Recording Telephone Interviews
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The Narrow Pipe (or How to Squeeze an Interview out of your Telephone Line)
Go into any office supply and pick up the cheapest pocket micro cassette recorder you can find. With all its hiss, distortion, flutter, and narrow bandwidth, its still far better than any recording you'll make over the telephone line. The fact is, that telephone audio quality is much worse than it was just 10 years ago. This is due to the propagation of cell phones, low cost long distance carriers, and internet telephony compression.

This article does not deal with the various methods of encoding and decoding using POTS or ISDN or IP CODECS to squeeze broadcast audio through the narrow phone line. Those products are designed for point-to-point communications. This article focuses on recording interviews and dealing with the limitations of bad caller audio.

Picture your recording or transmission media as a pipe. The larger the diameter of the pipe, the better the audio signal. While there are many dimensions of audio quality, let's just consider bandwidth and noise. A music CD can be compared to a very wide pipe, as low noise, full fidelity music flows freely. An analog tape recorder may offer similar bandwidth, but now tape hiss has moved the noise floor up, creating another challenge. To properly record on tape, you must record at a high enough level to keep above the noise floor, while avoiding clipping and distortion if your level is too high (a narrow pipe). Now consider a telephone call; the noise floor consists not only of AC power hum and noise, but also the noisy lower bits of non-linear encoding algorithms. Your bandwidth had been limited to 3.4 kHz, just wide enough to carry on an intelligible conversation. Top that off with the fact that the person on the other end of the phone is probably talking into a 20 cent microphone element in their telephone handset (a very narrow clogged rusty pipe). Now you might say "my telephone calls don't sound that bad". You must remember that in most cases you are listening to the call over a bandwidth limited, dynamic compressed earpiece in your telephone handset. Replace that earpiece with full fidelity headphones and an amplifier and you'll hear things you've never heard before. Submit this recording for broadcast and everyone will hear the noise.

There is not much you can do to counteract the artifacts of bad caller audio. On the other hand, there is no need to compound the problem by recording your voice through the caller audio path. If your telephone audio interface does not provide complete separation between send and receive, some of your voice will come back mixed with the caller's voice. This extra image of your voice will sound like you were also talking on a telephone.

Choose Your Path
There are two types of telephone recording interfaces; handset interfaces and phone line hybrids. Both types have their advantages, so you will have to choose the best match for your situation. Remember, these are audio interfaces, not recorders. These products give you a clean connection to your sound card, recorder, or mixer.

Handset Interfaces
The handset interface is a device that connects between the handset and base of your existing telephone. The main advantage here is that (JK Audio) handset interfaces will work with any telephone system, whether it's a simple one-line home telephone or a multi-line business digital PBX. Of course, since the audio is captured from the handset cord, the speakerphone is off limits. Besides the obvious portability and travel advantages, you also get to use any function that is available on the phone, such as conferencing, and line selection. A handset interface can turn an already familiar telephone set into an easy-to-use talk show console.

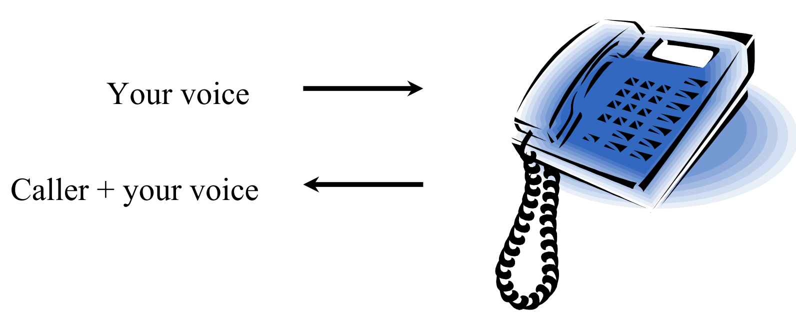
Audio quality on handset interfaces can go either way. Generally speaking, recordings made through a $20 phone will probably have noticeable artifacts, including added noise and bandwidth anomalies. On the other hand, a digital phone system in an office building may have lower noise than a phone line hybrid, since in many cases the audio path remains digital all the way from the phone company to the telephone set.

Telephone Hybrids
A telephone hybrid is a device that allows you to send and receive audio over a telephone line. Without getting into too much detail, it's important to note that you cannot send and receive audio to and from a telephone line without a hybrid. Phone lines carry both transmit and receive audio signals mixed together over two balanced wires. The hybrid separates transmit and receive audio signals, and safely isolates the customer from dangerous ring and supply voltages. Even the cheapest telephone contains a hybrid circuit.

Most stand-alone telephone hybrids do not contain a telephone keypad, so you will want to keep a simple telephone handy to initiate calls. The telephone hybrid has the added advantage of providing auto-answer of an incoming call.

Audio quality on a telephone hybrid is more predictable than a handset interface since the connection is directly on the phone network.

Common problem
Both handset interfaces and phone line hybrids face the same problem, the presence of your voice mixed with the caller's voice.



