

Audio interface box for less than \$10 to create good quality telephone audio on the radio.

I had three criterion.

1) Good audio quality

2) I wanted to use a telephone equipped with the ability to combine two lines for conference calls. So it was better to put the conversion box between the headset and the phone rather than the phone and the wall. That way the box will pick up the phone combined lines. I used a Panasonic KX-t3280 which I just had anyway, but in the case of handset substitution, a good quality phone is important, Sony is recommended by the big companies who spend far more to do this.

3) The listener has to hear the conversation in the room. There may be up to 2-5 others all on microphones so that output needs to be sent right into the phone, so the listener can be a part of the conversation. But to avoid feedback the listener's voice cannot be played back into the phone or squeal of feedback will result.

We have the advantage of using a \$500 Mackie mixer 1402 VLZ (they are in Washington state) that has a mix minus circuit so I can get an output from it that does not contain the telephone voice.

After studying some of the other circuits described below I came up with a simplified design, using just two radio shack transformers, a box some cable and connectors.

I sacrificed the handset cable from an old phone, figured out, using alligator clips and trial and error, which pair of the four wires went to the mouth piece and which to the listening part. Divided it up into two circuits. Both go through their own radio shack isolation transformers (radio Shack part # 273-1374) I got a Radio shack box (270-283a) with PC board installed, Attached the transformers plastic ties (I had to drill to hold the ties). Then to separate phono plugs. Carefully soldered it all up. More or less follow this layout without the capacitor. But making two of them in the same box.

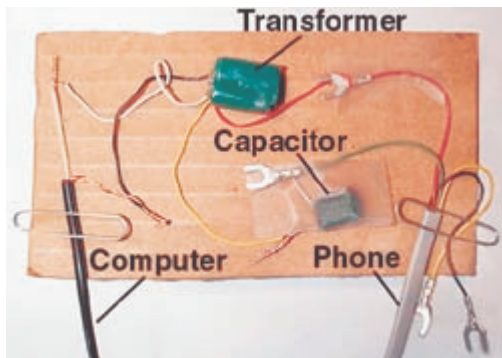


Image ripped from stuff to follow

I did noticed that there is a bit of an extra static charge so I might put a capacitor back in to knock that.

Back ground study

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## Gadgets

by Jay Rose

**Leonardo mixed his own paints. The least you can do is build your own simple audio tools.**

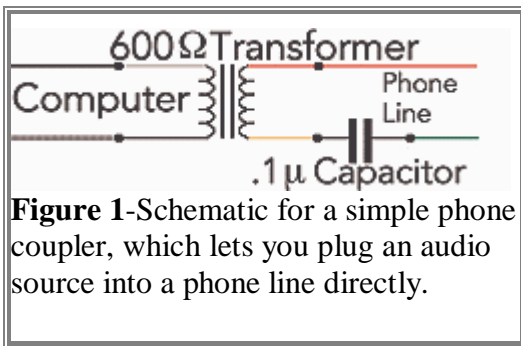
I am not a brilliant audio hardware designer. I've worked with a few of them, and don't consider myself worthy of carrying their voltmeters. But when I started in this business, audio was done in circuits instead of via digital signal processing in CPUs, and any respectable recording engineer had to understand basic electronics. Back then, it was expected that we'd build our own little circuits to solve basic studio problems. This article describes two of the simplest: a phone coupler that lets you plug into a phone line directly, and a compatibility switch to compare mono and stereo versions of your mix. They're cheap, ridiculously easy to build (and to understand), and still highly useful for the computer-based digital editor.

Most of these instructions include Radio Shack part numbers. Their products aren't necessarily the best or the most economical, but they've got a store in every shopping mall in the universe. If you're scared of soldering-and there's no reason you should be-just hand these designs to any high-school hacker: He (or she) probably already has most of the parts on hand.

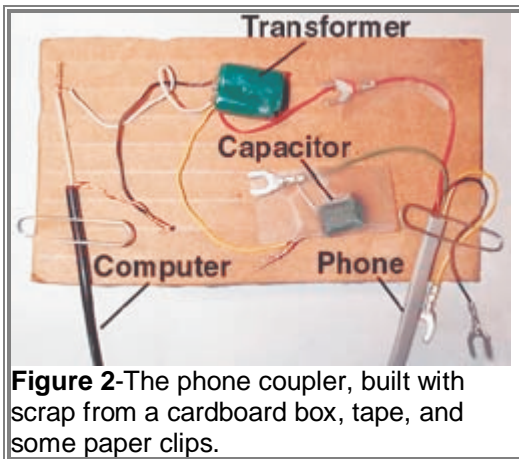
### Something phoney

Real-time ISDN audio codecs are great if you can afford them, but sometimes you just need to play a library cut or dialog edit to a distant client for content approval. This \$10 coupler lets you plug a computer or mixer into a phone line directly, with far better results than holding the handset up to a speaker. It's based on a circuit the phone company used to provide, and under ideal circumstances can sound as good as the connections at radio call-in shows.

The coupler doesn't work with the digital phone lines used by some businesses, but chances are your phones are analog, even if your sophisticated office system includes voice mail and call-forwarding. To find out, turn a phone over and look for a "Ringer Equivalent Number," or REN value. This means the phone wiring is analog. (Sometimes the REN label is hidden under a snap-on mounting bracket, so look carefully.) If you don't see an REN, you can still use the coupler with one of the \$50 handset modem adaptors sold for business travel (though some of them already have coupling audio jacks built-in).



**Figure 1**-Schematic for a simple phone coupler, which lets you plug an audio source into a phone line directly.



**Figure 2**—The phone coupler, built with scrap from a cardboard box, tape, and some paper clips.

The circuit (see **Figure 1**) is laughably simple, with only two components. A 1:1 600-ohm transformer isolates the computer and phone system from each other, while still letting audio pass through. Radio Shack's tiny version (part #273-1374) isn't exactly high-fidelity, but its quality matches the telephone system perfectly. A better sounding transformer might not work as well. A 0.1 microfarad capacitor (#272-1069) protects the transformer from the constant voltage that's always on a phone line, and keeps the coupler from interfering with normal dialing and hanging up. If you're getting parts from a more comprehensive supplier, a capacitor with a 100-volt rating will provide a safety margin.

