3-Way Analog Stereo Tone Control Kit

+/-20dB adjustment at 80Hz (Bass), 1KHz (Midband), and 8KHz (Treble).

Up to 20dB linear amplification. Choice of x1, x2, x5, or x10 overall gain.

Low noise and distortion of 0.00006% THD and S/N ratio of 86dB below a 10mV input, and 106dB below a 100mV input.

Requires bipolar power supply of minimum +/-5V to max +/-15V

Optional components included for Dynamic microphone (moving coil) sound amplification

Overview.

The circuit has a 2-Stage design: a linear amplifier stage (with gain chosen by user between 1 and 10) and an active filter amplifier (with additional +/-20dB gains at selected frequencies)

Input impedance is determined by resistor R1. Standard audio equipment is able to feed signal into input impedances of 10K (KiloOhms). So when R1 = 10K, the input impedance of the tone control will be 10K. Overall gain is determined approximately by ratio of R2/R1. Increasing R1 rises input impedance (good) but also degenerated gain (bad). Raising the value of both resistors allows increasing input impedance without gain reduction. However there are other considerations limiting the values of resistors that should be used. All resistors create electrical noise (hiss), due to Brownian motion of molecules inside materials. This noise is inversely proportional to ambient temperature and directly proportional to resistance value. Higher value resistors will produce more noise. So we will want to use the lowest possible resistor values that will do the job in any situation. Using resistor values of 1Meg or more is undesirable. The noise contributed by all resistors in the circuit is of the order of 3-4 times lower than the noise contributed by the integrated circuit OpAmps (and that noise is already extremely low), so it is not really a problem. We just need to be aware of the consequences of choosing high resistor values , though.

We need to keep the input impedance to at least 10K (preferably higher) so as to maintain compatibility with all sound sources.

For a gain of 1, the values of R1 and R2 will be both 22K. For a gain of 2, R2 will increase to 47K. For a gain of 5, R1 will be10K and R2 will be 47K. And for a gain of 10, R1 is 10K and R2 is 100K.

The 10K input impedance will accommodate electret or condenser type microphone inputs as well as crystal or ceramic units. These microphones typically produce signals of 100mV or more.

Dynamic microphones have a moving coil inside, and the impedance of that coil is of the order of 100 to 500 Ohms. They are low impedance sources, and work best when fed into low impedance amplifier inputs. For connection to such a microphone, R1 will be 1K. This allows greater latitude in the selection of feedback resistor R2, since we can stay at lower resistance values, and still achieve larger gains. Large gains are important for dynamic microphones, since their signal output is one of the weakest sound sources we will ever encounter: about 2-5mV rms. This is about 20 to 50 times lower than line level inputs which are typically 100mV. For such a setup, selecting R2 of 22K will give a gain of 22 , and 47K will give a gain of 47.
This will need to be determined experimentally, using your own dynamic microphone input.

Phono (record player) signals are not suited for direct connection to the tone control circuit.
This is due, on the one hand, to the output impedance of moving-magnet cartridges, which is 47K , and would be loaded by our circuit, and on the other hand, to the RIAA equalization required by vinyl records, which this circuit does not provide.
Moving magnet cartridges are the most common type - chances are 99%+ that any cartridge bought today (or in the past 30 years) is a moving magnet type. Moving coil cartridges, are very low impedance sources, and require dedicated preamplifiers to handle their very low signal outputs.

The tone control should generally be inserted in the signal path ***after*** the initial preamplifier that follows the phono cartridge, and provides impedance matching and RIAA equalization.

Although it is possible to use this circuit to amplify phono cartridge outputs, that would require increasing R1 to 47K, which, in turn would leave less latitude for obtaining gain from the first opamp stage. Feedback resistor R2 would then need to be increased to 500K or more, to amplify the 10mV phono cartridge signals to the 100-200mV optimal for line level. In fact , many power amplifiers would expect input levels of 1V rms to deliver full power, which means we need to get a 100 x gain from the first stage, bringing R2 into the multi-megohm range. High value resistors are undesirable, due to noise considerations, as mentioned before, and excessively large gains in the first opamp are also not desirable due to degradation in output impedance and harmonic distortion of the stage.

