Telephone line audio interface circuits

First of all, I must officially advise against connecting anything other to the telephone line than equipment approved for the purpose by the telephone company or some other regulatery body. Telephone company tend to be very strict about unauthorized gear hanging on their lines, and if something does go wrong with your gadget (like putting dangerous voltages to telephone line) you will be in deep trouble.

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How telephone works

A telephone uses an electric current to convey sound information from your home to that of a friend. When the two of you are talking on the telephone, the telephone company is sending a steady electric current through your telephones. The two telephones, yours and that of your friend, are sharing this steady current. But as you talk into your telephone's microphone, the current that your telephone draws from the telephone company fluctuates up and down. These fluctuations are directly related to the air pressure fluctuations that are the sound of your voice at the microphone.

Because the telephones are sharing the total current, any change in the current through your telephone causes a change in the current through your friend's telephone. Thus as you talk, the current through your friend's telephone fluctuates. A speaker in that telephone responds to these current fluctuations by compressing and rarefying the air. The resulting air pressure fluctuations reproduces the sound of your voice. Although the nature of telephones and the circuits connecting them have changed radically in the past few decades, the telephone system still functions in a manner that at least simulates this behavior.

The current which powers your telephone is generated from the 48V battery in the central office. The 48V voltage is sent to the telephone line through some resistors and indictors (typically there is 2000 to 4000 ohms in series with the

48V power source). The old ordinary offices had about 400 ohm line relay coils in series with the line. Here is a simplified picture of typical traditional telephone line interface: to

```
Telephone central                              Telephone equipment

Ground -----+
            |                          Hookswitch  /
          COIL 5H                            +--o/ o-------+
            |                                 |             )|| ___
          Resistor 200 ohm                    |         ____)||(   |
      2uF   |                            TIP  |        |    )||( Speaker
    --||----+--o---------------------------o--+       Mic   )||(___|
Audio               Line wire                          |    )||
    --||----+--o---------------------------o-----------+----+
      2uF   |                            RING
          Resistor 200 ohm
            |
          COIL 5H
            |
-48V DC ----+
```

In some old switches there rere no separate reistors after the relay couils. Just a relay with 2x500 ohm coils was used for both current limiting and the coils between the line wires and the power source.

When your telephone is in on-hook state the "TIP" is at about 0v, while "RING" is about -48v with respect to earth ground. When you go off hook, and current is drawn, TIP goes negative and RING goes positive (I mean less negative). A typical off hook condition is TIP at about -20v and ring at about -28v. This means that there is about 8V voltage between the wires going to telephone in normal operation condition. The DC-resistance of typical telephone equipment is in 200-300 ohm range and current flowing through the telephone is in 20-50 mA range.

Why 48V voltage is used in telephone systems ?

The -48V voltage was selected because it was enough to get through kilometers of thin telephone wire and still low enough to be safe (electrical safety regulations in many countries consider DC voltages lower than 50V to be safe low voltage circuits). 48V voltage is also easy to generate from normal lead acid batteries (4 x 12V car battery in series). Batteries are needed in telephone central to make sure that it operates also when mains voltage is cut and they also give very stable output voltage which is needed for reliable operation of all the circuit in the central office. Typically the CO actually runs off of the battery chargers with the batteries in parallel getting a floating charge.

The line feeding voltage was selected to be negative to make the electrochemical reactions on the wet telephone wiring to be less harmful. When the wires are at negative potential compared to the ground the metal ions go form the ground to the wire instead of the situation where positive voltage would cause metal from the wire to leave which causes quick corrosion.

Some countries use other voltages in typically 36V to 60V range. PBXes may use as low as 24 Volts and can possibly use positive feeding voltage instead of the negative one used in normal telephone network. Positive voltage is more commonly used in many electronics circuits, so it is easier to generate and electrolysis in telecommunications wiring is not a problem in typical environment inside office buildings.

Some older offices employ battery reversal (swap DC feed to tip and ring) to signal off-hook at the remote end.

What is sealing current ?

The current sent to telephone line as an another advantage besides that it supplies the operating power for your telephone. Telephone practice uses (or did use) twisted splices. These splices did not always make good connections. Placing a small DC bias on a long transmission pair is often done by telecommuncation carriers to reduce poor connections, and noisy lines. The DC bias is often refered to as a "sealing current". So putting DC current through the cable sealed the connection and so improved the transmission.

Why full duplex operation in single wire pair ?

Full-Duplex is a term used to describe a communications channel which is capable of both receiving and sending information simultaneously.

Telephone sets (ordinary analog ones) have only 2 wires, which carry both speaker and microphone signals. The signal path between two telephones, involving a call other than a local one, requires amplification using a 4-wire circuit. The cost and cabling required ruled out the idea of running a 4-wire circuit out to the subscribers' premises from the local exchange and an alternative solution had to be found. Hence, the 4-wire trunk circuits were converted to 2-wire local cabling, using a device called a "hybrid".

This function can send and receive audio signals at the same time is accomplished by designing the system so that there is a well balanced circuit in both ends of the wire which are capable or separating incoming audio from outgoing signal. This function is done by telephone hybrid circuit contained in the network interface of the telephone.

What is the bandwidth of the telephone line ?

A POTS line (in the US and Europe) has a bandwidth of 3kHz. A normal POTS line can transfer the frequencies between 400 Hz and 3.4 Khz. The frequency response is limited by the telephone transmission system (the actual wire from central office to your wall can usually do much more).

Nowadays POTS is sharply bandlimited due to the fact that the line almost always is digitally sampled at 8kHz at some point in the circuit. The absolute, theoretical limit (with perfect filters) is therefore 4kHz - but this isn't reality, 3.4 kHz maximum frequency is.

The bass frequency response is limted because of the limitations in telephone system components: transfromers and capacitors can be smaller if they don't have to deal with lowest frequencies. Other reason to drop out the lowest frequencies is to keep the possibly strong mains frequency (50 or 60 Hz and it's harmonics) hummign away from the audio signal you will hear.

Network interface details

The telephone has a circuit called network interface (also called voice network or telephone hybrid) which connects the microphone and speaker to the telephone line. Network interface circuitry is designed so that it sends only the current changes the other telephone causes to the speaker. The current changes which the telephone's own microphone generates are not send to the speaker. All this is accomplished using quite ingenious transformer circuitry. In theory the hybrid circuit can separate all incoming oudion from the audio sent out at the same time if all the impedances in the circuitry (hybrids on both ends and the wire impedance in between) are well matched. Unfortunately, the hybrid is by its very nature a "leaky" device. As voice signals pass from the 4-wire to the 2-wire portion of the network, the higher energy level in the 4-wire section is also reflected back on itself, creating the echoed speech. The because circuit does not work perfectly and you can still hear some of your own voice in the speaker

The actual amount of signal which is reflected back depends on how well the balance circuit of the hybrid matches the 2-wire line. In the vast majority of cases, the match is quite poor, resulting in a considerable level of signal being reflected back.

The signal which is reflected back is not always bad and in normal telephone some if it is really intentional by the design. The separation of the received and transmitted audio could be done much better with modern electronics than with old phones, but but people who use the telephone prefer to hear some of their own voice back. Radio Shack's "Understanding Telephone Electronics" (copyrighted around 1985 I think) calls this effect sidetone and gives the impression that this was indeed intentional in order for the speaker to determine how loud they were speaking with reference to the called party.

Signaling

Ringing signals

When the central office want to make your telephone ring it will send an AC ringing voltage to the line which will ring the bell in your telephone. Most of the world uses frequencies in 20..40 Hz range and voltage in 40..150 volts range.

The ringer is built so that it will no pass any DC current when it is connected to telephone line (traditionally there has been a capacitor in series with the bell coil). So only the AC ring singal can go though the bell and make it ring. The bell circuit is either designed so that it has high impedance in audio frequencies or it is disconnected from line when phone is picked off-hook.

For more information about telephone ringing take a look at my telephone ringing circuits web page.

Dialing

There are two types of dials in use around the world: pulse dialing and tone dialling.

The most common one is called pulse dialing (also called loop disconnect or rotary dialing). Pulse dialling is oldest form of dialing, it's been with us since the 1920's. Pulse dialing is traditionally accomplished with a rotary dial, which is a speed governed wheel with a cam that opens and closes a switch in series with your phone and the line. It works by actually disconnecting or "hanging up" the telephone at specific intervals. The mostly used standard is one disconnect per digit (so if you dial a "1," your telephone is "disconnected" once and if you dial "2" your telephone is "disconnected" twice and for zero the line is "disconnected" ten times) but there are also other systems used in some countries.

Tone dialing is more modern dialing method is usually called with names Touch-tone, Dual Tone Multi-Frequency (DTMF) or Multi-Frequency (MF) in Europe. Touch tone is fast and less prone to error than pulse dialing. Bell Labs developed DTMF in order to have a dialing system that could travel across microwave links and work rapidly with computer controlled exchanges. Touch-tone can therefore send signals around the world via the telephone lines, and can be used to control phone answering machines and computers (this is used in many automatic telephone services which you operate using your telephone keypad). Each transmitted digit consists of two separate audio tones that are mixed together (the four vertical columns on the keypad are known as the high group and the four horizontal rows as the low group). Standard DTMF dials will produce a tone as long as a key is depressed. No matter how long you press, the tone will be decoded as the appropriate digit. The shortest duration in which a digit can be sent and decoded is about 100 milliseconds (ms).

Other signals

The telephone central can send any different types of signals to the caller telling the status of telephone call. Those signals are typically audio tones generated by the central office. Typical this kind of tones are dialing tone (typically constant tone of around 400 Hz), calling tone (tone telling that the telephone in other end is ringing) or busy tone (usually like quickly on and off switched dialing tone). The exact tones used vary from country to country.

Detecting end of call

There is no guaranteed (single) way to determine when a call was terminated at the far end. Depending on the switch type you need to look at loop break (loss of loop current), change of DC polarity, dial tone, stutter dial tone, and/or silence. If you want to do something for unknown lines on unknown switches then you will need a combination of the above.

Safety issues of telephones

The telephones should be designed so that they do not cause danger to the user. The 48V DC voltage in telephone lines does not cause immediate danger to the user, but the AC ring signal (70-120V AC) can give a nasty shock. Telephone wires are also exposed to any different environmental effects (nearby lightning, ground potential differences in buildings, interference from power lines) which can cause that there are sometimes high voltage spikes on the telephone wires. Normal telephones are designed to be fully enclosed in insulating plastic case which provides isolation. The plastic case works nicely as isolation if there are no metal contacts in the telephone which are somehow connected to telephone line.

If the equipment has touchable metal surfaces or connections for power from other sources the equipment must provide proper electrical isolation between the telephone line. You have to provide 1500 volts between the telephone line and rest of your electronics. Typically computer modems does this isolation using transformers, optoisolators and relays.

The telephone company can't know what kind of foolishly designed gadgets their customer has hanging on the end of line, but it does specifically tell them, that at any time without warning and at their convenience they might just put a variety of voltages and currents on any given loop. If the device is not designed to meet the regulations it can cause dangers or problems in those situations. Equipments must be also designed to meet the safety rules so that they do not led dangerous voltages to enter telephone lines and causing a danger of being electrocuted to the telephone company workerd which do the wiring.

DO NOT PUT, in series or in parallel, into a telephone local loop:

- Batteries of any kind
- Polarized capacitors
- Diodes of any kind
- 1/4 watt resistors (or any other 10 cent resistor!)
- Lamps

With the exception of the lamps, all of the above are a safety hazard in addition to being very likely to make the phone line malfunction. In particular polarized capacitors (or any cap rated at less than 250 volts DC working volts), and batteries of any kind, should be avoided because of the potential for a an explosion. The other devices are merely a fire hazard. The resistors used in the real telephone circuits must have enough power handling capacity and be used so that they do not cause fire hazard (non-flammable resistors situated away from flammable materials).

The simple things are that the telephone line is a balanced transmission line which can have up to 120 ma of DC current from up to 56 VDC (actually in some cases up to 90 VDC) and up to 120 VAC RMS (ring voltage) in the way of various voltages and currents. Those voltages and currents can be any polarity and might be applied all at one time.

Telephone line parameters

Telephone line resistance, capacitance and inductance do not depend on the voltage or current on the line.

Line balance

For telephone local loops, crosstalk is related to how well balanced the circuit is. Loop current does not affect that balance, even if excessively high. If the balance is not gooe enought you can hear crosstalk form pther telephone lines or from other noise sources. The balance of the telephone line is determined by the circuits connected to telephone line ends (typically line transformers) and the quality of the telephone cable (wet cable can cause noticable balance problems if wires are in contact with the water).

Loop current effects

The detrimental effects of excessive loop current would be distortion caused by saturation of transformers ("repeat coils" in the vernacular). Within the range of acceptable loop current (up to 120mA), no transformer used in a telephone equipment should become saturated. If an inferior transformer is used, or if loop current were significantly higher than 120mA, then distortion could be expected. Neither situation is common.

Network Interface in telephone

The telephone has a circuit called network interface (also called voice network or telephone hybrid) which connects the microphone and speaker to the telephone line. Network interface circuitry is designed so that it sends only the current changes the other telephone causes to the speaker. The current changes which the telephone's own microphone generates are not send to the speaker. All this is accomplished using quite ingenious transformer circuitry. The circuit does not work perfectly and you can still hear some of your own voice in the speaker (it could be done better nowadays but people who use the telephone prefer to hear some of their own voice back).

Simplified traditional network interface

Normal telephone consist of ringer, dialing circuit and voice circuit. A traditional telephone voice circuit consisted of hybrid transformer, speaker, carbon microphone and one resistor.

```
              Hookswitch
                 /            Transformer
Tip   ---------o/  o------------+
                                |
                                 ) ||
                                 ) ||  _____   _  /
                                 ) || (     |_| |/
                   ______________) || (      _| |   Speaker
                  |              ) || (_____| |_|\  (32-64 ohm)
                  |              ) ||             \
               Carbon            ) ||
               Microphone       |
                  |            | |
                  |            | | Resistor
                  |            |_|
                  |             |
Ring -------------+-------------+
```

The circuit is designed so that the impedance at audio frequencies looks like about 600 ohms. The audio impedance is controlled by the transformer characteristics, carbon microphone, speaker impedance and the resistor in series with the transformer.

The DC resistance consist of the transformer coil in series with the resistor and part of the coil in series with carbon microphone. The carbon microphone is put to the transformer so that the changes in the current flowing through it do not generate voltage to the secondary coil where the speaker is connected.

Modern telephone circuit are much more complicated because they typically include compensation for the attenuation caused by long subscriber lines. This compensation is done so that the audio levels are controlled according the current flowing through the telephone (longer line has more resistance so there is less current which you get form 48V source through it).

Why carbon microphone in telephones ?

Carbon mikes were the first microphones and consisted of a small button of carbon powder connected to a metal diaphragm. When sound flexed the diaphragm, the carbon grains changed their electrical resistance. When a voltage source is applied between the microphone wires a variable current is generated. This is how the first telephones were constructed, and many phones to this day still use the idea. Carbon microphones have poor frequency response and bad signal-to-noise ratios and they are only suitable for telephones and such communication applications.

Typical European Network

The following network circuit schematic was shown in BUILDING AND USING PHONE PATCHES by Julian Macassey:

```
  A  \
 o--o.\---------o----o----o-------|
     .          | 0.1|    |       |
     .          |  uF|    |       |
     .          |   ---   \       |
     .          |   ---   / 1k    |
     .          |    |    \       |
     .          |    |    |       |
     .          |    -----o----)|||
     .          |              )|||
 S1  .          |              )||o------o-----
HKSW .          |         200  )|||   VR |    |
     .       TX O          Ohms)|||   60 |    |
     .          |              )||(    -----  |
     .          |              )||(     ^ ^   O RX
     .          ---------------|||(    -----  |
     .                    50  )||(60    |    |
     .                    Ohms)||(Ohms  |    |
  B \.                         )||(------o-----
 o--o\-------------------------)||
```

Note: I have edited the schematic by replacing the component numbers with the component values listed in component list.

Simplified U.S. Standard "425B"

This circuit is put here to show an example of the electronics inside typical traditional telephone which uses hybrid transformer circuit. Modern telephones usually have special ICs to do the same things without the transformer. The circuit tries just to be an example what's on inside typical old telephone for those who want to know how telephone works. Building this circuit is not a good idea because the circuit diagrams does not have all component values and the circuit is optimized only for telephones (it is not good for anything else).

This circuit is taken from UNDERSTANDING TELEPHONES article by Julian Macassey. Component values may vary between manufacturers. The circuit is designed to operate with standard telephone speaker (RX) and carbon microphone (TX). Connections for Dials, Ringers etc. not shown to keep the picture a little bit clearer.

```
                          +-------------------+
                        ..|...................|
                        . |                  .|
      Sidetone balancing. |  0.047uF 250V    .|
      impedance & loop  . |    | |           .|
      compensation. >>> . o----| |-------o   .|
                        . |    | |       |   .|
                        . |              |   .|
                        . |    |<| VR2   |   .|
                        . o----| |-------o---.|
                        . |    |>|          |.|
                        . |                 |.|
                        . |   68 Ohms       |.|
                        . o---\/\/\/-----|  |.|
                        ..|..............|..|.|
                          |              |  | |
                          |        .     |  | |
                          +----)||(------|---------o (GN) --+
                              1)||(5     |  | |             |
                Loop           )||(      |  | |             |
      TIP       Compensation  2)||(6     |  | |             |
   L1 o------ \------o---------)||(------o  | |          RX O  Speaker
             .       | (RR)   . ||       |  | |             |
             .       |          || 1.5uF |  | |             |
             .       \ 180      ||      --- | |             |
             .       / Ohms     ||      --- | +----o (R) ---+
             .       \          || 250V  |  |               |
             .       |          ||       |  |               |
             .  VR1 ---       . || .     |  |               |
             .      ^ ^    ----)||(------o--+            TX O  Microphone
             .      ---    |  3)||(7                        |
             .       |     |   )||(                         |
      RING   .       | (C) |  4)||(8       22 Ohms          |
   L2 o----- \-------o---------)||(---o----/\/\/---o (B) ---+
                           |          |
             ^             |          |
         Hookswitch        +----------+
```

The circuit is quite complicated because it is optimized for use in standard telephone which is used in various conditions. Varistors VR1 and VR2 are used for loop compensation circuit which tries to keep the telephone volumes (incoming and outgoing) at suitable levels even if the local loop attenuation varies. This compensation can be done because longer local loop which has more attenuation has also more resistance, so less current passes through the telephone. If the loop is very short there is more current passing through the telephone and

the varistors cause more signal attenuation inside the telephone hybrid.

The hybrid circuits in telephone sets are deliberately mismatched, so that you can hear yourself in the earpiece when you speak. This is called "sidetone".

Details of wiring inside telephones used in USA

The following wiring info is from wiring.inside.phones document from TELECOM Digest Archive

This should apply to all WE phones and ITT phones that use the standard dial/ringer/network block/ handset configuration. Everything basically talks to the network block. The network block contains the ringer capacitor, the induction coil that handles the handset, and very little else save some spare screw terminals. The network block can function as a standard line load [it looks electrically like a phone] when a line is connected across RR and C (These are the inputs to the coil). The ringing capacitor is across A and K contacts.

Handset connections: Green and White are earpiece leads which connect to R and GN respectively. Black and Red are mike leads and they connect to B and R respectively.

Ringer: Connect the single winding in series with the A-K capacitor and this whole thing across the line.

Rotary dial: Blue and Green go to interruptor (net F and RR)

Touch-tone dial: Green is + line in and connects to net F. Black is + line out and connects to net RR. Org/Blk is - line in and connects to net C. Red/Grn is output common and connects to net R. Blue is signal output and connects to net B.

Hookswitch: You'll find many variants of this in different units; some configurations switch both sides of the line, some only one, some switch out the ringer when off-hook. One switch switches the connection between L2 and C. Another switch switches the connection between L1 and RR.

Line in: Green and Red connect to L1 and L2. Try one polarity; if the touchtone dial doesn't work, then flip them.

Telecom hybrid circuits for other equipments than telephones

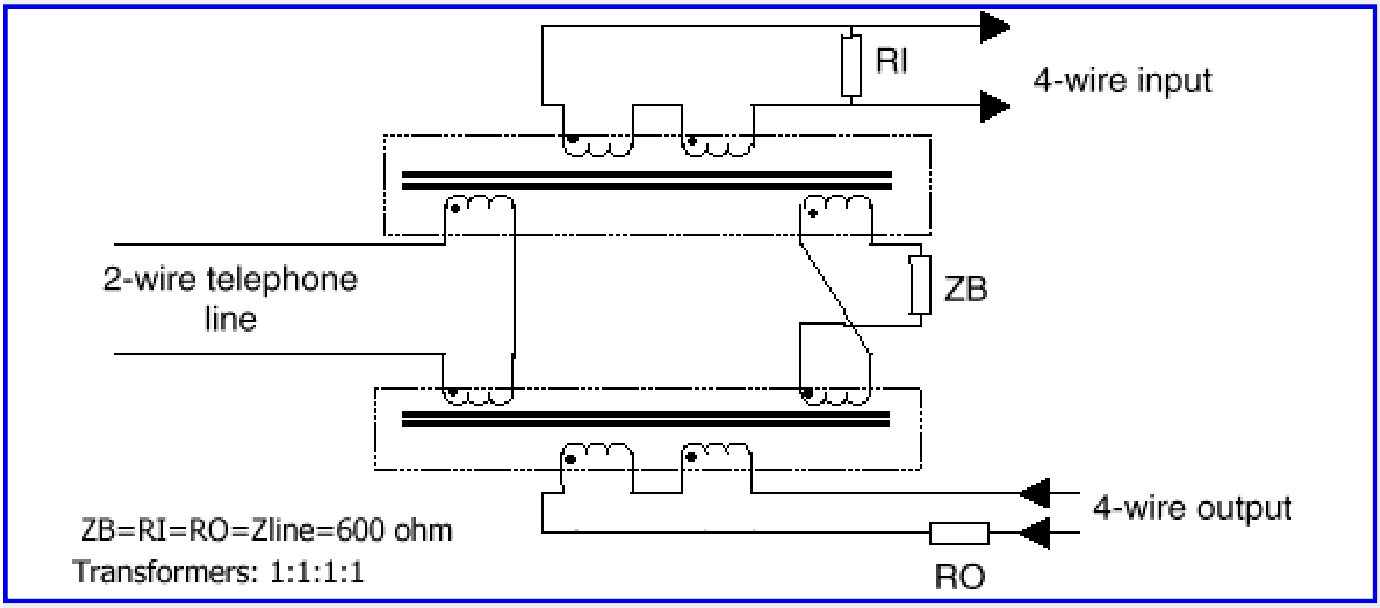
Traditional transformer hybrid circuit

The transformer type was the most used to make telephone hybrids (around 1964 or so) was four winding transformer. Two of those were needed for one hybrid circuit.

Richard Harrison gave me the follwong description how to make such hybrid circuit: To make the hybrid, strap two coils together in each transformer (series-aiding in each case). Call them primaries. One primary will serve as the 4-wire transmit connection. The other primary will serve as the 4-wire receive connection. Four coils, two on each transformer remain undedicated at this point. Connect the start terminal of a secondary coil on one transformer to the finish terminal of a like coil on the other transformer. The other two terminals of this pair of secondary coils will be dedicated to a balancing network.

Two coils now have no connections, yet. Connect the start terminal of the coil on one transformer to the start terminal of the coil on the other transformer. The other two terminals of this pair of secondary coils will be dedicated to the 2-wire line.

Because there is a polarity reversal in the interconnection in one of the two paths between the two transformers, no coupling will exist between the transmit and receive connections of the 4-wire paths (provided perfect balance in the line-balance network against the 2-wire line). The 2-wire line will,however, be coupled with the transmit and receive pairs of the 4-wire line. That is what the hybrid is supposed to do.



The advantages of the traditional circuit are, high isolation. No dc path exists between any lines. The circuit is completely passive and precision balance can produce almost any desired transhybrid loss. You should get very good results when you implement this circuit using high quality audio transformers (for example broadcast quality Western Electric 111-C "repeat coils", Lundahl Transformers Hybrid Transformers etc.).

Siemens and ITT resistive hybrids

This is a simplified circuit diagram you can made a simple 600 ohm hybrid as such. The circuit is indeed a Wheatstone bridge consisting of four 620 ohm impedances (one of them is telephone line in series with 2 uF DC blocking capacitor).

```
Phone line                                             Transmit      Receive

-----------------------------------|-------------------S1
line input/LA)                    620 ohm
                                   |-------------------------------R1
                                  620 ohm
                                   |-------------------S2
Line input/LB                     620 ohm
----------------------2uF---------|-------------------------------R2
```

Receiver connected to R1/R2 and transmitter is connected to S1/S2. Note that this circuit does not show any dc paths which would be needed for real telephone line hybrid. Loss on all ports are 6dB nominal. This hybrid design can be used for simple experimenting when measuring telephone equipments and such applications.

Better hybrids with two transformers have a typical loss of 3,5dB and 30dB isolation from TX to RX (but typically little isolation from RX to TX but that does not typically matter).

One transformer hybrid circuit for 2 wire to 4 wire conversion

This is a quite typical 4 wire to 2 wire conversion circuit which is shown in telecom books.

```
LINE 2W ---------------------/ II /------------- RX 4W
                             / II /
                             / II /
                             / II /
                             / II /
         TX 4W --------------/ II /            Receive
         Transmit            / II /
         TX 4W --+           / II /
                 |           / II /
                 |           / II /
LINE 2W ----------+-ZZZZ------/ II /------------- RX 4W
```

Here the wires marked with LINE 2W are the wires of the 2 wire duplex line. Wires marked with RX and TX belong to 4 wire line. RX is the pair where the received audio form 2 wire line comes. TX is the pair where the audio which is to be transmitted to 2 wire line are sent. The component market with ZZZZ models the telephone line impedance (typically around 600 ohms).

Notes about telephone transformers

Telephone line inmterfacing transformers are usually called 600:600 ohm transformers (or 1:1 ratio 600 ohm transfromers). The both markings tell that the transformer has (around) same number of turns on both primary and secondary coils and they are optimized to operate at 600 ohm load. The 600 ohm load does not tell the primary or secondary coil resistances or impedance, it just tells in what kind of application the transformer is designed to be used. The DC resistance of typical telephone line transformer coils is around in 40-150 ohm range and inductance is typically in range of few henries.

A 600:600 transformer is optimised for 600 ohms use, but of course will work over a range of impedances more or less well (for example you lose a whole octave at the low frequency end if the impedance is 1200 ohms).

Telephone line interfacing transformers are available in two major types "wet" and "dry". Typical modern transformers are "dry" type, because they perform well and anre small, but can'twithstand the line DC current going though them without saturation. "Dry" transformer can be used in application where line current is blocked not to go though the line transformer and if some current must be taken from the line, an alternate path is provided for it.

"Wet" transformers are are designed so that they can withstand the DC current present on the telephone line flowing though their primary without transformer saturation. Ther drawback of "wet" transformers are that they are typically bigger and have worse performance figures than "dry" transformers. "Wet" transformers are traditionally used in telephoine circuit, but nowadays they are more and more often relaced with "dry" transformers for economical and technical reasons (a high speed modem would not work well if it would use typical "wet" telephone transformer). The "wet" transformers have typically the maximum allowed direct current listed on their datasheet (more current than that will saturate the transformer core).

Telephone line transformers provide also isolation from the telephone line. Normally the voltages on telephone line are in order of 100V, but in some special cases there can be higher voltages present on the telephone line or between the equipment and telephone linne, so the transformer must withstand quite high voltages to be safe in such circuimtances. Typical telephone transformers are rated to have the isolation rating of around 1000-4000V range (look for the transformer datasheet and your local safety regulations to selector a type which has high enough isolation voltage rating).

Audio interfaces to telephone lines

The following two interfaces were designed for connecting small tape recording to telephone line for recording telephone conversations. The interfaces are connected in parallel with the telephone and they can be kept connected to telephone line all the time because they only pass the audio signal, not the DC which is used in telephone system to detect when phone is picked up. The original Philips and Norelco interfaces are type approved for use in Finland for connecting telephone line to tape recorder.