This coupler is so easy to build that I threw one together on a piece of scrap cardboard (see **Figure 2**), using cellophane tape and paper clips. The thick black wire on the left comes from the computer or mixer's line output. Find a cable with an appropriate plug, cut off one end, and use a knife or cutting pliers to pull off the outer covering. You'll see a metal shield-braided or twisted wires, or a foil wrapper-around a single insulated conductor. Separate the shield from the inner wire and twist it together, then strip back some of the insulation from the end of the inner wire (or buy a #42-2370 cable, which is already prepared for use). The gray telephone wire on the right is a #279-310, which has a standard modular plug on the other end. Follow the color codes in the photo and schematic in **Figure 1**, and connect the wires by twisting them together (soldering isn't necessary). Make sure the connections are secured so they don't touch each other, and be sure to insulate the wires on the telephone side of the circuit so nobody accidentally touches them: When the phone rings, these carry as much as 125 volts. It's low current, but can give you quite a tickle.

To use the coupler, just plug it into a sound source and a telephone jack. You'll need a regular phone on the same line to dial and monitor the call; the easiest way to hook them both up is a Y-connector (#279-357). Raise the source's volume until it just starts to distort in your phone's handset, then back it off a little. It's okay to leave the coupler connected between calls. You can also use this coupler to record telephone calls through a line-level input, but it's less than ideal. The local sound can be as much as 20dB louder than the distant caller. To even them out, you need a totally different circuit.

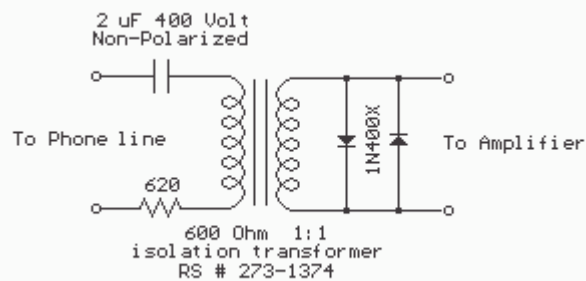
### **One-sided listening**

Even though most television mixes are stereo, mono

<http://livedv.com/magazine/1998/0998/rose0998.html>

## Telephone Audio Interface

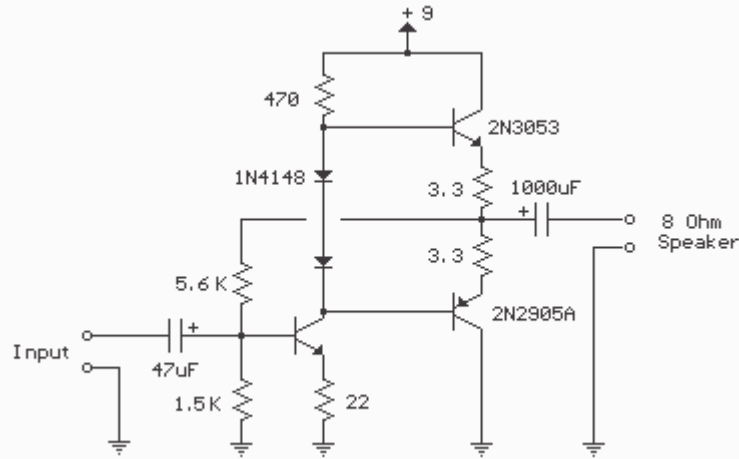
Audio from a telephone line can be obtained using a transformer and capacitor to isolate the line from external equipment. A non-polarized capacitor is placed in series with the transformer line connection to prevent DC current from flowing in the transformer winding which may prevent the line from returning to the on-hook state. The capacitor should have a voltage rating above the peak ring voltage of 90 volts plus the on-hook voltage of 48 volts, or 138 volts total. This was measured locally and may vary with location, a 400 volt or more rating is recommended. Audio level from the transformer is about 100 millivolts which can be connected to a high impedance amplifier or tape recorder input. The 3 transistor amplifier shown above can also be used. For overvoltage protection, two diodes are connected across the transformer secondary to limit the audio signal to 700 millivolts peak during the ringing signal. The diodes can be most any silicon type (1N400X / 1N4148 / 1N914 or other). The 620 ohm resistor serves to reduce loading of the line if the output is connected to a very low impedance.



[http://ourworld.compuserve.com/homepages/Bill\\_Bowden/page8.htm#phone.gif](http://ourworld.compuserve.com/homepages/Bill_Bowden/page8.htm#phone.gif)

## 3 Transistor Audio Amp (50 milliwatt)

Here is a little audio amplifier similar to what you might find in a small transistor radio. The input stage is biased so that the supply voltage is divided equally across the two complimentary output transistors which are slightly biased in conduction by the diodes between the bases. A 3.3 ohm resistor is used in series with the emitters of the output transistors to stabilize the bias current so it doesn't change much with temperature or with different transistors and diodes. As the bias current increases, the voltage between the emitter and base decreases, thus reducing the conduction. Input impedance is about 500 ohms and voltage gain is about 5 with an 8 ohm speaker attached. The voltage swing on the speaker is about 2 volts without distorting and power output is in the 50 milliwatt range. A higher supply voltage and the addition of heat sinks to the output transistors would provide more power. Circuit draws about 30 milliamps from a 9 volt supply.



