Design for

Simple Party Line Intercom System

Contents

cameratim.com

Brief outline of the design

A party line system is one where everybody can hear each other at the same time. For small productions, this makes things simpler for the operators, as there will only be the most basic of controls (individual volumes, and mike muting). And it makes things easier for the design, as all/most stations will have the same circuit design. You can add more stations simply by joining them to the party line.

The system consists of several intercom stations, each being self-contained. Each station amplifies their own microphone, and supplies it to the common party line. And each station amplifies the common party line, and supplies it to their own headphones. In doing so, they also amplify their own microphone to their own headphones, so the user hears themselves the same as everyone else does. All the basic station use the same circuit., though some stations could be more elaborate (e.g. stereo audio, for an audio operator station).

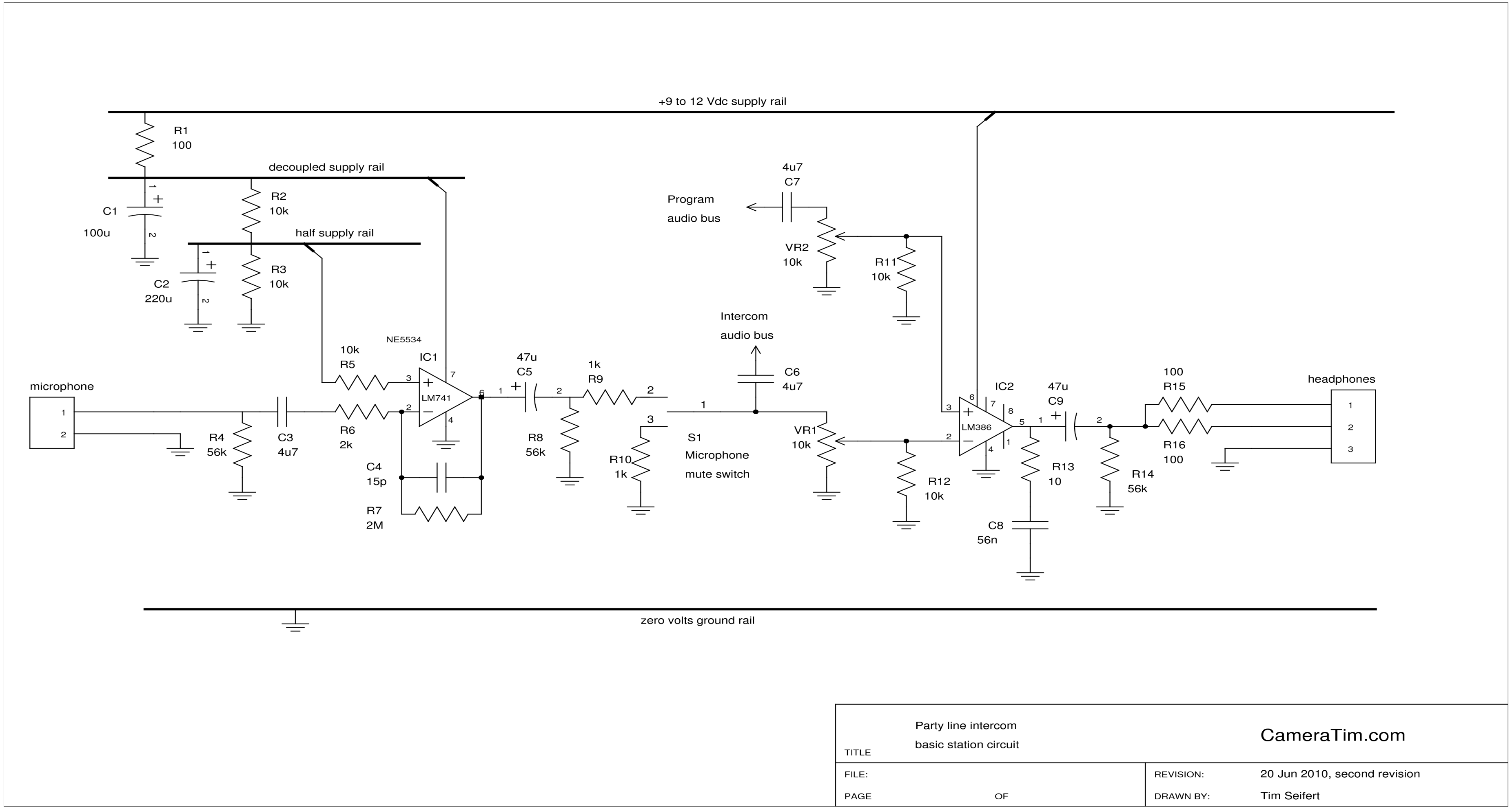
This has an advantage that people will not speak too loudly, because they'll sound too loud to themselves, but a disadvantage that it's easier to get acoustic feedback through headphones (squealing noises). A more complex circuit design could cancel the user's own voice from their own headphones, somewhat, but that carries the disadvantage that users may speak too loudly, expecting to be able to hear themselves more loudly, and it makes the circuit more complex (another amplifier would be needed, the nulling may need to be adjusted any time more or less stations are connected to the party line, and users would need to know how to use the extra control). Nulling should only be partial, as people expect to be able to hear themselves, and will raise their voice until they hear themselves at a comfortable level. To have a loudspeaker-based station in the control room, so that the director can speak into a desk microphone and not have to wear headphones, would *require* a design where the director's microphone is almost completely cancelled from the loudspeaker amplifier, and would preclude the use of other stations in the same room.