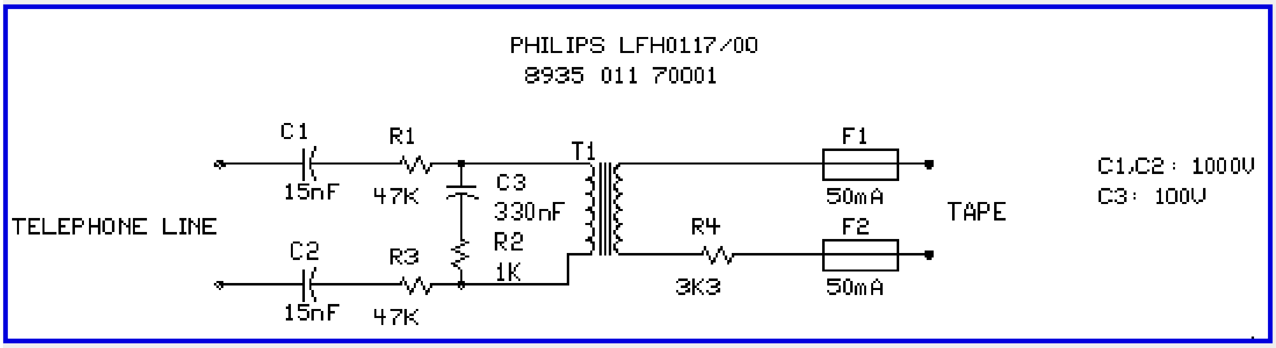
Those two circuits are useful because they don't pick up the line when they are inserted to telephone line and they pass the audio even when the line is not picked up. That makes then useful fo connecting computer sound cards and caller ID circuit to telephone line. If you worry about spikes coming from telephone line your circuits, you can use pair of diodes or zener diodes to limit the spikes on the transformer secondary.

If you can find this kind of interfacing circuit approved and ready-made, it would be safe to use ready-made interface. In this you don't have to worry about getting into problem from connecting non-approved circuits to telephone line. If you happen to live in USA you can use Xecom XE0068 Data Access Arrangement module which provides legal and quite low cost interface to the phone system with FCC Part 68 registration (the registration transfers to final product which uses this module). Xecom makes also similar DAA modules to meet the regulations in use in other countries also. CP Clare advertises quite actively it's CYBERGATE Telephone Line Interface DAA Modules in their web pages and some electronics design magazines.

Philips recording interface

The first circuit is Philips LFH0117/00 telephone recording adapter, which is not manufactured anymore I think. The circuit is quite typical telephone recording adapter design. I have used succesfully for getting audio from telephone line to soundcard and my stereo system.

The circuit is designed to be connected in parallel with normal telephone. This circuit does not provide any DC path or correct impedance matching to telephone line because the telephone provides them (this circuit is designed so that it disturbs the operation of telephone as little as possible so it has high impedance input).

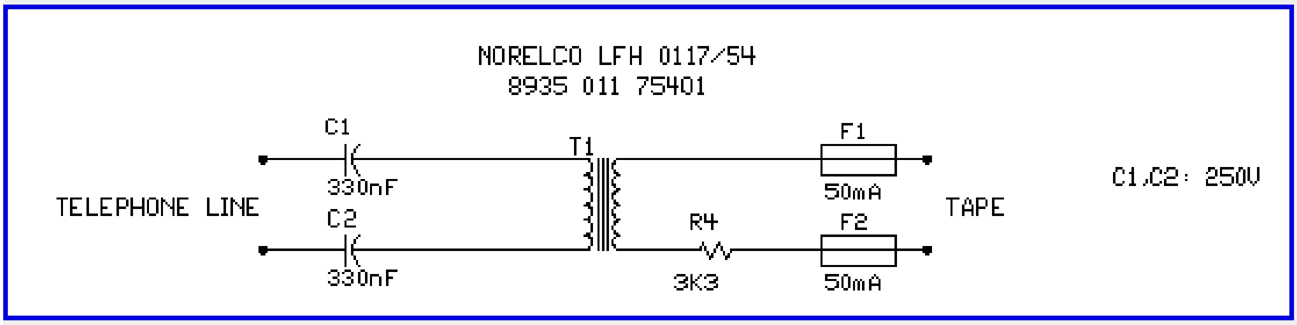


I found out the type of all other components than the transformer T1. T1 is the audio isolation transformer, which seems to have properties quite similar to typical 600:600 ohm telecom isolation transformer. The components F1 and F2 are 50mA fuses. The signal output level is suitable for microphone input because the resistors attenuate the voice signals form telephone line around 40 dB (some telephone equipment regulations in Finland needed this). On the picture below you can see a picture of the inside of the Philips LFH0117/00 telephone recording adapter:

In the circuit the two 15nF capacitors are blocking the line DC level and the low frequency for ringing. All other compnents are safety requirements, for lowering the noise and for matching the telephone equipment regulations (signal isolation from line, impedances etc.). This schematic is basically a very safe one: fuses are not necessary for proper function (just for extra safety) and the transformer provides galvanic isolation from telephone line. The circuit is designed to be connected to the microphone input of a recorder (the output signal level is typically few millivolts which is too low for any other type of input).

Norelco recording interface

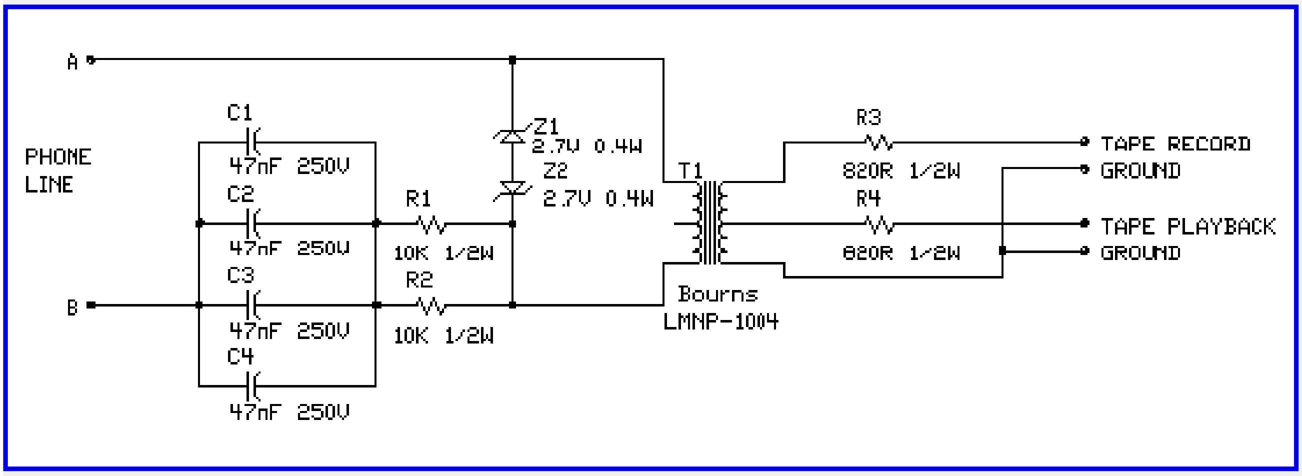
The second circuit is made by Norelco and is also designed to be used in parallel with existing telephone. This circuit has much less attenuation between telephone line and tape plug so you get stronger output signal out. I have used this circuit successfully for getting some audio from telephone line and sending some audio back to telephone line.



I found out the type of all other components than the transformer T1. T1 is the audio isolation transformer, which seems to have properties quite similar to typical 600:600 ohm telecommunications isolation transformer. The components F1 and F2 are 50mA fuses.

Recording interface from Tekniikan Maailma magazine

The third circuit was shown in an article written by Martti Koskinen in Tekniikan Maailma magazine issue 8/1994 pages 94-95. The circuit is designed for recording telephone conversations using normal tape recorder. The circuit has an option to also play back sound tot the telephone line from reparate connector. The transformer T1 is typical 600:600 ohm telephone isolation transformer with centre tapped secondary. This circuit is also designed to be used in parallel wiht existing telephone.



The capacitors C1 to C4 are connected in parallel to make about 200 nF capacitance. Four separate capacitors can be more easily fitted to one case than single 200 nF 250V capacitor which is quite large. I don't know the reason of why R1 and R2 are connected in parallel, because single 4.7 kohm resistor would do their job as well.

Marantz PMD recorder series

```
                    ||
   Tip   ---+-------||----+--------+
            |       ||    |        |      +----+----------+----- Audio ut
            |     470 nF  |        |  ||  |   _|_         |
            |             |         ) || (     \          |
            |             |         ) || (    /_\ zener   |
           | |           ---        ) || (     |         | |
           | | 470 ohm   --- 47 nF  ) || (    \ /        | | 4k7
           |_|  1/2W      |         ) || (    _\_ zener  |_|
            |             |        |  ||  |    |          |
            |             |        |      +----+----------+------ Audio ground
   Ring ----+-------------+--------+
```

Tip and ring are first shunted with a 470 ohm 1/2 W resistor (to allow the interface to sieze the line). Next, before the transformer primary, there is a 470nF series cap (high pass, and DC transformer isolation) and a 47nF shunt cap (low pass to limit the upper end). The transformer is a 1:1 600 ohm, I believe, though the schematic doesn't specify.

On the secondary there is first a shunt pair of back to back Zeners to limit the max voltage seen by the the recorder circuit, and then a shunt 4.7K ohm to ground.

Telephone audio interfaces from Usenet news

There have been presented many telephone interfacing circuits in Usenet newsgroups. Those circuit have been more or less good. Most of them works somehow, but fail to meet the technical specs needed from telephone circuits. More or less they have usually been very simple circuits

Simple audio interface

Jim Earl (jre@earldom.UUCP) has shown out very simple circuit for connecting telephone signal to SSI202 DTMF decoding chip audio input. The circuit is basic 600:600 ohm transformer isolation circuit with capacitor in series with primary circuit and potentiometer for setting the output signal level. The circuit us designed to be used in parallel with existing telephone or other telecom equipments.

```
                    .22 uf              10k pot
                     400v    ||(----------->
Phone line tip  o-----)(----)||(           <---o to SSI202 input
                            )||(           >
Phone line ring o-----------)||(-----------o---o ground
```

This circuit works as designed with the chip specified, because it has high impedance input. The design has one problem: If this circuit is connected to low impedance input and the output potentiometer is set to maximum level, the low impedance is reflected to primary side of the transformer, which is not good.

My recommendation is to change the capacitor size to to 0.1 uF and add one 4.7 kohm resistor in series with circuit input or output. That will keep the circuit from disturbing the telephone line when connected to other circuits.

```
                    .22 uf              10k pot
                     400v    ||(----------->
Phone line tip  o-----)(----)||(           <---o to SSI202 input
                            )||(           >
Phone line ring o---\/\/\/--)||(-----------o---o ground
                     4k7
```

Telephone to studio mixer interface

The second circuit from news is designed by tpappas@hamp.hampshire.edu. The circuit seems to be quite nice desing and should work nice with mixers which have imput impedance of 600 ohm (mic input) or higher. The transformer and 44 nF capacitor keeps the impedance seen from line high enough that not bad mismatching happens when connected to studio mixer. If I would be connecting something like this to my audio gear I would add some type of surge protection to the circuit (two zener diodes in output would be nice) or add external surge protector. But let's the original text to describe the circuit in more detail.

We use telephone audio in our studio all the time. And yes, it's an off the shelf design. I designed and built such a device with scrap door components. I used an audio coupling transformer and a capacitor. The primary windings add in series to 500 ohms. Instead of connecting them directly together I added a cap between them. I it was something like 0.047 micro farads with a 600 volt rating. And the secondary which is 500 ohms runs into the control room mixer.

```
           Tip   >------------/ II
                              / II /-----------<
      (primary winding 1)     / II /
                              / II /
                 >-----X------/ II /
                       I        II /
         0.047 uF      =        II / -----------CT   (secondary winding)
                       I        II /
                >------X------/ II /                      Output Side
                              / II /                        to Mixer
      (primary winding 2)     / II /
                              / II /-------------<
        Ring    >-------------/ II
```

Try this circuit it works great for us in the studio. The circuit is designed to be used in paprallel with existing

telephone. Just make sure you use properly rated components.

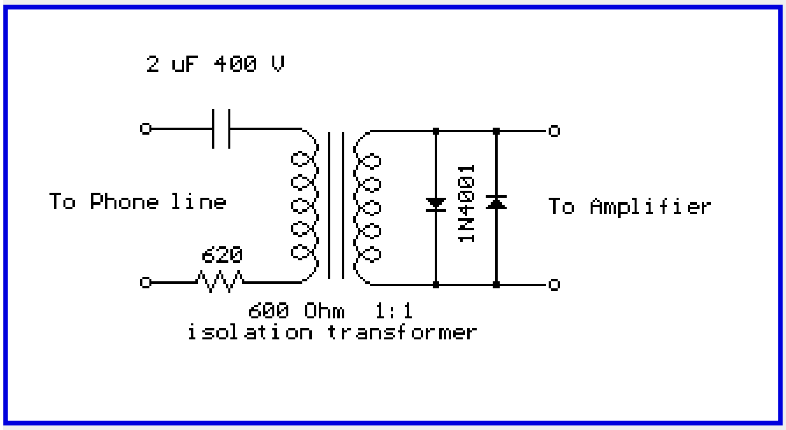
Other transformer isolated interfaces

I saw the following circuit idea [Bowden's Hobby Circuits] site and make my own modification of the circuit.

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 plus the on-hook voltage (typically 138V total), so a 400V capacitor 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 620 ohm resistor serves to reduce loading of the line if the output is connected to a very low impedance.

For overvoltage protection, two diodes are connected across the transformer secondary to limit the audio signal to 700 millivolts peak during the ringing signal.



Audio interfaces without transformer isolation

In some special case the audio interface is built without isolation transformers. In those cases the audio signal is passed from telephone line through the capacitor which blocks the DC from telephone line. This type of isolation works quite well in applications where you don't want to use the transformer but you still want to get some audio from the line. Typical application is called ID boxes.

A typical capacitor isolation circuit:

```
              C1  R1
Line ---------||-/\/\--o------O
                       |
                       \
                    R3 /       Audio out
                       \
              C2  R2   |
Line ---------||-/\/\--o------O
```

The capacitors C1 and C2 will block the DC and pass the audio signal to the output. The resistors R1 and R2 provide some protection against the spikes on the telephone line and make sure that the circuit is so high impedance that it does not disturb the telephone line operation. R1, R2 and R3 make together a voltage

division network which will attenuate the audio signal coming from telephone line to the desired signal output level.

The circuit should be connected to differential audio input. If the circuit is connected to single-ended input the circuit works worse and gets easily all kinds of interference. Capacitors C1 and C2 should be rated to handle the 1.5 kV pulses. The capacitors C1, C2, R1 and R2 should provide so high impedance to the telephone line that the telephone line balancing is not disturbed. You should also note that this circuit does not provide as good surge protection as transformer (surges can quite easily pass trough C1, C2, R1 and R2). This is not the preferred way to do the telephone line interface ! Preferred way is to use transformer isolation instead.

Example circuits

DTMF decoder schematic shown at http://www.ee.washington.edu/conselec/A95/projects/jjblome/links.htm used the following component values:

```
C1,C2     470 pF
R1,R2     10 kohm
R3        Not used
```

The output of the circuit was directly connected to the MC145447 IC differential audio input. The circuit isself was designed to be fitted inside an isolating plastic enclosure.

Caller ID detector schematic shown at http://www.helsinki.fi/~metsala/cid.txt used the following values:

```
C1, C2    10 nF
R1, R2    100 kohm
R3        (around 40 kohm)
```

The output of this circuit was directly fed to the differential input of MT8870C-1 DTMF decoder IC. The decoder IC was connected to the computer. In this applications is essential that the capacitors C1 and C2 can withstand the 1.5 kV surges which can be sometimes present in telephone line.

Simple telecom hybrid circuits

Telephone hybrid circuit is the circuit which is designed for converting 2-wire interface to 4-wire interface and is one of the basic building blocks of the telephone system. Telephone hybrid is the circuit which separates the transmitted and received audio which are sent both at the same wire pair in 2-wire normal telephone interface.

There are many different types of hybrid circuit in use. Traditionally telephones have used combination of special transformer and few additional components to keep incoming and outgoing signal separated from each other. Nowadays this is done more or less electronically.

In telephone central end hybrid circuits are needed when must be done any amplification to the signal. Traditionally the systems separate the incoming and outgoing signal, then they are amplified separately and sent to other telephone central using separate wires or otherwise separate communication channels. The oldest models of those circuits have been built from one or two transformers and some other balancing components to get best results. The problem have been how to get good balance to the hybrid circuit, said in other way how to separate incoming and outgoing signals as well as possible. Nowadays everybody is avoiding bulky and expensive special transformers and more and more electronics is used because it is cheaper. Modern hybrid circuits consists only of one audio isolation transformer, two operational amplifiers, resistors and some capacitors. and the most modern approaches try to avoid that transformer altogether by using active electronics circuits in telephone line side to do the job and optocouplers to do the isolation where needed.

Many different system circuit have been used and I am showing here just one basic transformer based circuit which easy to understand and is useful for many experiments.

```
       Primary 600 ohm                  Secondary 600ohm+600 ohm

           Tip   >------------/ II /------------< audio to telephone line
                              / II /             (low impedance output)
                              / II /
                              / II /
                              / II /
                              / II /------600ohm--Ground
                              / II /
                              / II /
                              / II /
                              / II /
         Ring   >-------------/ II /-------------> audio to mixer
                                                   (high impedance input)
```

This first circuit is a traditional simple hybrid circuit which have been earlier successfully used in many telephone circuit (for example modems). The circuit works so that the 600 ohm resistor in the center pin of the secondary is seen as 600 ohm impedance load in primary circuit. The end of the secondary which is connected to low impedance audio output (for example amplifier made for driving small speaker) must be always connected to amplifier or ground to make the circuit work as expected. The audio signal output from the circuit must be fed to high impedance (>10 kohm) audio input to make sure that the operation of the circuit is not disturbed.

The circuit gives quite acceptable separation between incoming and outgoing signals when all impedances are set correctly. The 600 ohm impedance is kind of idealistic value and does not fully reflect the reality. In real life the impedance of the telephone line or telephone is not exactly 600 ohm and the transformer has it's losses. A 600 ohm resistor is anyway quite a good starting point.

If transmitted and received signals mix with each other, you will have to fiddle with the balancing network. For experiments I can suggest fitting 1 kohm variable resistor to the pace of 600 ohm resistor for experimenting which impedance value gives the best results. You may also want to try other type of line impedance simulation circuits if you know what what matches your system better. If the impedances presented by both the send and receive sides are the same the hybrid circuit will work quite well. You will find that the send and receive signals don't interfere with each other, but both come and go from and to the line.

If you are thinking of connecting this circuit to telephone line or otherwise sending DC current through the primary of the transformer remember to use a transformer which can handle the DC without saturating (telephone transformers made for "wet" circuits). And remember that there are strict rules what the equipment you connect to telephone line must meet and you are not allowed to connect anything not approved to public telephone system.

Modified circuit

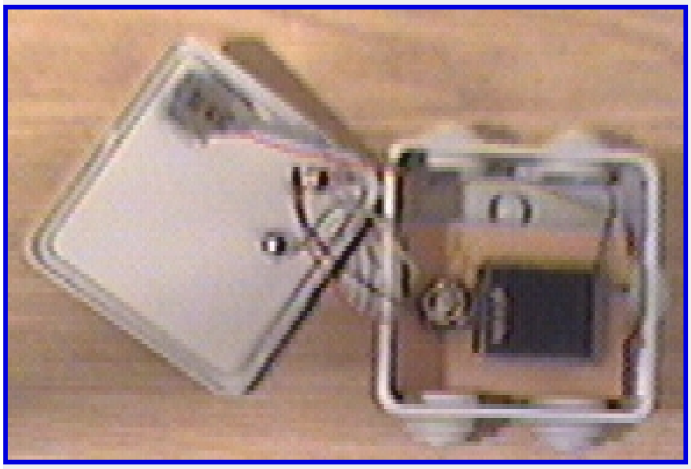
The following circuit is for telephone line interfacing when using a 600 ohm to 600 ohm transformer with center tapped output:

```
       Primary 600 ohm                  Secondary 600ohm centre-tapped
                                        (same as 150ohm+150ohm secondary)
```

```
          Tip   >------------/ II /------------< audio to telephone line
                             / II /              (low impedance output)
                             / II /
                             / II /
                             / II /
                             / II /------150ohm--Ground
                             / II /
                             / II /
                             / II /
                             / II /
        Ring   >-------------/ II /-------------> audio to mixer
                                                  (high impedance input)
```

This circuit works the same as that circuit above, but uses standard telephone line transformers easily available. One common transformer type has 600 ohm primary and 600 ohm centre-tapped secondary. In centre-tapped secondary each secondary side presents 150 ohm impedance. By using 150 ohm resistor connected to secondary centre pin, the primary sees 600 ohm impedance. Experimenters can try for example 470 ohm variable resistor instead of 150 ohm fixed resistor to test which value gives the best results.

You can see this type of circuit built to a small plastic box at the picture below:



Soundcard to telephone line interface

The following circuit a a modifed version of the circuit above. This circuit includes suitable audio output which can be fed to a PC soundcard and a on/off hook switch:

```
      Primary 600 ohm                Secondary 600ohm centre-tapped
                                     (same as 150ohm+150ohm secondary)
               /
     TIP ----o/ o------------/ II /------------< soundcard speker output
           ON/OFF HOOK       / II /
            SWITCH           / II /
                             / II /         +-< speker connector ground
                             / II /         |
                             / II /---150ohm-+
                             / II /         |
                             / II /         |
                             / II /         +-> line input connector ground
```

```
                              / II /
     RING --------------------/ II /------------> soundcard line level input
```

Remember that in telephony applications the signals levels must be adjusted carefully and sure must be made that the circuit is in good enough balance that there is no annoying feedback in the whole system. Warning that this ciruict is a little bit simplified interface diagram. This diagram lacks for example overvoltage protection on the audio output and also limiting circuits which stop too large signal levels to enter the telephone network.

Hybrid circuits for interfacing telephone equipments and not incoming line

Sometimes is is useful to connect telephone or a modem to hybrid circuit instead of connecting hybrid circuit to telephone network. This can be easily done by connecting the telephone/modem in series with the primary of the hybrid circuit and then supplying well filtered +12V power to that circuit to give power to the telephone in the circuit and it is a good idea to feed the operating current to modem also, because they are usually designed to operate when then line current is present. The picture below tries to clear out the connection:

```
       Primary 600 ohm                Secondary 600ohm centre-tapped
                                      (same as 150ohm+150ohm secondary)
              telephone
      +12V-------or-----------/ II /------------< audio to telephone line
                modem         / II /             (low impedance output)
                              / II /
                              / II /
                              / II /
                              / II /------150ohm--Ground
                              / II /
                              / II /
                              / II /
                              / II /
    GROUND--------------------/ II /-------------> audio to mixer
                                                   (high impedance input)
```

Telephone or modem and hybrid circuit provide the 600 ohm termination for each other to operate correctly. The transformer used in hybrid circuit must be a type which can handle at least 40 mA DC current without saturation. It is a good idea to use 470 ohm trimmer in place of 150 ohm resistor especially if you are trying to get the best performance with ordinary telephone. In other way the circuits work in the same way as the hybrid circuits above.

The possible uses for this circuits might be converting normal 2 wire modem for operating in 4 wire circuit (for example for connecting to radio link) or using normal telehone as microphone and speaker for computer soundcard when doing Internet telephony. If you are planning to connect the circuit to your soundcard use the following wiring:

```
       Primary 600 ohm                Secondary 600ohm centre-tapped
                                      (same as 150ohm+150ohm secondary)
              telephone
    +9-12V-------or-----------/ II /------------< soundcard speker output
                modem         / II /
                              / II /
                              / II /         +-< speker connector ground
                              / II /         |
                              / II /---150ohm-+
```

```
                              / II /          |
                              / II /          |
                              / II /          +-> line input connector ground
                              / II /
    GROUND--------------------/ II /------------> soundcard line level input
```

Remember that in telephony applications the signals levels must be adjusted carefully and sure must be made that the circuit is in good enough balance that there is no annoying feedback in the whole system.

Operational amplifier based hybrid circuits

Modern modems use hybrid circuits built from operational amplifiers, resistors and one 600:600 ohm isolation transformer. With operational amplifier circuit the circuit can be made cheaper and performing better.

```
         Primary 600 ohm        Secondary 600ohm



                                    ____
    LINE --------------/ II /--+--|____|------+---< Audio to telephone line
                       / II /  |  600 ohm     |
                       / II /  |             | |
                       / II /  |             | | 600 ohm
                       / II /  |              |
                       / II /  +--> diff <----+
                       / II /  opamp circuit  |
                       / II /   for received | |
                       / II /      audio     | | 600 ohm *
                       / II /                 |
    LINE --------------/ II /-----------------+---- Ground
```

The source for audio signal which is transmitted to the telephone line should be low impedance to ensure that the impedance matching to telephone line is correct. For receiving audio a differential amplifier must be used to separate the incoming signal form outgoing signal, but differential amplifier is very easy to implement using operational amplifiers.

The performance of the circuit can be made better by replacing the 600 ohm resistor which is marked by * with some better model for the telephone line seen through the isolation amplifier. A better model provides better isolation between incoming and outgoing audio signals.

A quick note to mixing desk users: professional mixing desks nowadays have differential inputs and low impedance outputs. This makes it very easy to experiment with this type of circuit if you happen to own a good audio mixer.

Here is the full operational amplifier based hybrid circuit diagram (theoretical circuit):

```
             |\  op amp buffer
   xmit------| >-------+-----------------+
             |/        |                 |
                       |                 |
                       \                 \
```

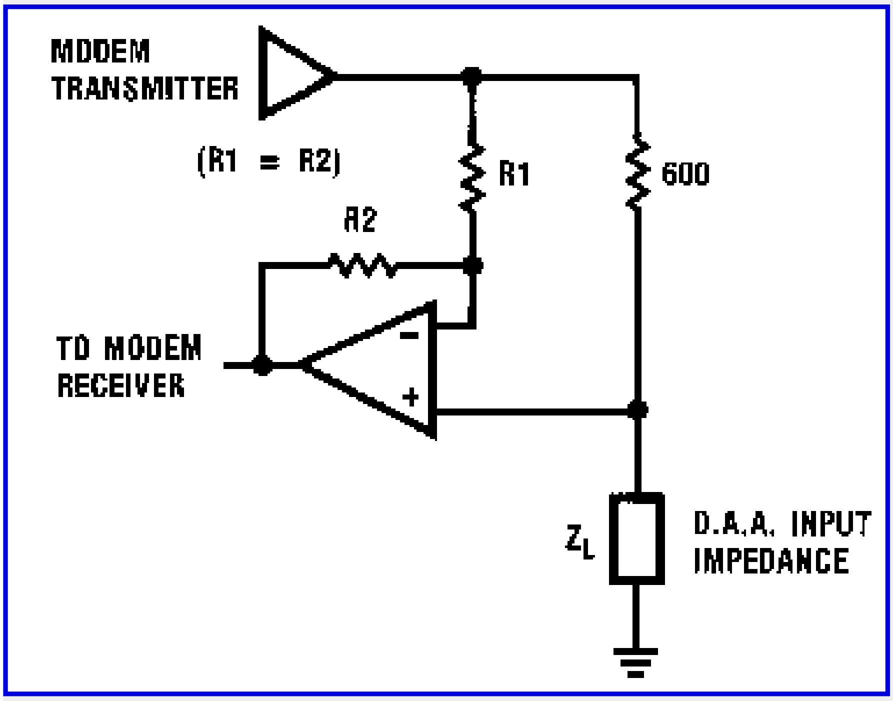
```
                       / R1               / R2
                       \                  \
                       /                  /
                 /|    |                  |
         op amp /-|---------------------+
   rcv-------- <  |    |                  |
                \+|----+                  |
                 \|    |                  | xfrmr
                       \            -->   )||(-------------- Tip
                       /             |    )||(                (to telco)
                       \ R3        (Rn)   )||(
                       /             |    )||(-------------- Ring
                       |            -->   |
                       |                  |
                      gnd                gnd
```

In the above diagram Rn = the telco network impedance as seen at the other side of the transformer.

