Microphone Phase Tutorial

Part One — Dealing with Phase/Polarity Issues

By Jay Graydon

I offer this tutorial gratis for the San Luis Obispo County and LA County schools music departments. Please fell free to share with other schools and anyone you wish.

Further note this tutorial does not cover many basics needed for the novice recording engineer. If a novice, please ask your teacher questions as needed.

After around 40 years of recording, you might think I would know the physical/audio recording studio path in full. I thought so until around three years ago after discovering a loose end. Ironically, after the discovery and then talking with some of the top recording engineers in LA explaining the details, none had thought about the logic and were pleased I shared the information.

Note: best to share your discoveries or concepts passed along to you by other engineers with all of your recording engineer friends. The odds are very good you shall receive knowledge in return.

I would love to state the discoveries at this point but before doing so, the following is important as to clear up some basics as well as to keep things in chronological order.

Ok, if you are impatient, skip down to “Accidental Discovery of Inverted Phase Sonics” and “Important Phase Discovery” but make sure to read this tutorial in full after doing so or you may miss important details.

Phase and Polarity

(Note the following is explained in full in the "Studio Phase Tutorial (Part#1)" as to perform complete studio phase tests.)

You'll hear people talk using the terms phase and polarity to describe the same phenomenon. Technically speaking, phase usually is frequency-dependent (such as *comb filtering* issues), whereas polarity affects all frequencies equally (such as electronic path wiring).

<u>In the context of this tutorial, polarity is technically more correct than phase, but I will use phase because that is the term most recording engineers and musicians use. Ok, some of the time I will note polarity as well in parenthesis as a reminder that is the correct term.</u>

We assume that every mic, cable, mixer electronic path, outboard gear, recorder electronic path, speaker cables, speakers and headphones are wired in such a way that keeps phase integrity intact (absolute phase). In the most basic wiring path, there is a "hot wire" (positive) and a ground wire (negative). Reversing this anywhere in the signal path will lead to inverse phase (inverse polarity).

With balanced lines, there are three leads ("hot," cold, and ground). According to an IEC standard, pin 1 is ground, pin 2 hot, and pin 3 cold (also called neutral). However, in an amazing fit of ignorance, some manufacturers wire pin 3 hot and pin 2 cold. This inherently reverses the phase when connected to any equipment that is properly wired.

When combining any two sound sources — speakers, mixers modules, whatever — it's important that they be in phase. As a classic example, assume you have a sharp drum sound (like a bass drum) going through a set of stereo speakers. When the bass drum is hit (kicked with the "beater"), the speakers should push air outward, which we perceive as sound. However, if the wires going to the speakers are reversed, then the speaker will pull air inward at initial attack.

Generally, it was considered that when listening to a speaker by itself, reversing the phase (polarity) didn't make any audible difference. However, preserving absolute phase *does* make a difference, and that when you hear something like a kick (bass) drum or the bass instrument, it doesn't sound right unless it's pushing air toward you at the initial attack. There are many valid reasons to retain a proper phase relationship anyway, so why not?

There is no disagreement, though, that combining sound sources that are *out of phase* with each other can be a big problem. For example, if one of the speakers in a stereo system is *out of phase* with respect to the other, then as one is pushing air, the other will be pulling it. Like any other *out of phase* waves, they cancel.

This needn't just happen at the output stage. Mics and direct boxes can be wired out of phase, two mixer channels could be out of phase with respect to the other and so on. Anything wired incorrectly throughout the electronic chain could be the culprit.

The bottom line — most important that the total recording path from the mics through the monitor speakers are wired "in phase" (absolute phase). Best to refer to the "Studio Phase Tutorial" for complete test details.

Discovering The Sound Of “Out Of Phase” And “Inverted Phase” Signals

Most important to hear how *out of phase* signals degrade the sound. We will start with a total phase reversal (one of two channels reversed in phase) and then reverse the phase on both channels.

One of Two Speakers Intentionally Wired Reverse Phase

Set your mixer monitor mode as to monitor a CD, or any recorded information you prefer. In any case, make sure the sound source has some common information regarding both (stereo) channels in “center position” (such as bass drum, bass, snare drum, etc.) Easy to discover this fact when playing the sound source as you will be sitting equally between both speakers.

We mention using a previously recorded CD for the following example. If using another source, simply replace.