A ceramic or crystal phono cartridge (used on many consumer - non-HiFi record players) would have much higher a signal output of 100-200mV, and could be used as input to the tone control.
There still remains the issue of providing RIAA compensation, however doing so later in the signal path is not impossible, and would even increase overall S/N ratio of the system (by retaining high frequency emphasis throughout the signal path, and de-emphasizing at higher signal levels.)

Construction

Soldering to a double sided PCB requires more heat and more time than to a single sided PCB.
You have to heat 2 copper pads (top + bottom) the copper channel that connects them, and the component lead to flow temperature before solder will melt. And you have more solder to heat and melt. Once melted, you must continue applying heat for 2 seconds to allow solder to reach the opposite side.
More than twice as much solder is required. Enough solder to make 2 solder cones , and fill the channel. Removal will be substantially more difficult so avoid mistakes.
Some components (capacitors / resistors) may have multiple solder pads for inserting pins. There may be pads at, at 5mm, 7.5mm and 10mm distance. Some of these pads are electrically shorted. This is to accommodate parts of different pin spacing. It is crucial that you insert the component with its pins to pads that are NOT shorted. Be sure to look on both sides of the PCB to avoid using 2 shorted pads, or you will short-circuit the component and the circuit will not work.

All components, including potentiometers will be installed on the top side of the PCB. This is the printed side, with part number legends. Ideally, the potentiometers should be soldered in position on the PCB after they have already been secured to the front panel with their mounting nuts. Stressing the pots by forcing them to fit through front panel holes after they have been soldered, is likely to crack them. It is therefore safer to first loosely screw the pots to the front panel, then position them and insert their pins into the PCB. Then solder only ONE pin (a middle pin) of each pot to the PCB. You can still, at this point, re-heat the single pin solder joint to re-position the pot slightly if needed and relieve any stress. Tighten the nuts to the front panel, and if everything is good, then solder the remaining pins of the pots. It is only after this , that any internal mounting of the PCB to the chassis (using the 4 corner holes) should be marked and drilled. This may generally not be necessary, as the 3 potentiometer mounting screws can well support the circuit, and the pots are the only part of the circuit that undergoes mechanical stress.
IC's should only be installed into their sockets after you are finished handling the board, to avoid undue exposure to ESD. All the other components can be mounted either topside or bottom side, if convenience requires it.

The only important positioning rule is that the potentiometers and the IC's go on the printed side, or the circuit will not work.

Start by inserting all resistors first. Resistors only, nothing else. Components will have to be inserted and soldered strictly in the order of their height above the PCB.
A good order for soldering is:
1. solder one end of each component first - this is an anchor joint - it may be imperfect, because the component may move,
2. solder the opposite end,
3. reheat the first soldered end and reflow it to perfection.