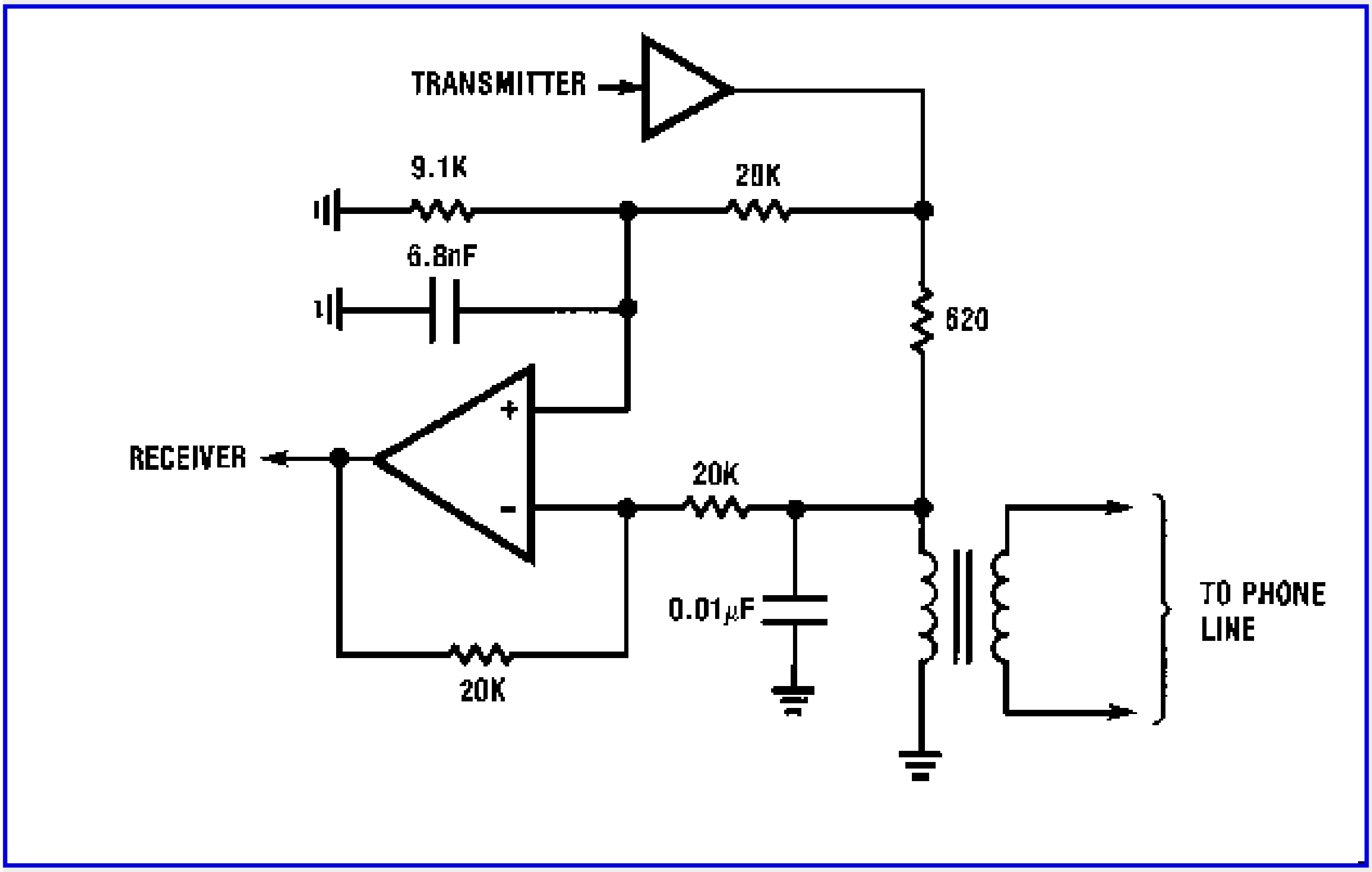
This circuit is an example of an "active" hybrid. Essentially it is a balanced network. If the ratio of Rl/R3 = R2/Rn, then you have infinite return loss - that is, you should have none of your transmit signal appearing on your receive line (when this happens, this signal is called side-tone). Yet the receive signal from the "far-end" will appear on the receive line. In other words, two signals can use the same two-wire interface, yet are seperable. Not theat the resistors which define the amplification of the opamps are not drawn here, so if you are plannign to build this circuit you will have to add them.

Unfortunately, telco line impedances can vary quite a bit, so the ratio of R2/Rn rarely equals R1/R3 except in situations where the designer has tight control over loop lengths and terminations. Any imbalance in the balanced network creates sidetone - a small amount of the transmit signal will appear on the receive line. In typical situations the sidetone can be attenuated aroun 20-30 dB with a well designed hybrid circuit.

Another typical way to implement a hybrid circuit is to build an optoamplifier circuit which takes the signal over the transformer coil and subtracts the transmitted signal from it. The following operational amplifier circuit does this:



The circut below is partly redrawn optimized hybrid circuit from National Semicondictor application note "Optimum Hybrid Design" from 1985 (that application note is no longer available).



The transformer in this circuit is 600:600 ohm telephone line transformer. For best results you have to adapt the component values slightly to match the line impedance and the transformer you are using.

That upper amplifier (the triangle with one input and output wire) is just a buffer amplifier with amplification factor of one. Signal from transmitter is connected to the positive input of opamp. The negative input of that opamp is conected to the opamp output.

More details implementing telephone line interface

Telephone line interface has to provide two functions when it is off-hook:

- Provide DC path for current flowing in telephone line. Normally there flows about 20-50 mA current in telephone line and telephone regulations typically sepcify that the DC resistance must be less than 400 ohms.
- Provide proper termination for telephone audio frequencies (300-3400 Hz). This is typically specified to be 600 ohms.

"Wet" transformer

Traditionally those two functions are accomplished in modems and their telephone interfaces are accomplished by "wet" telephone transformer. Wet type means that the transformer is designed to handle the DC current (typically 20-50 mA) properly and does not saturate at this DC current. Typically "wet" transformers are more expensive, bigger and have worse specs than "dry" transformers (which do not have to withstands any DC current). The proper termination in modems is provided by the electronic hybrid circuit connected to of the transformer. Another possibility is to use transformer which with center tap and build simple transformer and resistor hybrid circuit around it.

Here is a typical circuit for "wet transformer:

```
         Hookswitch
             /
   Tip   ---o/ o--------------------+
                                    |      +------->
                                    |  ||  |
                                     ) || (
                                     ) || (
                                     ) || (      To hybrid circuit which
                                     ) || (      prodived 600 ohm termination
                                     ) || (
                                    |  ||  |
                                    |      +------->
   Ring ---------------------------+
```

The circuit operation is quite straightforward. when the hookswitch is closed the telephone line DC current starts to flow trough the transformer primary coil. The DC resistance of the circuit is determined by the resistance of the transformer primary coil (typically in 60-200 ohm range in 600:600 ohm telecommunication transformers). The transformer is 1:1 transformer designed to operate at 600 ohm impedances, so 600 ohm termination provided in the secondary reflects as 600 ohm to primary (this in not totally accurate because transformer has some losses so you need a little smaller than 600 ohm resistance in secondary so that it looks 600 ohms in primary).

"Dry" transformers

Dry transformers are transformers which are not designed to handle DC current flowing through them (if you put DC through them they saturate and do not work correctly as transformers). 600:600 ohm dry transformers are very useful for example in modems because they are available in small sizes (even so small that can be fitted inside PCMCIA modem card) and can have very good performance figures.

Because the dry transformer can't stand DC then in telephone application where there is DC present the DC must be blocked by suitable capacitor (usually 2..10 uF) and alternate path for DC must be provided. Here is an example circuit:

```
         Hookswitch      Capacitor
             /               ||
   Tip   ---o/ o-----+--------||---+
                     |        ||   |      +------->
                     |             |  ||  |
                     |              ) || (
                    DC              ) || (
                   PATH             ) || (      To hybrid circuit which
                     |              ) || (      prodived 600 ohm termination
                     |              ) || (
                     |             |  ||  |
                     |             |      +------->
   Ring -------------+-------------+
```

The DC path must be designed so that it will pass DC well but provides high impedance to telephone audio frequencies (so that it does not disturb the impedance matching done elsewhere). A large inductance coil can be used in this but it is not practical because you wanted to get rid of that bulky "wet" transformer using small "dry" transformer instead, so you don't want an expensive and bulky coil in your circuit.

Fortunately coils can be simulated electronically using gyrator circuit. With gyrator it is very easy to have a simulated coil which has low DC resistance and the circuit looks like high inductance coil (few Henries simulated coil can be made easily). Another possibility is to use constant current sinking circuitry. Constant current circuit provides path to DC current but has very high impedance (before using constant current circuitry take a look if your telephone regulations allow constant current operation or you can make the circuit to work inside the specs in varying line conditions).

When you add electronics to transformer primary side remember that those must work at both line polarities. A bridge rectifier will help to make sure that the current going to DC path circuit is always at correct polarity. Another thing to consider is overvoltage protection because your circuit in the transformer primary side has to withstand the spikes which exist in telephone lines primary side. Make also the circuit so that it is not damaged by a little more current than normally present in telephone line (sometimes there are overcurrent situations and you don't want your circuit to break down too easily).

Secondary overvoltage protection

In telephone circuits there are situations where there is some high voltage spikes on the line. If such overvoltage get through the transformer it can destory the electronics connected to the transformer secondary unless there is some overvoltage protection. Even a normal telephone ring signal going through the transformer can cause harmful voltages on transformer secondary.

Fortunatly the protection on the transformer secondary is usually quite easy, because the transformer itself redices the energy which can pass through it. Typically a pair of zener diodes (voltage of few volts) connected to the transformer secondary can do the protection nicely.

```
   -+
    |      +----+----------- Audio in/
    |  ||  |   _|_
     ) || (     \
     ) || (    /_\ zener
     ) || (     |
     ) || (    \ /
     ) || (    _\_ zener
    |  ||  |    |
    |      +----+----------- Audio ground
   -+
```

The zener diodes will limit the signals on the transformer output to around the zener diode voltage + 0.7 V range.

In audio output circuits where the signal levels are fraction of volt range you can use normal diodes (like 1N4148) in the following way to do the protection:

```
   -+
    |      +----------+----- Audio out
    |  ||  |          |
     ) || (     Diode |
     ) || (     +-|>|-+
```

```
     ) || (     |     |
     ) || (     +-|<|-+
     ) || (     | Diode
    |  ||  |    |
    |      +----+----------- Audio ground
   -+
```

The diodes on the secondary of the transformer connected in this way will limit the signal levels to less than 0.7 Vpp.

Problems in linking telephone hybrid to audio system

Every sound engineer has had to deal with telephone lines at one time or another. Linking the phone conversation to audio system like taking calls to radio studio can be more problematic than you first thought. You can get a good view of the scenario at article Phone Line Basics article from JK Audio.

The main problem in making the audio connection is that the telephone line is full-duplex interface implemented using single twisted pair. When an announcer speaks, his/her voice travels through the phone line output of the phone patch (transformer, analog hybrid, digital hybrid, etc.), to the caller, and back to the studio into the telephone line input of the phone patch. (You can hear this leakage in the earpiece of your telephone handset. Just listen to how much of your own voice comes back to you!)

Level matching on local and remote voice

Typical commercial telephone hybrid allows the equalizing of levels of local and remote voices. Typically a hybrid needs adjustment for every new connection because of impedance changes. Today automatic digital hybrids are used for equalizing local and remote telephone conversations.

Trans-hybrid loss and announcer voice distortion

Trans-hybrid loss is that portion of the announcer's voice that leaks through the hybrid to its audio output. The higher this spec, in db, the better isolation in the device. This leakage is distorted and phase shifted after its long journey. In the studio, the announcer audio is mixed at the console with the phone patch (caller) output to create the on-air mix. When you use a poor phone patch, its output includes a distorted, phase shifted version of the announcer signal. When this leakage is combined with the clean announcer audio, a "hollow" or "tinny" sound is produced as some frequencies are more affected by phase cancellation than others.

The greater the trans-hybrid loss, the less announcer audio that leaks into the hybrid output and the less the announcer voice distortion. Ideally, the output of the hybrid should consist of the caller audio only. Digital hybrids have signal processing electronics to get better trans-hybrid loss figures than which are available with simple analogue solutions. You have to decide what's best for your application and your budget. There are different requirement depending the application (broadcast, teleconference or remote training). For links to telephone hybrid circuit check Hardware for Audio/Video Conferencing at http://www.cs.columbia.edu/~hgs/rtp/hardware.html.

It hase been suggestions that ISDN be used as it is full duplex is a good one, but it might be only practical if both sides of the telco path have ISDN. When calling between plain old telephone service (POTS) and ISDN, the above problems remains.

Echo problem in long distance calls

Echo is caused because of the coupling between incoming and outgoing audio in the telephone circuit and the delay in the telephone line (especially in long distance calls). Echoed back audio is usually caused by an impedance mismatch at a 2/4-wire conversion point (such as a codec-annex-hybrid, analog CO line interface) and by acoustic feedback (feedback from spaker to microphone in handset, acoustic echo in hands-free phones). Thus there is echo; ISDN or other digital telephone set on an all-digital connection would not cause echo because of conversion mismatch, but if normal handset or hadfree telephone is used the acoustic echo is still possible.

Echo doesn't become audible until the delay in the circuit exceeds a certain threshold value which depends on the losses in the circuit. Even milliseconds of terrestrial echo can be annoying, but typically the echo is not annoying if the delay stays below 25ms. Old Bell standards said that on calls of more than 1800 miles, an echo suppressor was used. In general, you need echo cancellation when the delay exceeds some subjective value in the 30-50 ms range.

As it is practically impossible to prevent echo (by perfectly matching the impedance in line circuits and by acoustically insulating all phones), it either has to be suppressed or cancelled when it does occur. For this reason, echo cancellers are deployed by telephone company on long-haul routes that, when used, bring the total circuit delay to above the echo threshold value determined by line loss. These echo cancellers are deployed on both sides of such long-haul routes and the echo canceller at the remote end of the call is responsible for ensuring that you don't hear any echo.

For more information on how echo cancelling works, please consult ITU-T recommendation G.165 or some good telecommunication book. The morale is therefore that if you hear echo, you can't do practically anything about it, as both the cause of the problem and the solution to it lie at the remote end of the connection (typically at the telephone company equipments). If the connection you're talking about is across a private network, make sure that the echo cancellers are correctly dimensioned because wrongly dimensioned echo canceller will be totally ineffective.

Metallic sounding caller voice problem

If your telephone connection is though a digital PBX or digital switch (typical nowadays) then you might encounter a problem that the voice which might sound OK on telephone but sound "metallic" when you connect it to the mixing desk through your high quality hybrid circuit. The metallic sound problem is an aliasing problem cause by the digital telephone system where there is not much filtering after the D/A converter which outputs the sound. The absence of the output filters causes that there are high frequency noise components added to the output audio signal.

The audio sounds fine on normal telephone because it can only playback the normal telephone audio range. The problem is audible with your hybrid circuit of that circuit has wider bandwidth than normal telephone. The solution to make this signa sound normal telephone is to remove everything above 4 kHz by a sharp lowpass filter. You can try if your mixing desk channel equalizers are effective enough to remove this problem. When you start equalizing the signal from telephone hybrid then you can also remove the bass frequencies also (there is usable sound information below 200 Hz on normal telephone line) so you can also get rid of the possible low frequency noise (mains 50 Hz or 60 Hz) which is sometimes present on telephone line.

Helpful tips for telephone hybrid circuit designers

Transformer equivalent circuit

Transformer equivalent circuit is very useful tool when you need to analyse the circuit operation using mathematical methods. Transformers are usually modeled using "t" equivalent circuits.

Here is the "t" equivalent circuit for a 1:1 audio isolation transformer:

```
          ------R1---L1-----+----L2----R2------
 Primary                    |                   Secondary
 Side                       Lm                  Side
```

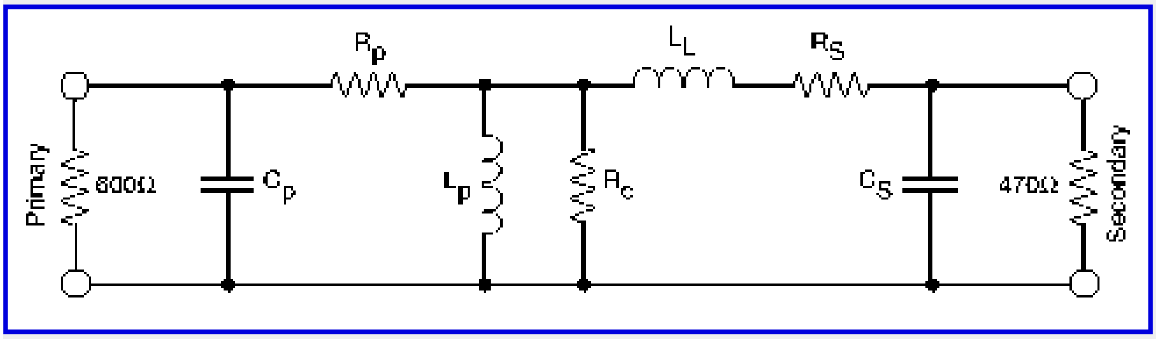
```
                         |
        -----------------+-----------------
```

- R1,R2 = primary and secondary winding (copper) resistance. Typically about 50-100 ohms. Not necessarily equal.
- L1,L2 = primary and secondary leakage inductances. About 5 mH in "dry" tranformers. Not necessarily equal.
- Lm = mutual inductance, about 2H. (about the same as the self inductance or shunt inductance)

This model can be applied to telecommunication because are typically 1:1 audio isolation transformers designed to operate at 600 ohm impedance. For the model above you can measure easily the primary and secondary copper resistance (coil DC resistance). If you have the transformer datasheet you can usually find the values of those all parameters used in this model.

This model models the transformer behavior in all except two things: isolation and possible core saturation. If your intuition needs to see the isolation in the circuit model you can thinks that you have an ideal 1:1 isolation transformer after this circuit. For core saturation in telecommunication applications you don't run the transformer core to saturation or near it so you don't have to model it.

If more accurate equivalent model is needed for the transformer, then you can use the following model for the transformer:



For Midcom 671-8005 tranformer the model has following parameters:

```
Cp = Primary Capacitance            150 pF
Rp = Primary D.C. Resistance        108 ohm
Lp = Primary Leakage Inductance     0.224 H
Rc = Core Loss Resistance           18 kohm
Ll = Secondary Leakage Inductance   5.38 mH
Rs = Secondary D.C. Resistance      120 ohm
Cs = Secondary Capacitance          180 pF
```

The primary leakage inductance is quite large in this transformer because it is a "wet" type which can handle up to 100 mA DC on the primary and is physically small. The information for this transformer model is taken from Silicon Systems K-Series Modem Design Manual from 1992.

Measuring return loss figures

Return loss is a measure of match between the impedance of the line termination and the line itself. If the impedance of the line is Zo and the termination or load is Zl then the return loss is given by the formula:

RL = 20 * log ((Zl-Zo) / (Zo+Zl))

The log function in the formula above is logarithm of 10.

The return loss must meet the regulations in the whole specified frequency range. The measurements can be quite easily made using a variable frequency signal sinewave generator and the reference impedance Zo (can be built easily from resistors and capacitors). The following circuit can be used to measure the return loss:

```
 Vin  ------------+----------------+
                  |                |
                600 ohm            Zo
                  |                |
                  +-----Vout-------+
                  |                |
                600 ohm            Zl
                  |                |
  GND ------------+----------------+
```

If you want to test the device with the signal level of Vin then you put the voltage 2*Vin to the circuit from the signal generator (the input impedance of the circuit is around 600 ohms if Zo and Zl are near 600 ohms). Connect the reference impedance Zo and the measured telephone interface circuit Zl to this measurement circuit. Connect multimeter to the circuit to place marked with Vout to measure the Vout voltage. In ideally balanced circuit this voltage is always zero. Make sure that your multimeter can measure the AC voltages in the frequency range you are using accurately (some multimeters have very large measurement error when frequencies go much higher than few hundred Hz).

Using the circuit is very simple. Just apply the input signal and measure the output voltage. Do as many measurements as necessary to cover the whole specified frequency range. When you have made the measurements you can calculate the return loss using following formula:

RL=20 * log (2*Vout/Vin)

If you want to measure telephone equipment which need some DC current flowing through the circuit you try to measure you have to use a little bit more complicated circuit to do that. You can separate the DC signal from the measurement circuit using capacitor (10 uF capacitor does not cause much error on telephone 300-3400 Hz frequency range, for lower frequencies use higher value). The power to the measured telephone or other equipment must be fed from separate power supply and run through AC block circuit which prevents the power source for short circuiting the AC signals. This AC blocking circuit can be a large coil (preferably more than 5 henries), gyrator circuit or constant current source.

```
  Vin ------------+----------------+       +----< power to the telephone
                  |                |       |
                600 ohm            Zo    AC Block
                  |                |  C    |
                  +-----Vout-------+--||---+
                  |                  10u   |
                600 ohm                    Zl
                  |                        |
  GND ------------+------------------------+
```

The measurements can be done with this circuit in the same way as the original circuit. The only thing you muts consider is the possible measurement errors caused by the capacitor and AC blocking circuit. You must make sure that Z >> 1/(2*pi*f*C). 10 uF is a good value to start because it has maximum resistance of about 50 ohms in telephone audio spectrum (300-3400 Hz).

Distortion figures

Distortion figures of the transformers have effect on voice quality on the telephone circuit. Normal telephone voice communications are not very sensitive to distortion, but modem communications are very sensitive to it. The distortion of telephone line interfacing transformer can be caused by many factors, but is specially sensitive to the performance of the magnetic lamination within the transformer. If the transformer passes some DC current on the primary or secondary coils, the distortion figures will usually get worse when current increases and many transformer do not perform in any usable way if there is any DC current around. If the transformer must handle DC current, you need a transformer designed to withstand some DC current ("wet") to keep the distortion in some usable range in this kind of circuits. Typically low end modems use "wet" transformers and high end (fast) modems use "dry" transformers.

The transmission speed of a modem is a function of many different design parameters. The performance of Modem Isolation Transformer (MIT) is one of the hardware design aspects which constrain the modem's transmission speed. Specifically, the MIT's signal distortion is the main constraint on modem speed. The distortion of the MIT can be thought of as any change in the waveshape between the secondary (output) signal from the original primary (input) signal. Significant distortion can cause problems with the signal transmission from one telecom circuit to another.

The following table (from CP Clare Databook page 61) relates the transmission baud rate, the ITU designation, and the maximum allowable THD (total harmonic distortion) of the Modem Isolation Transformer.

```
Data speed  ITU designation    Max THD

 9600         V.32               - 71 dB
14400        V.32bis             - 76 dB
28800        V.34                - 82 dB
```

As the actual modem transmission speed is a function of many variables, this table is meant to be used only as a general guideline relating THD and baud rate.

Connecting hybrids and telephone equipments

Sometimes there is need to connect normal telephone equipments directly to the telephone hybrid circuit without any connection to public telephone network. This kind of interfacing is needed for exampel for telephone equipment measurements using hybrids or for itnerfacing telephones to computers through a hybrid circuit. There are few different ways to do the interconnection of hybrid and telephone equipment.

Simple interconnection with no power provided to the line

Simplest inteconnection is just wiring the telephone equipment to the hybrid. This kind of simple interconnection works for cases where the telephone equipment does not need any telephone line loop current to operate (normal telephone operated on loop current and can not be used in this way).

```
   Green o--------------------0
TELEPHONE                     Hybrid
EQUIPMENT                     circuit
    Red  o--------------------o
```

Simplest powered circuit

This circuit is suitable for simple telephone equipments like normal telephones connected to transformer based telephone hybrids which can withstand normal telephone line DC current (use "wet" type transformer which can withstand at least 50 mA DC without saturation).

```
                Battery
                + 12V  -
   Green o-------|'|'|'------------0
TELEPHONE                         Hybrid
EQUIPMENT                         circuit
    Red  o-------------------------o
```

The circuit work so that the battery voltage powers the telephone equipment. The current taken from it is limited by teh resistance in telephone itself and the DC resistance of the hybbrid. If you fear of excessivue current, you can put a 220 ohm 1W resistor in seriws with the power supply. This will limit the current below 50 mA in all cases and does not cause too much impdeance mismatch to the circuit.

General hybrid interface

This is a general circuit suitable for interfacing "dry" hybrid circuits to practically any telephoen equipments (works also for "wet" hybrids).

```
           Line current
             feed +
               |          C1
               |         2.2 uF
               |          ||
   Green o-----+----------||-------0
TELEPHONE                 ||     Hybrid
EQUIPMENT                        circuit
    Red  o-----+-------------------o
               |
            Line current
             feed -
```

This circuit uses a capacitor C1 to isolate the line current fed to the equipment from the hybrid circuit but still passes the audio signals. The C1 shoudl have a voltage rating so high that it can withstand the voltages which might be present in the line. The value of C1 is not very critical, all values from 2 uF to 50 uF will work well. A "dry" capaictor type like polyrpopylene or duch is preferred capacitor type to be used.

The current feed is an external circuit which is used to supply the current to the telephone equipment in use. For normal tephone equipments and ideal current source with nominal current in 20..30 mA range and the open circuit voltage in range 12..48V would be ideal. NOTE: The source must be current source type. Normal voltage sources like batteries or normal DC power supplied does not wotk for this because of their low internal impedance which would just short-circuit the audio.

If you do not have a suitable ideal current source, you can use other methods for making "close enough" substitute for telephone applications. The closest thign to a traditional power supplied by telephone company would be a 48V power source fed through around 2 kohm resistor and 2H inductor. If you use lower resistance values you can use lower voltages. The 2H coil is needed to keep the impedance on audio frequencies high so that the power supply does not "short circuit" the audio signal or cause serious impedance mistaches.

If the actual impedance matches are not very important, them you can try methids like 12V power source fedh through the coil of small 12V relay or through 680 ohm 1W resistor. Both methods work in some cases, but can cause impedance mistaches which can cause poor operation of the hybrid (the isolation between incoming and outgoing audio signals will not be very good).

Components for telephone line interfacing

If you are looking for components relays and transformers for making telephone interface, check the following companies:

- Clare
- Midcom
- Prem Magnetics
- Bourns.

Using ready-made type approved interface can make designing small volume telecommunication product more easily, but unfortunately those ready made DAAs are usually more expensive than the discrete components. The following companies make DAA products:

- Xecom makes miniature telephone interface modules
- Cermetek has a selection of DAA products
- Siemens makes optically isolated DAA module DAA2000

Telephone line technical details

Telephone line frequency response

Typical telephone line has frequency response of 300 Hz to 3400 Hz. The signal starts to attenuate in the frequencies below 300 Hz because of the AC coupling of audio signals (audio signal goes trough capacitors and transformers). The high frequency response is limited by the transformer and the available bandwidth in the telephone transmission system (in digital telephone network the telephone audio is sampled at 8 kHz sample rate).

On the figure below you can see a typical frequency response of telephone line:

```
              _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
      -20   _                                   - - _             1
      -22                                             -           3
      -24                                               _         5
      -26                                                         7
      -28                                                 _       9
      -30                                                        11
      -32                                                        13
      -34                                                        15
      -36                                                   -    17
     Level =================================================== Attenuation (dB)
            0 0 0 0 0 0 1 1 1 1 1 1 1 2 2 2 2 2 2 3 3 3 3 3 3
            1 3 4 6 7 9 0 2 3 5 6 8 9 1 2 4 5 7 8 0 1 3 4 6 7
Frequency   5 0 5 0 5 0 5 0 5 0 5 0 5 0 5 0 5 0 5 0 5 0 5 0 5
            0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
```

The frequency response above is for a typical good telephone line. In real life situations the high and lof frequencies can be more attenuated. The frequency curve information was taken from Dialup Line Quality in Houston web page.

The frequency response of the line depends on the line length. When line gets long high-tones drop-off much more quickly than the low tones (with the obvious effect on speech). It's not all-that difficult to tell the distance an analog phone is from a central office (assuming an analog line is the connector): if there are no highs... it's far.