1. Play an 8 bar section of a well-recorded CD and note this musical section. This CD should be a recording with a wide frequency range.
2. Now switch off the speakers stereo power amp.
3. Reverse the wires going to one of the speakers (swap the + and - wires either at the speaker or amplifier). Make sure to note which speaker (or power amp channel) has been reversed.
4. Switch the speaker’s stereo power amp back on.
5. Play the noted CD 8 bar section.

You will notice that the sound is very strange. The bass frequencies will virtually disappear (very small sounding) and the overall sound is hard to listen to meaning it bothers your ears. If this is the sound you hear, keep in mind the problem is caused by one speaker cabinet pushing air outward (correct phase) while the other is pulling air inward (inverted phase) thus causing common signal waveform cancellation. (Common signal means a sound source that is mixed equally to both stereo channels such as lead vocal, bass, bass (kick) drum, snare drum, etc.). If that was the sound you heard before reversing the speaker wire in step #3, one of the speakers was previously wired out of phase. In either case, here is what to do.

- If your speaker system was originally “in phase” before switching the speaker wire (or power amp speaker wire) in step #3, switch off the speakers stereo power amp and swap back the + and - speaker connection to their original connection.
- If your system was originally “out of phase” before switching the speaker wire (or power amp wire), in step #3, leave the swapped wiring for now.

In any case, you have the speakers wired as to allow for “common phase” but the phase may not be absolute! This will be explained below in, “Both Speakers Intentionally Wired Reverse Phase”.

(Note: if your speaker system uses powered speakers (the speaker power amps are “built in”), if you are testing a studio monitor system, you need a way to reverse the phase on one of the two channels. If using a mixer that includes phase switches, plug the sound source (CD, whatever) into two mixer channels and reverse the phase on one channel. Simply flip the phase switch on one mixer input channel when listening.

Another option would be to un-patch one of the two cables between the sound source (stereo system, or mixer) and speakers — replace with a cable that reverses the phase. If the cable is balanced (three wires), on one connector end, reverse pin 2 and 3 wires. If an unbalanced line, on one connector end, reverse the hot and ground pin. (If not clear on this, see section, “Wiring Cables Reverse Phase.”)

In either case, perform the above test. Mark this cable as *reversed phase* as to not accidentally use unless needing to reverse phase in a path.

Both Speakers Intentionally Wired Reverse Phase

1. Play an 8 bar section of a well-recorded CD and note this section played. This CD should be a recording with a wide frequency range.
2. Now switch off the speakers stereo power amp.
3. Reverse the wires going to both of the speakers (swap the + and - wires either at the speaker or amplifier).
4. Switch the speaker's stereo power amp back on.
5. Play the noted CD 8 bar section.

When both stereo channels are reversed from the stereo system speaker terminal outputs to the speaker terminal inputs, the sonic problem is not nearly as bad as one of the speakers reversed in phase. Even so, the audio will not sound as good in comparison to absolute phase (correct wiring throughout the audio path). In this case, both speaker cabinets are not pushing air forward at initial attack. Instead, the speaker cabinets are pulling air inward causing a slight loss of low-end frequency level as well as low frequency "punch" (impact).

Again, this type of inverted phase may be difficult to hear at first. Repeat the above 5 steps until you notice a difference. When you notice a difference before or after swapping both speaker plus (+) and minus (-) wires, perform the following.

- If your speaker system was originally "in phase" before switching the speaker wires (or power amp speaker wires) in step #3, switch off the speakers stereo power amp and swap back the + and - speaker connection to their original connection.
- If your system was originally "inverted phase" before switching the speaker wires (or power amp wires), in step #3, leave the swapped wiring for now. See Studio Phase Tutorial (Part #1) as to continue the most important tests.

Regarding home stereo systems, stereo systems in stores, novice home recording studios, etc, if the amp to speaker connectors use individual screw on wire terminals (typical), if not installed by someone that knows how wire is coded, a wild guess is at least 10% of all such systems in the world have a reversed phase problem (out of phase) let alone inverted phase. The reason is simple — many consumers do not read manuals in full and may not understand how speaker wire is coded. With that in mind, when connecting speaker wire, *out of phase* will be in play on one of two speakers or both connections will be *inverted phase*. If more than two speakers are connected (separate speaker selection channel, multiple speakers using the same terminals, or a home theater set up), the odds of incorrect phase connections become higher.

