase with the IBP:

"Well, I'm sure it works fine with lining up the tracks, but the cool thing is that you can sit in your control room and twiddle the knobs until it sounds right while you're listening to the changes, kind of like an EQ. That doesn't necessarily mean the tracks are exactly aligned. In the case of the speaker/SM91 combo for the kick I was able to dial in exactly the right amount of punchy low end without it getting woofy even though the tracks might have not been totally aligned. Also, sometimes I find it hard to 'see' the relationship between waveforms, as in the example above, the speaker did not have any high end content and the SM91 only had some small amount of low end, so the waveforms don't look very similar. Also, I'd rather hear what it sounds like before it gets printed."

Well that's it for user tips and rants but please e-mail me any interesting things you find and I will add it to future manuals.

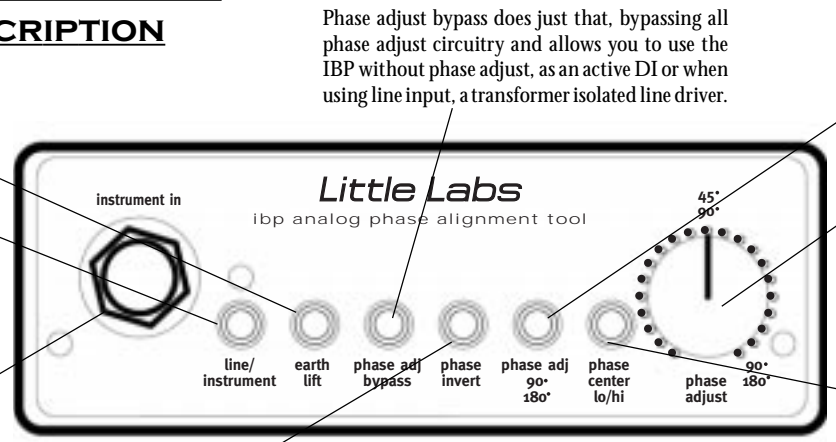
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## FRONT AND REAR IBP ANALOG PHASE ALIGNMENT TOOL DESCRIPTION

Earth lift lifts the ground (pin 1) of the output xlr and the sleeve of the re-amp out jack.

line/instrument selects between the line level xlr input on the rear (unity gain) and when pushed in the instrument in on the front (hi Z 10M with gain) for using as a direct box.

Instrument level (hi Z 10M with gain) input. This input is selected when the line/instrument switch on the front is in the in position. It will send to the rear buffered output with the line/instrument switch in either position.



Phase adjust bypass does just that, bypassing all phase adjust circuitry and allows you to use the IBP without phase adjust, as an active DI or when using line input, a transformer isolated line driver.

Phase adjust 90 degrees /180 degrees selects between two stages (button out) or one stage of phase adjust. This allows a more extreme phase adjustment with a full swing of as much as 180 degrees or a less extreme 90 degrees of phase adjustment.

Phase adjust knob determines the degree of phase adjust at a frequency set by the phase center lo/hi button position.

Phase center lo/hi selects between a higher bandwidth phase adjust (button out) and a primarily lower frequency phase adjust (button in). This and the phase adjust button create the phase curve. It is important to note phase is frequency dependent. For example depending on the two sources you are combining, and whether one source is direct and one is miked- between room reflections distance from the cabinet etc. determines what the curve would need to be. (e.g. you need to get the G note in phase as well the A note).

Phase invert is a simple polarity reversal as you would find on a mixing console out is 0 degrees, in is 180 degrees.

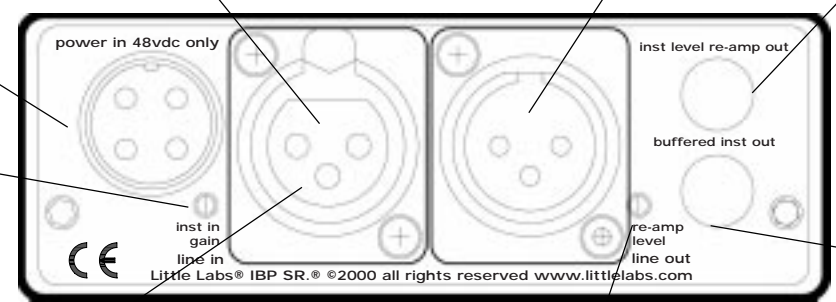
Transformer balanced line level output pin 2 hot. Can feed balanced or unbalanced inputs. The line output and the re-amp output share a transformer and are intended to be used separately. Max output before distortion +32dB phase adjust bypassed +19dB to +26dB phase adjust in.