Take also note that the telephone equipment has a huge effect on the speech quality. For example carbon and electret handset microphones have radically different frequency responses. The frequency response and overall sound quality of carbon microphones used in old telephones are not very good. Many modern telephones with electret microphones give better sound quality.

Telephone line details in different countries

Normal telephone line is theoretically designed to be 600 ohm resistive impedance. This 600 ohm is kept as international reference for designing telephone line equipment (typically the signal powers are measured to 600 ohm load). In practice the telephone line does lot look like pure 600 ohm resistance. The cable and equipments used by the telephone companies have effect what the real impedance is.

Telephone equipment which is designed to operate with 600 ohm loads will operate with those real-life lines, but it's performance is worse than in ideal situation. Typically the modems are designed for 600 ohm reference impedance because they can handle the sidetone, but for best performance the telephones are designed to the exact line impedance.

When best performance is needed the circuit should be exactly matched to the impedance of the real telephone lines. Matching the hybrid circuit to the real line impedance (instead of 600 ohm) will improve the feedback typically by 3-6dB. 20dB sidetone is easy to achieve, but 30dB is also not too difficult provided you can measure the line impedance and take steps to build a correct balancing network.

Different countries have different characteristics on the telephone line parameters. Here are some impedance models for typical lines in different countries:

USA

Normal telephone subscriber lines in USA (0.4-0,6mm subscriber PE insulated vaseline filled cable) are 770 ohm resistor (with 2uf series capacitor) and 47nF parallel capacity.

```
            2 uF
             ||
    ----+-----||--------+
        |    ||         |
        |              | |
       ---             | | 770 ohms
       --- 47 nF       | |
        |               |
    ----+---------------+
```

This diagram is referred to 800Hz, but impedance is rather complex, and varies from high value at low frequency and drops to ca. 150 ohm on 10kHz and 120-125 ohm above 100kHz.

Some telephone lines can have higher impedance (typically 1100 ohms in lines with loading coils or telephone air cables).

Finland

The equipments connected to public telephone network in Finland must meet NET4 (ETS 300 001) technical specs. All power specs and return loss measurements are taken so that the reference impedance is 600 ohm resistive.

```
   ---+
      |
     | |
     | | 600 ohm
     | |
      |
   ---+
```

The return loss of the terminal equipment must be greater than 10 dB when compared to 600 ohm reference. This measurement applies to telephones, modems and other terminal equipments. NET4 technical specs are European specs and they are used in many European countries (NET4 is actually a collection of different specs in use in different countries). Telecommunications Administration Centre in Finland also mentions in it's regulation THK 20 I/1997 M that the telephone line equipment can be measured against the 600 ohm resistance mentioned in NET4 or complex impedance of Z = 270 + (750 //150 nF). Here is a picture of that complex reference impedance:

```
                 750 ohm
                  _____
    270 ohm   +--|_____|--+
     _____    |           |
  --|_____|---+           +-----
              |    ||     |
              +----||-----+
                   ||
                 150 nF
```

Typical cable used in for subscriber lines has following characteristics: 0.5 mm diameter wire, loop resistance 182 ohm/km and pair capacitance 39 nf/km.

Because the telephones designed to meet the needed characteristics measured against 600 ohm reference impedance does not always work satisfactory when connected to telephone switches. A better results can be obtained if the phone meets all other NET4 regulations, except the return loss is matched to TPL06 regulations. The reference model used in TLP06 for telephone line:

```
                 910 ohm
                  _____
    270 ohm   +--|_____|--+
     _____    |           |
  --|_____|---+           +-----
              |    ||     |
              +----||-----+
                   ||
                 120 nF
```

The return loss to this reference model must be greater than 15 dB.

Loading coils

Loading coils are lumped inductance added in series with the telephone line to compensate for the mutual capacitance of the cable pair(s). They are placed at specific intervals on loops of 18,000 feet or greater to improve voice grade transmission. Placing load coils at other than the specified intervals actually degrades voice grade transmission. It is generally accepted that the upper cutoff frequency of these devices is 3000 Hz. Therefore loaded loops do not lend themselves well to high frequency or high data rate transmission. Therefore loaded loops do not lend themselves well to high frequency or high data rate transmission.

Loading coils introduce phase delays which are fine for voice but unacceptable for high speed data and are best confined to the past, or to very long local voice loops where they can't be done without. With the advent of ISDN and other high bitrate digital transmission technologies, many telephone companies are attempting to limit loop lengths to 18k' or less and so eliminate need for loading coils.

Simulating telephone line

Resistor and capacitor network simulation models

The most traditional way to simulate telephone line is to use resistor and capacitor networks to simulate the attenuation caused by the telephone line. A typical model for this type of telephone line simulator is a resistor and capacitor network which looks like this:

```
          R               R
         _____           _____
  o------|_____|---+----|_____|------o
                   |
                 -----
                 -----  C
                   |
  o----------------+-----------------o
```

The resistor R and capacitor C values depend on the cable characteristics. Old Swedish telephone equipment regulations have listed the following values for simulating a typical local loop cable:

Length	Cable diameter	R	C
0.5 km	0.4 mm	70 ohm	20 nF
1.0 km	0.4 mm	140 ohm	40 nF
0.5 km	0.5 mm	45 ohm	20 nF
1.0 km	0.5 mm	90 ohm	40 nF

The circuit can be modified for simulating symmetrical cable better in some measurement by dividing the resistance to four wires. This arrangement leads to following circuit:

```
          R/2             R/2
         _____           _____
  o------|_____|---+----|_____|------o
                   |
                 ----- C
          R/2    -----    R/2
```

```
              _____     |      _____
o------|_____|---+----|_____|------o
```

Build a telephone line test system from telephone cable

Find an old spool of 25 pair cable, preferably pulp insulated, from the back of the warehouse. Punch down each end to opposite sides of a 66 block, only on one side start with pair *two*, and bring pair one down to the last two terminals, after pair 25. Stick in bridging clips across all but the bottom pair of the block. In this way you get quite easily very long line to test quite easily. Attach one end of this to a cheap phone line simulator (all it needs to provide is battery, dial tone, and ring voltage.) Buy a couple of test clips from your usual supplier, and now you have a fairly easy to use test device for cheap.

If you need to simulate also the interference which can go to the cable, use an old office fan and an interference source by putting this right next to the spool of cable and turn it on during testing.

If you are doing worst case testing, you should use junky cable from the recycling bin and stick in loading coils at the appropriate intervals. To be really nasty, take a couple dozen feet of your telephone this cable, create leaks to the cable and put that cable into water (you can add some salt and dirt to the water for more realistic situation). Add this into the middle of your test circuit someplace. Now you have something that is beginning to approach the real world cable plant worst case.

General model for twisted pair line for simulation purposes

It is not too difficult to model the parameters of the pair of wires if the cable in not too long. If your transmission line is not extremely long (by "long" I mean, say, more than a few wavelengths at the highest frequency of interest), you could build up a transmission line model with R, L, and C lumped elements. Each "lump" would be a resistor and inductor in series for each wire, and a capacitor in shunt, as shown below:

```
     R        L
---/\/\----()()()---+-------
                   _|_
                   ___  C
     R        L     |
---/\/\----()()()---+-------
```

The more of these lumps you cascade, the better the approximation to a real, distributed transmission line. Note that this model won't properly account for skin effect unless you can make the resistors and inductors frequency dependent. This means that a lumped approximation consisting of N RLC sections will work quite well up to a certain cutoff frequency. If R, L and C are the component values in each section (not the same as the per km values) then the input Z of the line will go to zero at the same frequency that the gain hits the "brickwall". This frequency will be about 2 * (1/ (2 pi sqrt(lc))). The simulation is perhaps usable up to half this frequency (An N section simulation is actually a 2N pole low pass filter. Think about it!). If you're only interested in doing voice band stuff then you might only need a few sections, but if you need to use high frequency (HDSL at around 500 kHz), you will need much more sections (over 100).

For getting the R, L and C values for the model can be done in many ways. Capacitance per foot between pairs and to the shield is specified by the cable manufacturers and you can calculate inductance from that and impedance. Usually the resistance of the cable is also specified. For a given R, L, C (the per km values), the number of sections needed is proportional to the product of the attenuation (in dB) and the bandwidth required.

If you want you can measure the capacitance and inductance of say a metre length of the cable and divide into say six discrete sections. To determine the coupling, form the cable into a loop and measure the inductance of one conductor (about 1.6 uH) with the other conductor first open and then short-circuited. You can calculate the coupling from these values. Use as low a frequency as possible to minimise any capacitance effects.

It gets a little more complicated if you want to model the frequency dependent resistance (due to skin effect) or frequency dependent conductance (due to dielectric loss). To model the conductance, replace the shunt capacitors with:

```
   |
   +---+
   |   R2
   C1  |
   |   C2
   +---+
   |
```

This gives a reasonable approximation to dielectric loss. The dielectric loss is modededeld by R2 (R2 = 1/G). The capacitor C2 in series is here just to block out DC, because real cables look like capacitors at low frequencies.

To model skin effect, replace the series RL sections with:

```
 ----L1--R1--+--L2--+----
             |      |
             +--R2--+
```

Most commercial line simulators work this way.

The coupling between 2 different pairs is more difficult to model. To model coupling between pairs, then, you'd have to put little capacitors from one transmission line to the other and couple the inductors. Of course it would be a majoe project to set the values of those capacitors and the coupling coefficients for the inductors. Those values depends on the "lay" of the pairs within the jacket, and may be significantly different from cable to cable. This meanst that you have to measure the cable you are actually going to use, or pay for 'star-quad' which is carefully constructed to meet low coupling specifications between pairs.

TRANSMISSION SYSTEMS FOR COMMUNICATIONS, revised 4th edition, Bell Telephone Laboratories (1971) gives the followign information on typical cable characteristics:

"The primary constants of twisted pair cables are subject to manufacturing deviations, and change with the physical environment such as temperature, moisture, and mechanical stress. The inductance, L, is of the order 1 mH/mile for low frequencies and the capacitance, C, has two standard values of 0.066 and 0.083 uF per mile although lower capacitance cables are under development.

Of the primary constants, only C is relatively independent of frequency; L decreases to about 70 percent of its initial value as frequency increases from 50 kHZ to 1 MHz and is stable beyond; G is very small for PIC (polyethylene insulated cables) and roughly proportional to frequency for pulp insulation; and R, approximately constant over the voiceband, is proportional to the square root of frequency at higher frequencies where skin effect and proximity effect dominate."

Tips:

If you're just doing a computer simulation, then simulate an equivalent unbalanced (half) line; this reduces the computation required.

```
---Zt/2------R/4N--L/4N---------R/2N--L/2N--------R/4N--L/4N---------
 |                         |                |                   |
 V1                       2C/N             2C/N               Zt/2
 |                         |                |                   |
---------------------------------------------------------------------

             ^^^^^^^^^^^^^^^^   ^^^^^^^^^^^^^^    ^^^^^^^^^^
                  (one)              (N - 1)         (one)
                                    Sections
```

Where R is the total line resistance (over the length required), C is the total line capacitance, L is the total line inductance, N is the number of sections. The conductance is assumed to be zero.

Testing standards

To be able to do any repeatable testing one must have control over the equipment in a controlled environment. Also to correlation you results to any test done by other individuals, one must test to a given set of "standards". For USA standards TSB37A and TSB38 deal on telephone lines and modem testing (Loop 1 condition for them is EAI 1, which is a 2kft of 26ga). For longer loop there can be up to 5 loading coils in the line.

Other countries have also standards on the testing conditions but I have not found references to them.

Other technical regulations for telephone line terminal equipments

The following specs are taken from the European NET4 (ETS 300 001, second edition, April 1994) regulations. ETS 300 001 is, basically, a big collection of the various European countries parameters. I have put some parts of the specs (Finnish part) to here:

On-hook

- DC resistance must be at least 1 Mohm when measured at 100 V voltage
- Isolation resistance must be at least 5 Mohm from line to touchable metal parts measured at 100 V voltage
- The impedance in voice frequency (200-3400 Hz) must be greater than 10 kohm when measured with 0.5V RMS audio signal

Off-hook

- Isoltation resistance must be at leats 5 Mohm from line to touchable metal parts measured at 100 V voltage
- DC-resistance must be less than 400 ohm in current values between 20 mA and 50 mA
- If the terminal equipment used constant current principle then the current must be in 20-50 mA region in all conditions
- The impedance of the terminal equipment must be so matched that the return loss is greater than 10 dB compared to 600 ohm reference
- When the terminal equipment transmit voice or music the mean signal level must not be greater than -10 dBm level in any 10s timeslot
- When other signals are sent to line the signal must not be greater than -10 dBm in any 200 ms timeslot
- The signal level between 3400 Hz and 12 kHz must be attenuated 12 dB/oct and the signal level in frequencies greater than 12 kHz must be less than -55 dBm
- Common mode rejection must be greater than 40 dB in 40-300 Hz region, greater than 50 dB in 300-600 Hz region and greater than 55 dB in 600-3400 Hz region

Note on signal levels: 0 dBm means 0.775 Vrms level, so -10 dBm is around 0.2 Vrms.

Equipment in series with telephone

- The series resistance must be less than 200 ohm
- The attenuation of audio signal must be less than 1 dB at 800 Hz

There are also many other technical specs in NET4 document, but those are the most critical to hybrid circuits. When building telephone signals you should also understand the telephone equipment electrical safety regulation in EN 41 003 standard. This means generally that the equipment must withstand 2-3 kV surge and DC test between the telephone line. The equipment does not be able to cause dangerous voltages to the telephone line or to touchable parts in any probable single component failure. And many other safety regulations.

Interference in the telephone line signal

Noise on telephone lines are often caused by longitudinal waves causing EMC problems at one end (possibly subscriber end), improving line balance will improve this, or you can attenuate the longitudinal current. Sometimes the line seem unbalanced toward the ground at subscriber end, this may cause some hum problems.

Using a bifilar wound audio choke will often improve the balance and the undesired current, however it cannot improve noise caused by bad joints. Simplest bifilar coil for reducing radio frequency interference can made by coiling the telephone wire around a ferrite core. Another of bifilar wound choke is an audio 600 ohm 1:1 transformer, connected in series, such that you choose the two 'in-phase' sides of windings for input and the opposite sides for output/subscriber side. The choke should preferably be heavy, 1kg weight is find, too low core may cause it to saturate and such will not operate well. When you connect anythign in series with the telephone line make sure that the series-connected filerr does not cause impedance mismatches or too much series resistance to the line (check what is said about equipment in series with telephone line and stay within those limits).

Other filtering methods are not very effective in reducing the noise in telephone lines. If your filter removes certain frequencies it will affect the telephone speech quality and make sure that modern modems will not work well anymore.

Other related document

- Disassembly Of A Touch-tone Phone
- Telephone ringing circuits
- Telephone privacy adapter
- Understanding telephones

Final words

Please read ITU-T (former CCITT) recommendations and make sure to follow any instructions from the telephone company before you connect any of your own equipment to the local line. Read also your national regulations on telephone equipments.

Remeber that in case you cause any trouble, you may have to pay for faultfinding and other problems!

Tomi Engdahl <Tomi.Engdahl@iki.fi>

Telephone ringing circuits

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Preface

Telephone circuit gain always interrest, because telephones are everywhere and quite often there are old telephone luying around somewhere. Those telephones can be used for many interresting experiments including small home intercom: connect telephones in series or parallel and feed suitable operating current (about 20 mA) to them through resistor from power supply.

The most problematic to home experimenter is how to get telephone ringing because the ringing voltage is over 50V and not at standard mains frequency (50/60Hz). Sometimes you want to get the information that telephone is ringing to your own circuits. This text tries to clear out those problems.

What is ring signal ?

The telephone company sends a ringing signal which is an AC waveform. Although the common frequency used in the United States is 20 HZ and in Europe is typically 25 Hz, it can be any frequency between 15 and 68 Hz. Most of the world uses frequencies between 20 and 40 Hz. The voltage at the subscribers end depends upon loop length and number of ringers attached to the line; it could be between 40 and 150 Volts. The ringing cadence - the timing of ringing to pause - varies from telephone company to company.

The usual arrangement is to feed the 75 V a.c. ringing current (backed by earth) down one wire of the phone line. On the other wire is placed a slugged relay (or equivalent) which is backed by -48V d.c. When you pick up the phone, the relay operates to the loop d.c. current and trips the ringing current. It also triggers a further device to put the transmission bridge in circuit to enable speech to take place, together with supervision of the calling and called loops. The ralay needs to be a slugged relay to prevent premature ring trip by the a.c. ringing current.

In USA minimum ring voltage supplied is 40Vrms (delivered into a 5 REN load). This is the must detect limit. There is also a minimum must ignore value of 10Vrms. Milage on individual PBX's will vary greatly. But most guarantee to deliver 40Vrms into a 3 to 5 REN load.

When the telephone ring signal is sen to the telephone, the ring voltage is not applied constanly to the line. Typically ring timing is 2 seconds on and 4 seconds off in the US. In the UK ring timing goes .4 sec on, .2 sec off, .4 sec on, 2 sec off then repeats. In toher countries the ring timign cna vary from country to country (even from operator to operator) and you should check the local regulations if you want to get to know the actual ring signal timing in use.

For more information, check Understanding Telephones article by Julian Macassey at http://www.egyed.com/phonework.html and appropriate BellCore documents.

What is REN ?

REN stand for Ringer Equivalen Number. It is a measurment of how mugh ringing power certain telephone equipment takes. REN numbers are used in USA to determine how many telephoen equipments you can connect to same telephone line and still get them ringing properly (typical line can drive about 3-5 REN load).

The definition of 1 REN is the ringer power required by one ringer of an AT&T standard 500 series telephone set in single-party configuration (ringer placed ACROSS the line). One place to find the exact info: get a copy of 47CFR Part 68 - this is the FCC technical specs (and other info) regarding the PSTN (public switched telephone network). This info also may be available from the FCC's web site.

What is ringing tone ?

Ringing tone is the ringing that can be heard while the receiver is on-hook and somebody tries to call you. The terms used for describing this telephone ringing are not always very clear whet they mean, because the same term has been used in differnet places to mean different things. ITU-T Q.9 indicates the preferred term is "ringing tone", but that "ringback tone" is used in the USA. On the other hand, Bellcore (and the old Bell System), used "audible ringing tone" in many of their documents. In 5ESS switch documentation (according some news articles), RINGBACK is used only to describe various ways (other than a normal terminating call) by which a subscriber's telephone may be rung. Usually people say "ringback" in place of "ringing tone".

What is distinctive ringing

Distinctive ringing is a system where different ringing tone patterns can tel different thing about the telephone calls. Typical applications are PBXs where you can identify if the call is from inside buildign or from outside by hgearing different ring pattern. Aother applications are when multiple phone numbers are assigned to one physical line and the rign pattern tells which number of them has been called.

Distinctive Ringing and Call Waiting patterns and timing use in USA are covered in GR-506-CORE. Use of multiple patterns to identify the CALLED party (multiple DNs per line) is covered in the basic LSSGR (GR-505 and GR-506 in particular), in the ability to assign ringing patterns to numbers and to Centrex services. ANSI T1.401 identifies some other requirements for distinctive ringing involving inter-exchange carriers.

Normal telephone wiring

In normal telephoen wiring (used in Finland, USA and very amny other countries) the telephoen audio and sing signals share the same wire pair. Typical wiring for 6 pin modular connector:

```
1
2
3 a-wire
4 b-wire
5
6
```

A and B wires make the pair which telephone used. Typicslly the modular connectos used in telephone have only 2 or 4 pins installed. Normally unused pins are used for wiring more than one line to same connector or for some special applications.

There are also many other types of telephone line connectors in use, but nowadays this modular connector is the most common in telephone terminal equipmens like telephones with removable cord, modems and FAX machines.

Special cases in ring signal wiring

On some coutries the ring signal is fed to the customer telephoes using one extra wire. The UK wiring tetails are available in separate UK wiring document.

Ringer circuits in telephones

Classical bell type ringer

The most classical telephone ringer circuit is a mechanical bell controlled by an electronic coil. The circut consists of the bell coil and a capacitor (usually 470 nF to 2 uF rated for 250V or more) in series with it. This circuit is connected in parallel to other telephone electronics. The capacitor in the circuit stops the DC in to pass through the bell coil, but it lets the ring voltage through easily. Because of mechanical nature of the ring circuit, it is very sensitive to the frequency of ring voltage and other than the resonance frequency of the bell system (usually around 20-25 Hz) do not generate satisfactory ring.

The coil has usually so high impedance that it does not disturb the telephone audio circuit operation when telephone is off-hook. Other possiblity is that the ring circuit is disconnected when the telephone is picked off-hook.

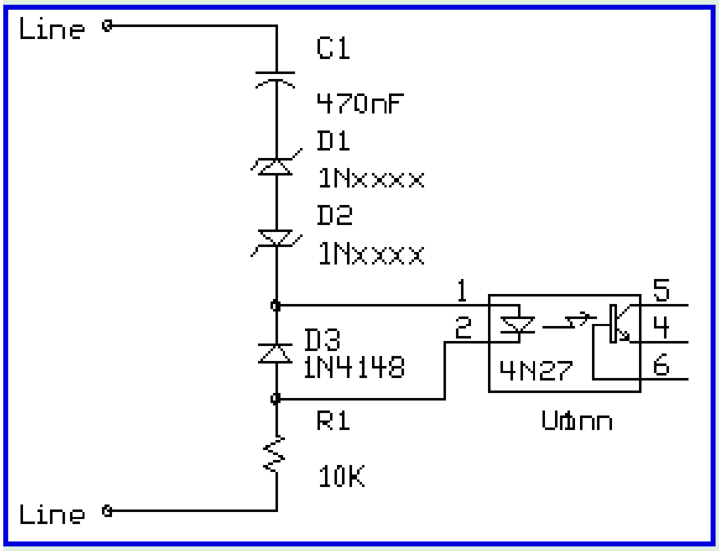
Electronic ringers

The ringer circuits in the modern telephones have the same basic idea, but the coil controlled bell is replaced by modern electronic ringing chip and small speaker. The capacitor is still used in series with ring IC input to make only AC pass to the ring chip. The electronic ringing circuits are not sensitive to the ringing voltage and they easily ring with ring signal frequencies between 16 Hz and 60 Hz.

Ring detection circuits in modems

In computer modems the logical signal from ringing is needed instead of ringing tone. The ring circuit must pass the ring signal information to modem electronics and still provide electrical isolation between telephone line and modem electronics. This ring detection is usually done using one optoisolator circuit, which replaces the raditional ring circuit. The optoisolator output can be easily connected digital electronics, but the optoisolator input side needs more electronics: one capacitor for not letting DC to pass through optoisolator, one resistor to limit the cirrent passing through optoisolator LED and one reverse conencted diode in parallel with optoisolator LED to prevent negative voltages from damaging the LED. This is the basic ring detection circuit.

Usually there is also two zener diodes (usually 10-20V models) to make sure that the ring detection circuit does not detect too small AC signals in the line as ring signal. In the picture below you see a very typical ring detector circuit for modems. The circuit just gives the idea how modem ring detector circuit work. The actual component component values selection must be so that the circuit meets the national telephone regulations (this can be usually easily done by using suitable zener diodes and maybe chancing the resistor value a little).



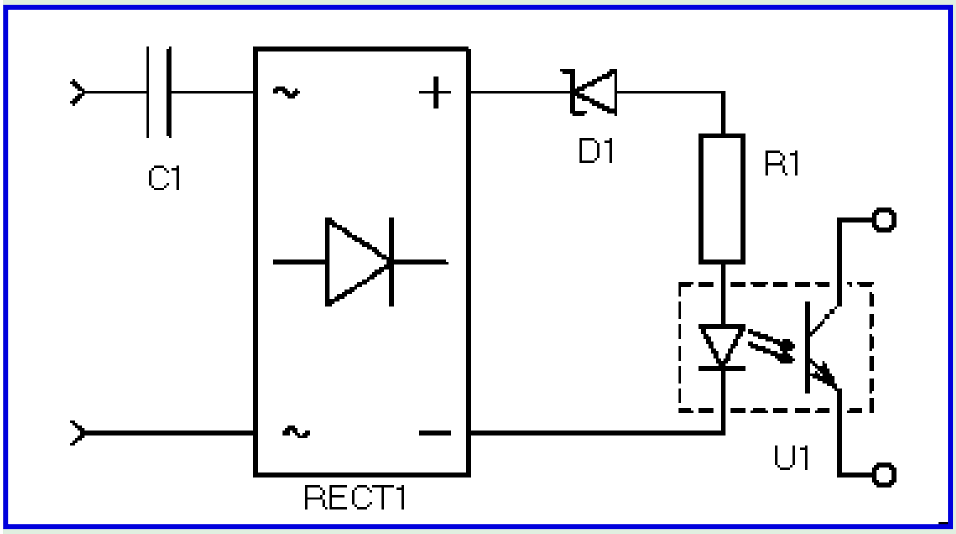
Component list:

```
C1      470 nF 250V AC
R1      10 kohm 1W
D1,D2   10-20V zener diode (any value in this range), 400 mW power rating
D3      1N4148 diode or equivalent
U1      4N27 optoisolator or similar
```

NOTE: You can get the circuit work by taking out D1 and D2 and replacing them with a short circuit. The circuit works after then, but it is possible that in this case some low voltage noise on the line can cause the circuit to ring. Different countries have different specifications on how low voltages should not cause a telephone to ring at all.

PS. If you are interrested in using theis cirucit as basis for controlling some high power cirucitry take a look at http://www.aaroncake.net/circuits/pflash.htm for a circuit example how to drive a relay when ring is detected.

Another apprach for ring detecting is to use a full wave rectifier circuit to convert the AC sign signal to the DC suitable for optoisolator and then put current limiting resistor and zener diode to the rectifier output.



Component list:

```
C1     470 nF 250V AC
R1     10 kohm 1W
D1     10-20V zener diode (any value in this range), 400 mW power rating
RECT1  Rectivifier bridge 200V voltage ratign, at least 0.1 current rating
U1     4N27 or CNY17 optoisolator
```

Other ideas to detect telephone ringing

One idea which is proposed in many sources is to use small neon bulb (like those used as lights in some mains switches) for detecting the ring signal. The circcuit proposed is to connect one neon bulb and 47kohm resistors in series and connect this to telephone line. The neon bulb has about 60V trigger voltage to start conducting, so standard 48V telephone battery voltage does not light it. When the AC ring signal is added to that voltage, the voltage is enough to light the neon bulb. The neon bulb can be used as visual indicator or electronics can sense it with LDR photoresistor or phototransistor.

If you don't want to build your own circuit from neon bulb and resistor, there is an even easier solution is to go down to the hardware store and get a "pigtail" tester. It has two nice leads that one normally pokes into the wall outlet to test for voltage. Wire it instead to the phone line. This saves the hassle of trying to find the container for the neon lamp, and the resistor (which is VERY necessary, take my word for it).

One modem schematic I have seen used quite special method for detecting ringing signals: It had a small capacitor in parallel with on-hook/off-hook control relay contacts. This capacitor let some small part of the sound and ring signals pass to the telephone transformer. In this way those ring signals can be detected as small signal pulses in transformer secondary (and this circuit can be also used for Caller ID signal detection). The capacitor was so small that the impedance seen from telephone line stays high enough not to disturb other equipments in the same telephone line when modem is no on-line.

What telephone regulations say about telephone ringers

European NET4 telephone line terminal equipment specs define the following specs for the telephoen ringing detector circuit.

- The impedance in voice frequency (200-3400 Hz) must be greater than 10 kohm when measured with 0.5V RMS audio signal
- The current taken by the ringer must be equal or less than 5 mA at 35 V ring voltage and equal or ledd than 10.7 mA at 75V ring voltage. The measurments are made using 25 Hz ring current frequnecy.
- Ring detector must work on ring signal which is 44-58V DC summed with 25+-3Hz AC ring signal in voltage range 35-75 V. The feeding resistance for ring generator is 800-1710 Hz.
- Ring detector must not detect ring signal which is 44-58V DC summed with 20-3400 Hz AC ring signal which is less than 10 V. The feeding resistance for ring generator is 800-1710 Hz.

If the equipment is automatically responding the equipment must wait at least 1s from the ring detection until it goes off-hook.