Listen-only stations can be easily made by omitting the microphone amplifier, only a basic line level to speaker level amplification is required, and almost any amplifier could do that, as the party line audio is nothing special.

This design only has two active components, making repairs cheap and easy should a power supply problem cause damage (e.g. reversed batteries). The stations can be powered from a central supply, or local to each station. This design only uses a few milliamps, so it's suitable for many hours of battery life using ordinary 9 volt transistor radio batteries. The usual LM386 power amplifier ICs have a maximum supply voltage of 12 volts, so the circuitry shouldn't be powered from anything higher. The camera power supply can be used, though a separate supply means that you can use an intercom independently of the camera, such as while setting up, and makes it less likely that a problem on the intercom could upset the camera.

Most component values are not critical. They were picked, with a bit of trial and error, from the spare parts that I already had.

Schematic



R1, R15 & R16 are each 100Ω, and R13 is 10Ω. A NE5534A has been the best affordable choice for IC1. If one of the 5534 ICs are used, a 22pF cap should be placed across pins 5 & 8, as close to the IC as possible.

Circuit description

R1 and C1 decouple the microphone amplifier from the main supply. This amp has quite high gain, and can easily go into oscillation (with itself, or with the earphone amplifier, together). The decoupling helps to prevent that.

R2 and R3 provide a half supply voltage, with C2 as supply decoupling, to bias the microphone amplifier (IC1) to mid-supply through R5. This allows the amplified microphone signal to use almost the entire supply voltage. It won't normally be driven that hard, but that leaves plenty of headroom so that clipping shouldn't happen.

The microphone signal is supplied through C3 to IC1. C3 blocks DC, with R4 keeping the microphone side of the capacitor discharged at all times. This is to avoid destroying dynamic microphones.

If an electret microphone is used, instead, then R4 should be connected to a positive voltage (either the half-rail supply, or the decoupled supply rail), instead of to the ground rail. And the value adjusted to suit the microphone that you have (probably somewhere around 5kΩ). An electret microphone will, typically, have a higher audio output than a dynamic microphone, as it's pre-amplified. So, with a system with a mixture of dynamic and electret microphones, you might want to reduce the microphone amplifier gain, or move the electret microphones a bit further away from the mouth, to equalise the different audio levels.