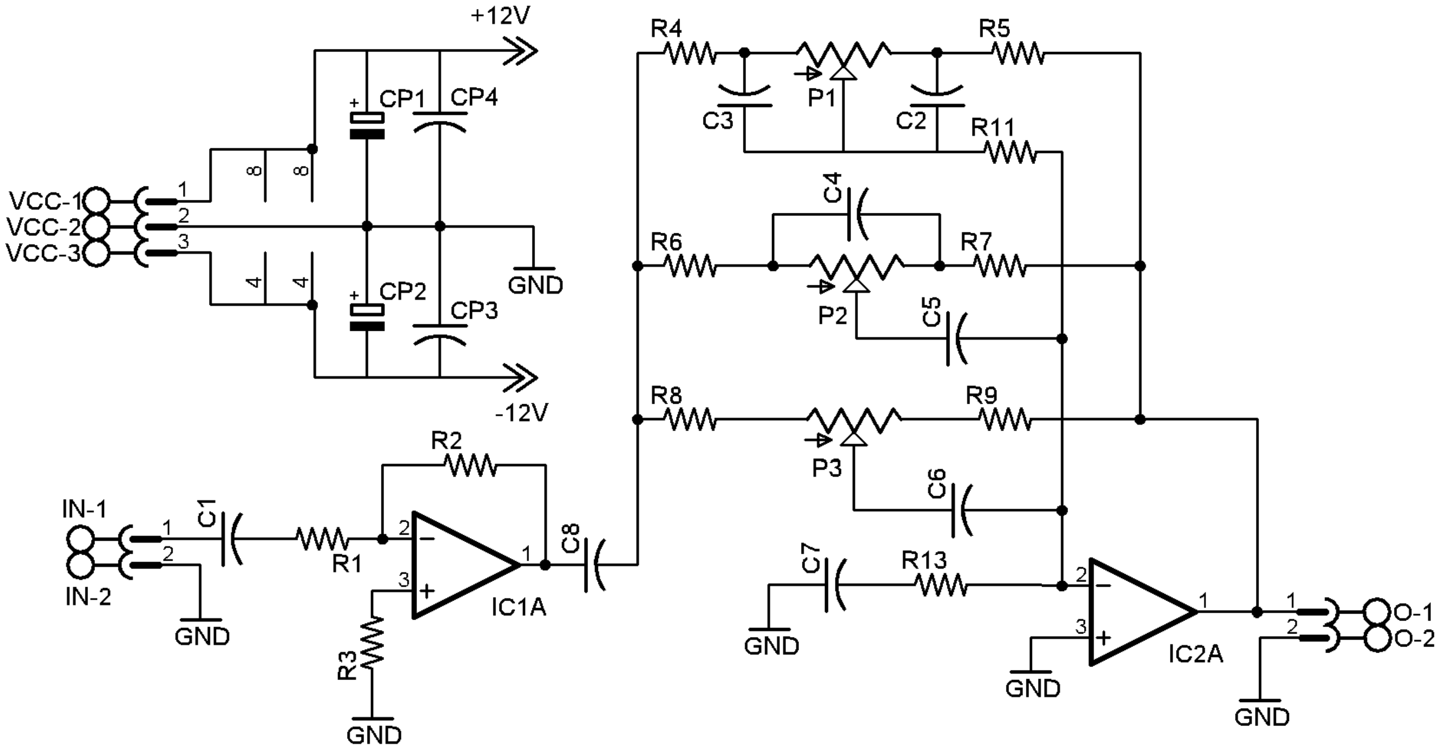
By first soldering one end of all resistors, you anchor all of them so they don't move.
Because components are initially free to move, this first solder joint may turn out frosty, as the part may have moved slightly during cooling. This is ok, it is only an anchor joint. It will keep the component fixed so the opposite end will get a perfect joint.
You then solder the opposite pin of each part, in order, trying to spend no more than 10 seconds with the soldering iron at each pad. This second solder joint is done while the part is firmly held, so you have no excuse not to make a perfectly shiny cone.
After all parts have their second lead soldered, return and reheat all the first joints, to allow them to re-flow without movement, creating perfect solder joints.
By only soldering one end of each component at a time, you give time to all parts to cool down between jobs, not overheating any of them.

Perfect solder joints are extremely important here, because high quality audio requires the best connections. Non-acid (greasy) flux paste can be used, to enure quick, good joints. Extremely small amounts of flux are sufficient to facilitate soldering. Excess flux then needs to be removed from the PCB with a solvent like acetone after the board is completed.