Power in 4 pin male connector for 48 volt DC 250 ma power supply.

Balanced line level input pin 2 hot. Can be fed from unbalanced or balanced sources.

The re-amp output is for feeding guitar level inputs. This output is effected by the phase adjust. It is transformer isolated allowing you to lift the ground in case of hum. It can be used on the instrument in as a phase adjusted guitar thru or as a re-amp box bringing your line level signal to guitar level with or without phase adjust. The line output and the re-amp output share a transformer and are intended to be used separately. When installing in a rack and you intend to extend the re-amp output, use a trs jack and plug(stereo type 1/4"), this will allow the ground lift switch feature to function properly for both the line out and re-amp out.

Gain trim pot for the Di. This can be reached with a flat blade "greenie" tool, it is a ten turn potentiometer and can be set from 12 to 26 db of gain, turning clockwise is more gain.



Balanced line level input pin 2 hot. This input is selected when the line/instrument switch on the front is in the out position.

Level adjust for the reamp output. This can be reached with a flat blade "greenie" tool, it is a ten turn potentiometer, turning clockwise is louder.

On the rear of the unit you also have a buffered instrument out which follows the instrument input and is designed for running long guitar cables. This output is unaffected by the phase adjust and can be used completely independent of the balanced line in / line out section of the unit. In other words you can be using the phase adjust on line level signals and use this as a guitar cord line driver simultaneously.

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## A CLOSER LOOK AT TIME AND PHASE CORRECTED SPEAKER SYSTEMS

by  
Jonathan W. Little  
as written in July of 1978

(Not Required Reading)

### Abstract

This article looks in depth at what time and phase alignment is, how it can be done, and why some systems solve the problem only partially. Its intent is to take the mystery out of this new subject and let any prospective buyer of a time and phase aligned system know what he is getting. It should also aid the do-it-yourselfer in designing a time and phase aligned type speaker system.

Time and phase alignment is a relatively new refinement, which has been included in some speaker systems. It has generally become accepted as a positive design attribute in the audio industry. Fortunately, it has not met an ill fate, like some fads in audio, such as quad. Subjective comments, though, on the effect of time and phase aligned speakers are often vague and vary greatly. Advertising statements such as, "The difference is imperceptible to most people, but not to the recording engineer," I feel are confusing and seem to make the whole idea look like a gimmick to sell a speaker system. Who, in the audio industry would like to admit they hear like "most people"? A more objective look at time and phase alignment, I feel, will make people more confident about the refinement. Time and phase alignment does actually make a speaker system more accurate. But, not all speaker systems claiming names to the effect of time and phase alignment, do time and phase align the signal completely. This paper will take a closer look at the problems of time and phase alignment and how they can be solved.

For our readers who are unfamiliar with what phase and time alignment does, here is a brief explanation. A properly time and phase aligned speaker system insures that the original sound, will arrive at our ears in the same time and phase relationship to that which was electrically present at the input terminals of the crossover. For example, say we have someone playing an A on the A string of a violin. The acoustic spectrum would look something like that shown in Fig.1

We play this instrument through a non time and phase-corrected speaker system, like that shown in Fig. 2, which has been crossed over at 800 Hz. Some of the frequencies in the spectrum will arrive at our ears late and some would arrive too early, depending on the following variables. The most obvious is the physical offset between the low frequency driver's voice coil and the high frequency driver's voice coil. Another variable that must be taken into account is the phase shift introduced by the filters in the crossover. This can cause a phase offset between crossover outputs. The last and most often overlooked variable is the frequency dependent phase offsets present in each individual driver. With these problems not corrected, a thoroughly screwed up representation of the electrical signal present at the input of the crossover would result at the listening ear. If we defined the timbre of the violin as the