IC1 amplifies the microphone to the party line level (loud enough to avoid noise problems on the line, and loud enough for simple headphone amplifier design). The gain is set by R6 & R7, with R6 primarily concerned with input impedance, and R7 for negative feedback. C4 reduces the gain at high frequencies, to help reduce audible hiss and avoid ultrasonic feedback. R6 was deliberately chosen as a lowish value, so it loads down dynamic microphones and deadens them, somewhat. You only want to hear someone speaking closely to an intercom microphone, not the whole room around it. R7 and C4 must be close to IC1, to avoid feedback.

Not shown on the circuit is a compensation capacitor, of around 22pF, required between pins 5 & 8 on IC1, if the IC is a NE5534. Other ICs may not need such a capacitor. I recommend using a NE5534A, as it produces significantly less audible hiss than than LM741 or TL071 ICs, has better audio, and it isn't as expensive as some other alternative ICs. With the NE5534A being better, for audio, than the NE5534.

The output of the amplifier is AC coupled through C5, with R8 keeping it discharged on the other side, at all times. R9 sets the output impedance of the amplifier to the party-line, and provides stability to the output of IC1. It's somewhat lowish as a method of keeping noise down on the party line, but not so low that connecting half a dozen stations to the line will be a problem. It feeds the microphone mute switch (S1), which connects the party line (AC coupled through C6) to either the microphone amplifier, or a dummy load (R10). This keeps the party line impedance fairly consistent. R9 should be close to the output of IC1, to avoid oscillation problems.

The output from the microphone amplifier (when not muted) and the party line audio goes to VR1 to control the volume of the intercom received in the local headphones. The wiper of the pot goes directly to one of the inputs on IC2. R12 is across the input in case of damage to the volume pot, such as if the intercom station gets frequently dropped. If the pot goes open circuit, the IC input is still loaded down, and shouldn't start to oscillate or make other horrible noises.

A second audio line is similarly brought into IC2. C7 AC couples it to VR2, which connects to the other input pin of IC2. R11 is across the input for the same reason as R12. If a second audio line is not required, then omit C7 and VR2, but keep R11 to load the input down. Or alternatively, the microphone amplifier output could, also, be connected to the other input, for nulling it out of the local headphones. In this case, VR2 would be the null control. And S1 should be changed to a DPDT switch, with one half muting the signal to the party line, and the other half muting it to the second input.

The second input audio could be program audio, or to monitor another one-way intercom signal. It doesn't matter that the two inputs to IC2 are the inverting and non-inverting inputs, as the two audio lines will be completely independent audio sources.

IC2 is the headphone amplifier. It takes the two inputs, and amplifies them up to a level suitable for driving tiny speakers. R13 and C8 form a Zoebel network for amplifier stability. C9 blocks DC from the amplifier (it's biassed at half supply) to the headphones. The left and right sides of stereo headphones are connected through R15 and R16. These resistors drop the audio signal voltage down to a suitable level, and also reduce the noise from the amplifier, at the same time. They provide a suitable output impedance for most headphones, protect the amplifier from shorts to ground, such as when plugging mono headphone plugs into a stereo socket. And can protect the amplifier from shorts between two headphones amplifiers, such as in a stereo intercom unit, with two output circuits.

The default gain of the LM386 amplifier is just about perfect for this circuit, it can drive low and medium impedance headphones quite adequately. If IC2 is something other than an LM386, the inputs to it may need capacitors between the volume controls and the IC inputs, for AC coupling. It may need DC biassing on the IC inputs. And it may need a different gain.

The choice of input and output amplifier levels means that it should be possible to turn the volume controls completely up without creating a signal so loud as to cause hearing damage or discomfort, nor feedback to start (although that may happen with some headsets design, particularly if the headphones aren't being worn, and the microphone is dangling too close to the earpieces). The normal volume control position should be less than full.

You might want to use higher resistance volume pots than the 10kΩ pots that I used in the prototypes (they were what I had in my spare parts collection), they will load the party line down when there's several stations connected. Try 25kΩ or 50kΩ. Likewise with the resistor across the wiper (which can be completely omitted, if you prefer, or tweaked to modify the taper of the volume control).