[Menu](#)

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## ePanorama.net - Ground Loops

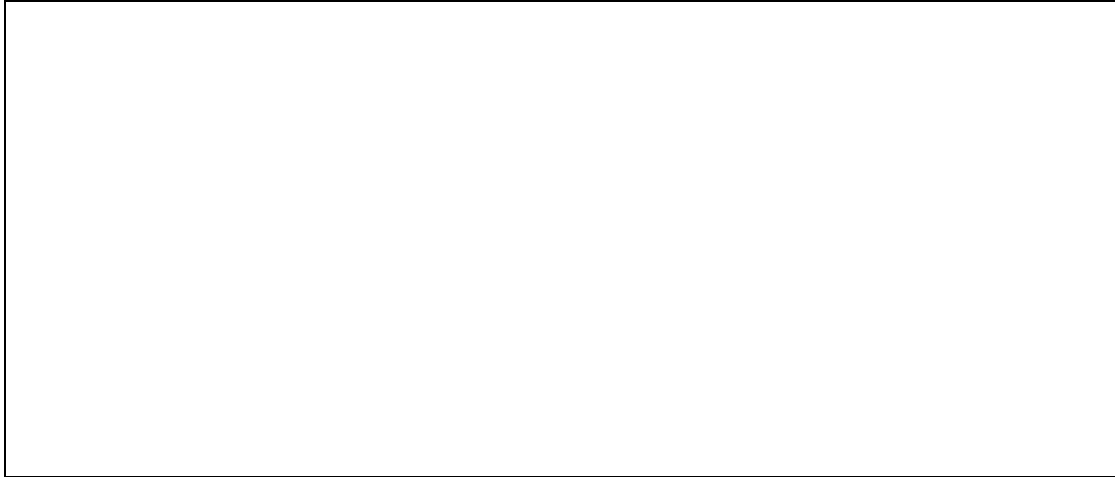
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### How to build your own audio isolator

#### Building yourself an audio isolation transformers

If you want to build one yourself, you have to get two audio transformers which have 1:1 transformation ratio and greater than 1 kohm impedance. There are high quality audio transformers in the market that meet those specs, but those can be quite expensive. Another option is to use 600:600 ohm isolation transformers widely available for telecommunications and other uses. Those are not that high quality as good audio transformers, but can be well adequate for many not so demanding multimedia applications like computer audio if suitable transformer is selected.

#### Isolation for unbalanced lines



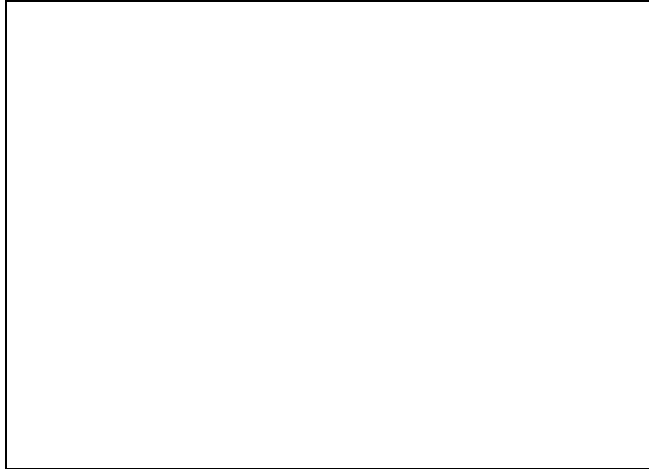
Notice one thing in the transformer wiring: Many transformers have the coil starting marked with dot. If you put the audio signal to the transformer in such way that the center wire in RCA connector is always connected to the coil end marked as coil start you will get a nice isolator. If you for some reason connect the wires in one side of the transformer in the wrong way, your transformer will cause a 180 degree phase shift to your audio signal. If some of the signals in your audio system will get this kind of phase shift and some other will not, you will face all kinds of sound quality problems. If you have this kind of system, it is best to test that the audio isolation transformer will not cause unintended phase shift to the signal.

The transformer used for this should have a flat frequency response over whole audio frequency range, should have high enough impedance at lowest frequencies and should not cause noticeable distortion. If the impedance drops on low frequencies, this can cause attenuation of the lowest frequencies if the transformer is connected to the equipment which have a very high output impedance.

This kind of isolation transformer circuit should be best connected on the receiving end (equipment input) end of the cable to get best performance. If the cable is not very long (not many meters), in practice it does not matter to which end of the cable this transformer isolation circuit is connected.

### **Isolation for balanced lines**

For balanced lines equipped with XLR connectors the audio signals are transported between signal pins 2 and 3. So the isolation transformer should be wired so that the primary is wired between input XLR pins 2 and 3 and the secondary is wired between output XLR pins 2 and 3. The grounding pin 1 of the output XLR connector can be wired to metal case of the isolator, but do not connect the input XLR connector pin 1 to anything.



The transformer to be used in the circuit above should be 600:600 ohm audio transformer which can handle the signal levels you are about to transfer through the isolator without causing too much distortion or other problems (not very low signal level microphone transformer for line signals etc.). The transformer which is used should have very flat frequency response over the whole audio frequency range (20Hz..20kHz) and even more. This kind of isolation transformer circuit should be best connected on the receiving end (equipment input) end of the cable to get best performance. If the cable is not very long (not many meters), in practice it does not matter to which end of the cable this transformer isolation circuit is connected.

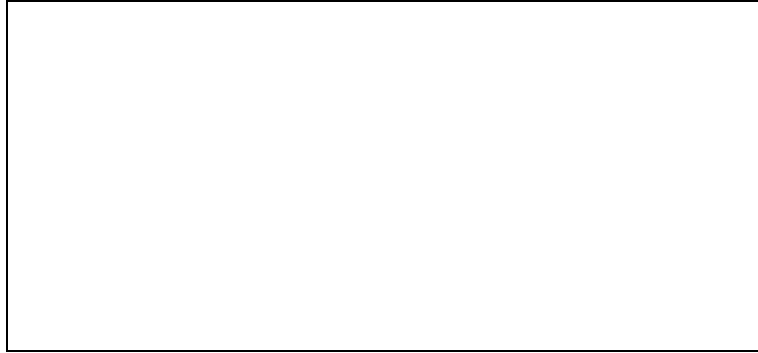
[Jensen Transformers](#) has a good application note [JT-11SSP-6M 600 Ohm Line Isolation Application](#) which describes how to build a very nice isolation transformer for 600 ohm balanced audio lines. Other simpler design is described at [JT-11-DM Basic Hum Stopper Transformer isolation box](#). Both of the designs use more components than my simple design but are expected to perform better under strong RF fields.