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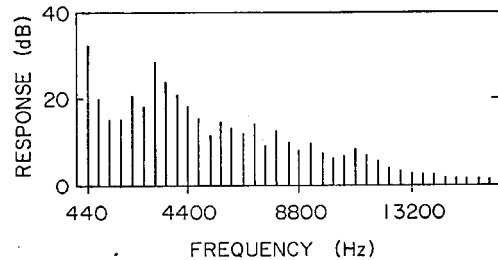
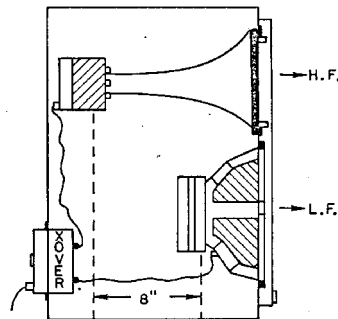
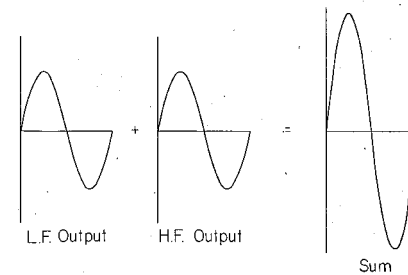


Fig. 1 - Acoustic Spectrum of the A string of a violin for the tone 440 Hz.

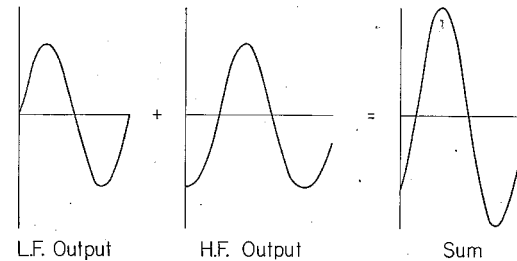


2)Fig. 2 - Side view of our proposed speaker system. Voice coils have an offset of 8 inches.

number, intensity, distribution, and phase relations of its frequency components then in our proposed speaker system, shown in Fig. 2, the phase relation of the components would arrive at our ears shifted relative to the input (of the crossover). An inaccurate representation of the violin's timbre would result. The same holds true for any instrument the sound spectrum of which has a harmonic structure. Time and Phase offset is usually most severe and easily heard at the crossover frequency. This is because two drivers are required to reproduce the same frequency, each at the same amplitude but 3dB down with respect to the full power pass band level. When combined in air, the frequency from each driver should sum by 3dB. (See Fig. 3). This gives a flat system amplitude response. If one driver reproduces the frequency too early or too late, a sum of



3)Fig. 3 - Signals at the crossover frequency as it is reproduced by the high frequency and low frequency driver in a properly time and phase aligned system.

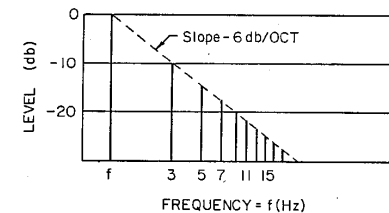


4)Fig. 4 - Signals at the crossover frequency as it is reproduced by the high frequency and low frequency driver in our example of a non-time and phase aligned speaker. Notice the sum of the high frequency driver's waveform is smaller than that of the time and phase aligned speaker system's in Fig. 3. (In fact 2.3 db down.)

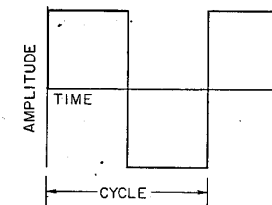
less than 3dB could result. (See Fig. 4). This would cause a dip in the speaker system's amplitude response. (This has often been mistaken by engineers who thought when one driver reproduced the frequency too late a skewed waveform-like "adding harmonics in some proportion", would result. The physics of the problem just does not work that way.) Take, for example, our speaker system referred to before in Fig. 2. The high frequency driver's voice coil lies eight inches behind the low frequency driver's voice coil. If the crossover is fed with a signal from an audio oscillator tuned to the crossover frequency, one finds that the low pass section retards the phase by 45° whereas the high pass section advances the phase by 45°. (This is easily seen on a dual beam scope). Therefore,