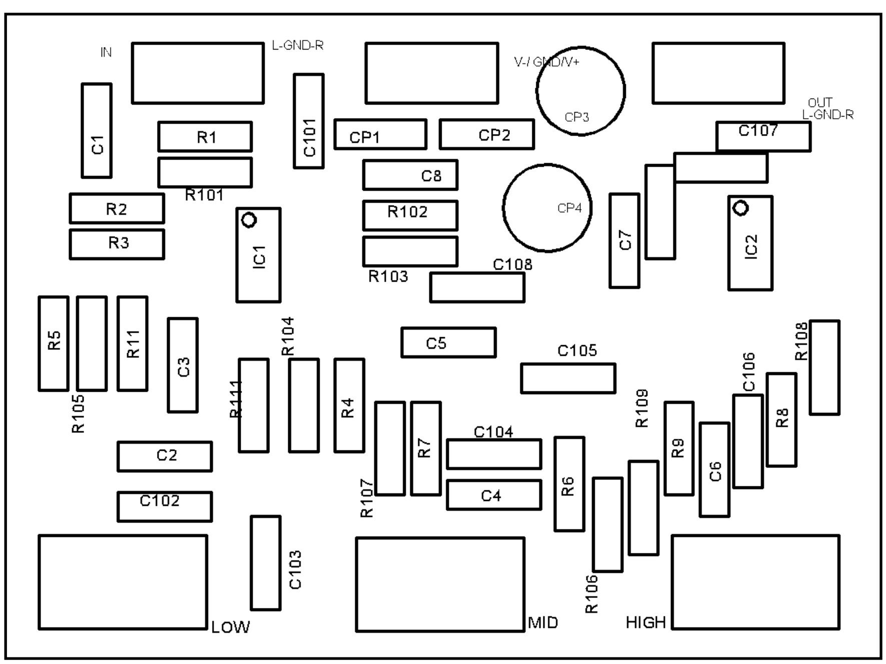
Add the IC sockets next. Insert them into the PCB with the PIN 1 index marker to match the index painted on the PCB. This is important. Once inserted , the IC sockets will obscure the white legend on the PCB, and the socket marker will be the only reminder you have of which way to insert the PIN 1 end of the IC. Place the PCB face down, so it is seated on the IC sockets. Solder one pin of each socket to anchor them. Then solder the remaining pins. Try to NOT solder them in order. Heating 3 adjacent pins for 10 seconds each can supply enough heat to melt the plastic socket frame. Go from one pin to an opposite pin, and then to the other IC socket, before returning to an adjacent pin. Do not insert the IC's into the sockets at this time.

Continue adding the next taller components to the PCB. Capacitors, then terminal blocks, then potentiometers. You only need to bend resistor and capacitor wires slightly outward on the bottom side, to keep them from falling. Bending their leads too far from the vertical plane, will make it more difficult to properly snip all excess leads after soldering, and will also risk them touching adjacent component leads, or directing the solder sideways and causing a short-circuit.

K330 3-way tone control schematic diagram
Once channel only



K330 3-way tone control PCB component placement



Electrolytic capacitor, CP3 mounts with its negative (-) terminal towards the RIGHT. CP4 mounts with its negative pin towards the TOP.

Be careful both handling and soldering the potentiometers. Excessive force on the pins can cause the phenolic substrate to crack. Potentiometer pins should fit easily in their holes. Once soldered, do NOT twist, push, bend or otherwise try to adjust the position or angle of the potentiometers. The pins are very firmly soldered into the PCB and WILL NOT move, the only thing that will give are the phenolic pin supports, which may crack. Such a crack will kill the potentiometer. The phenolic supports carry the electrical contacts and any crack will result in a broken conductor path. No repair is possible, and removing the damaged pot will be a veritable challenge, possibly damaging the PCB.
So, ensure the front panel holes in the enclosure are correctly positioned so the potentiometer shafts will fit without stress or force. Once the pots are secured to both the front panel and the PCB, do not twist or nudge them. Allow some play in the fasteners that secure the PCB inside the case (if using them) , such as with rubber or flexible grommets, so you will not stress the potentiometers when securing the PCB.

The screw terminals allow solderless, removeable connections between the circuit and the enclosure jacks. More than one wire can connect to each terminal. Shielded wires should be used to connect the sound input and output lines from the PCB to the enclosure terminations.

The fasteners that secure the PCB to the enclosure should not make grounding connections. PCB ground should connect to enclosure ground at one point only. Ideal fasteners for the PCB will be nylon standoff snap-in fasteners.

Schematic diagram included on previous page shows one channel of the circuit only and power components. Second channel component numbers are the same as those for the first channel, but with an "A" added at the end. For purposes of this circuit, the "A" designated components belong to the RIGHT channel, and the plain number components belong to the LEFT channel. Parts starting with "CP" belong to the power supply and are not channel-specific.