Interconnection

The party line terminals of all stations are connected together, along with the common ground. Or, if audio coupling transformers are used for balanced audio, just the two signal lines from each station's transformer need joining together. Joining the audio lines together, balanced or unbalanced, is all that's required for the intercom stations to work together.

They can be powered individually, but it may be more convenient for them to run from a central power source (mains-derived, or batteries). They only use a few milliamps, each, so they could easily be individually battery powered, but it gets expensive and time-consuming to deal with multiple batteries. Using a separate supply than the camera's helps avoid video problems by keeping the intercom isolated from the video wiring, but it's certainly possible to use the same 12 volt supply for camera and intercom.

I've deliberately not shown a power supply circuit, because building mains supplies is inherently dangerous. If you don't already know how to build one safely, then buy a pre-built supply, and connect that to your intercom.

Because the stations are all virtually the same, there's no need to have a master station, other than for convenience's sake. You could easily just have a junction box for joining all the audio and power connections, and every station could be a belt-pack style station.

If you're going to use the second audio channel feature, then the audio source needs to be capable of driving multiple stations, or you'll need a buffer amplifier for it. This is a case where having a master station can be useful, it can house this buffer amplifier, and a central power supply.

Our set-up does use a master station (actually, it has two stations built into it), and it has buffer amplifiers for the program audio monitoring, and a central 12 volt power supply for everything. There's more than one buffer amplifier, because the stations in our main unit have stereo earphone amplifiers, allowing the audio operator to wear just one headset for both intercom and audio monitoring (those stations have the earphone section of the intercom circuit built twice over, once for each side). The rest of the stations just get fed a mono feed, of left and right summed together, because that's all they need.

The buffer circuits are fairly similar to the microphone amplifier, though with a few changes (mostly) related to gain. For left and right audio buffers, the input impedance resistor (R6) is increased to 100kΩ, the feedback resistor (R7) increased to 100kΩ (for unity gain), the high-frequency roll-off capacitor (C4) omitted. A left + right mix amp takes the output from the left and right buffers, with a 100kΩ from each buffer to the mix amp input (like twin R6 resistors), the feedback resistor (R7) reduced to 50kΩ (because, otherwise, left + right would be double signal level), and no C4. If you want balanced inputs or outputs, use coupling transformers. It's a simple solution that also avoids earth loops.

Other notes

We've been using this circuit for about two years, now. It's been modified a bit, as we've discovered problems (oscillation, hiss being a bit bad, etc.), and this final design has worked quite well with a variety of different headsets. In the past, we'd had problems with (other) systems being too complicated for the users, headphone volumes being too quiet to hear, and the ambient racket overdriving the intercom microphone amplifiers. But now we have something that's really simple to use, and has adequate volume levels to properly hear ourselves speaking while video recording a concert put on in a theatre (that was somewhat louder than they really needed to be), using the ever popular beyerdynamic DT109 headsets, without any problems. And the stations are fairly cheap and quick to build, and don't use difficult to find parts.

This circuit has worked well using ordinary shielded unbalanced wiring across 50 metre camera cables wrapped around 240 volt mains cables without too much hum or noise. It's mostly unnoticeable, compared to ambient background sound in the room, and less noisy than several other intercom systems that I've used. Though, for harsh environments, the simple answer is to use transformer coupling in each station, with twisted pair or balanced cable between the stations. Cheap $3 audio coupling transformers are good enough, it's only intercom audio. I've done this to use 100 metres of unshielded CAT5 network cable between some stations.

We avoided using a push-to-talk system, opting for open-mike, and designing the system to work well in noisy environments, as this is easier for the camera crew (they don't have to find buttons to press when they urgently need to say something, and they might bump their shots around while pressing buttons). Having said that, it's better, for everyone else, if unmanned stations are left muted. And it would be easy enough to extend the mute switch wiring out of the intercom to a small button mounted onto the tripod arm, connected to the station via a switched TRS jack, so that the station would still work without an external switch plugged in.