Most likely, the audio novice/consumer who connected the speaker wires will not know a problem exists!

Note over the years I have heard *out of phase* problems in many stores, etc. For years, I told the person in charge there was a wiring problem and explained what to do. Just trying to help out.

Accidental Discovery of Inverted Phase Sonics

The outcome of the following accidental blind test will prove very important as things unfold.

In the days of a full analog recording path, in the early 1980’s I requested my studio tech bypass the input and output transformers in my AMPEX ATR 2-track recorder. Note that removing the transformers makes for a clearer and *punchy* sound.

All should have been fine since simply a wiring jump eliminating the transformers. After my tech bypassed the transformers, I began mixing a song for a Manhattan Transfer project I was producing/engineering.

When mixing, I always monitor through the 2-track recorder in *input mode* and back through the mixer monitor section. A few reasons to do so. The wire/connectors from the mixer to the 2-track recorder electronics and wire/connectors back to the mixer surely colors the sound in comparison to monitoring the mixer 2-track monitor bus. Further, monitoring through the 2-track monitor chain path is important as to hear if any electronic problems show up inside that path.

When I started mixing the song, I occasionally switch the monitor through the mixer monitor bus for a brief moment as to hear the sonic difference (to check for possible electronic problems in either path — always best to check).

I noticed the mixer monitor bus sounded bigger in the low frequencies and had more punch compared to monitoring through the AMPEX ATR 2-track recorder. Common to hear a slight difference meaning a very slightly darker sound through the 2-track recorder monitor path but never a lack of low frequency information and punch — this difference was serious causing an audio red flag!

I called my tech and asked he come to the studio as soon as he had a time slot open. When he arrived, I played the multi track recorder mix switching from the 2-track input monitor path to the mixer monitor bus. My tech could not hear the difference. Well, I could but he said I was crazy.

I was sure there was an electronic problem and since the only electronic path that had been altered was the transformer bypass within the 2-track recorder. I asked he investigate the electronic path.

Using a test oscillator and a dual trace oscilloscope, it was proven the phase (polarity) was reversed in the AMPEX ATR 2-track recorder!!! Ouch. The AMPEX ATR recorder schematic did not show a phase swap (inverted phase on both channels) if bypassing the transformers!!!

After my tech swapped the wires to correct phase (swapping pin 2 to pin 3 in the balanced line path on both channels), problem solved!!!

I played the multi track recorder and compared the signal through the 2-track recorder and the mixer monitor bus and all was well. Sweet!!! I mixed the song, checked playback and clearly noted the problem was solved.

Bottom line — My tech of the era is brilliant but does not have the capability to hear sonic nuances. I then realized there are degrees of sonic hearing. A while back in time the term “Golden Ears” popped up. I assume I have that gift but as all sensory gifts, you do not find that out until it becomes apparent some people can’t do the same. (Note there are degrees of such gifts and noted in the Perfect Pitch/Relative Pitch tutorial).

So what has been established in the above accidental blind test is inverted phase (polarity) causes a slight loss of low frequency information and *punch*. KEEP THIS IN MIND REGARDING THE FOLLOWING!!!

The Acoustic and Electronic Recording Path

Most of you know the following path but I need to note as to tie into the next section.

1. A mic diaphragm translates acoustic sonic energy into electronic energy.
2. The mic diaphragm vibrates in regard to the frequency cycle of the sound source and turns the mechanical vibration into electronic energy.
3. The output of the mic is routed into a mixer module input.
4. The mic signal is amplified within the mic pre amp and again in the line stage.
5. The signal is assigned to a record track via a bus or mixer module direct output.
6. The recorder track returns through a mixer module used for the monitor path.
7. Mixer monitor output(s) route to the monitor speaker power amp(s).
8. Power amp output(s) route to the monitor speaker(s).
9. The monitor speaker(s) reproduce the waveform initiated into the mic diaphragm by vibrating thus pushing air outward and pulling inward on each cycle.

So if the electronic path is correct wiring meaning the path is correct phase (polarity) through the whole chain, when any sound source signal is induced into the mic diaphragm, the result should be the monitor speaker(s) push air outwards at initial attack on the front end of every waveform cycle.