Component values are not shown on the schematic, but on the last page of this leaflet. Some component values (such as the working voltage of the capacitors) may change in future versions of the kit, and this will be reflected on the last page.

Power Supply

A rectified and filtered bipolar (dual rail) power supply is needed. Minimum voltage is +/-5V, and maximum +/-17V. A +/-18V supply will allow a maximum output voltage of 11V rms from the circuit, while a +/-12V supply will allow a maximum output of 7V rms.

The maximum current draw is 65mA, and this will only happen if the circuit is operating at +/-18V supply. Filtering capacitors of at least 3000uF per rail should be provided in the power supply, if unregulated.

Although a regulated supply can be used, regulation is not likely to bring any noticeable benefit, so long as the power transformer is capable of 100mA output.

Signal levels and power supply

The supply voltage determines the maximum signal levels that can be processed without clipping (distortion). A 1Vrms input signal will have a peak voltage of 1.41V. If a gain of 5 is chosen for the input stage, the signal output from the first opamp will be 1.41 x 5 =7.05V peak. The IC's cannot swing up to the supply rails. They can only come to within 2V of the supply rails, and if distortion is to be kept low, they can only come within 3V of the rails. That means that for a 7.05V peak signal to be reproduced, the supply voltage to the opamp would need to be at least +/-10.05V.
If you use a supply voltage of +/- 8V, the signal will be clipped (peaks of waves will be flattened), which results in very audible distortion.
So logically, you might want to use the highest supply voltage possible to prevent running out of signal headroom.
The IC's can withstand a maximum of +/-18V supply, but to be safe you should not go over +/-17V. With a 3V headroom, this allows them to handle a maximum signal of about 10V rms.

Similarly, the input stage of the power amplifier (or other devices that follow this unit) can be overloaded into distortion if the signal level is too high compared to the power supply voltage used in that equipment. Often the only way to find out is by trying.

It is theoretically desirable to try to work with the largest signal levels possible at every processing stage, since the noise added by every component is constant regardless of signal level, and
thus higher signals will yield higher S/N ratios.

The choice of input gain however, needs to take into account the peak values of the input signal and the tolerance of driven equipment to large signals.
A signal that is specified at 1V rms may have brief peaks that exceed 15Vrms. These may happen only once or twice in a song, but if they do, distortion may occur.
Music dynamic range can exceed 80dB (this is what CD's bragged about when they replaced vinyl), which means the loudest sound can have a voltage level 10,000 times higher than the softest sound. You do the math, and you can see that finding a signal level that both brings out the quietest sounds, and does not overload equipment can be tricky. Having to switch between multiple input signal sources that have intrinsically different signal levels, generally forces manufacturers to maintain signal levels conservatively low to avoid clipping.

Since you have the luxury of making some decisions about your input gain, you may want to test your setup with actual music, while installing gain resistors temporarily at the input stage.
Resistors R1 and R2 can be soldered in a way that makes removal easier. Do not push the resistor all the way down to the PCB . Allow its leads to protrude to the bottom of the PCB only enough to be soldered, and

leave the resistor with long leads on the top side of the PCB, towering above other components. You can thus remove it if you want to change gain setting.

Going further

Although some level of experimentation is possible with component values, to achieve various frequency response schemes, this can only be done within a narrow range.

The 3-way tone control is a combination of high-pass, low-pass, and band-pass active filters. It is not a parametric equalizer. Although traditional (Baxandall) bass/treble controls have always had simple mathematics, and were easy to design, adding a third bandpass filter into the circuit is not trivial. In fact, the mathematics increase in complexity so much, that they become impossible to resolve except by iterative process. Arbitrary values are thrown in to the equations, and many iterations are calculated until relatively consistent part values are determined.
This complexity arises from the fact that each filter interacts to some degree with the operation of the others. The default values provided with this circuit result in minimal frequency interactions, which are inaudible in use. However, departing from these values, begins to cause increasingly audible interactions. (You could almost say that the component values designed for this circuit lie in a “sweet spot” and are nearly the only values that will work)

In practice, you could move the working frequencies of the 3 bands by up to a factor of 2 up or down, without significant interaction. Certainly shifting ***up*** the corner frequency of the treble control, or shifting ***down*** the frequency of the bass control are modifications that can be made without worry.
Any frequency change would need to involve changing some capacitor values. Changing resistor values is out of the question. A very delicate balance was achieved with the resistor values, and those should not be disturbed.