## **Circuits I have built**

### **Circuit using telephone line transformers**

I built my first isolators using two high quality telephone line coupling transformers which have 600 ohm impedance. This is the most commonly transformer type used in high-speed modems. Best of those are quite wideband devices (far more bandwidth than usual 300-3400 Hz as used in telephone). Using two of those transformers and few RCA connectors made quite satisfactory (but not really hifi) audio isolator. The connection is easy: connect primary side of the transformer to one audio connector and secondary to other.

I used EOP Z1612 transformers in my test circuit and got quite acceptable frequency response of  $\pm 1$  dB from 40 Hz to 20 kHz as you can see in figure below. The bass frequency below 40 Hz is not good.



The frequency measurements were made with Nacamichi T-100 Audio Analyzer and the isolator circuit was connected between it's 600 ohm output and 50 kohm input. I don't know if EOP Z1612 transformers are still available at [Farnell](#), but you can try. If you are looking for other high quality transformers which could be a used, I would try ETAL P2001. I haven't been able to test their performance in this application, but they have proven to good transformers in other laboratory test and applications. Avoid cheapest telephone and audio transformers, because their performance is very poor at frequencies over about 5 kHz (for example Radio Shack (273-1374)).

### **Transformer isolation using audio transformer**

One way I got my hands on the component of old cenral radio systems amplifiers. Those amplifiers had a transformer isolated balanced audio input with impedance of around 1 kohm. I took the high quality input transformer from the preamplifier board and tried how well they would work in my applications. The text on the top of the transformer indicated that those transformers were 1000:1000 ohm audio transformers. The measurements showed that those transformers perform better than those telephone transformer.

I took a project box and pair of XLR connectors to make a nice set of audio isolation/balancing transformers for professional audio system. I keep those boxes with the audio wiring set and use them as problem solvers if there is some humming problem in the system which can't be easily solved in other ways.

### **Notes on selecting audio transformer**

If you want the best sound quality (not much sound degration) you must select a good audio transformer. Any transformer of the proper ratio will give you a high impedance input, but not any transformer will sound good. Cheap transformers add distortion, mostly at low frequencies, and it's not the "good" kind of distortion. They also tend to have poor high frequency response.

Makign good audio transformers is not easy. Lower impedance transformer are generally easilier that higher impedance. High Z primary is difficult to wind because the capacitance between windings starts to become a big issue with that many turns. And that kills your high end.

There was a day when audio transformers were commodity items and they were in everything, but now they aren't so common and as a result, the cheaper ones aren't in the catalogues any more, and the better ones cost more than they used to because there aren't the economies of scale.

If you get a cheap transformer, you will eventually get depressed with the way it sounds, and you'll be spending the money on a Jensen, a Reichenbach, Sowter, or Lundahl (or any other reputable manufacturer) anyway. So you might as well just get it now and save the wasted time effort and money.

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*author: Tomi Engdahl*

[http://www.epanorama.net/documents/groundloop/audio\\_isolator\\_building.html](http://www.epanorama.net/documents/groundloop/audio_isolator_building.html)

## Isolator transformer for audio lines

*Copyright Tomi Engdahl 1996-1998*

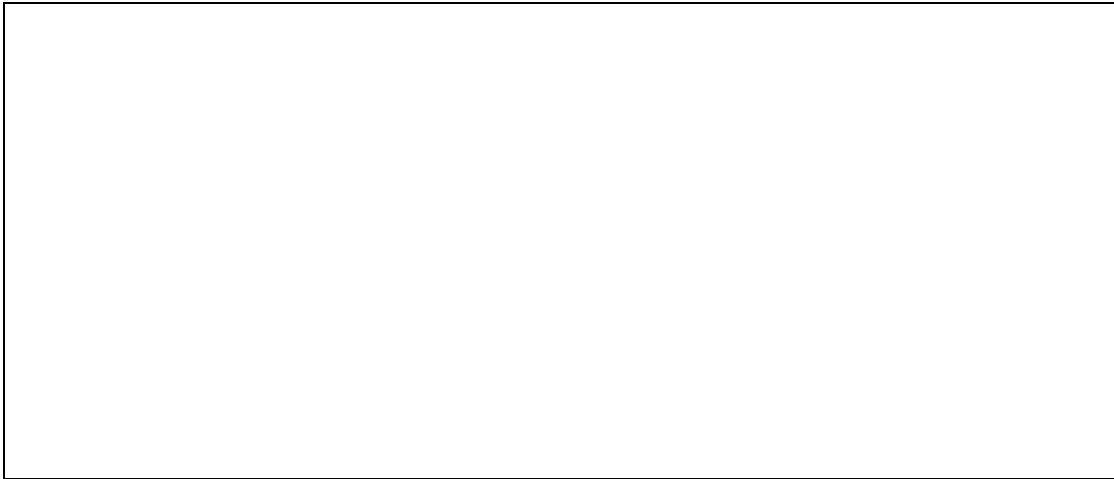
If you want to do the ground loop elimination in audio path, you have to cut the galvanic connection but pass the whole audio range. The simplest and most common way to do the isolation is use audio transformer which is ment for audio use. Transformers for audio use have some problems like distorted bass response and attenuating in high-frequency response. Basically a transformer slows down upper frequencies and allow the low frequencies to pass first, creating what we perceive as a "fat/warm" tone. Inadequate frequency response on the low end (rolloff at like 20Hz), causes low frequencies to be "slowed", allowing the upper frequencies to be heard first, this is perceived as "barky/brittle". High-quality audio transformers cover whole audio band with good response, but those are quite expensive.