Important Phase Discovery

Finally, time for the discovery that is the crux of part one of the tutorial.

The last paragraph in the above section states, *“So if the electronic path is correct wiring meaning the path is correct phase (polarity) through the whole chain, when any sound source signal is induced into the mic diaphragm, the result should be the monitor speaker(s) push air outwards at initial attack on the front end of every waveform cycle.”*

<u>This may be correct or not!!! Until a few years ago I had not pondered the first part of the chain in full, which is the movement of air into the mic diaphragm!!!</u>

<u>The above path would be correct phase (polarity) as along as the air movement is towards the mic diaphragm at initial attack. If the air movement is moving away from the mic diaphragm at initial attack, the phase (polarity) is inverted!!!</u>

Microphone Diaphragm/Air Movement

The mic diaphragm sees the sound source air pressure arrive in one of two basic directions — either towards or away.

1. If the sound source signal air moves towards the mic diaphragm, the mic diaphragm translates the initial attack into a positive up-sloping electronic waveform. If the studio electronic path is phase coherent, when the mic’s electronic energy routs through the mixer, through the recorder, and finally through the studio monitor speakers, the speakers push air forward at the initial waveform attack, which is what we want.

2. If the sound source signal air moves away from the mic diaphragm, the mic diaphragm translates the initial attack into a negative down-sloping electronic waveform. If the studio electronic path is phase coherent, when the mikes electronic energy routs through the mixer, through the recorder, and finally through the studio monitor speakers, the speakers push air *backward* at the initial waveform attack, which is *not* what we want.

<u>Correct phase (polarity) example</u>: With the bass drum miked in the standard fashion (mic inside the drum pointing towards the drum head), when struck with the beater, the bass drum head pushes air towards the mic diaphragm thus leading to correct phase (polarity).

In this case, the mic diaphragm translates the initial attack into a positive up-sloping electronic waveform, which once again is what we want.

<u>Inverse phase (polarity) example</u>: With the snare drum miked in the standard fashion (from above the top drum head), the initial attack from the drum stick to the snare head

pushes the snare head down causing air to move away from the mic diaphragm. The result is inverse phase (polarity).

In this case, the waveform would “slope down” at initial attack causing the studio monitor speakers to “suck in (pull in)” instead of pushing air forward. Again, not what we want since this slightly sucks back low frequencies and causes less “punch” (see sidebar) at initial attack. If the sound source does not have much low frequency information (such as drum cymbals), not a big deal but still best to have the waveform rise at initial attack.

Sidebar:
“Punch” is very fast movement of air generated by the speakers, which is heard by the listener and possibly physically felt. Such an example of “punch” felt is in a rock concert setting utilizing a mega watt PA system. If in close proximity to the PA speakers, when the beater strikes the bass drum, you have felt the major amount of sonic air movement in your chest.

So we want the studio monitor speakers to move air forward regarding all sound sources at initial attack as this allows for more “punch” and better low-end frequency response.

Degrees of Phase

I have been talking about two basic directions of sonic air movement in regard to the mic diaphragm — forward or away. What if the mic diaphragm is positioned off center (at an angle) in regard to a sound source? The phase in not absolute (it rarely is when miking a sound source). This opens up a can of worms in physics and electronic land.

The brief summary is on the scale table of 360 degrees of phase, 180 degrees is out of phase (reverse polarity).

Example: I typically mic a guitar amp speaker around and inch back from the grill cloth and an inch 22 degrees left of center in regard to the speaker cone thus leading to 22 degrees off absolute phase. The sonic information into the mic diaphragm is still a positive waveform since not near 180 degrees out of phase. A slight loss of *punch* is in play but the miking angle is needed to get a good sound from the guitar amp speaker — this is a needed trade off in sonic land and very typical.

The bottom line is we want the mic to see a sound source waveform on the positive side of 180 degrees. Yes, the closer to zero the better.

Good news!!!

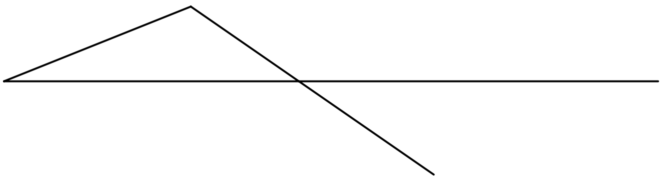
If a waveform is negative at initial attack, it can be flipped upside down (positive) after recording by simply reversing the phase (polarity) electronically in the audio path. You could reverse the phase (polarity) to the recorder beforehand if using a mixer with input module phase switches. More on this in the section, “Checking the waveform for attack phase”.