We opted for the simplicity of just one audio output for most stations, feeding the same audio to both ears of stereo headphones. This means that you can use headsets with one or two earphones, or operators can wear dual-muff headphones with one ear uncovered, and they won't miss out on hearing something (intercom or program audio). It's also easier to understand someone talking to you when you can hear them in both ears. Contrast that with how many systems use split earphones, with intercom on one side, and program audio on the other. They must have dual amplifiers, the operator must wear dual-muff headsets, and they must have both ears covered.

When monitoring audio, be sure to pick a source that is not delayed, such as one of the outputs from the audio mixer. The output from a HDD/DVD recorder may be delayed, this is very confusing for people listening to the audio, and makes it very hard for camera operators to find shots, by themselves, anticipating the director's needs, by listening to the program sound.

I recommend using a single multi-pin connector for the headset, for the convenience of having just one plug to connect them up. Or, different types of connectors for the microphone and headphones, so that users don't plug things into the wrong sockets. One common approach is a 3-pin XLR for the microphone, and a normal headphone plug for the headphones (this also allows a communications headset to be used by a commentator, plugging them directly into standard audio connectors).

It's well worth buying decent headphones, the fiddly little computer headsets made for VOIPing are not very good. They break easily, and fall off when you move your head around. If you intend to borrow or hire communication headsets, rather than buy them—as they can be very expensive—you may want to use male 4-pin or female 5-pin XLR connectors. The usual pinouts can be found on the pinouts page.

Likewise for using different types of sockets for the party line and headset. Using 3-pin XLR connectors for the party line is convenient, for being able to use any handy microphone cable for your intercom, but you must label them if they carry power as well as audio, so that they don't get connected to something else and damage it. Though using something else means that you must make up special cables just for your intercom.

For connectors carrying power, the general rule is "no power on exposed pins." This avoids damage caused when disconnected leads come into contact with something else. Use female connectors for the side supplying power. Or find a type of connector that completely covers the pins on the male connectors.

Pinouts

If you want to use this intercom with professional headsets (bought, borrowed, or hired), you may want to fit your stations with the types of connectors that usually match those headsets. You will find the 4-pin XLRs used on many beltpack intercom systems, and the 5-pin XLRs used on Sony broadcast cameras, and a few other manufacturers, as well.

4-pin XLR male headset connector wiring

1. Microphone shield
2. Microphone audio
3. Earphones – or shield
4. Earphones +

Since 4-pin XLRs are also used for power connectors on many professional cameras (females on the supply, males on the camera, with pin 1 being ground, pins 2 & 3 not connected, and pin 4 being +12 volts), be sure to use male connectors on the stations and female connectors on the headsets, so that expensive headphones can't be accidentally connected to a power supply. But it's better if you use different connector types for all the different purposes (camera power, headsets, station interconnecting). Such as 5-pin connectors for the headsets, 4-pin connectors between stations.

5-pin XLR female headset connector wiring

1. Microphone shield
2. Microphone audio
3. Earphones common, or shield
4. Earphones left audio
5. Earphones right audio

Since it's expensive, and hard to find, XLR connectors with more than 5 pins, we used DIN plugs to connect the stations together. A 6-pin DIN plug is all you need for separate power, intercom, and program audio lines. Less pins will be needed if you share pins for common parts of signal path wiring.

One solution for minimal inter-station cabling is to use twin shielded cable, figure-eight style, where the shielding for each audio line is separate (the shields are isolated from each other), using the two shields for the power supply connections. And you can use the 4-pin XLR connectors, the same as are commonly used for professional video equipment power supplies, using the two spare pins for the audio lines. Though, this is not a good idea if you're using 4-pin XLRs for the headsets, as it's too easy to plug things into the wrong sockets. Likewise, it's not a good idea if your video equipment uses 4-pin XLRs, and it uses all 4 pins in the connector.

4-pin XLR female base unit / junction box wiring, & 4-pin XLR male station wiring

1. Power supply ground, and intercom shield
2. Intercom audio
3. Program audio
4. +12 volt power supply, and program audio shield

Remember: No voltages on exposed pins. Use female sockets on the supply side, male connectors on the station side.