There are ready made circuits available from shops selling car audio stuff (ground loops are usually problem also in car environment). If you live in USA, take a look in Radio Shack's catalog on car electronics or check the [Radio Shack Product Support](#) pages which have specifications of Ground LP Isolator (270-0054) which cost about \$15US. For more professional product check JK Audio Pureformer Stereo Isolation Transformer. Those products seem to be quite suitable for solving ground loop problems in consumer audio systems, but I have not tested them myself. Europeans should take a look at their nearest dealer which carries Monacor products, because Monacor's new catalogue lists FGA-40 (Best.-Nr. 06.4370) Ground Isolators which are 1:1 audio isolation with 10 kohm impedance (look quite good on the catalogue).

For professional audio use buy high quality commercial audio isolation/balancing transformers (those are very handy to keep around to solve unexpected ground loop problems). DI-boxes are also used to solve ground loop problems in a PA situations where different instruments are connected to mixing desk. Most DI boxes are active and are almost useless are getting rid of earth loops and stopping buzzes & hums etc. A GOOD passive DI which provides isolation is the only way to go.

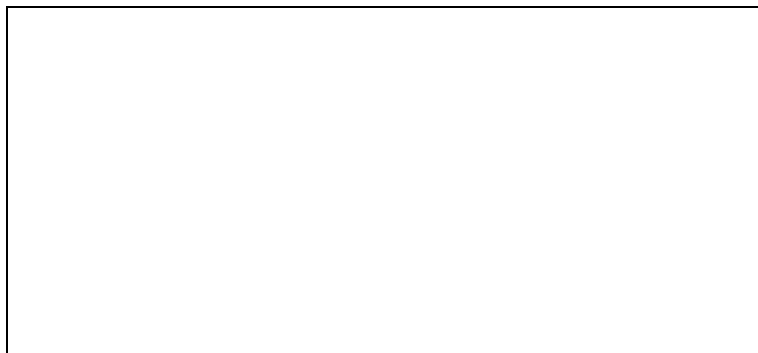
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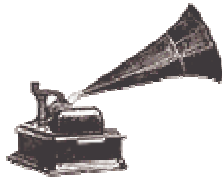
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## Notes about signal phasing

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# The OOPS Effect

## What is OOPS?

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**OOPS** stands for **Out Of Phase Stereo**. It is a simple technique used to process the two channels of modern stereo recordings into a "new", third channel, enabling us uncover "hidden" sounds in stereo recordings. The resulting OOPSed signal is a single channel, mono signal.

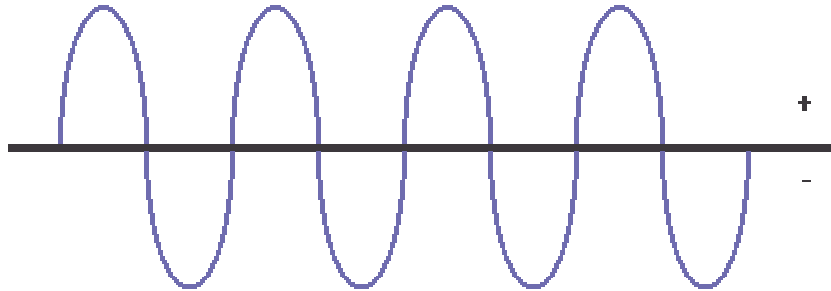
This process is also known as "Left Minus Right". When Quadraphonic recording was the rage in the seventies, OOPS or Left Minus Right was used as a cheap way of creating a new, third channel to increase the stereo listening experience.

The OOPS effect can be used on all of your stereo recordings to hear them in a way you've never heard them before. The OOPS process has also been used by some unscrupulous bootleggers to create and sell "new mixes" of previously available material.

## Why OOPS Works

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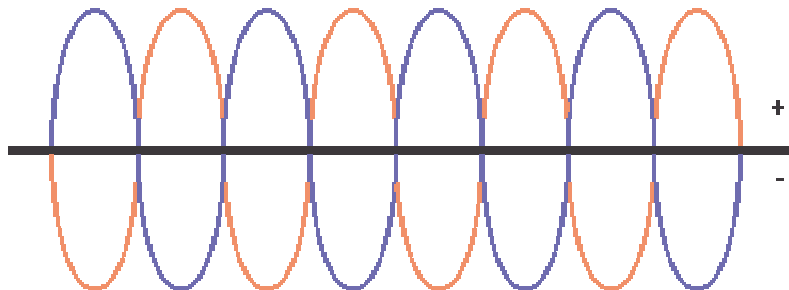
The speakers (or headphones) of your stereo system convert an electrical signal into sound, by moving the speaker cone in relation to the positive and negative waves in the signal.



In the simple example of a pure tone, the electrical signal makes the speaker "oscillate" (move back and forth rapidly) by moving one way, outwards for example, on the positive (+) halves of the electrical waves, and move back the other way on the negative halves. The speaker cone movement makes the air in front of it move, and this frequently oscillating air reaches your ears as sound. This single wave is a "frequency" (how frequently the sound moves back and forth) and you hear it as a tone.

Recorded sound is made up of many of these electrical waves in various combinations relating to the frequencies in the sounds being recorded.

Now, imagine two signals that are identical, except they are "out of phase" with each other, that is, that when one signal is having a positive wave peak, the other is having an identical negative peak.



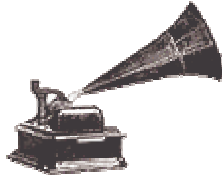
If these signals are mixed together, they will cancel each other out. The positive peak of the first signal tries to make the speaker move out, but as it does, it is counteracted by the equal but opposite negative peak from the second signal, which tries to pull the speaker cone back in, so no sound results.