Basic Waveform Shapes

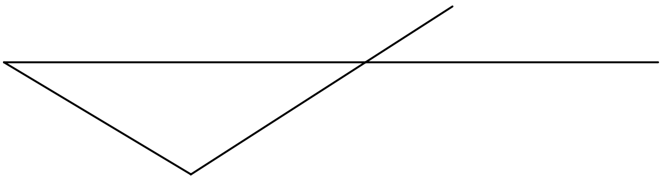
Some text will seem redundant but I really need to make all of this clear as possible. Please excuse my terrible drawing technique in the following.

Lets look at a simple waveform shapes. This will look like a saw tooth waveform since easy to draw in *word* format. No matter what the waveform shape, it is either positive or negative at initial attack.

Positive wave form shape for one cycle.



Negative wave form shape for one cycle.



As you have noticed in the waveform drawings, after the first half pulse of the waveform, the 2nd half pulse shifts to the other side of zero. I.e., if the waveform slopes down at initial attack (negative as noted in the above example), it rises on the 2nd half of the pulse immediately thereafter (positive).

Using an example of a low frequency source at 100 cycles per second (bass, (kick) drum, or any low frequency sound source), if the first half of the waveform pulse is positive the 2nd half of the waveform pulse would be "negative". This would repeat 100 times per second.

Some will say that since the waveform change happens so quickly, what's the big deal? "Even if the first pulse is negative, the 2nd pulse will be positive very quickly.

Yea, that seems to be logical in theory but there is surely a sonic difference as noted in the story "Accidental Discovery of Inverted Phase Sonics" and in other examples.

Checking the waveform for attack phase

Hard disk recorder with individual track waveform readout

In this era of hard disk recording, since most are using a system with many tracks, always best to record source to it's own track(s). In this case, if inverted phase on any track, very easy to fix.

Let's use a snare drum for the example. As we know, the snare head is typically miked from above and as explained, the mic would see a negative waveform slope at initial attack.

On the hard disk recorder track waveform readout, go to the snare track and look at the very first snare hit. Magnify the waveform (stretch out if needed in a non-destructible mode) as to see how the initial waveform attack looks. Also look at the next few hits as well just to be sure you are seeing the snare waveform and not leakage from other drums (see *important* below for more details). As mentioned, the odds are good it will have a negative slope at attack.

The Fix: If the waveform is sloping down at initial attack, go to the edit mode for your system that allows you to invert the phase of the complete snare track. Do so and check the first few hits again, and the very last hit to make sure the edit worked.

Important: Real drums — If you had recorded a real drummer previously, there will be some audio leakage into the snare mic from other drums and cymbals, and possible other instruments. Simply look for the first waveform that shows a huge spike compared to waveforms that are much smaller in height. Isolate the first waveform that has a big spike and start playback on that hit just to make sure it is the snare.

Important: Sampled drums — If looking at a recorded drum machine sample, or sampler, etc., no leakage so the waveform should be the snare on each hit.

Since you are viewing waveforms, do the same with all tracks. Reverse phase as needed.

(Note the tracks in which you have reversed phase and always listen as to make sure it sounds good. When we get into the full drum set (Microphone Phase Tutorial Part Two — Dealing with Drum Phase/Polarity Issues), there are times when flipping phase may need to be set with a reversed waveform. Yea, breaking the rules but this has to do with comb filtering issues.)

Digital tape recorder or analog tape recorder

Since such recorders do not allow for track waveform readouts, if you do not have Pro Tools or the like as to bounce over to see waveform shape, we need to use a sampler that allows waveform viewing.

Again we use a snare drum for the example. As we know, the snare head is typically miked from above and as explained, the mic would see a negative waveform slope at initial attack.