Telephone ringer classification

In USA FCC regulations need the ringer type to be specified on the device. The possible types are Class A and Class B. Class B ringers will respond to ringing frequencies of between 17 and 68 Hertz while Class A ringers will respond to betwwen 16 and 33 Hertz. Class A devices are those typical old telephone bells and practically all electronic ringers are B type. Nearly all of the devices made to connect to the phone lines today are of the Class B type. The telephone ringer type on your device (if you live in USA) is printed on the FCC sticker on the bottom with a REN number on it. You'll see something like .9B (= REN 0.9 Class B) or 1.0A (= REN 1.0 Class A).

How to make telephone ring

The following ideas are simple circuits, which generate ringing voltage at mains frequency (50 or 60Hz depending on country). They will ring modern telephones very well, but the rign sound might not be actually the same as with right ringing signal. If that is not a problem, then go on. The ring signal at 50 or 60 Hz does not work with old telephones which have mechanical bells in them.

Direct connection to mains

This approach has been proposed many times at rec.theatre.stagescraft newsgroups but I strongly suggest not to use it. Mains voltage (120V AC 60Hz) used in USA makes the modern telephones ring, but it is dangerous to make direct connection to mains voltage. And if you don't use any type of current limiting, the telephone will cause dangerous short circuit when it is picked up. The telephone will destroy and put out smoke.

50/60Hz ring voltage generated from mains voltage

If you want to use very simple circuit for ringing, I would suggest following combination: a small ready made AC adapter which puts out AC and a small transformer connected to it. If you use suitable tranformer combination, you will get nice 70-90V AC voltage at you mains voltage frequency (50 or 60 Hz). Ready made wall adapter will provide provide isolation from mains voltage and also limit the current in short circuit situation.

Suitable comination for example is wall adapter which outputs 8-9V AC at 200-500 mA connected to transformer which has 120V primary, 12V secondary and power handling capacity of few watts. The wall adapter is connected to transformer's 12V secondary through a button. When the button is pressed, there is about 70-90V AC available at transformer's primary winding. For current limiting it is a good idea to put 1 kohm 3W resistor in series with transformer's secondary. If you can't find transformer I told earlier, remeber that many transformers with 220V primary winding have center tap connection for 110V voltage wiring. And if there is not centre tapped 220V transformer, you can always use 220V to 24V transformer. If your wall transformer has different rating, the scale the transformer's values according that. The component values in this circuit are not critical, but keep in mind that the voltage of transformer's secondary must be grater thatn the output voltage of the wall adapter.

And for your safety, build this circuit to a good box in which you have telephone connector on one side. And be careful with the circuit not to get shocked because the 50/60Hz ring voltage is more dangerous than normal ring voltage.

Methods for generating good ring voltage

Ringer module

The easiest way to get real ringing module. Those units are available from some companies which make DC/DC converters for telecommunication industry. Migh not be the easiest component to get.

70V line PA amplifier

The output voltage of PA amplifiers designed for driving 70V speaker system speakers have enough output voltage and power for ringing telephones. If you have old this type of amplifier lying somewhere, you can connect the amplifier input to fuction generator and output to telephone through 1 kohm 3W resistor. When you set the function generator to generate sine wave at 20-25 Hz at suitable level for amplifier, you have an adjustable level ring generator. Usually those amplifiers are not good at playing back frequencies below 50 Hz, so you might have to try higher frequencies if that does not work as expected.

Normal audio amplifier and transformer

Very nice variable amplitude ring generator can be built from audio amplifier designed for driwing 4 or 8 ohm speakers and have output power of 3W or more, 10 ohm 10 W resistor, 220V to 12V transformer (few watts), 1000 ohm 3W resistor and function generator.

```
              ___________      10 ohm         1000 ohm
             |           |----/\/\/\--+ ||(---/\/\/\---
             |           |            | ||(
Sinewave----| Amplifier |            )||(          Ring voltage out
             |           |            | ||(
             |___________|------------+ ||(------------
                                   Transformer
                                    12V:220V
```

The circuit is easy to build. Connect 10 ohm resistor in series with transformer's secondary winding and 1000 ohm resistor in series with primary winding. Connect the primary winding side of the transformer to amplifier's speaker output. Connect the telephone to the secondary side. The resistors are in the circuit to limit the current and to keep the impedance high enough for the amplifier.

When you have done this, connect you function generator to amplifier's input and set it to generate 20-25 Hz sine wave at suitable level for amplifier's input. Turn down the volume of the amplifer. Turn the amplifier on. Turn the volume up until you hear telephone ringing well. You can check the ringing voltage with multimeter if you vat to make it to exactly right level.

Modified power inverter circuit

It is possible to make 17 - 25Hz a.c. from d.c. A simple multivibrator will do it. You then need a power transistor or similar to give the high-current output. A suitable circuit can be modified from typical power inverter circuit by changing the timing components to make the frequency to 20-25 Hz range. Then the transformer needs to be selected so that it matches this application (for 12V operation take a mains centre-tapped 60V (30+30V) secondary and 230V primary).

Dedicated ringing generator circuit

There have been telephone ringer circuit in major electronics magazines and circuit books. Those circuit are good idea when you want to build the circuit from base components.

There are commercial units specifically made fro ringing telephone. TELE-Q is a device designed for ringing telephone theatre effect. That unit is available from Norcostco for little over 100 US dollars. I have no experience in this product but it has been suggested in many usenet news articles.

Maplin Electronics has a phone ringer electronics kit which can give out UK and USA type ring styles. It has been reported to work uite well with any modern telephone, though it has said to struggle slightly to drive old fashioned bell types which need lots of ring current.

There are also telephone line simulators available from some tecom equipment manufacturers. Those telephone line simulator boxes also usually include the ringer circuit. Two examples are Viking Electronics Line Simulator/Ringdown Circuit and Jech Tech Phone Helper. Usually complete line simulators are more expensive than simple ringer circuit but they have more uses also (you can make two telephones an intercom etc.).

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From telecom@eecs.nwu.edu Wed Aug 7 00:47:09 1991
Received: from hub.eecs.nwu.edu by gaak.LCS.MIT.EDU via TCP with SMTP
        id AA19091; Wed, 7 Aug 91 00:46:57 EDT
Resent-Message-Id: <9108070446.AA19091@gaak.LCS.MIT.EDU>
Received: from trout.nosc.mil by delta.eecs.nwu.edu id aa29672;
          5 Aug 91 8:43 CDT
Received: by trout.nosc.mil (5.59/1.27)
        id AA15410; Mon, 5 Aug 91 06:40:37 PDT
Received: by jartel.info.com (/\=-/\ Smail3.1.18.1 #18.7)
        id <m0k758G-00018IC@jartel.info.com>; Mon, 5 Aug 91 06:38 PDT
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        id AA10327; 5 Aug 91 06:28:44 PDT (Mon)
Received: by denwa.info.com (5.59/smail2.5) with UUCP
        id AA10322; 5 Aug 91 06:28:28 PDT (Mon)
Received: by bongo.info.com (smail2.5)
        id AA04642; 5 Aug 91 06:20:12 PDT (Mon)
Reply-To: julian@bongo.info.com
X-Mailer: Mail User's Shell (6.4 2/14/89)
To: telecom@eecs.nwu.edu
Subject: Phone Patches
Message-Id: <9108050620.AA04638@bongo.info.com>
Date: 5 Aug 91 06:20:06 PDT (Mon)
From: Julian Macassey <julian@bongo.info.com>
Resent-Date: Tue, 6 Aug 91 23:50:11 CDT
Resent-From: telecom@eecs.nwu.edu
Resent-To: ptownson@gaak.LCS.MIT.EDU
Status: RO

Dear Patrick,
Here is an article I wrote about phone patches. If you think it is
worth it, stuff it in the archives.

------cut, slash, deforest ----------------------

BUILDING AND USING PHONE PATCHES

From simple to elegant, patches help make the connection

By

Julian Macassey, N6ARE

First Published in Ham Radio Magazine
October 1985.

In telephone company parlance, a patch is any connection between a phone line and another communications device, whether it be a radio, a tape recorder, a data device (such as a modem), or even another phone line.

Radio Amateurs, on the other hand, tend to limit the meaning of "patch" to the connection of transmitters or receivers to the phone line for phone conversations. But there's more to it - Amateurs can and do use phone patches for purposes other than telephone conversations. One particularly effective application is for checking TVI and RFI complaints; simply set the transmitter on VOX, go to the site of the interference complaint, and then key your transmitter via the phone line. Doing this will indicate whether your transmitter is or is not the source of the problem. If it is, you can use this method to test the measures you've taken to correct the problem.

A phone line is, simply speaking, a 600-ohm balanced feed device - which also happens to be how professional audio can be described. Most modern Amateur transmitters have 600-ohm unbalanced inputs; most cassette recorders have a 600-Ohm unbalanced input; the "tape" outputs on home stereos are also 600-ohm unbalanced. All this makes patching relatively simple. While there are various degrees of sophistication and complexity in patching, in an emergency, patches can be easily put together using readily available components. Before starting to build a patch, however, it might be helpful to read last month's article on understanding phone lines.

The Simple Patch

The simplest way to patch a phone line to another piece of equipment is to use a couple of capacitors to block the phone line DC. While this simple approach will work in a pinch, it will tend to introduce hum to the line because of the unbalance introduced. The capacitors used should be nonpolar, at least 2-ohm F, and rated at 250 volts or better (see fig.1).

To hold the line, the patch should provide a DC load by means of a resistor (R6) or by simply leaving a phone off the hook. The receiver output may need a DC load (R7) to prevent the output stage from "motorboating." Use two capacitors to maintain the balance.

With all patches hum can be lessened by reversing the phone wires. A well-made patch will have no discernible hum.

The Basic Phone Patch

Because a phone line is balanced and carries DC as well as an AC signal, a patch should include a DC block, a balun, and a DC load to hold the line. The best component for doing this is a 600-ohm 1:1 transformer such as those used in professional audio and for coupling modem signals to the phone line, available from most electronics supply houses. Old telephone answering machines are also a good source of 600-ohm transformers. Some transformers are rated at 600-900 ohms or 900-900 ohms; these are also acceptable. Make sure that the transformer has a large enough core, because DC current will be flowing through it. (Some small-core transformers become saturated and distort the signal.)

In section 68.304 of the FCC Part 68 regulations, it states that a coupling transformer should withstand a 60 Hz 1kV signal for one minute with less than 10 mA leakage. For casual use this may seem unimportant, but it provides good protection against any destructive high voltage that may come down the phone line, and into the Amateur's equipment. A 130 to 250 volt Metal Oxide Varistor (MOV) across the phone line will provide further protection if needed.

The DC resistance of the transformer winding may be so low that it hogs most of the phone line current. Therefore, while using a phone in parallel for monitoring and dialing - which is recommended - the audio level on the incoming line may be too low. Resistors R1A and R1B (see fig.2) will act as current limiters and allow the DC to flow through the phone where it's needed. If possible, these resistors should be carbon composition types.

To keep the line balanced, use two resistors of the same value and adjust the values by listening to the dial tone on a telephone handset. There should be little or no drop in volume when the patch transformer is switched across the phone line.

One of these transformers, or even two capacitors, can be used to patch two phone lines together, should there be a need to allow two distant parties to converse. There will be losses through the transformer so the audio level will degrade, but with two good connections this will not be a problem.

On the other side of the transformer - which could be called the secondary winding - choose one pin as the ground and attach the shields of the microphone and headphone cables to it. Attach the inner conductors to the other pin. The receiver output will work well into the 600-ohm winding, and if transmitting simplex or just putting receiver audio on the line there will be no crosstalk or feedback problems. In some cases, the audio amplifier in a receiver does not have enough output to feed the phone line at an adequate level; this can be handled by using the transformer with two secondaries (see the "improved" patch below) or by coupling a 8:1 kilohm transformer between the audio output and 600-ohm transformer. If RF is getting into the transmitter input, a capacitor (C1) across the secondary should help. A good value for the lower bands and AM broadcast interference is 0.1 uF. For higher frequencies, 0.01 uF usually gets rid of the problem. Unshielded transformers are sensitive to hum fields and building any patch into a steel box will help alleviate hum as well as RFI.

The Improved Phone Patch

Several enhancements can be made to the basic phone patch to improve operation. The first is the addition of a double-pole double-throw switch to reverse the polarity of the phone line to reduce hum. This may not be necessary with a patch at the same location with the same equipment, but if it is, experiment with the polarity of the transformer connections and adjust for the least hum. Most of the time the balance will be so good that switching line polarity makes no difference. The switch should have a center "off" position or use a separate double-pole single throw switch to disconnect from the line. The two secondaries on the "improved" patch (fig.3) should be checked for balance by connecting the receiver and transmitter and checking for hum while transmitting and receiving. Switch the shield and inner conductors of the secondaries for minimum hum.

Many transmitters do not offer easy access to the microphone gain control. There may also be too much level from the patch to make adjustment of the transmit level easy. Placing R10 across the transformer allows easy adjustment of the level. It can be set so that when switching from the station microphone to the patch the transmitter microphone gain control does not need to be adjusted. This will also work on the basic 600-ohm 1:1 transformer. Most of the time a 1 kilohm potentiometer - logarithmic if possible - will work well. If not, a linear potentiometer will do. A 2.5kilohm potentiometer may provide better control.

Deluxe Operation and VOX

Using VOX with a phone patch may cause a problem with receive audio going down the line and into the transmit input, triggering the VOX. There may not be enough Anti-VOX adjustment to compensate for this. The usual solution for this problem is to use a hybrid transformer, a special telephone transformer with a phasing network to null out the transmit audio and keep it off the receive line. Most telephones employ a similar transformer and circuit so that callers will not deafen themselves with their own voices. These devices are called "networks" (see figs. 4 and 5).

A network can be removed from an old phone and modified into a deluxe patch, or the phone can be left intact and connections made to the line and handset cords. The line cord should be coupled to a 600-ohm 1:1 transformer to keep the ground off the line. Note, in the network schematics, that the receiver and transmitter have a common connection; when coupling into radios or other unbalanced devices, make this the ground connection.

There may be confusion about terms used in the network. The telephone receiver is receiving the phone line audio, and the transmitter is transmitting the caller's voice. For phone patch use, a telephone receive line is coupled to the transmitter and the transmit line is coupled to the radio receiver. This is a fast way to put together a phone patch and may be adequate for VOX use.

A better patch can be built by using a network removed from a phone or purchased from a local telephone supply house. This approach offers the added advantage of being able to adjust or null the sidetone. The circled letters in figs. 4 and 6 refer to the markings on the network terminal block. These letters are common to all United States networks made by Western Electric (AT & T), ITT, Automatic Electric, Comdial, Stromberg Carlson, and ATC.

To make sidetone adjustable, remove R4 (R5 in European networks) and replace it with R11 (for European networks use R12). The Western Electric Network comes point-to-point wired and sealed in a can; the other networks are mounted on PCBs. To remove R4 from the Western Electric network, the can has to be opened by bending the holding tabs. Don't be surprised to find that the network has been potted in a very sticky, odious paste that has the texture of hot chewing gum and the odor of unwashed shirts. (This material - alleged to be manufactured according to a secret formula - will not wash off with soap and water. The phone company has a solvent for it, but because one of the secret ingredients is said to be beeswax, ordinary beeswax solvents such as gum turpentine, mineral turpentine (paint thinner or white spirit) and kerosene will work.) To remove the bulk of the potting compound, heat the opened can for 30 minutes in a 300 degree F (148 degree C) oven, or apply heat from a hot hairdryer or heatgun. You can also put the can out in the hot sun under a sheet of glass. Don't use too much heat because the plastic terminal strip may melt. Even with a film of compound remaining on it, the network can be worked on.

Using a Patch

For efficient use, a patch should have a telephone connected in parallel with it. This enables the operator to dial, answer, and monitor calls to and from the patch, as well as use the handset for joining in conversations or giving IDs.

One useful modification to the control telephone is adding a mute switch to the handset transmitter. This allows monitoring calls without letting room noise intrude on the line. It's also a good modification for high noise environments, where ambient noise enters through the handset transmitter and is heard in the receiver, masking the incoming call. Muting the transmitter makes calls surprisingly easy to hear. The mute switch can be a momentary switch used as a "Push-To-Talk" (PTT) or a Single Pole Single Throw (SPST) mounted on the body of the phone for long-term monitoring. The switch should be wired as Normally Closed, so that the transmitter element is muted by shorting across it (see fig.4). This makes the mute "clickless." If the monitor phone uses an electret or dynamic transmitter it should still be wired as shown in fig.4.

Transmit and receive levels on the phone line are a source of confusion that even telephone companies and regulatory agencies tend to be vague about. The levels, which can be measured in various ways, vary. But all phone companies and regulatory agencies aim for the same goals; enough level for intelligibility, but not enough to cause crosstalk. The most trouble-free way to set the outgoing level on the patch is to adjust the feed onto the phone line until it sounds slightly louder than the voice from the distant party on the phone line. If the level out from the patch is not high enough, the distant party will ask for repeats and tend to speak louder to compensate for a "bad line." In this case, adjust the level to the patch until the other party lowers his or her voice. The best way to get a feel for the level needed is to practice monitoring on the handset by feeding a broadcast station down the phone line to another Amateur who can give meaningful signal reports. It's difficult to send too much level down the phone while monitoring because the signal would simply be too loud to listen to comfortably. The major problem is sending too little signal down the line.

Coupling the phone line into the radio transmitter is not much more difficult than adjusting a microphone to work with a radio transmitter. Depending on the setup, the RF output indication on a wattmeter, the ALC on the transmitter or even listening to the transmitted signal on a monitor receiver will help in adjusting the audio into the radio transmitter. Phone lines can be noisy, and running too much level into the transmitter and relying on the ALC to set the modulation can cause a fair amount of white noise to be transmitted. Watching the RF output while there are no voice or control signals on the line will help in adjusting for this. VOX operation can alleviate the problem of noise being transmitted during speech pauses.

A hybrid patch used for VOX operation needs to be adjusted carefully for good performance. If it has a null adjustment, this should be set before adjusting the VOX controls. Using a separate receiver/transmitter setup is the easiest to adjust the patch. The phone line should be attached to a silent termination: the easiest way to do this is to dial part of a number; another way to do it is call a cooperative friend. Tune the shack receiver to a "talk" broadcast station or use the BFO as a heterodyne. With the transmitter keyed into a dummy load, set the null adjustment potentiometer R11 (R12 for European phones) for a minimum RF output on the transmitter. Using a transceiver, place an oscilloscope or audio voltmeter across the microphone input terminals and, while receiving a signal, adjust for the lowest voltage. For proper operation, it's important that the phone be connected to the patch during these adjustments since the hybrid relies on all inputs and outputs being terminated.

Reference

1. Julian Macassey, N6ARE, "Understanding Telephones," ham radio, September 1985, page 38

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Rogers, Tom, You and Your Telephone, Howard W. Sams & Co., Inc., Indianapolis, Indiana 46206. ISBN No. 0-672-21744-9.

Bell System Technical Reference 48005; Telephones, January, 1980.

British Standard Specification for General Requirements for Apparatus for Connection to the British Telecommunications Public Switched Telephone Network. BS 6305.

Certification Standard for Voice-Type Terminal Equipment and Connectors, No.CS-01 and No.CS-03, Department of Communications, Government of Canada.

FCC Rules and Regulations: Part 68 - connection of Terminal Equipment to the Telephone Network, United States Government Printing Office, 1982.

End of Text

Fig 1. Simple Phone Patch

```
Tip  \                C5
O----.\---o---------o----||----------O
     .    |         |
     .    |         /
    S1.   |        R6\       Shielded
     .    |         /       Wire
Ring \    |         |    C5 To Transmitter
O---- \--------o----o----||---o------O
          |    |            |
```

```
        |    |              |
        |    |            -----
        |    |             ---
        |    |              -
        |    |         C5
        |    |---------||---o------O
        |                   |
        |                   \ Shileded
        |                 R7/ Wire
        |                   \ To Receiver
        |              C5   |
        ---------------||---o------O
                            |
                            |
                          -----
                           ---
                            -


          ------------------------------

           Fig 2. Basic Phone Patch



  Tip \     R1A
 O---o.\o--/\/\/-----o-----  -------o----o----O
     .          |    |  |      |    | To Tx
     .          |    |  |      |    |
     .          |    )||(      |    |
     .         ---   )||(   C1---   -----O
   S1.       MOV ^ ^ T1)||(     ---     To Rx
     .         ---   )||(      |
     .          |    )||(      | Shielded
     .          |    |  |      | Cable
  Ring\     R1B     |    |  |      |    Common
 O---o \o--/\/\/-----o-----  -------o----o----O
                                   |
                                   |
                                 -----
                                  ---
                                   -

                      -----------------
```

Fig. 3 Improved Phone Patch

```
        ----
Tip   \ |  |    R1A
o----o.\o  o---/\/\/--o-------||(-----------------o
     |.    |         |      |||(
     |.    |         |      |||( 8 Ohms   To RX
     |.    |         |      |||(   Shielded cable
   --|.    |         |      |||(------------o----o
   |  .    |         |   T2 )||            |
   | |-----|     MOV ---     )||(----o----o  |
   | |.             ^ ^     )||(    |    |  |
   | |. S2 Hksw      ---     )||(600 | C1 \  | R10
   --------           | 600  )||(   ---   /<------o
     |.    |         | Ohms |||(   ---   \  | To TX
     |.    |         |      |||(Ohms|    /  | Shielded
 Ring|\    |   R1B    |      |||(    |    |  | Cable
o----o \o  o--/\/\/---o-------  (----o----o--o----o
        |  |                         |
        ----                         -----
                                     ---
                                      -
```

NOTE: S2 Hook Switch is also a polarity reversal switch.

Fig 4. Typical U.S. Network (425B). Note: Circled letters are marked on Network Interconnection block terminals. Component values may vary slightly between manufacturers.

```
                    |-------------------|
                  ..|..................|
                  . |                 .|
Sidetone balancing. |    C3           .|
impedance & loop  . |    | |          .|
compensation. >>> . o----| |-------o   .|
```

```
                   . |    | |       |   .|
                   . |              |   .|
                   . |    |<| VR2   |   .|
                   . o----| |-------o---.|
                   . |    |>|          |.|
                   . |                 |.|
                   . |   R4            |.|
                   . o---\/\/\/-----|  |.|
                   ..|..............|..|.|
                     |              |  | |
                     |       . (GN)|  | |
                 (R) -----)||(------|-------o-----|
                     TA1 1)||(5 TC  |  | |  |     |
           Loop          )||(      |  | |  |     |
TIP    \      Compensation  2)||(6     |  | | ---    |
o-----o.\----------o---------)||(------o  | | ^ ^ RX O
      .          |   (RR) . ||       |  | | ---    |
      .          |          ||      |  | |  |VR60 |
      .          \ 180      ||   C2 --- | |  |     |
      .          / Ohms     ||      --- | |--o-----o
      .  (F) C4   \         ||      |  |   |     |
  S1   .   o--||---|        ||      |  |   |     |
 HKSW  .          ---       . || .     |  |    o     |
      .         ^ ^   -----)||(------o---   \   TX O
      .      VR1 ---   |   3)||(7           S3|     |
      .          |    |TA2 )||(  TB          |     |
 RING \.          |    |   4)||(8       R3    |     |
o----o \-----------o---------)||(---o----/\/\/-o------
    (L2)              | (C)      |        (B)
          ^           |          |
      Hookswitch       ------------
```

Fig. 5. Typical European Network

```
    A  \
   o--o.\---------o----o----o-------|
      .          |    |    |       |
      .          | C4 |    |       |
      .          |   ---   \       |
```

```
    .        |   ---   / R5    |
    .        |    |    \       |
    .        |    |    |       |
    .        |    -----o----)|||
    .        |              )|||
  S1  .         |           )||o------o-----
 HKSW .         |        200  )|||   VR |    |
    .      TX O        Ohms)|||   60 |    |
    .        |              )||(    -----  |
    .        |              )||(     ^ ^   O RX
    .        ---------------|||(    -----  |
    .                  50  )||(60    |    |
    .                  Ohms)||(Ohms  |    |
  B \.                     )||(------o-----
 o--o\-----------------------)||




             ----------------------------
```

Fig. 6. Deluxe Phone Patch

```
               |-------------------|
               |                   |
               |                   |
               |    C3             |
               |    | |            |
               o----| |-------o    |
               |    | |       |    |
               |              |    |
               |    |<| VR2   |    |
               o----| |-------o--- |
               |    |>|          | |
               |                 | |
               |   R4            | |
               o---\/\/\/-----|  | |
               |     ^ or R11|  | |
               |     |-------|  | |
               |       . (GN)|  | |
            (R) -----)||(------|-------------
               TA1 1)||(5 TC  |  | |      |
```

```
                         )||(     |  | |      |
TIP    \    R1A          2)||(6    |  | |   R12/   To TX
o-----o.\--/\/\/---o---------)||(------o  | |      \<---------
      .        |   (RR) . ||      |  | |      /
      .        |        ||      |  | |      |
      .        \ R2      ||   C2 --- | |--o----|---|------
      .        /        ||     --- |   |   |  ---
      .        \        ||      |  |   | R12\   -
  S1   .        |        ||      |  |   |   /<---------
 HKSW  .        ---      . || .    |  |   ---   \   To RX
      .        ^ ^   -----)||(------o---   ---   |
      .     VR1 ---   |   3)||(7        C1 |   |
      .        |   |TA2 )||(  TB        |   |
 RING \.   R1B    |   |   4)||(8      R3    |   |
o----o \---/\/\----o---------)||(---o----/\/\/-o------
    (L2)            | (C)      |        (B)
                   |         |
                    ------------
```

Note: T1 600 Ohm 1:1 Transformer would be between R1 and the line.

Parts List

Item	Description
C1	0.1 uF (see text)
C2	1.5 to 2.0uF (Depending on manufacturer)
C3	0.47 uF Not used in all networks
C4	0.1 uF
C5	2.0 uF 250 Volt Mylar Film (see text)
MOV	130 to 250 Volt MOV (see text)
R1A,B	100 to 270 Ohms (see text)
R2	180 to 220 Ohms (depending on manufacturer)
R3	22 Ohms
R4	47 to 110 Ohms (depending on manufacturer)
R5	1 Kilo Ohm
R6	1 Kilo Ohm (see text)
R7	10 Ohm (see text)

R10 1 Kilo Ohm potentiometer (see text)
R11 200 Ohm potentiometer (see text)
R12 2 Kilo Ohm potentiometer (see text)
S1 DPST or Hookswitch
S3 NC Momentary switch (see text)
T1 600 Ohm 1:1 transformer
T2 600 Ohm primary. 600 Ohm and 8 Ohm secondary (see text)
T3 Network Transformer
VR1 Silicon Carbide Varistor or Back-to-back Zener
VR2 Silicon Carbide Varistor or Back-to-back Zener
VR60 Silicon Carbide Varistor or Back-to-back Zener

END

--
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***** 4101 0
Received: from USC-ECLC.ARPA by BBNCCA ; 14 May 85 16:11:15 EDT
Received: from ECLB by ECLC with ECLnet; Tue 14 May 85 13:09:52-PDT
Received: from RUTGERS by USC-ECLB; Tue 14 May 85 13:08:10-PDT
Date: 14 May 85 16:04:49 EDT
From: *Hobbit* <AWalker@RUTGERS.ARPA>
Subject: Wiring
To: Telecom@RUTGERS.ARPA
cc: Hobbit@RUTGERS.ARPA

It's rather difficult to send out schematic diagrams to a network of people using regular old ascii terminals, but since wiring inside most fones is pretty standard, a description should do the trick. This applies to *all* WE phones and ITT phones that use the standard dial/ringer/network block/ handset configuration. I've rebuilt lots of these suckers, and can confidently say that they're all the same.

Everything basically talks to the network block. The network block contains the ringer capacitor, the induction coil that handles the handset, and very little else save some spare screw terminals. Left to itself, the network block can function as a standard line load [it looks electrically like a phone] when a line is connected across RR and C. These are the inputs to the coil. The ringing capacitor is indeed across A and K as someone mentioned. In addition, older blocks have a smaller capacitor across F and RR, to decrease sparking across rotary dial contacts.

Handset:
Green and White: Earpiece leads. These connect to net R and GN respectively.
Black and Red: Mike leads. Connect to net B and R respectively.