The presence of your voice on the handset receiver is known as "Sidetone". This audio path gives the user immediate feedback, letting them know that they are speaking into the handset at an appropriate level. Simple recording taps will not allow you to adjust the level of your voice versus the caller's voice. In addition, your voice will have that special "tin can" or telephone sound.

Telephone hybrids face a problem similar to the sidetone mentioned in the telephone handset interface. It is nearly impossible to design a hybrid circuit that will completely separate transmit and receive audio signals from an analog telephone line. The resulting crosstalk is called line echo, resulting in your voice mixed with the caller's voice.

Broadcast quality recordings can be made using low cost telephone hybrids or handset interfaces provided the interview is made with clean verbal exchanges. Many interviews that you hear on the radio are recorded with simple gear and a bit of editing. The most important step is to start with a professional microphone for your voice. You must have a good recording of your voice on one channel of your recording device. The object is to mute the caller audio channel whenever you speak. This will remove your voice as it comes back through the telephone interface, and remove background noise when the caller is not talking. In both cases you are simply dropping the audio level of the caller audio channel.

You can save yourself a bit of editing by using a telephone interface that provides complete separation of your voice from the caller's voice. A digital hybrid uses a DSP to analyze and remove any traces of transmit (talent) audio from the caller audio. The digital hybrid not only provides the separation needed for excellent interviews, but this technology also allows for real-time conversation during a broadcast.

JK Audio Handset Interfaces

QuickTap - Sometimes the simple approach is the best match for the application or budget. A recording interface that taps into the audio signal on the coily handset cord can actually sound pretty good. Our model QuickTap allows you to make a nice, clean recording of both sides of the conversation. The quality of our design, versus other products on the market, results in a recording with minimal hum and noise. Both voices will be mixed together on one output jack, at about the same level.
http://www.jkaudio.com/quicktap.htm

Voice Path / THAT-2 - If you need to send audio into the phone line, as well as record your conversation, we offer our models Voice Path and THAT-2. Both of these products connect between the handset and base of your telephone. Voice Path is designed more for a permanent installation, while THAT-2 is built for portable, rugged use. On both units, a switch selects between talking on the handset or sending a signal from your audio player. The send audio source is typically the headphone output of an MP3 player, sound card, or an audio mixer. The caller output is always a mix of either the send jack input or the handset mic, mixed with the caller's voice.
http://www.jkaudio.com/voice-path.htm
http://www.jkaudio.com/that-2.htm

Innkeeper PBX - While handset taps are the most convenient, universal way of connecting to both simple consumer and complex business telephones, they have the limitation of the mixed send and caller signal. Find a way to remove the sidetone, and the resulting caller audio will

make a great interview recording. To this end, our model innkeeper PBX contains a DSP running a sophisticated algorithm that removes the sidetone signal, as well as any phone line echo of the send signal. This all-in-one product functions as a simple interview console. Connect a microphone and your sound card directly to the back connectors. The stereo output jack contains your voice on the left channel (in full fidelity) and only the caller's voice on the right channel.
This product represents the state of the art in interview recording. The complete separation of send and caller audio gives you the ability to adjust levels independently, and prevents your voice from sounding like a tin can.
http://www.jkaudio.com/innkeeper-pbx.htm

JK Audio Phone Line Hybrids
AutoHybrid - Our simple AutoHybrid offers audio performance very similar to our model THAT-2 handset interface. This is a passive interface, so a mixer is required to boost send and receive signals to an acceptable level. The audio signal that you send into this unit will come back mixed with the caller signal, at about the same level as the caller. Since AutoHybrid connects directly to the phone line, it has the capability of automatically answering and disconnecting from an incoming call.
http://www.jkaudio.com/autohybrid.htm

Broadcast Host/Guest Module - Our model Broadcast Host offers the same capability as the innkeeper PBX, but connects directly to a phone line instead of a telephone handset jack. Broadcast Host is referred to as a Digital Hybrid. It contains a DSP running a sophisticated algorithm that removes the send signal phone line echo from the caller output. Broadcast Host offers the same level of separation as innkeeper PBX, with the added advantage of auto-answer / auto disconnect.

The decision between innkeeper PBX and Broadcast Host may simply come down to your ability to get to an analog phone line. Remember, Broadcast Host is a single line device without a keypad. You will need a single line telephone or choose the optional Guest Module remote keypad to place outgoing calls.
http://www.jkaudio.com/broadcast-host.htm
http://www.jkaudio.com/guest-module-1.htm

Innkeeper 1x - Our model innkeeper 1x is our top of the line digital hybrid. Innkeeper 1x adds a few features that automatically improve the quality of the caller audio. Features like Caller ducking, AGC, and Presence. Caller ducking automatically drops the caller's voice whenever you speak. Automatic Gain Control adjusts the caller's level; reducing loud passages, and bringing up quiet callers. Presence is a custom filter that boosts lows and highs for a richer sound. Innkeeper 1x also offers additional monitoring and remote control capability.
http://www.jkaudio.com/innkeeper1x.htm
http://www.jkaudio.com/guest-module-1.htm
http://www.jkaudio.com/riu-ip.htm