1. Patch the snare output from the recorder into the sampler input. If a digital recorder with digital outputs, patch into the samplers digital input. If analog routing is needed, simply patch the snare into the left input. There are many ways to patch such as assigning the snare track to a bus on the mixer and patch the bus output into the sampler. Others patch options as well regarding your setup.
2. Play your recorder and record the first few snare hits into the sampler using “force mode” — This will allow you to start recording into the sampler before the first snare hit occurs. (If using “arm mode”, the sampler may choke off a bit of the initial attack).
3. Go to the sample edit page as to view the first hit waveform. (Note: do not truncate the sample as this may also choke off some of the initial attack).
4. In “sample edit” mode (the page you are on), scroll the waveform to the point of initial attack — this will be obvious since the waveform will show a serious slope either up or down. Now stretch out and magnify the waveform (using the edit tools within the edit mode) as to see the exact shape of the very beginning of the waveform. Also look at the next few hits as well just to be sure. As mentioned, the odds are good it will have a negative slope at attack.
5. Note the waveform shape.

Redundant but I really need to make sure this is clear. When looking at the attack of the snare hit, note the direction of the first slope. If your studio audio path is wired to correct electronic polarity, the odds are good that the snare waveform would slope down.

Important: regarding step #4 above, if not using recorded sampled drums but a real drummer you had recorded, there will be some audio leakage into the snare mic from other drums and cymbals. Simply look for waveforms that show a huge spike compared to waveforms that are much smaller. Isolate the first one that has a big spike and start playback on that hit (or loosely truncate meaning leave a little room before the waveform hit) just to make sure it is the snare.

Since you have the sampler set up, might as well check all other tracks using steps 1 through 5 noting your findings on your recorder track sheet. In the track box, write down if the waveform slope was positive or negative at attack. When finished checking, the sampler is no longer needed for the following so pull the patches to the sampler and put back patches to the recorder (if any were pulled).

The fix: In this case, a few options.

Now that you have the initial waveform slope shape noted, any waveform the starts “negative” needs to be electronically reversed.

Mixer module phase switches: If your mixer has phase switches on each module, simple — reverse the phase on initial attack down sloping waveforms. If your mixer modules do not include phase switches (one per input module), if your mixer has a patch bay allowing recorder output to line input patch points, reverse the phase using a “phase reversed” patch cord (see “Wiring Cables Reverse Phase”. I will include this section last as for easy reference).

If no patch bay or patch points, if your recorder uses analog cables, you will need a phase reversed cable from the recorder outputs to mixer line inputs temporally replacing the “in phase” cables. (Again, see “Wiring Cables Reverse Phase”. I will include this section last as for easy reference).

Important: Note the tracks in which you have reversed phase and always listen as to make sure it sounds correct. When we get into the full drum set, there are times when flipping phase may need to be set with a reversed waveform. Yea, breaking the rules but this has to do with comb filtering issues.

Sound Sources Miked From Above That Cause Inverted Phase

Drums: All drums that are typically miked from above. I.e., snare drum, toms, high hat and overhead mics (cymbals and overall drums).

Remember that the bass drum is not an issue here as the mic sees the air move towards the mic diaphragm at initial attack. Note the same applies to a snare drum miked under the bottom head with the mic pointing up. So much more on drum phase in Microphone Phase Tutorial Part Two — Dealing with Drum Phase/Polarity Issues.

Percussion drum instruments: Conga Drums, Bongos, Timbales, etc. The rule of thumb is any drumhead miked from above pushes air away at attack will invert the phase.

The same goes for hand percussion instruments such a shakers and Cabasa. The start motion could be anywhere is phase land. No matter since this is extreme high frequency information. Yea, the waveform phase still shows an upward or down motion at initial attack. If sloping down, might as well set to a rising waveform since we can!

Mallets: Vibes, Marimba, etc. I have not tested for phase but logic leads to reversing the phase.

Upright acoustic piano: Unlike the grand piano design, the hammers hit the strings in front causing an inverted waveform.

Instruments that are off center in phase land: Any acoustic string instrument in which the note is played (sounded) sideways in regard to the mic position, which would be in the neighborhood of around 90 degrees off phase. Such instruments are guitar, violin, etc.

Fighting The Speaker Air Movement

Think about this. Let’s say that a bass drum and a bass instrument are played at the exact same time or better yet, a drum machine/sequencer programmed with many bass drum hits and bass notes occurring at the same time. Let’s say that either the bass or bass drum was recorded reversed phase caused by a phase wiring problem in the studio audio chain.