Ringer [two-winding]:
Black and Red: To line. Connect to L1 and L2 [or whever your line comes in].
Grey and Grey/red [these may vary; they are the "other two" wires, anyway]:
    Connect to net A and K. The circuit thus formed runs from one side of the line to one ringer winding, thru the A-K cap, thru the other ringer winding, to the other side of the line. This configuration has infinite DC resistance, but picks up the AC ring voltage.
Ringer [one-winding, rare]: Connect the single winding [two wired] in series with the A-K capacitor somehow, and this whole thing across the line as above.

Rotary dial:
Blue and Green: Interruptor. Connect to net F and RR.
White [2]: Earpiece suppress. Connect to net B and GN if desired.

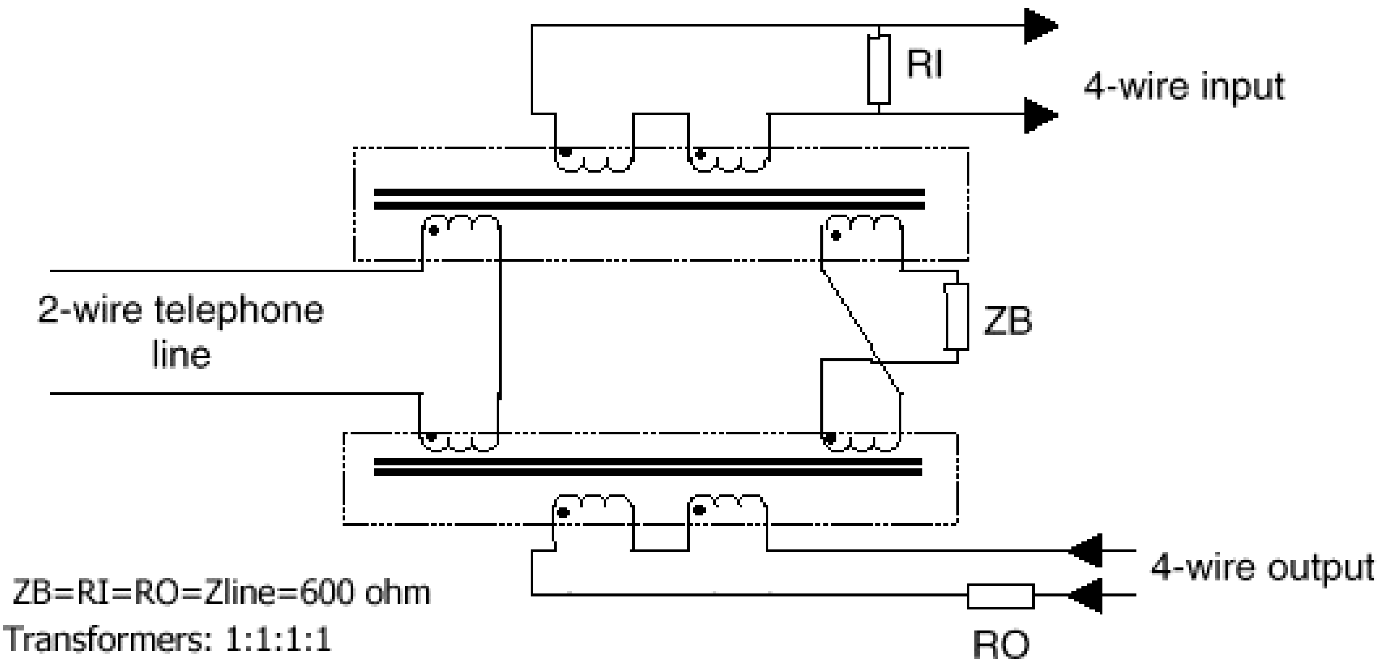
Touch-tone dial:
Green: + Line in. Connect to net F.
Black: + Line out. Connect to net RR.
Org/Blk: - Line in. Connect to net C.
Red/Grn: output common. Connect to net R.
Blue: output. Connect to net B.
*Note: the above 5 connections will give you a "bare-bones" dial configuration without features. Features are mike disconnect, earpiece suppress, etc which are done simply by routing leads to these through the extra contacts on the dial instead of directly. If you want the features, modify the wiring as follows. If your network block doesn't have the S and T terminals, you have an old one designed for rotary dials, and you'll have to do kludges.
Earpiece mute:
Move Handset lead at White to net S. Also connect Dial White-Blue to net S.
Connect Dial White to net GN. This routes the earpiece through the dial switching mechanism which resistifies the circuit on button press.
Mike disable:
Move Handset Red to T. Also connect Dial Red to T. This completely disables the mike on button press. Make sure Dial Red-Green is connected to R if you do this mod!

Hookswitch:
You'll find many variants of this in different units; some configurations switch both sides of the line, some only one, some switch out the ringer when off-hook [which isn't necessary, really]. The following should work:
Yellow: Connect to net L2. This is where the line enters.
Brown: Connect to net C.
Green: Connect to net L1. This is the other side of the line.
White: Connect to F. This is switched line power to the dial and the rest.
Red: Connect to R. This, with Black, is shorting earpiece mute.
Black: Connect to GN.

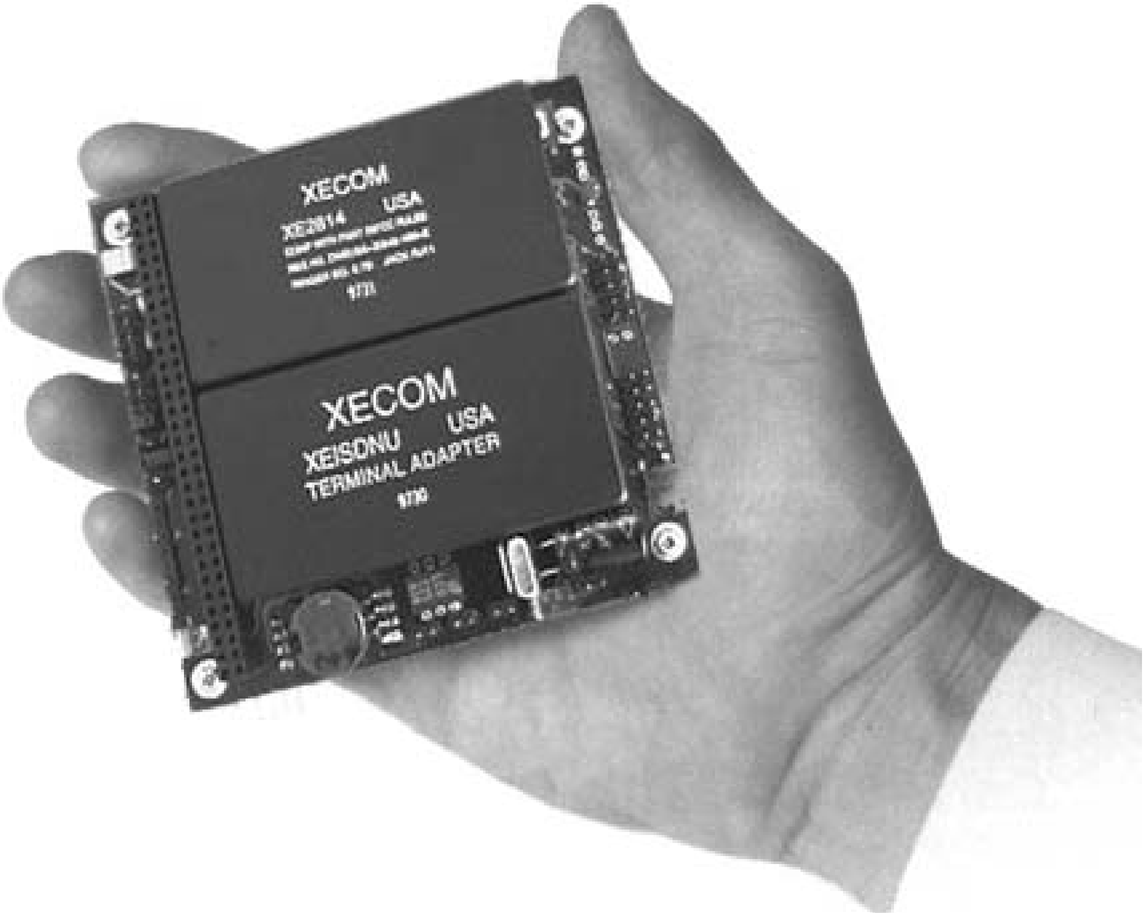
Line in:
Green and Red connect to L1 and L2. Try one polarity; if the touchtone dial doesn't work, then flip them. Rotary dials, of course, don't matter.

If someone sees errors in this, please notify the list with the correction...

_H*


RI
4-wire input
2-wire telephone
line
ZB
4-wire output
ZB=RI=RO=Zline=600 ohm
Transformers: 1:1:1:1
RO

Welcome to
Xecom New Website!



Click here to go to Xecom New Website.

CLARE An IXYS Company

GO

NEWS

Clare expands AC-Power Switch Family by Offering 600V & 800V Devices for Improved Power Control & Efficiency

February 2, 2006 - Beverly, MA - Clare, Inc., an IXYS company (NASDAQ: SYXI - News), today announced that it has expanded its family of AC-Power Switches with the CPC1962 and CPC1972, offering designers peak blocking voltages of 600 and 800 respectively. The devices use optical-coupling inputs, with dual power Silicon Controlled Rectifier (SCR) outputs. In addition, tightly controlled zero-cross circuitry ensures switching of 120 to 240Volt AC loads without the generation of transients. Read more

Ninety-Eight Percent of Clare's Standard Products Are RoHS Compliant

November 11, 2005 - Beverly, MA - Clare, Inc., an IXYS company (NASDAQ: SYXI - News), today announced that greater than ninety-eight percent of all standard ICs offered are RoHS compliant, and that plans are on schedule to have all of the company's standard ICs fully compliant by early next year, well ahead of the July, 2006 deadline. Clare's early compliance is to ensure that customers in turn meet their lead free initiatives. Read more

Clare Releases Latest Single Pole OptoMOS® Solid State Relay with built-in Current-Limiting Circuitry: Saves Board Space

Beverly, MA - May 10, 2005 - Clare, Inc., an IXYS company (NASDAQ: SYXI - News), announces the immediate production release of the CPC1510, a current limiting, normally open, 1-Form-A optically isolated Solid State Relay (SSR) that both replaces electromechanical devices and enhances the performance of wireline-interface applications. Read more

CPC1510 Website Page

CPC1510 Data Sheet

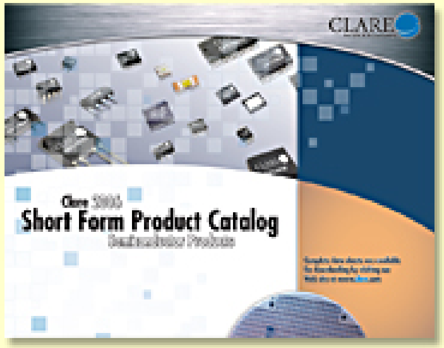
Clare Releases Latest LITELINK™ III Phone Line IC with New Caller ID and Ringing Detect Features for the Growing VoIP Gateway Market

Beverly, MA - March 8, 2005 - Clare, Inc., an IXYS company (NASDAQ: SYXI - News), announced the immediate availability of the CPC5622A, the newest member of the LITELINK™ III Phone Line Interface IC family for direct access arrangements (DAAs) in telephony applications. Read more

CPC5622A Website Page

CPC5622A Data Sheet

Available Online Collateral



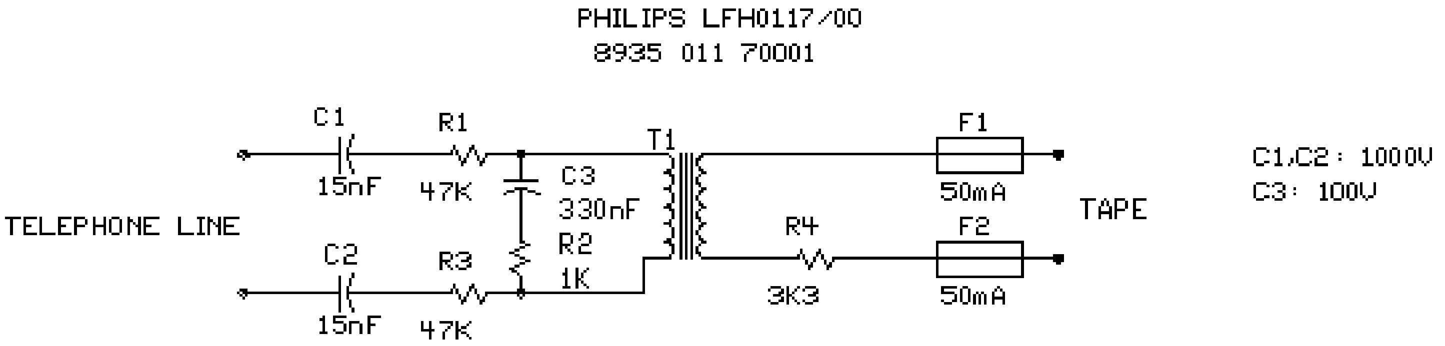
Download Short Form Catalog 2006 | Download Clare Product Briefs | Clare Advertisements

Solid State Relay Cross Reference

Sales and Marketing/ Customer Service/ Webmaster

Clare, 78 Cherry Hill Drive, Beverly, MA 01915, Phone: 978-524-6700, Fax: 978-524-4700

Minimum Site Requirements: *Internet Explorer 5.5*, *Netscape 6.2*, *and Flash 5 player*


PHILIPS LFH0117/00
8935 011 70001
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C1
15nF
R1
47K
C2
15nF
R3
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C3
330nF
R2
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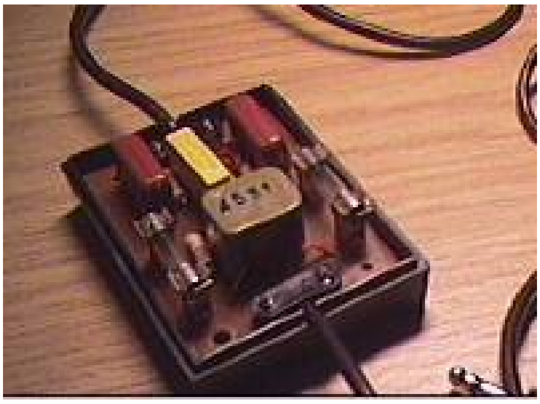
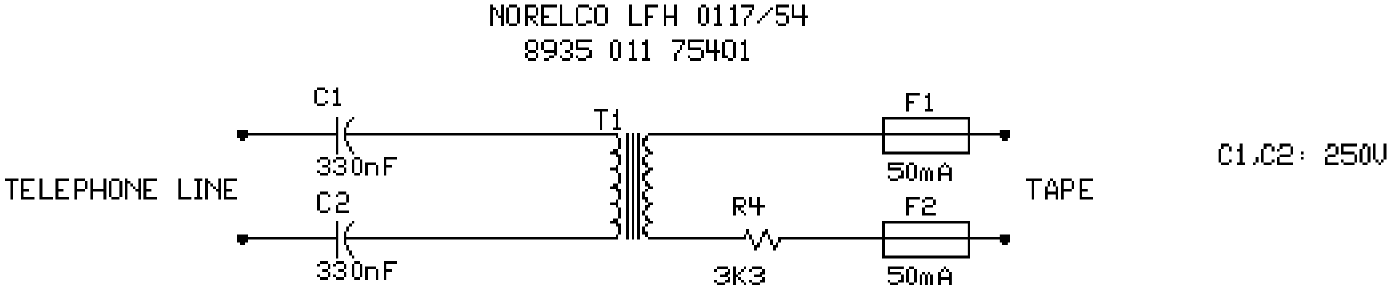
C1,C2: 1000V
C3: 100V




NORELCO LFH 0117/54
8935 011 75401
C1
330nF
C2
330nF
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R4
3K3
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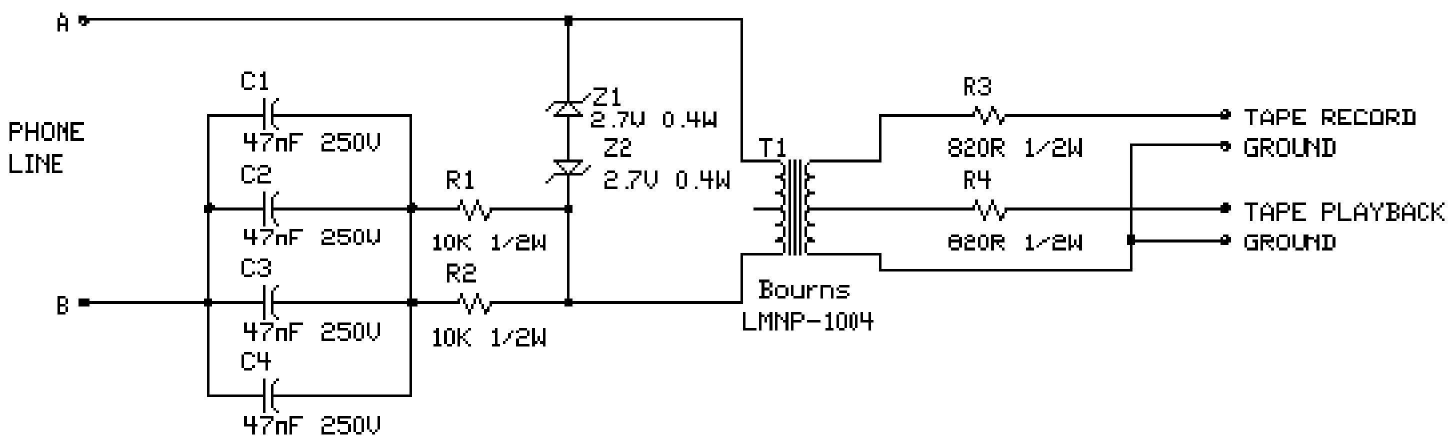
F2
50mA
TAPE
C1,C2: 250V


A
B
PHONE
LINE
C1
47nF 250V
C2
47nF 250V
C3
47nF 250V
C4
47nF 250V
R1
10K 1/2W
R2
10K 1/2W
Z1
2.7V 0.4W
Z2
2.7V 0.4W
T1
Bourns
LMNP-1004
R3
820R 1/2W
R4
820R 1/2W
TAPE RECORD
GROUND
TAPE PLAYBACK
GROUND

Bowden's Hobby Circuits

A small collection of electronic circuits for the hobbyist or student. Site includes over 100 circuit diagrams, links to related sites, commercial kits and projects, newsgroups and educational areas. Most of the circuits can be built with common components available from Radio Shack or salvaged from scrap electronic equipment. Most all of the circuits have been built and tested and believed to perform as described, however possible mistakes may be found.

Additions and Corrections (01/06/06)

Digital/Computer

- 16 Bit PC Serial Port Receiver (CMOS)
- 24 bit ISA card
  Installs into your computer. Parts, plans, schematics and programmng available. Also may be purchased as a kit.
- 32 Bit Serial Receiver (57.6 K Baud TTL & CMOS)
- Parallel Port Relay Interface Circuit
- Reading Data From The Parallel Port
- 1 Second Time Base From Crystal Osc.
- 32.768 KHz Oscillator Using A Common Watch Crystal
- Digital Electronic Lock

Telephone

Circuits

- Use Old Telephones as Intercom
- Telephone In-Use LED Indicator
- Telephone In-Use Relay Circuit
- Telephone Ring Generator Using 60Hz Power Transformer
- Telephone Ring Generator Using Switching Power Supply
- Telephone Audio Interface

LED Circuits

- 40 LED Bicycle Light (555 Timer - 6 volt)
- 28 LED Clock
- 72 LED Clock
- 10 to 100 LED Sequencer Using a Matrix
- PIC Digital Clock Timer
- Binary Coded Decimal (BCD) Clock
- Digital LED Clock and Timer
- Astable Multivibrator
- Expandable 16 Stage LED Sequencer
- 16 Stage Bi-Directional LED Sequencer
- 10 Stage LED Sequencer

Power Circuits

- Simple Variable Voltage Source
- LM317 Variable Voltage Regulator
- LM317 Regulator With Pass Transistor
- Variable 3 - 24 Volt / 3 Amp Power Supply
- Variable Voltage and Current Power Supply
- 2 Cell Lithium Ion Battery Charger
- 2 watt switching power supply
- Low Power DC to DC Converter
- Constant Current Battery Charger
- 120VAC Lamp Chaser Using Solid State Relays
- 120 VAC Lamp Dimmer (full wave SCR)
- Varying Brightness AC Lamp
- 12 Volt Lamp Fader (Automatic)
- 1.5 Hour Lamp Fader (Sunset Lamp)
- 12 Volt Lamp Dimmer (using a pot)
- Interfacing 5 volt CMOS to 12 volt loads
- Low Voltage , High Current Time Delay Circuit
- High Current Regulated Power Supply
- Thermostat for 1KW Space Heater (SCR controlled)

Miscellaneous

- Triangle and Squarewave Generator
- Transistor Schmitt Trigger Oscillator
- Descrete Set/Reset Flip Flop
- Discrete Bistable Flip Flop

- 18 Stage LED Sequencer
- 9 Second LED Relay Timer
- 9 Second Digital Readout Timer
- Two Transistor LED Flasher
- 1.5 Volt LED Flashers
- LED Photo Sensor Circuit
- Fading Red Eyes
- 12 Volt Lamp Fader (LEDs or incandescent)
- LED Battery Condition Indicator
- 8 Stage LED VU Meter
- Battery Equal Charge Indicator
- IR Remote Control Tester
- AC Line Powered LEDs
- Line Powered White LEDs
- LED Traffic Lights

Analog / RF Circuits

- LED Decibel Meter
- Whistle On - Whistle Off
- Long Loopstick AM Radio Antenna
- Micro Power AM Broadcast Transmitter
- FM Beacon Transmitter (88-108

- Toggle Switch Debounced Pushbutton
- High Current MOSFET Flip Flop With Debounced Pushbutton
- Monostable Flip Flops (one shot)
- Ignition Coil High Voltage Circuit
- Ignition Coil Buzz Box
- Capacitor Discharge Ignition Circuit
- Generating Long Time Delays
- Flashing Neons (NE-51 / NE-2)
- Sequencing Neons (NE-51 / NE-2)
- 555 Tone Generator (8 Ohm Speaker)
- Generating Negative 5 Volts from 9 Volt Battery
- Touch Activated Lamp
- Game Show Who's First Indicator Lights
- Salt Water Battery
- Transistor, Diode, IC outlines

Circuits Controlling Relays

- Relay Toggle Circuit Using a Dual 555 timer (556) - This is probably the best toggle circuit.
- Relay Toggle Circuit Using a Single Transistor and Push Button
- Relay Toggle Circuit Using a MOSFET and Push Button
- CMOS Toggle Flip Flop Using Push Button (CD4013) - This is fairly complex and not the best toggle circuit.
- CMOS Toggle Flip Flop Using Laser Pointer
- 555 Timer Monostable Circuit Using Pushbutton
- Generating a Delayed Pulse With a dual 555 Timer
- Light Activated Relay (toggled)
- Photo Electric Street Light
- Power-On Time Delay Relay Circuit
- Power-Off Time Delay Relay Circuit
- Electronic Thermostat Relay Circuit
- AC Line Current Detector
- Pinewood Derby Finish Line Lights
- Pinewood Derby Finish Line Using a Computer - Scores times and places.
- Controlling relays with logic voltages
- ....

MHz)

- Op-Amp Basics - The text information for the basic Op-Amp operation, 2nd order filters and bandpass filters was obtained in part from the paper back book "Design of Active Filters, With Experiments" by Howard M. Berlin, 1977. The book is out of print but possibly can be found at used book stores, or through Amazon.com
- Active 2nd Order Filters
- Bandpass Filter (Single Op-Amp)
- Low Power Op-Amp - Audio Amp (Intercom)
- Crystal Radio Circuits
- Simple Op-Amp Radio
- Low Frequency Sinewave Generator
- Simple 3 Transistor Audio Amp (50 milliwatt)
- Improved 3 Transistor Audio Amp (80 milliwatt)
- RC Notch Filter (Twin T)
- Analog Milliamp Meter Used as Voltmeter

Semiconductor Data Sheets

TI Semiconductors (search)	National Semiconductors (search)	Motorola
NTE	www.datasheets4u.com	

Java Script Calculators

Resistor Color Code Calculator - Graphical resistor color code calculator by Danny Goodman. Uses pulldown menus and a realistic picture of a resistor.
Another Resistor Color Code Calculator - This one uses check boxes instead of pulldown menus and also calculates the equivalent value of two resistors in parallel. My own creation.
Ohm's Law Calculator - Java Script to solve Ohm's Law for Voltage, Current, Resistance and Power. Enter any two unknowns and solve for the other two.
Voltage Divider Calculator - Solves voltage, current, and power dissipation problems for two element resistive voltage dividers.
L or C Reactance Calculator - Java Script to calculate capacitive or inductive reactance and resonant frequency. For ideal devices only, resistance not included.
Allen Newman's Impedance Calculator - Solves passive series RLC networks, for reactance, impedance and phase angle.
RC Time Calculator - Java Script to solve R and C values for given values of time or instantaneous voltages.
RL Time Calculator - Java Script to solve R and L values for given values of time or instantaneous current.
555 Timer - Frequency and Time Interval Calculator - Calculates positive and negative time intervals for the 555 timer based on R and C values. Also contains descriptions and operation of each input and output of the timer and schematics for the two basic modes of operation (monostable or "one-shot" and astable or "rectangular wave oscillator"). Also contains a pictorial diagram of the timer connected as a LED flasher and a table of connections for the 556 timer (dual 555 timer).
LED Series Resistor Calculator - Finds the series resistance needed for various series LED combinations and supply voltages.
The Electronics Calculator Website - Several calculators for Ohm's law, capacitor or inductor impedance, tuned circuits and RC time constants.
Several JavaScript Calculators by John Owen. - Audio op-amp filter, Op-amp circuit, Decibels, Zener Diodes and more...
Air Core Inductance Calculator - Calculates number of turns and layers needed for air core inductors.
Gregorian Calendar - Displays any month from Oct 1582 forward.

Links to Other Hobby Electronics Sites and Useful Information

Hobby Projects to Build, That Work Color Organ, Wattmeter, 12 volt to 120VAC inverter and more.
Don Klipstein's LED Website - Lots of useful LED information, FAQs, and sources for the brightest and most efficient LEDs.
Circuit Archive - University of Washington Circuit Archive, lots of good circuits and links.
EDUCYPEDIA - The educational encyclopedia (Electronics section)
Tomi Engdahl's Electronic Info - Links to a wide variety of analog and digital circuits.
Imagineering On-Line Magazine ,The Design Corner, over 100 circuits in pdf format.
Tom Loredo's Electronics Bookmarks - Many resources for electronics hardware and software.
Harry's Homebrew Homepage - For building amateur radio equipment. Antennas, Receivers, Transmitters and other useful circuits.
Links for FM Transmitter Kits, Circuits, Electronics ...
Electronics Links and Resources -Links to circuits, components, educational sites and more..
Tony's Website - R/C Gadgets and electronic circuits for the hobbyist.
Steve Walz's FTP Site - FTP Resource Site, 1000 Files in 50 Directories.
Wenzel & Associates (Circuits) - Technical Library, Hobby Circuits.
Samuel M. Goldwasser Homepage - Silicon Sam's Technology Resource - FAQs, Links, Troubshooting & Repair, Laser info, Circuits.
Beyond Logic - Information on the PC Parallel, Serial and USB ports.
How Stuff Works - Interesting site on how things work, but you will have to clear your screen of many pop up ads.
Deep Cycle Battery Frequently Asked Questions
www.saroff.com Has some useful search engines for locating parts and data sheets.

Wonton Soup Recipe This is copied recipe from a local Chineese take out place. Costs $4 to buy it, but only $2 to make it.