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the high frequency output is  $90^\circ$  ahead of the low frequency output. With the crossover phase offset and the voice coil's physical offset, we can calculate the overall phase offset between drivers at the crossover frequency. From this overall phase offset we can calculate the dip in amplitude at the crossover frequency. From this overall phase offset we can calculate the dip in amplitude at the crossover frequency that would result. In our system we have an overall phase offset between drivers of  $81^\circ$  or  $280\mu$  sec of time offset at the crossover frequency, thus causing an amplitude response dip of 2.3dB. (See Fig. 4.) The crossover frequency time offset and amplitude inaccuracies could in theory be corrected, in this case, by moving the high frequency driver so it's voice coil lies behind the low frequency driver's voice coil by 4.2 inches. The distance the high frequency driver needs to be moved is determined by the phase offset between crossover outputs at the crossover frequency. Our system's crossover exhibits a phase offset between its outputs of  $90^\circ$  and has a crossover frequency of 800 Hz. So the high frequency driver should be moved so its voice coil lies behind the low frequency drivers voice coil a distance corresponding to  $90^\circ$  or an 800 Hz wavelength, or 4.2 inches. This is a very simple way of correcting the most noticeable of the time and phase offsets. Apparently, most consumer speaker systems boasting of phase coherent response use this method of correction. But this does not solve the problem completely. The individual drivers have frequency dependent phase offsets in themselves, and even if the drivers were perfect and exhibited no such phase offset (such a driver is yet to be developed) we would not have the ultimate answer. Physically moving the driver will solve the problem at the crossover frequency, but the components of a signal with frequency content above and below that will arrive at different times. The effect would be less audible, perhaps, but out of time, nevertheless. The only way to really align the drivers (without regard to each individual driver's inherent frequency dependent phase offsets) is to correct the crossover's output phase offsets in the crossover itself. Due to the laws of physics, as frequency goes up wavelengths get shorter, and as frequency goes down wavelengths get longer. Therefore in our violin with a fundamental of 440 Hz, to obtain perfect alignment each successive harmonic would have to be reproduced by a separate high frequency driver each of which was physically located a smaller and smaller distance behind the low frequency driver. In this way the phase relationship for each of the harmonics would be kept correct. The only practical



5) Fig. 5. - Harmonic composition of the square wave illustrated below.



way to correct for this is to get rid of all phase offsets between crossover outputs, and to align the drivers' voice coil's in the same vertical plane.

Designing a crossover with no phase offset between the outputs is not a simple task. To my knowledge most commercially available covers exhibit some degree of phase offset between the outputs. You can always check by feeding the crossover with a tone and examining the outputs on a dual beam scope. An even better and more accurate test method would be to feed a square wave (refer to Fig. 5) to the input. Then sum the outputs of the crossover and feed them into a scope. A crossover with no phase offset would show a perfect square wave. The square wave method is more accurate, because the phase offset would not show using a pure sine wave, if the crossover outputs were offset by  $360^\circ$ . But, it would show using a pulse or square wave. If you are thoroughly disgusted with the results of your test, please don't junk your crossover. You would be much more disgusted if you put a microphone in front of your speaker system and looked

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at your, aah, - at least what was once - a square wave.

For a crossover to achieve cut-off or attenuation at a given frequency, a phase shift must occur between the input and output of the filter. This would be fine if each filter in the crossover shifted the phase in the same direction and to the same degree. Low pass and high pass filters, though, in simple crossovers unfortunately usually rotate the phase in opposite directions. Therefore, a phase offset between the outputs of the crossover would result. For you electrical engineers out there, why not design a device that rotates the phase in the exact opposite direction and degree as your filter, but without attenuation. Placing this device at the output of the filter would compensate for the phase shift in the filter and lick the phase-offset problem in your crossover. This should get a lot of you thinking!