Now the good part!

In a stereo recording, some of the sounds are recorded on the left channel, and some are recorded on the right channel. However, because of the mixing done during the recording process, some sounds are on BOTH channels. It is very common in modern stereo recordings, for example, for the instruments to be in stereo (different) on both sides, and the vocal to be "in the middle", by having it mixed into both channels.

Now, if you took one of the channels of the above example and combined it OUT OF PHASE (plus to minus and minus to plus) with the other channel, whatever was THE SAME in both channels would cancel out, and you would hear only what was DIFFERENT in both channels. In our example, you would cancel the vocal, and hear only the instrumental accompaniment.

[Here are some instructions](#) on various ways to hear OOPS in your very own home!



# The OOPS Effect

## How To Hear OOPS On Your Stereo

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There are several ways to listen to a stereo recording using the OOPS technique. Some are safer to your equipment than others. If you're thinking about doing any of these hookups, please read this page carefully!

### Method #1: Buy New Equipment

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Commercially made Surround-Sound amplifiers and adapters use enhanced Left-Minus-Right circuits to create the same thing as the OOPS effect, and they're coming down in price, so if you can find one, that would probably be the easiest way to hear OOPS.

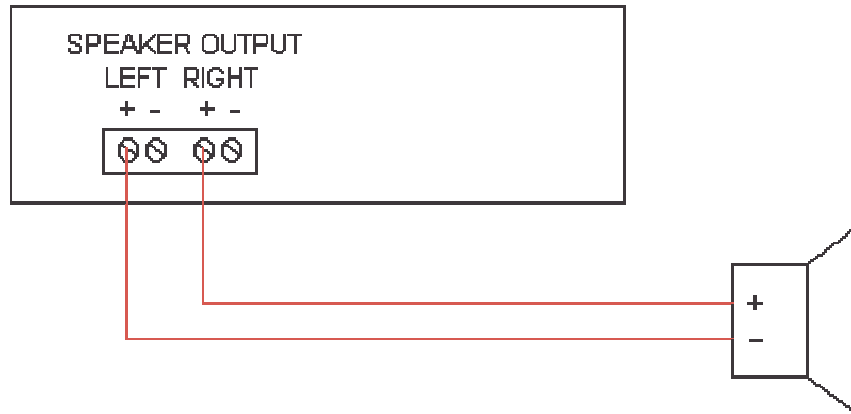
To hear OOPS, hook up the rear speakers to the surround-sound unit, turn off the "front" speakers and listen to the "rear" speaker(s) only.

### Method #2: Connect Outputs

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You can use your existing stereo system to listen to a stereo recording using OOPS without purchasing any new equipment. To do so, you need to make a special physical speaker hook-up.

**WARNING!** *I have gotten one report from a user who said following the below procedure permanently damaged his amplifier. Although I cannot guarantee your system will suffer no damage when doing this, I have never personally had such a problem doing this physical hookup on many different systems. If you're at all in doubt about trying this, please try one of the other "safe methods" listed below.*



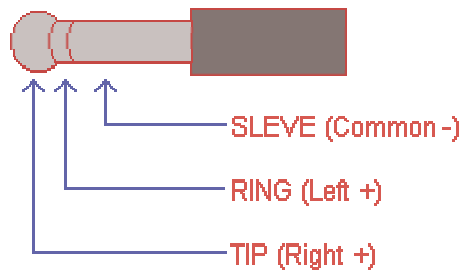
Using one speaker, hook up one speaker lead to the positive (+) connection of the left channel, and hook up the other speaker lead to the positive (+) connection of the right channel.

Now, put on your stereo recording. Use the BALANCE control of your stereo amplifier to adjust the signals so that they cancel out the most of the common sounds and you hear the most OOPSed sound.

Using the above hookup, a third speaker can be used in conjunction with your standard two speakers still connected in the normal manner to create "surround-sound" for your home stereo, without having to buy any other equipment.

### Method #2A: Headphones

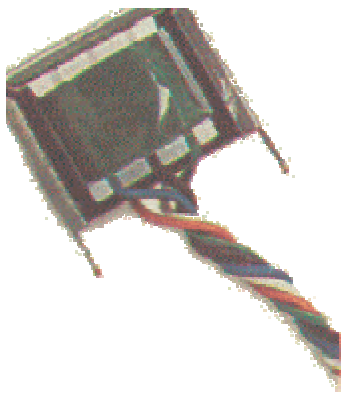
You can do the same physical hookup as above but using your headphones instead of your speakers. To do so, you need to disconnect the "SLEEVE" portion of the stereo headphone plug. (This is the connection closest to the handle of the plug.)



A good way to accomplish this without permanently destroying your headphones is to make (or modify) a headphone extension cable that is missing the sleeve connection. Another easy way to do this is by covering the sleeve portion of the plug with scotch tape. Please note this is as potentially dangerous as the speaker hookup described above.

### Method #3: Connect Inputs

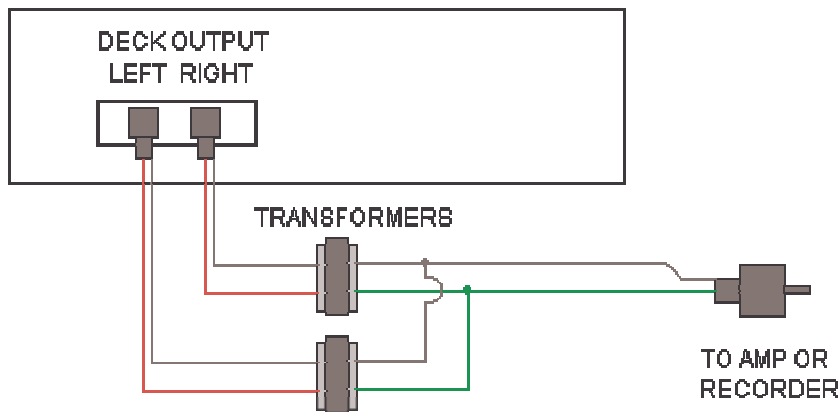
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Two isolation transformers, which can be bought at any electronics supply store, can be used to safely combine the signals of the stereo channels in a much more safer way than Method #2 above. In addition, you can also use this method to record the OOPS effect on tape!