The result surely messes with the speaker woofer vibration movement at initial attack dispersing air in two directions at once. Loss of low frequencies and *punch*!

OK, the bass drum and the bass instrument attacks would most likely not occur at the same time which puts this is in the category of comb filtering. If I have time, I will write a tutorial on comb filtering phase issues.

Wiring Cables Reverse Phase

Wiring Two Wire Connector Cables Reverse Phase

Typical two wire connectors are either 1/4” TS phone connectors (tip and sleeve) or RCA connectors.

1/4” TS phone connectors

Whether the connector has two male jacks or a male and female, on one connector end simply reverse the hot and ground wires.

Wire in the following manner:

Using a cable that has a stranded ground wire and a shielded wire, on one connector, solder the shielded wire to the hot pin on the connector. On the other connector, reverse the wiring. Solder the ground wire to the hot pin and the insolated wire to the ground terminal.

(Regarding a female 1/4” TS phone connector, keep in mind the inner most contact should be the “hot” (positive) and the other contact point should be the shielded ground (negative) wire. If this is confusing, plug in a male connector — the male tip equals the “inner most contact.”)

RCA connectors

The same as the above and note regarding a male RCA connector, keep in mind the protruding tip should be the “hot” (positive) wire and the outside circular shell should be the shielded ground (negative) wire.

Other two wire connectors

If using connectors such as mini patch cord cables, etc., follow the above procedures.

Wiring Three Wire Connector Cables Reverse Phase

1/4” TRS (tip-ring-sleeve) phone connector

In most cases, a 1/4” TRS (tip-ring-sleeve) phone connector patch cord will have male connectors on both ends.

Regarding a 1/4” TRS three wire male phone connector, keep in mind the outer tip equals the hot wire, the middle ring equals the neutral wire, and the inner sleeve equals the shielded ground wire.

Regarding a 1/4” TRS three wire female phone connector, keep in mind the inner most connection equals the hot wire, the middle connection equals the neutral wire, and the outer most connection equals the shielded ground wire. If this is confusing, plug in a male connector — the tip equals the “inner most connection.”

We assume this will be a male to male cable. If not, refer the above paragraph.

Wire in the following manner:

Using a cable that has a stranded ground wire and two shielded wires, here we go.

1. On one connector, solder one of the shielded wires (use the white wire) to the hot pin on the connector. This would be the lug located closest the thin plug shaft.
2. Solder the other shielded wire to the center pin (“middle ring” on the plug). (Note the color but let’s say green for this example). This would be the lug located just behind the one mentioned above.
3. Solder the stranded ground wire to the ground point.

On the other connector end, we are going to reverse the hot pin and middle pin.

1. Solder the green shielded wire to the hot pin on the connector. This would be the lug located closest the thin plug shaft.
2. Solder the white shielded wire to the center pin (“middle ring” on the plug). This would be the lug located just behind the one mentioned above.
3. Solder the stranded ground wire to the ground point.

TT patch bay cables

In most cases, such a patch cable uses male connectors on both ends.

This is the same configuration pin wise as a TRS connector so see the above regarding "1/4" TRS (tip-ring-sleeve) phone connector".

XLR connectors

No matter is male or female, the pin out is the same meaning they are numbered.

1. On one connector, solder the white wire to pin 2 (the hot pin).
2. Solder the other shielded wire (green for our example) to the pin 3 (the neutral pin).
3. Solder the stranded ground wire to pin 1 (the ground pin).

On the other connector end, we are going to reverse pin 2 and pin 3.

1. Solder the white wire to pin 3 (the hot pin).
2. Solder the other shielded wire (green for our example) to the pin 2 (the neutral pin).
3. Solder the stranded ground wire to pin 1 (the ground pin).

Important: Always mark these cables with white artist tape, and using a pen, mark REVERSE PHASE! These cables are only to be used when needed to reverse phase (polarity)!

The Wrap Up

When working in real time engineering a recording session with musicians and/or singers *on the clock*, it is most difficult to think slowly through the audio chain of events. When thinking about concepts not under the pressure of being *on the clock* in the studio, I come up with new ideas from time to time such as the crux of this tutorial. Try doing this and you will be amazed what you will discover.

You may want to go on line to www.jaygraydon.com from time to time since I may continue to write tutorials on recording techniques.