Now we've taken care of the phase offset in the crossover. We would have a theoretically perfect phase and time aligned system - if we were using drivers with no frequency dependent phase offsets inherent in themselves. But, as Richard Heyser points out in his 1969 Journal of the Audio Engineering Society articles: "A loudspeaker, when considered as a transducer of electrical signals to acoustic pressure, has a transfer function which has a frequency dependent amplitude and phase response". Heyser goes on to examine different loudspeakers with the low frequency drivers measured in some type of enclosure. Examining these graphics points out that a relatively flat frequency response does not necessarily mean that you have a flat phase response. Unfortunately, this is not generally appreciated because the major loudspeaker system companies are still not including frequency vs. phase response graphs with their loudspeaker systems. Whereas amplitude vs. frequency response graphs has been given for a good twenty years. In order to solve the phase problems inherent in today's drivers some sort of electrical phase equalization is necessary. Some loudspeaker companies incorporate this phase equalization in the crossover to correct the phase offsets in the driver cabinet combination. (This electrical phase equalization is often referred to as 'proprietary instrumentation'). JWL Sound, a company located in Palo Alto, California (Note: Little Labs previous name and no, I never made the product) is developing a graphic type equalizer with time & phase vs. frequency, as well as the conventional amplitude vs. frequency

control. With this equalizer, any speaker system could be time and phase aligned except for worst case type folded horn systems which would require some kind of additional time delay. Acousta-voicing is a professionally accepted practice in studio monitor systems. It corrects for any minor amplitude vs. frequency variations of the speaker system and the absorptive qualities of the room. So why not phase equalization as well? Of course, the measurement technique for acousta voicing is relatively simple. Measuring the phase response of a speaker in a room is a bit more difficult. It would require time delay spectrometry that requires a license and some elaborate test gear. The equalizer's phase and amplitude section's are not independent but interact with one another, so some time would have to be spent fiddling between the two to get the red response. Once set up, though, as in acousta-voicing, you would not have to touch the equalizer again.

So there you have it, the ultimate finesse of a time and phase aligned speaker system. Now, for the real test, is your hearing like "the recording engineer", or "most people"? Enough of this gibberish! You know now whether it is audible to you or not, that those harmonics are arriving at your ears at the right time. That should make any audio addict, such as myself, happy. Now, does that cabinet need extra bracing, is that horn ringing, am I using a low enough resistance speaker cable, is that amplifier clipping those transients? Ahh.... well, one problem solved in the quest for audio perfection.

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## DISASSEMBLY

There are people who could care less about the insides of an electronic device. Then there are people such as me, who open their new toys to peer at the guts, sometimes before even plugging it in. If you are going to pull the IBP apart please follow these instructions, this way you will be able to put it back together.

1. With a 1/16" Allen hex, remove the two rear panel screws.
2. The rear panel is now held in place by the male and female XLR body half screw. Use a very small screwdriver and twist the screw inside the center of the XLR counter-clockwise. It should click and the rear panel and the plastic frame should come right off.
3. Unsolder the wires attached to the power connector (if necessary).
4. Slide the black textured steel cover carefully off of the base extrusion.
5. This will expose the insides to your leering eyes. Nicely made, eh.
6. If you need access to under the circuit board, use a 1/16" Allen hex to remove the front-screened panel screws. The circuit board should now easily slide out from the base extrusion.
7. Unless you have to replace the pad or switch assembly, I do not recommend removing the front utility (not screened) panel. If you must, gently but firmly pull off the knob.
8. Use a 1/2" nut driver to remove the front phone jack nut.
9. Slide out utility panel and plastic frame and then remove the spacer washer from phone jack and pad assembly. Do not lose these.
10. There you have it!

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## **ASSEMBLY**

1. On the front phone plug bushing install spacer washer consisting of a 3/8" fiber shoulder washer on the phone plug bushing and slide black utility panel with black plastic frame attached, onto the phone plug and through switch holes.
2. Attach nut to phone jack firmly with a 1/2" nut driver.
3. Place screened decorative front panel on top of utility panel, slide circuit board into base extrusion and screw in the two Allen button head screws with 1/16" Allen hex driver.
4. The front of the direct box is now one piece. If necessary slightly bend led leads so they sit flush with the front panel.
5. Gently but firmly push knob back on.
6. Slide the textured black cover into the base extrusion.
7. With plastic frame attached slide rear cover onto xlrs and phone plug bushing.
8. While slightly squeezing the front and rear panels of the direct box together, making sure everything is squarely in place, use a small screwdriver and twist the screw inside the center of each XLR clockwise. It should click and the rear panel and the plastic frame should be held in place.
9. Screw in the two Allen head screws with 1/16" Allen hex driver.
10. Thats it!

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