Electronics Educational Sites

- PIC Microcontroller Tutorial A very good introduction to PIC micros that includes 13 tutorial pages to get you started programming PICs. You will also need a programmer to load the finished program (.HEX file) into the PIC. The DOS programmer software and schematic for the programmer can be downloaded from David Tait's PIC archiveThe file you need is PIC84V05.ZIP. The file contains DOS software and programmer schematic that will work with the PIC16C84, PIC16F84 or the newer PIC16F628 which can be purchased on e-bay for about $2.68 or obtained from www.glitchbuster.com

- Lessons In Electric Circuits - A free series of textbooks on the subjects of electricity and electronics.
- Play Hookey Basic ideas about op-amps, analog circuits, optics, computers and digital logic.
- Alex's Electronic Resource Library An online guide to useful electrical and electronic information.
- Electronics Tutorials A good comprehensive site with detailed examples and book recommendations.
- Basic Electronics Tutorial (Iguana Labs)
- The Art of Electronics (Purchase the book) 1125 large format pages, 80 component-selection

Commercial Electronic Sites, Kits and Projects

- TheLEDLight.com Luxeon LEDs, LED bulbs, fixtures, flashlights, lanterns, clusters, arrays, and more...
- lsdiodes.com Basic collection of bright LEDs in several colors at low prices. Brightness ranges from 4000 mcd to 12,000 mcd for the white LEDs.
- Lighting Components LED Corp. LED lamps, light bulbs, clusters and lighting components.
- JDR Microcontrollers Books, Kits, Test Equipment and more.
- Alltronics Electronic Kits
- KitZ Electronic Kits
- Electronics USA - LED Digital and Binary Clocks, LED Timers (Up and down counting), LED flashlights, and a few other items, as kits or assembled.
- Centerpointe Electronics Store
- Hobbytron - The largest selection of fun electronics and toys!
- Almost All Digital Electronic Kits
- Hallbar Electronic Kits and Projects
- Kits-R-Us Electronic Kits
- Microchip Technology Inc. (PIC) Controllers
- Mental Automation's products ECAD tools: schematic capture, printed circuit board design/ layout, autorouting, circuit simulation, SPICE, and CD ROM Encyclopedia of Electronic Circuits, all running under Windows.
- Express Printed Circuit Boards Download PCB software to layout your project, E-mail the resulting file and receive delivery of finished circuit boards via Federal Express in 3 working days.
- PCB Express 1-Day Delivery, Economical, Easy to order and track, Lot charge as low as $80, E-Mail Help
- C. Crane Company LED Flashlights, windup radios, and other specialty items.

- **A few open links are available in this section.** Up to 25 words of text describing your site can be included.
  email if interested

tables,1500 figures, extensive practical advice, back-of-the-envelope techniques, exhaustive 4000-entry index.

- CircuitMaker Student Version (3.4M) Get ahead in class with the FREE CircuitMaker Student Version. You will be able to build and simulate circuits in a fraction of the time it takes in the lab.
- DC Circuits Department of Physics, University of Guelph.
- Elements of AC Electricity
- Electronics For Beginners

Newsgroups

- Science.Electronics.Basics
- sci.electronics.components
- sci.electronics.design
- sci.electronics.repair

Electronic Component Suppliers

- Mouser Electronics
- DIGI-KEY Corporation
- Allied Electronics
- Surplus Traders (Solar panels)
- Jameco
- Hosfelt Electronics
- Radio Shack
- Electronix Express - Components and test equipment for schools, colleges and industry. Very nice site.
- Dan's Small Parts and Kits -Good selection of small components at low cost.
- Bill's Surplus Parts For Sale- Resistors, capacitors, semiconductors and a few other items. I don't have a large

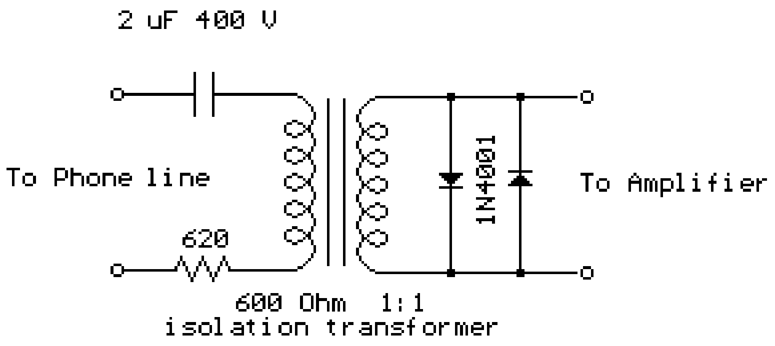
•	inventory, just more than I need.

Go to top

Having trouble building a circuit similar to something on this site?

Try posting a message with circuit details to one of the electronic newsgroups, either sci.electronics.basics or sci.electronics.design Many readers of those groups will offer ideas and a few specifics at no charge. If you need more detailed help and follow up advice, maybe I can help.

Send a detailed description of your circuit and objective. If it is within my expertise and relates to the projects on this site, I will work with you via email to help solve the problem. I can also test and refine the circuit on a vector board for a nominal fee.


2 uF 400 V
To Phone line
620
1N4001
To Amplifier
600 Ohm 1:1
isolation transformer

Consumer Electronics Design

EE498 -- Autumn 1995

Caller ID / DTMF Audio Decoder Sources & Links

Sources

- Weeder Technologies, Telephone Caller ID Pro-Kit documentation
  - Caller ID schematic part 1
  - Caller ID schematic part 2
- Popular Electronics Magazine, September 1995
  - DTMF Decoder/Logger schematic
- Datasheet: MC145436, Motorola Semiconductor
- Datasheet: ISD1000A, Information Storage Devices
- MicroChip PC16C55
- WWW
  - CallerID Archive
  - Caller ID FAQ
  - Motorola Semiconductors
  - Information Storage Devices
  - DTMF Info
  - DTMF FAQ

Other WWW - Links

- voice chips
- EE498 HomePage

Back to Project HomePage

12/15/95

Jason Blomenkamp, Jim Mitchell, Robert Ober

A-TILAAJAN TUNNISTUS, ELI CALLED-ID SUOMESSA:

Softapohjainen A-tilaajan tunnistus. Toimii digitaalikeskus-
numeroissa (muista pyytää puhelinyhtiötä kytkemään numeroosi
a-tilaajan tunnistus).

Palikka liitetään LPT1 porttiin. Palikka koodaa dtfm-signaalin
digitaaliseksi, joka luetaan lpt1-portin kautta mikrolle.

CALLER ID PALIKKA: (osien hinta muutamia kymppejä)

```
                             +---------+--------+--+-D1-<+5V
>-C1--R1-+---------+    MT8870C-1    |       |      | |
      |      |    ______________  |       C3     | +-D2-<+5V
      |      +----1        18--+       |      | |
>-C2--R4-?--+--+--------2        17------------+      | +-D3-<+5V
      R3 R2 +-R5-----3        16------R6----+      |
      +--+-----------4        15---->StD          |
                5        14---->Q4           |
                6        13---->Q3           |
          +--------7        12---->Q2           |
          +-X1-----8        11---->Q1           |
             +---9        10---------------------+
             |   ______________
            Vss (maa)

     R1=100k       C1=10nF    C1 ja C2  kytketään
     R2=60k        C2=10nF    puhelinlinjaan
     R3=37,5k      C3=100nF
     R4=100k       X1=3.579545 MHz +/-0.1%
     R5=100k       D1-D3=BAT42 tai BAT50 yms
     R6=390k
      R=+/-1%        C=+/-5%
```

PIIRI	LPT1	piikki	Selitys
StD	Act-piikkiin	10	(data valmiina luettavaksi)
Q4	Busy-piikkiin	11	(databitti 4)
Q3	PEnd-piikkiin	12	.
Q2	Sel-piikkiin	13	.
Q1	Err-piikkiin	15	(databitti 1)
+5v D1	data-piikkiin	2	+5v jännite piirille
+5v D2	data-piikkiin	3	+5v jännite piirille
+5v D3	data-piikkiin	4	+5v jännite piirille
Vss	Grd	25	Grd

Virta syötetään datalinjasta. Diodit rinnan +5v:lle
(muista kytkeä oikein päin).

Mukana seuraa dos ja windows softa, sekä testi, joilla
voi tarkistaa kytkennän näpyttelemällä omaa puhelinta.

CID .EXE dosversio: Caller ID
CIDP .EXE dosversio: Testaa kytkentä
CIDWIN .EXE winversio: Caller ID
CIDPWIN.EXE winversio: Testaa kytkentä

Dos-Cid on testikäyttöä lukuunottamatta käyttökelvoton, koska
ohjelma ei tee muuta kuin lukee DTFM-dekoderia.

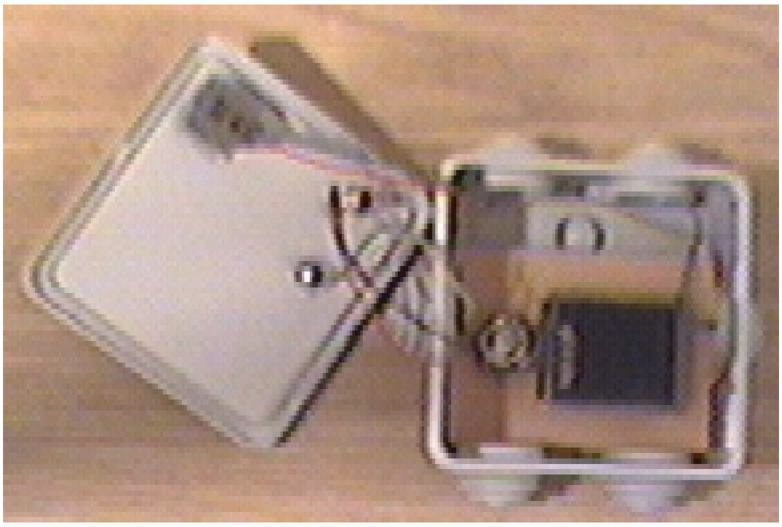
Win-Cid on käyttökelpoinen ohjelma, koska se toimii 'taustalla'
ja näyttää soittajan numerot. Koska win saattaa varata koneen
huomion niin täydellisesti, että IRQ7 ei ehdi palvella CID'iä
silloin kuin 'dataportissa' on tulevan soiton aloituskoodi,
jää osa soitoista näyttämättä.

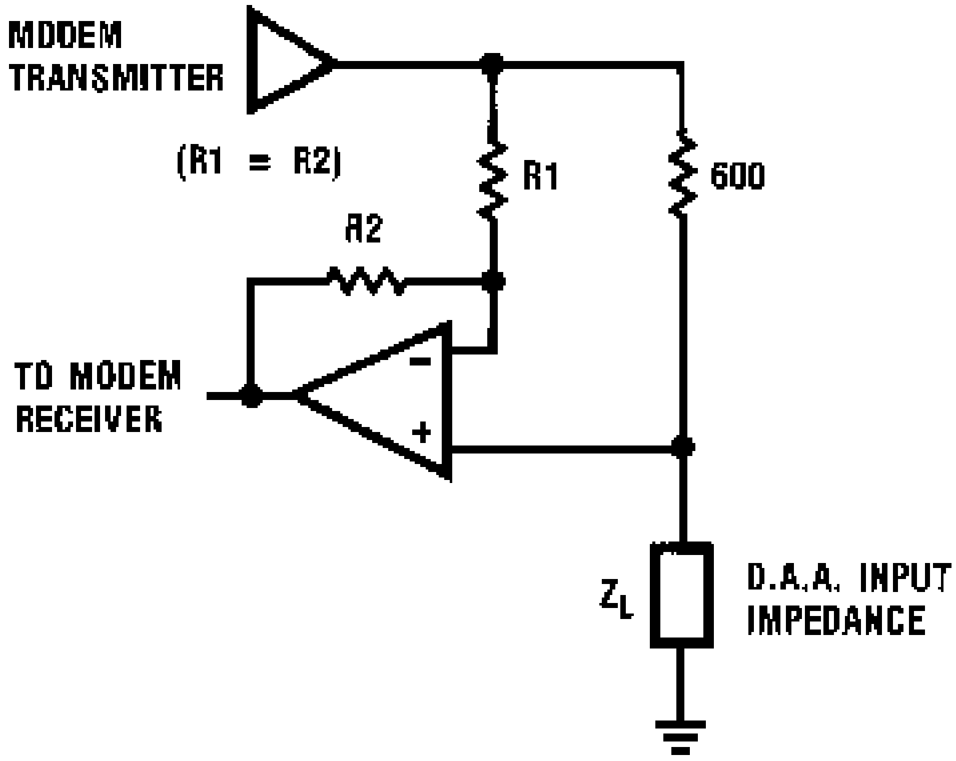
Ohjelman toimintaperiaate:
Act-piikin kautta välitetään IRQ7 linjalla keskeytys
keskeytysohjaimelle, joka ilmoittaa siitä prosessorille.
Prosessori lukaisee LPT1 portin statusrekisterin. Jos
validi aloituskoodi löytyy, estetään keskeytykset ja
aletaan pollaamaan LPT1 statusrekisteriä. Dftm-koodi
tulee n.300-500ms aikana ennen ekaa soittoa, jonka ajan
kone on valjastettu palvelemaan VAIN keskeytystä.

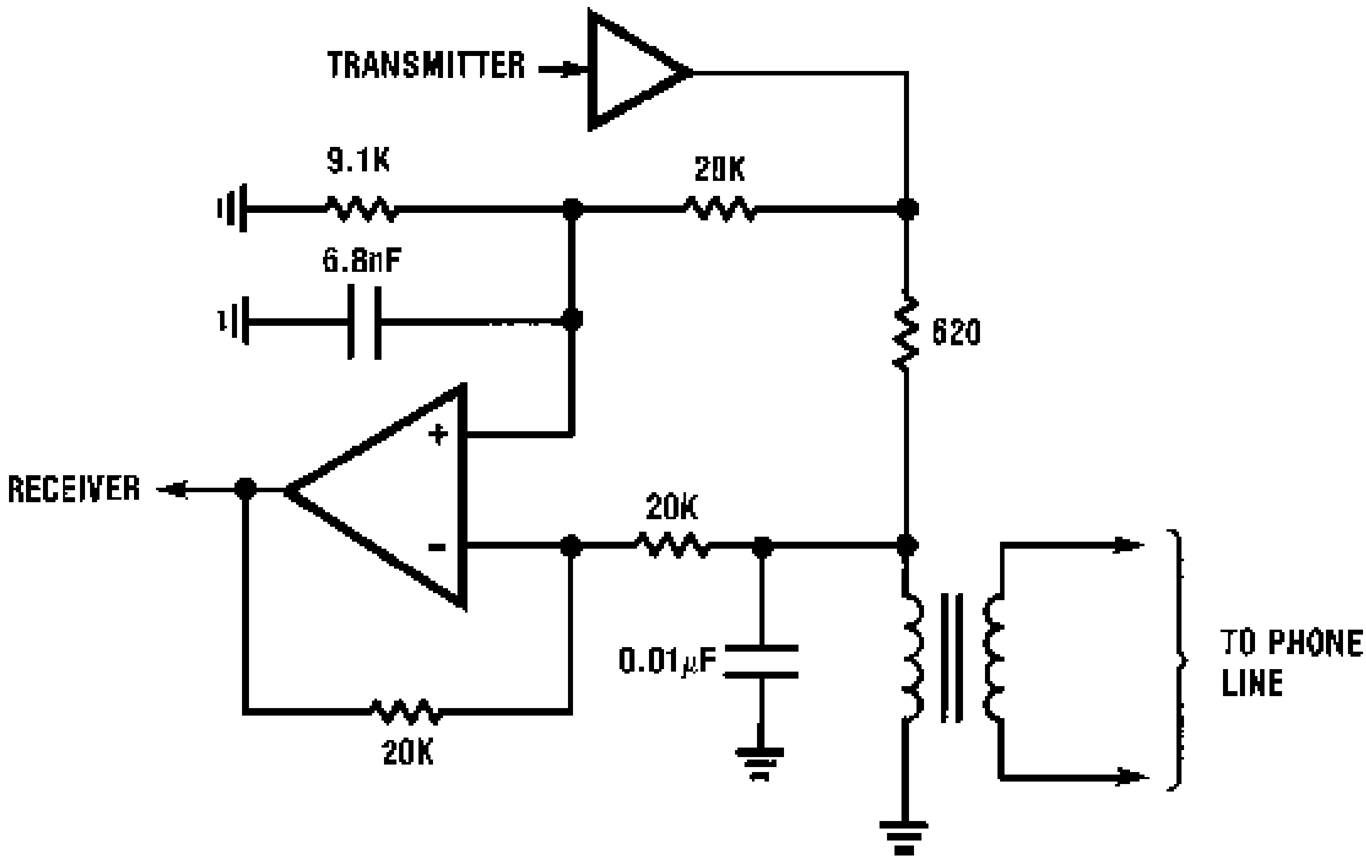
Mukana seuraava kytkentä on vain yksi mahdollisuus.
Softalle on ihan sama, miltä piiriltä se datan saa,
kunhan kytket sen ylläolevan mukaisesti.

Ohjelmat on tehty vain huvin vuoksi ja omaksi iloksi.
Jos niistä on Sinulle jotain iloa, voit käyttää niitä
ilman huonoa omaatuntoa.

By Per Elfstrom email: elfstrom@siilinjarvi.fi




MODEM
TRANSMITTER
(R1 = R2)
R1
600
R2
TO MODEM
RECEIVER
D.A.A. INPUT
IMPEDANCE
Z_L


TRANSMITTER
9.1K
20K
6.8nF
620
RECEIVER
20K
0.01μF
20K
TO PHONE LINE

ePanorama.net

Gyrator circuit: simulate large coils electronically

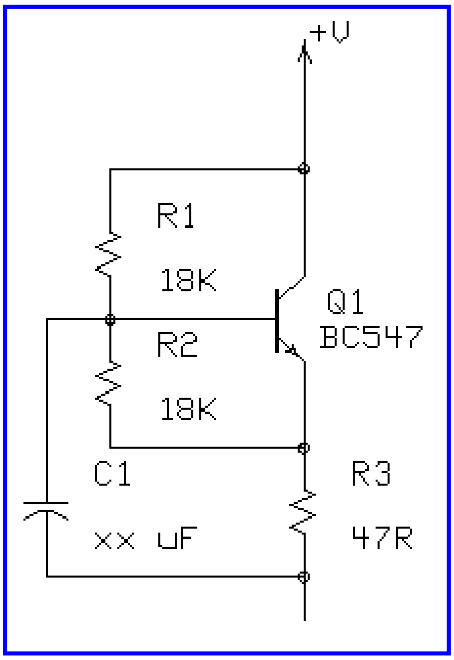
Sometimes coils with large inductance is needed in telecommunication systems and filters. Coils with inductance of many henries are not usually very useful in modern small electronics circuits. To overcome there is one solution which can be used in many applications: simulate the behauvior of coils using electronics.

What's a gyrator ?

A gyrator converts an impedance into its inverse. This lets you replace an inductor with a capacitor, a couple op amps, and some resistors. A very handy gadget to have, especially if you're trying to put large value inductors into a very small package.

Simple circuit

This simulation of coils can be done in many ways, but the circuit below is maybe one of the simplest and most inexpensive.



This circuit is one transistor gyrator circuit which can be easily adjusted to simulate different inductance coils just by changing the value of capacitor C2. The value of inductance the circuit simulates is approximally simulates has value of 1 henry for every 2 microfarads in C2. So if you want to make 5 henry coil simulator, you just use 10 uF electrolytic.

The circuit needs some current passing through it to operate correctly. The circuit is suitable fo example to be used as telecommunication holding coil or line current feeding coil.

The idea for this circuit is from P. Strict and is was published in Electronics World + Wireless World magazine September 1993 page 754.

Last modified: May 28, 2002

JK Audio

Telephone Audio Interface Products

Featured Articles:

[Radio World Magazine User Report on Innkeeper 4]

[Radio World Magazine User report on Broadcast Host]

[BE Radio Magazine Field Report on ComPack]

[Radio World Magazine User Report on Innkeeper PBX]

[Inkeeper PBX Wins Radio World 2003 Cool Stuff Award]

[ComPack Connects for Field Work]

[Radio World Magazine User Report on ComPack]

[Radio World Magazine Product Evaluation of ComPack]

[Radio World Magazine Review of CellTap and Daptor Two]

[Radio Guide Magazine Article - Phone Line Basics Revisited]

[Daptor One wins BE Radio NAB 2001 Pick Hit Award]

[Radio World Magazine User Report on Daptor One]

[Radio World Magazine Product Evaluation of Innkeeper 1]

[Radio World Magazine Product Evaluation of Inline Patch]

[Radio World Magazine Buyers Guide Report on RemoteMix 3 and Sony MD]

[Radio World Magazine reviews RemoteMix C+ and RemoteMix 3]

[Radio World Magazine User Report of Inline Patch]

[What Exactly is "Telephone Quality" Audio?]

[Radio World Magazine reviews Telephone Handset Interfaces]

[QuickTap Recommended by Computer Telephony Magazine]

Hot News:

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Visit JK Audio at NAB 2006 in Las Vegas, April 22-27.
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NEW Four IFB - Road Tough IFB for Television Field Production! [more->]

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NEW Four IFB - Road Tough IFB for Television Field Production! more->

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Product Applications:

Radio Applications:
Products for reporting, interviews, and remote broadcasts.
more->

AV/Telecom Applications:
Products for voice prompts, presentations, conference & PA interface. more->

TV / Film Applications:
Products for IFB feeds and telephone audio on set.
more->

Computer Applications:
Products for webcasting, soundcard to phone audio & noise reduction
more->

Hot Products:

Broadcast Host
Turns your desktop into a professional broadcast center.
more->

Innkeeper PBX
Converts your multi-line PBX type telephone system into a professional, talk show console.
more->

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Hardware for Audio and Video Conferencing

Below are some pointers to hardware commonly used for MBone-style events. Listing does not imply endorsement.

- Desktop Videoconferencing Products
- CAIRN Conferencing hardware recommendations

- PC Hardware for High-Performance Network Experiments

- Cameras
- Scan converters
- MPEG capture cards
- Firewire (IEEE 1394) cards
- Whiteboards
- Frame grabbers
- Sound cards
- Feedback control
- Audio I/O
- Microphones
- Audio mixers
- Speakers
- Audio Test Tools
- Telephone Interfaces
- Delay and Effect Units
- Distributors for A/V equipment

References

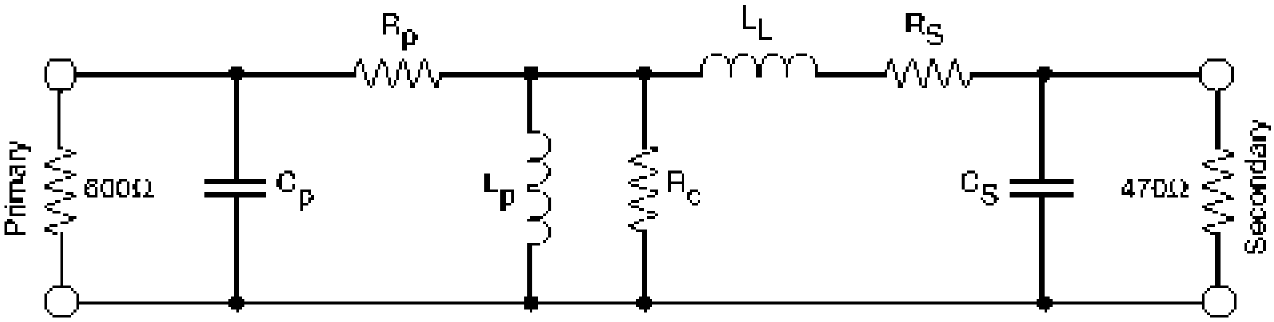
Microphones should be placed close to the speaker to reduce background noise, certainly not in the ceiling. For class rooms, microphones should be no further than 5 feet from the speaker.

The SuperJANET project provides additional advice on setting up audio/video conferencing facilities.

- Kai's Sound Handbook
- Rane professional audio reference
- Audio FAQ copy
- Worldwide TV Standards - A Web Guide
- Glossary of Technical Theatre Terms

- Educators' Survival Guide to TV Production Equipment and Setup -- Glossary
- Film glossary

Last modified: 1997-03-29 by Henning Schulzrinne


Rp
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RS
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600Ω
Cp
Lp
Rc
CS
470Ω
Secondary


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- Chip Resistors
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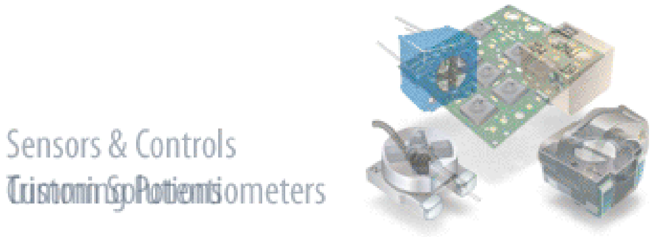
- Diodes
- Integrated Passive & Active Devices
- Linear Motion Potentiometers
- Magnetics
- Microelectronic Modules
- Modular Contacts
- Networks
- Panel Controls
- Precision Potentiometers
- Rotary Encoders
- Specialty Resistors
- Switch Power DC/DC Converters
- Switches
- Transistors & Thyristors
- Turns Counting Dials
- Trimming Potentiometers
- Application Notes
- Obsolete Products
- Design Kits
- All Products



News and Events

. Counterfeit BournsÂ® Thyristor Surge Protection (TISPÂ®) Devices

. Safety Issues with Counterfeit BournsÂ® Trimmers!

. BournsÂ® RoHS Roadmap (PDF)

. Bourns Releases Extended Range of Non-Isolated DC/DC Converters

Cermetek
microelectronics

- COMPANY
- PRODUCTS
- WHERE TO BUY
- TECHNICAL INFO
- NEWSROOM
- CONTACT US
- SITEMAP

Welcome to Cermetek

Cermetek Microelectronics, Inc. manufactures a broad line of Internet appliance modems, modem components and telephone line interfaces for the data communications industry. The company principally sells to the industrial OEM, through a comprehensive Rep and Distributor based sales force.

Cermetek's Internet products consist of a broad line of embeddable and freestanding industrial grade Internet appliance modems. Offered with these products is development software called iNetWizard, which facilitates design in. To support connectivity to the Internet, the company hosts a low cost Internet Service Provider (ISP).

WHAT'S NEW

11.01.03
New iModem Replaces Coventional Modems in Internet Appliance Applications

05.01.03
Cermetek Announces World's Smallest

01.01.02
Cermetek Introduces Low Cost Internet Appliance Modem

SIEMENS

→ siemens.com

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[Erweiterte] Suche

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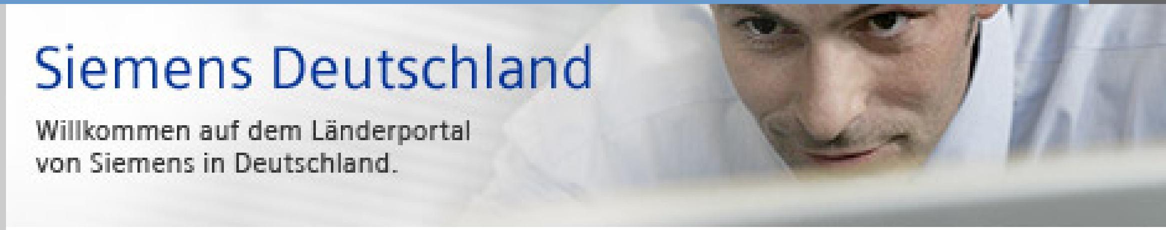
Bitte auswählen...

Geschäftskunden:

Hier haben sie schnellen Zugang zu häufig gewünschten Informationen:

Bitte auswählen...



Kurz vorgestellt

Der richtige Dreh

Drehzahlmesser helfen, Materialschäden und technische Probleme an Förderanlagen zu vermeiden. Der Milltronics ZSS ist ein hoch belastbarer Drehzahl-/Stillstandschalter. Das robuste Gerät ist unempfindlich gegen Staub, Schmutz, Anbackungen und Feuchtigkeit. Es funktioniert auch unter härtesten Bedingungen - zum Beispiel im Bergbau.