Use two 600 ohm 1:1 (that's pronounced "one to one") isolation transformers (Radio Shack Part #273-1374; \$3.99 each in the 1997 catalog).

Connect the output wires of the left channel of the cassette deck, cd player or tuner (hereinafter known as "deck") to the input wires of one of the transformers. Connect the output wires of the right channel of the deck to the input wires of the second transformer. You can then connect the output wires of the transformers together, and then connect this pair of wires to the input of an amp or a tape recorder, etc.



Connect the output wires together in one polarity and you get a mono signal. Turn one of the pair of output wires around so now they are out of phase with each other, and you get OOPS!

If your deck has output level controls (many don't), adjust them while listening to a stereo recording so that they cancel out the most of the common sounds and you hear the most OOPSed sound.

#### **Method #3A: Turntable**

A turntable can also be hooked up via an input adapter as described above, but it must first be connected through a turntable pre-amp.

#### **Method #4: Process a Computer Sound File**

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This requires the use of a computer that can record and play back sounds, and associated software that can manipulate these sound files.

First, record all or part of the stereo recording you want to hear in OOPS on your computer in stereo, and save it as a file.

Next, use your sound editing software to read the file. The first thing you need to do is select one channel only, so that the next change will be made to one channel and not the other. Then select the "editing" or "special effects" option known as "invert", and invert the selected channel.

Next, reselect both channels, and use the function in your software to pan both channels into one, making it into one mono signal. Since one of the channels was inverted before you did this, when you listen back to this mono signal, you'll now hear the track with the OOPS effect!

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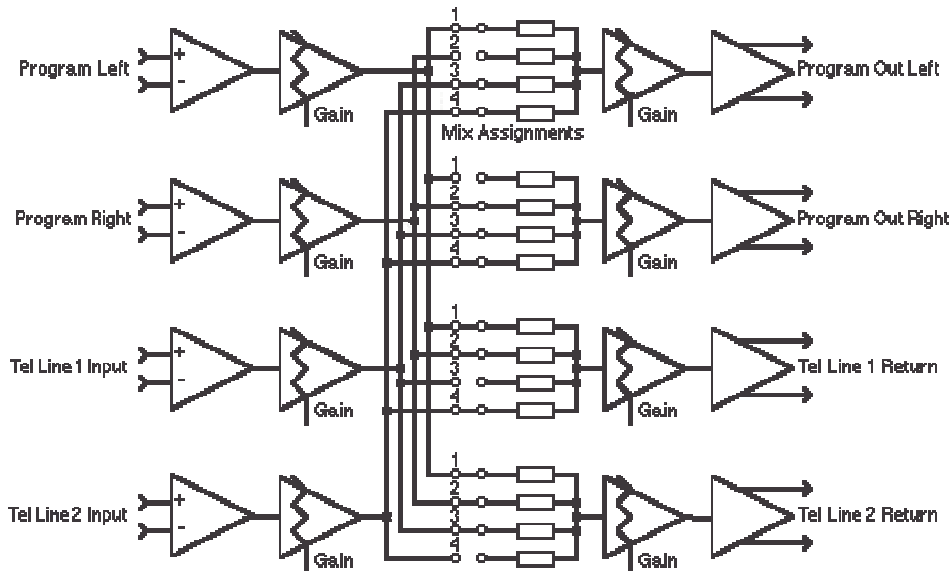
## **The IFA-10 as a Mix Minus**

### **Application Note #6**

#### **Introduction**

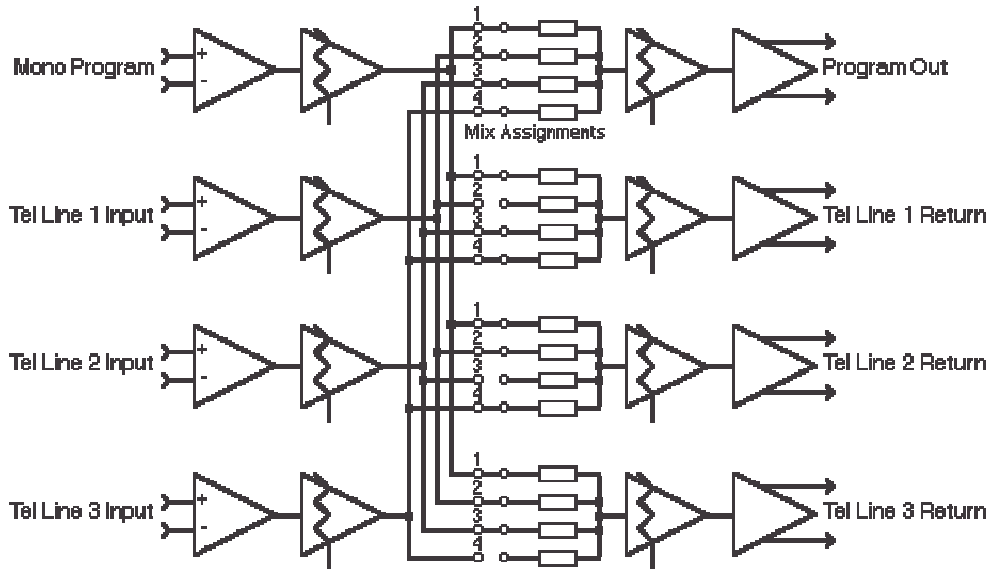
Radio talk shows are currently enjoying unprecedented success. Witness the phenomenal success of hosts such as Dr. Dean Edell, Rush Limbaugh, Larry King, not to mention comedy car repair experts. Is it any wonder that local hosts are clamoring for the limelight as well? If your station has plans to add a talkshow, you will face the problem of creating a mix-minus, a signal that returns all of the program audio, except the caller's own voice, to the caller. Otherwise, echo, caused by Telco system signal delays, can confuse the caller.