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Presse

17.03.2006 Siemens VAI rüstet neue Metallbeschichtungsanlage bei Steelscape, USA, aus

→ mehr

17.03.2006 Siemens rüstet die Olympia-Linie der Pekinger Metro mit Leit- und Sicherungstechnik aus

→ mehr

16.03.2006 Siemens VAI erhält Aufträge für die Modernisierung von Beizanlagen bei Posco, Korea

→ mehr

Siemens Weltweit

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→ Impressum

Blickpunkt

→ SiemensForum

→ Siemens Journal

→ Siemens Geschäftsbericht 2005

→ Jobs & Karriere Magazin

Bild der Woche

Kommunikation zwischen Maschinen
Der Siemens Bereich Communications präsentiert auch in diesem Jahr seine innovativen Produkte und Lösungen auf der CeBIT. Eine sehr interessante Geschäftslinie ist „M2M One" – individuelle Machine-to-Machine-Lösungen für Unternehmen.
→ mehr

Siemens Websites im Überblick

Sie suchen Online-Portale und Marktplätze von Siemens? Hier finden Sie unter anderem die Websites aller Siemens Geschäftsbereiche. → Übersicht

Aktuelles aus Deutschland

15.03.2006 Siemens baut neues Hochgeschwindigkeitsnetz von T-Com mit auf
→ mehr

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Europa – Die Macht von morgen?
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21.03.2006 -
MSR Anlagenmodernisierung (Hannover)
→ mehr

An On-line Source of Houston Information

Dialup Line Quality in Houston, Texas

This page was last updated at . Comments on these results should be directed to William Garfield, wdg@hal-pc.org. Bill recently decided to discountinue this work, so this page has been reformatted to inform people of this change.

- Expecting 56K?

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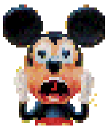


Disassembly Of A Touch-tone Phone

INTRODUCTION[1]

Touch-tone dialing is a method of sending signals from telephone customer's premises to central offices and beyond. The idea of touch-tone dialing was first introduced in 1964. Today, most of the telephones in the in this country use touch-tone phone. We found there are four major parts: dial circuits, switch-hook, ringer, and handset which includes the transmitter and the receiver. However, we studied only the dial circuit, switch-hook, and the signal on the telephone line. Our concentration is on studying the dial circuit, which is the frequency encoder. The principals of a touch-tone telephone is as follows:

(1) All the signaling energy is in the voice frequency band, making it possible to transmit signaling information to any point in the telephone network to which voice can be transmitted, that is, "end to end" which means that connection of phone signal from one telephone to another would no longer need an operator.

(2) Touch-tone dialing is faster, reducing the dialing time for users and, equally important, reducing the holding time for central office common equipment.

(3) It provides a means for transmitting more than ten distinct signals: twelve in all standard implementations.

(4) It is a more convenient signaling method, as attested by users.

Background Information On The Phone[1]

Since Touch-Tone scheme was the most outstanding progress of the telephone technology, our works were mainly concentrate on the signaling response of the Touch-Tone Encoder. Essentially, the idea of a touch tone is to employ four by three matrices of frequencies if twelve buttons required. Each button of the dial valued two components (one is imposed on the other) of frequencies in the matrices at the same time so that when the button is pushed, this value of signal is sent to communication office and they can be detected which button is being dial. And these two signals would be automatically transmitted to the other end of the telephone. Basically, the only objective of building a touch-tone was to find a way to reduce timing, probably through a scheme that would involve a push-button person -machine interface and to satisfy the "end to end " connections. that is, from one customer's premises, through one or more central offices, to the premises of another customer, anywhere that voice can be sent. And the requirements that satisfy this is that the signal must not contain an out of band component such as a dc step and to sustained rather than damped signals must be used to maintain adequate signal to noise margins for the wider range of transmission losses when two customer loops are involved.

Choice Of Code

When only voice frequencies are employed, protection against talk-off must rely heavily on statistical tools. This protection is required only during inter digital intervals; speech interference with valid signals can be avoided by transmitter disablement when a push-button is operated. Since signals with a simple structure are prone to frequent imitation by speech and music, some form of multifrequency code particularly difficult of imitation is indicated. If the signal frequencies are restricted in binary fashion to being either present of absent, the greatest economy in frequency space results from the use of all combinations of N frequencies, yielding n=2 to the N power of different signals. However, some of these are no more than single frequencies and are therefore undesirable from the standpoint of talk-off. Another drawback is that as many as N frequencies must be transmitted simultaneously; these involve an N-fold sharing of a restricted amplitude range and may also be costly to generate. If n=10, N would need to be at least four. At the expense of using up more frequency range, one is led to a P out of N code, yielding n=N!/P!(N-P)! combinations. There is statistical advantage in knowing that there are always P components in fall valid signals, no more and no less.

Choice Of Frequency

Attenuation and delays distortion characteristics of typical combinations of transmission circuits were such that it is desirable to keep the frequencies of an a telephone signaling system within the 700 to 1700 Hz range. The choice of the frequency spacing depends on part on the accuracy's of the signal frequencies. It was expected that signals generated at the station set could be held within 1.5% of their nominal frequency values and that the pass bands of the receive selective circuits could be maintained with 0.5% of their nominal ranges. On the basis of these numbers, the selective circuits of central office receivers need to have recognition bands of at least +,-2% about the nominal frequencies.

Button Frequencies:

In doing this part of the lab, we borrowed the telephones (EEB 221) at John Ofstad's office and at the stockroom. In several days, we used their phone during their normal operating hours. They were extreme patient and helpful with us, and we'd like to thank them.

In following our work, for one thing, we were to display the individual signals in time domain from each dialing button. We found out that button of " 1, 3, , 5, 6, 8, 9" are possesses a very similar wave form in the time domain (button 2) (button 3) (button 4). Their amplitudes are about two volts. Because we don't have the equipment, we could not transform them into frequency domain in order to make a comparison. The signal of 0, 4, and 7 also possess similar form in the whole; however, there are quite a bit noise at the negative side of the signal. The most distinctive and interesting signal is from the button 2. It has a totally different shape than the rest of the others. Its amplitude expended at the two sides and shrinks at the middle, but unfortunately we fail to find out the reason for this difference.

Switchhook:

In our telephone set, there are seven electrical contacts controlled by the mechanic switchhook. When we tried to see how the contacts are connected, we accidentally broke the plastic that pushes on the contacts. Therefore, we have to find our on our own which connections are on hook and off hook. What we did was we tried the different connections, and see which connections generate the dial tone. We label the contacts 1-7 for simplicity. When the handset is hanged up, the 1 - 2 contact pair and the 6 -7 contact pair connect (the other pairs are released). This breaks the connection of dial circuit and the telephone line, but connects the ringer to the telephone line. That is why we cannot dial when we hang up, but we can hear the ringer rings for an incoming call. When the handset is picked up, the 2 - 3 contact pair and the 4 -5 contact pair connect (the 1 -2 pair and the 6 -7 pair are released). This connects the dial circuit to the phone line, so the user can dial out, listen, and speak.

Dial circuit:

When we check the dial circuit, there are two chips that attract our attention. The first one is a square with 4 legs, and on the face it writes: 3579 KSS8J. This does not mean much, but we guessed that it is a oscillator. The second one is a chip made by Texas Instrument, the part number is TCM5094N. We went to the UW Engineering Library, and found the information that we need from the data book (Nonlinear IC in the reference section): it is a tone encoder with single/dual tone, and it uses a 3.58MHz oscillator. Therefore, our guess is correct that the KSS8J is really an oscillator! And 3579 indicates the oscillation frequency! On the TCM5094N chip, we found that some of the pins are labeled row 1 to row 4 and column 1 to column 4. At first we didn't know what that mean, but key pad in the telephone reminded us. This is the frequency generator for each row and column in the key pad. There are a total 16 combinations. However, there are only 12 buttons on our telephone. Therefore we guessed that one of the column pin is not used. Indeed when we traced the pins, we found there are the column 4 pin is not used.

(click here to see the dial circuit)

Fig.*"Did you hear the dial tone?"*

Again, we had to used the oscilloscope near John's office to find out the waveform when there is an incoming call. If you can't hang up, the caller cannot call you. It is that simple. When there is an incoming call, the AC signal amplitude would increase to about 45 volt, and the user can hear the ringer rings. See the following figure.

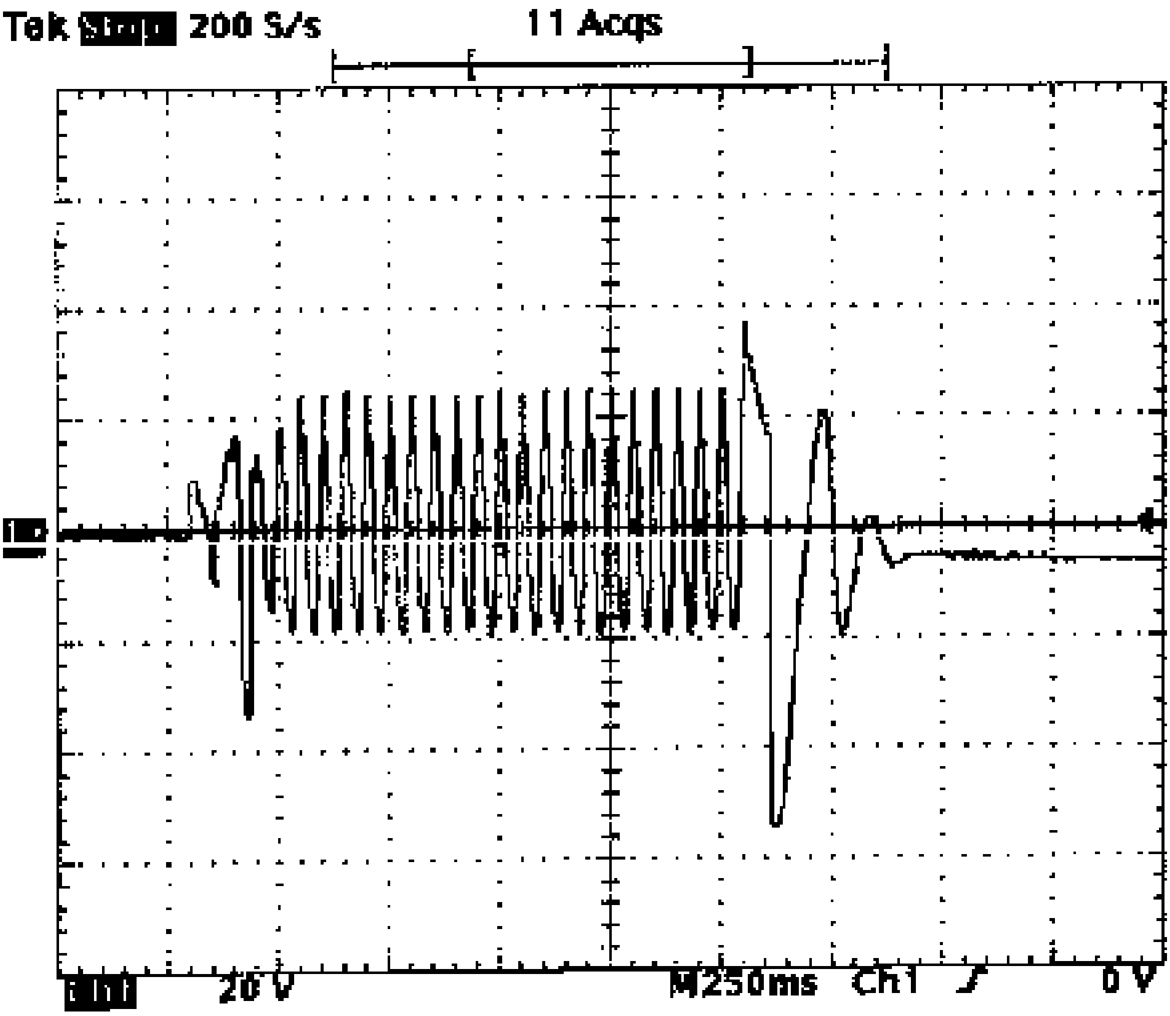


Fig.*"Now the bell rings!"*

Reference

1. Leo Schenker, *Telephone Signaling Systems, Touch-Tone*, Encyclopedia of Physical Science and Technology, Vol. 16, Academic Press, Inc., 1992

[Back to the final project home page]

[Back to the EE498 home page]

Telephone privacy adapter

Design and copyright by Tomi Engdahl 1996

Summary of circuit features

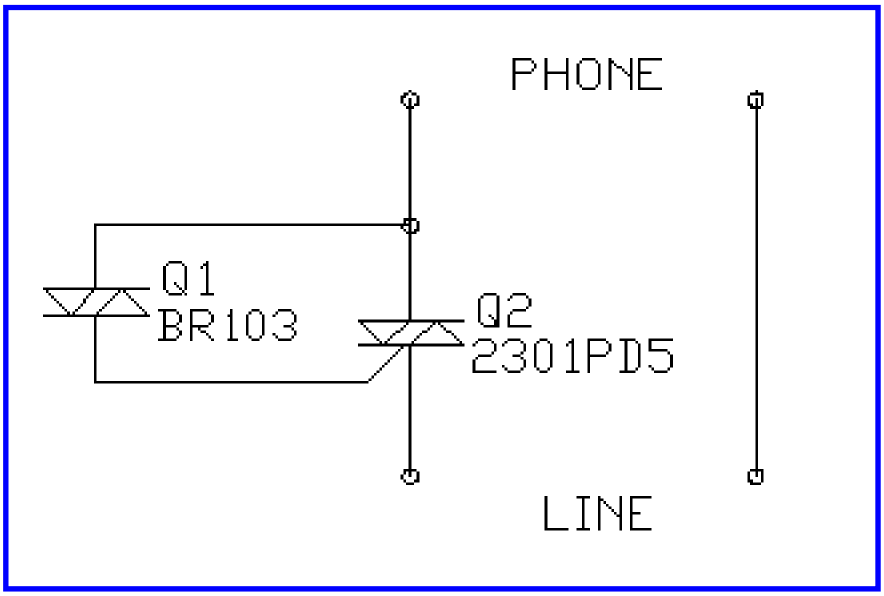
- Brief description of operation: Prevents telephone call to be heard form other telehones in the same subscriber line
- Circuit protection: Works on both line polarities, no special overvoltage protection
- Circuit complexity: Very simple and easy to build, no need for circuit board
- Circuit performance: Briefly tested with one telephone and telephone exchange
- Operation principle: When one telephone is picked up the circuits prevent other telephones to be picked up (thy do not get any line current).
- Circuit use: Installed between telephone line an telephone. You need one circuit per every telephone.
- Availability of components: Q1 (BR103) is widely available, Q2 might be hard to get
- Design testing: Built quickly from the parts found on my home lab, seemed to work on short test with telephone line simulator and line connected to PBX, I have received a report that this circuit works well in Australia
- Applications: Telephone accessory, stop telephones interrupting your modem calls
- Power supply: No need for extra power supply, takes power from telephone line
- Component cost: Few dollars
- Safety considerations: Should be be built to an insulating case
- Special notes: Not approved to be connected to public telephone network

Circuit description

This circuit is a simple circuit which prevents picking up other telephone when one telephone is in use. This can be done easily by installing this type of circuit between any telephone and the telephone line.

This type of function is very useful when you don't want other people from disturbing your modem connection or listening to your telephone calls by picking up other telephone connected to same line.

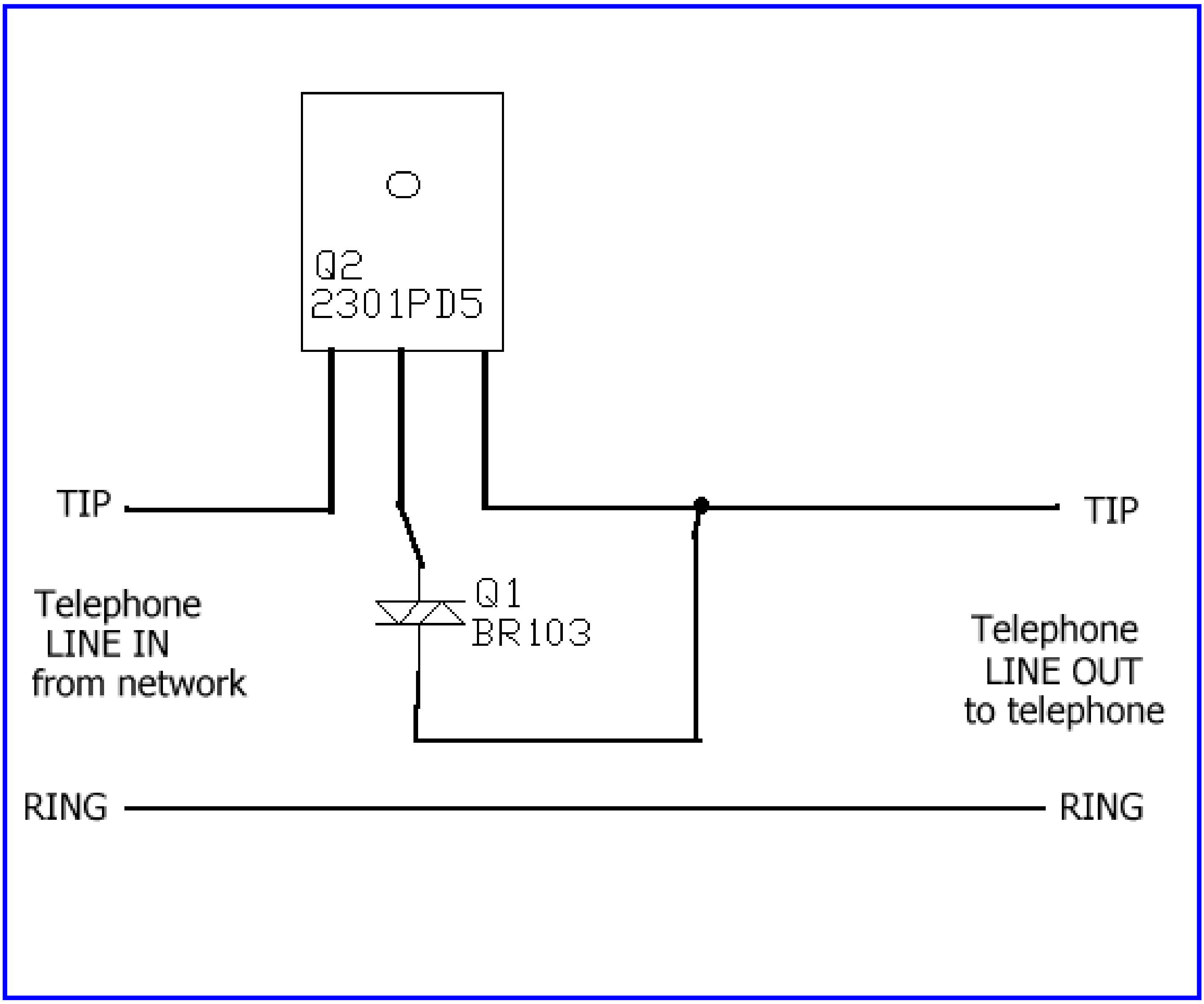
The idea of the circuit is to sense the voltage in the telephone line when the telephone is picked up. If that voltage is higher than about 30V (normal on-hook voltages is about 48V) then the circuit lets the telephone to work normally. If the voltage is lower 30V it prevents the current from going to telephone line to telehone (normally the voltage in line is about 6-10V when one telephone is off-hook). The circuit is designed so that is passes the ring voltage to all telephones without problems.



The circuit is very simple circuit built from one DIAC and one TRIAC. When telephone is picked up it will not get any operating current unless the TRIAC Q2 in series with telephone conducts. The triggering of the TRIAC Q2 is done through DIAC Q1, which will trigget the triac if there is more than about 30V voltage between TRIAC Q2 leads connected to telephone line wires. When TRIAC Q2 start to conduct it will conduct as long as there is any current flowing through it. So TRIAC Q2 conducts until the the telehone handset is put on-hook (call has ended).

This circuit is very similar to the operation of commercial adapters, but remeber that this adapter is not type approved for connection to public telephone network. The component values are just what I used in my prototype and you can replace that triac with nearly any type which will handle atleast 200V, can be triggered easily and keeps on conducting at currents as low as 15 mA.

The drawing below shows how to wire the circuit components and the whole circuit to the telephone line.



Modification ideas

I saw one article in sci.electronics.design which mentioned that an article in Elektor Electronics Dec/93 described a similar circuit idea. They used the following components:

- Triac: TIC206D
- Diac: BR100

When you appply this to my circircuit you will end up to following component list:

```
Q1   BR100
Q2   TIC106D
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NOTE: The pinout of TIC206D is different than the one used in the original circuit, so the component connection drawing is different. No wiring drawing of this modified version is available.

Comments on the circuits

AuVIP (adams@auvip.net.au) has sent me the following comments on this circuit to me (put here with permission):
"It works. I'm in Australia and it works very well. Components were either readily available or easily able to be substituted. Especially the TRIAC."

Other ways to do the circuit

There is one commercion single component designed to do the same as Q1 and Q2 in my circuit. That component is HS20 bilateral silicon switch which consists of zener diode and triac intergrated to same component. A telephone privacy adapter circuit using HS20 has been shown in Electronics Now and Poptronics magazines and that circuit is very simple because that two wire HS20 component is just wired in series with one of the telephone line wires. I have never tested HS20 component, so I can't tell how well it works but I believe that the circuit works because it is published in two magazines. According Poptronics magazine HS20 bilateral switch can be bought for around $4 from:

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SolarWorks
Grandprarie
TX75052
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According some usenet articles the HS-20 DIACs which are just about impossible to find and they might be discontinued.

The circuit might work with just DIAC

Huw Finney mentioned me that the circuit would work also by using just one normal DIAC in seris with the telephone and not usign any extra components. I have not personally tried it, but with suitable DIAC that might work.

Most diacs are used for pulse generation, probably in the order of amps, and are designed as such. I think the published ratings for static (DC) use are a bit on the consevative side, take a 1N4148 sized diac and say 100 to 200 mW dissipation and about 2V across the diac we are left with 50 to 100 mA, more than enough!

So in engineering sense it seems that just only a DIAC seems to be 'good enough' for this application. I have not personally tested this alternative, but this might be a worth to try.

Commercial circuits is a safe choice

If you rellay need this type of circuis for everyday use I recommend buying a type-approved commercial unit. If you live in Finland and want to buy a ready made commercial unit, go to a shop which sells telephone accessories and ask for PrivaPhone. This commercial unit will cost about 50 Finnish marks (a little over $10). Aastra Telecom makes also quite similar product BusyLine Switch which has also some extra features.

Tomi Engdahl <then@delta.hut.fi>

The WWW Home Page of Tomi Engdahl

Suomenkieliset sivut (pages in Finnish)

Minä itse

Kirjoittamiani dokumentteja

- AV-tekniikan opetusmateriaalia
- Kirjoittamiani artikkeleja elektroniikasta, teletekniikasta, tietotekniikan ja AV-tekniikasta
- sfnet.harrastus.audio+video usein kysytyt kysymykset (FAQ)
- sfnet.harrastus.elektroniikka usein kysytyt kysymykset (FAQ)
- Audiojuttuja - asiaa mm. Bilteman jakosuotimista ja muistakin tuotteista

Muuta hyödyllistä ja hauskaa

- Hyödyllisiä WWW-palveluita
- Hupilinkkejä

Pages in English

Interresting stuff you are propably looking for:

- Electronics Info Link Page (now www.epanorama.net)
- Electronics document collection
- My Electronics Circuits
- PC Hardware Pages

Documents in English

- Electronics documents written by Tomi Engdahl
- Ground loop resources page
- Joysticks and other game controllers
- Showing TV picture on VGA monitor page (FAQ)
- VGA to TV conversion information and circuit
- VGA to workstation monitor FAQ

Other pages:

- Computer Programming Links
- Virtual Reality Link Page
- Computer Graphics Link Page
- Multimedia Technology Links
- Cryptography and Cryptology Link Page
- Laser and Light Effects Link Page
- DJ Link Page
- Making WWW documents

Some information about me

- Information about my web pages

Tomi Engdahl <tomi.engdahl@iki.fi>

JK Audio

Telephone Audio Interface Products

Field-Friendly Phone Interfaces

Appeared in Radio World 4/1/98

by Paul Kaminski

The dial-up "plain old telephone system" or POTS, remains the least-expensive system for transmission or reception of audio, albeit with limited audio frequency response. With last-minute remotes, breaking news stories, increased program demands and budgetary pressures, some radio users still depend on POTS for transmission and reception. Most people producing such programs or news coverage are not technicians. An easy way to hook equipment to the phone system is a big plus.

Handset answer

This article focuses primarily on handset replacement products. Calls to manufacturers and dealers turned up the model mentioned here. If you make or use a model not shown, let me know and I'll write about it in a future article.

Handset replacement devices plug up or into the handset port on a POTS telephone. They replace the telephone handset microphone, and sometimes the receiver, either with an enhanced microphone or with a provision that lets you connect a broadcast mic and headphones to the telephone instrument. The replacement devices do not act like a telephone hybrid, splitting the signal into separate transmit and receive paths. Nor, with a few exceptions, do they plug up to standard telephone RJ-11 jacks, or work with phones in which the dial pad is in the handset. You still need to dial with the dial pad of the instrument. The handset, by design, allows leakage between the send and receive paths ("side tone").

John Lynch, a sales representative for Broadcast Supply West [Worldwide] of Tacoma, Wash., said handset replacement devices are not designed to record or put two way conversations on the air. "It can be done, but the quality will never be as good as that available with a telephone hybrid," he said. If you do one thing at a time with them, just send or receive audio, you can get acceptable results, Lynch said. According to Lynch, handset replacement devices are attractive to broadcasters on a budget, because they allow those broadcasters to "do a remote using any telephone that has a handset that can be disconnected...

...JK Audio makes similar passive handset interfaces. The THAT-1 and THAT-2 allow users to send or receive audio, from equipment ranging from full-blown mixers to cassette recorders. The THAT-1 has a simple interface, with RCA jacks for inputs and outputs and a handset mute switch that replaces the handset on an electronic phone (not the old carbon mic style).

The THAT-2 adds volume controls, XLR line in and line out, and a handset switch that allows it electrically to match handsets from old-style carbon mics to digital PBX. The XLR mic-level telephone output allows an easy interface to professional equipment. Both units provide a mix of send and receive audio on their outputs.

JK Audio makes a receive-only unit called the QuickTap IFB that provides and XLR line-level connection and a 1/8-inch mono output jack for listen-line audio ? a good tool when you need to feed pre-delay cue to a headphone amp or a talent ear piece.

Mixing, too...

...You also can choose units that include mixing capability.

JK Audio addresses the high-dollar end of this market, with the RemoteMix, RemoteMix C+ and the RemoteMix 3. The latter model blurs the line between handset replacement devices and telephone hybrids. If you need a three-input portable mixer and handset replacement device/phone line hybrid (hooks to both RJ-11 and handset jacks) with VU meter, squawk-box intercom style talkback and a touch-tone dial, this is your box.

The next JK iteration of the RemoteMix 3 is called the 3.m, which adds a clean mixer feed (without telephone receive audio) to its other capabilities, without an increase in price. With an audio response of 8Hz to 15 kHz, it works well for basic field mixing to record to a MiniDisc or High-end tape recorder. JK also has an even more advanced model, the Remote Mix 3x4, which allows not only a hot backup system for an ISDN/POTS codec or RPU, but also the ability to produce a remote call-in talk show in the field.

Versatility

The versatility of these devices can help broadcasters solve telephone and remote-feed problems using POTS. While the system audio response is band-width-limited, the transmitted audio benefits from the addition of a better microphone. Your news product can benefit, too. If your competitor covers an event with a voice report, and you cover the same event with a wraparound report, including actuality of the subject matter, expert or newsmaker, your station will be perceived as the one that brings an extra perspective to the story.

To summarize the reasons for handset replacement devices: The cost is low compared to POTS codecs, frequency extenders, RPUs and satellite transmission equipment. Some units are less than $100.

They offer relatively easy connection to any hard-wired telephone with a detachable headset. With digital PBXs, it may be the only option if you go hard-wired.

They allow recording or transmitting of enhanced dial-up POTS audio (but not both simultaneously with any quality).

They allow a hot backup to an RPU/POTS or ISDN codec or other feed (reduce make-goods on that must-run, big-dollar remote).

Higher-end handset replacement devices ($260 and higher) include rudimentary mixing of mic-and line-level send audio, a separate line-level feed, an output for recording from the telephone line and other features, depending on the box.

The marginal cost of transmission per minute is low compared to cellular phones.

Paul Kaminski is the Motor Sports Radio Network news director and host of its syndicated weekly programs "Race Talk" and "Radio Road Test."

If you have questions or comments, you can contact us toll free at 800-552-8346.

We appreciate your business.