If you have a production console with auxiliary buses that can be assigned to create a mix-minus, you are in luck. However, most simple on-air broadcast consoles do not have that luxury. And note that it takes one auxiliary mix-bus per phone line to create the mix-minus within the console. The IFA-10 is the answer for a simple two or three phone line talk show mix-minus. The IFA-10 is a 4 IN by 4 OUT assignable mixer, using removable internal jumpers to create the desired mixing array.

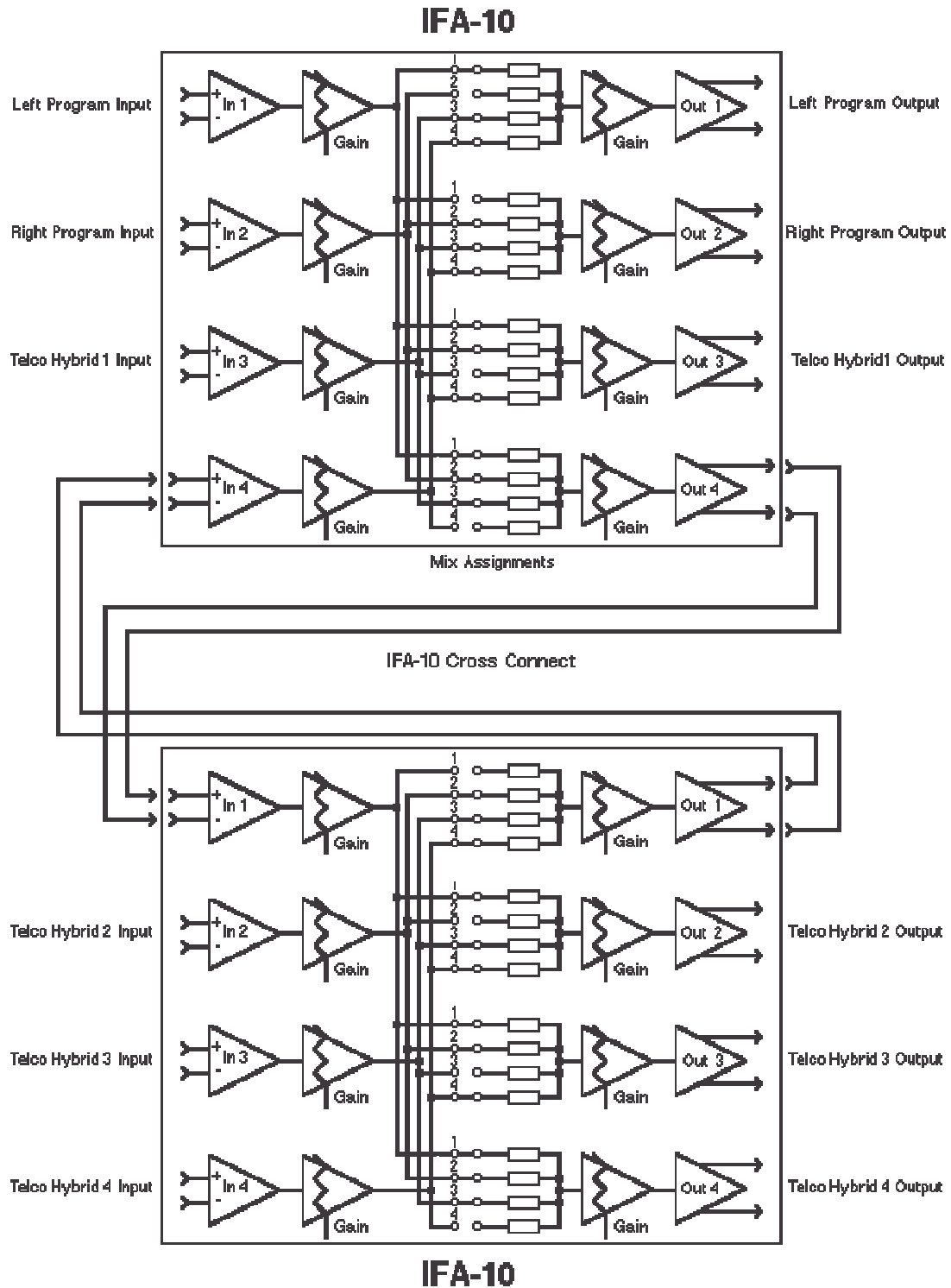


**Fig. 1.0 A Two Phone Stereo Mix-Minus**

Figure 1.0 above shows a block diagram of a mix-minus system for use with a stereo console, with the IFA-10 downstream from the console. All of the audio for the host and local guests is mixed in the console. The output of the console is fed to the IFA-10 at the Program Left and Right input positions. By following the schematic diagram you will see that the program audio follows across to the respective Program outputs. Additionally, telco audio is mixed with program audio. At the Tel Line 1 and 2 outputs you will see that they receive the program audio and the audio from the other Tel lines, but not audio from their own line. Of course you will need a Telco "Hybrid" for each of the phone lines and appropriate controls for the phone lines. Gain control for the incoming telco levels is also important. Metering at the program outputs is a must. Figure 2.0 below shows the configuration for a mono mix-minus and Fig. 3.0 shows what can be done by interconnecting two IFA-10s.



**Fig. 2.0 A Three Phone Mono Mix-Minus**



**Fig 3.0 A Four Phone Stereo Mix-Minus**

Larger mix-minus systems are easily possible using System 1000™ technology. For example, we recently designed a 48 by 20 system for one customer and we can do the same for you!

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