Essentially, doubling a capacitor value in the bass or treble controls, lowers the corner frequency of that control by a factor of 2 (an octave). Conversely, halving the capacitor value, increases the frequency by a factor of 2.

The most common modification involves moving the bass frequency down from 60 Hz to 45 or 30Hz, to allow controlling of frequencies that feed a sub-woofer, and avoiding muddy bass when increasing low-pass response. The alternate 0.068uF capacitors provided, can be used in place of the 0.047uF for C2 and C3 in the schematic to lower the bass frequency to 45Hz. You can further lower this frequency to 30 Hz, by providing your own, 0.1uF capacitors.

For the high-pass filter, decrease C6 to raise the band where the treble control is acting, and increase it to lower the band. Use a 1.5:1 or 2:1 ratio at most. Be aware that lowering this band by more than 1.5:1 will clash with the mid-range control.

For the mid-range control, increase or decrease C4 and C5 together, maintaining the value of C5 always about 5 times larger than C4. This will move the center frequency up or down. Again, shift frequency no more than a 1.5:1 or 2:1 ratio at most.

Omitting the center band components (R6, R7, C4, C5, P2) will create a 2-band circuit, without any adverse effect on the remaining controls. Greater latitude in moving the bass and treble frequencies will be possible in that case.

Although it is conceivable that you could allow selection of input impedance and gain (R1 and/or R2) by front panel mounted switches, if such a setup is desired, then the wires connecting to those switches would need to be shielded wires. Noise pickup becomes a possibility, so extreme care is required. Two shielded cable segments will be needed for each resistor that needs to be switched. The core of each cable will go the each resistor lead, and the shields will all be connected to the PCB ground at a single point. Only one end of the shield should connect to ground. The opposite end remains unconnected.
The potentiometers can likewise also be connected to the PCB via shielded wires, if the circuit needs to be positioned farther from the front panel.

Noise pickup from the power supply (60Hz and 120Hz hum) can be prevented by enclosing the power supply in a ferrous metal shield, connected to ground.
If a metal enclosure is not possible, a shielding cage can be constructed from plastic sheet (or even cardboard) which has adhesive copper or iron foil applied on it. Electrical connection to the foil can be made using a machine screw and nut, since soldering would melt or char the underlying plastic.
The circuit can first be tested with it's power supply unshielded, to determine if a hum problem really exists, before planning for a shield. Many times this is not necessary.
The power supply can also be kept outside the circuit enclosure, feeding only the filtered DC to the tone control case.

Component removal

De-soldering components from a 2-sided PCB can be a challenge. It helps to take a 2-step approach: first remove as much solder as possible from each joint by using a solder sucking pump and/or wire braid. Then re-heat each joint on bottom side, while gently pulling up on the component lead with long-nose pliers from the opposite side. Do not start pulling until the solder in the joint has been flowing for at least 2 seconds. Opposite side solder tends to melt later than heated-side solder, and you risk pulling away the copper pad if you pull prematurely.
It is always best to **think first**, **plan ahead**, so you won't have to remove anything. If you think a component installation may be temporary, leave long leads, then remove by snipping off the leads just above the PCB. Desoldering the lead fragments is then an easier task.

3-Way Tone Control Kit K330

See the latest documentation at:
www.siliconwiz.com/k330/doc.pdf
Please download, save, and print this file. Above URL may change in the future.

Parts List

Part values and markings, one channel shown.
Second channel uses same part numbers, with suffix "A" .

C1, C8 1uF 1uJ 100 x4
C2, C3 0.047 xx473J x4
C2, C3 0.068 xx683J x4 (alternative)
C4, C6 0.0047 xx472J x4
C5 0.022 xx223J x2
C7 0.001 xx102J x2
CP3, CP4 0.1 ceramic 104 x2
CP1, CP2 50-100uF 25V+ electrolytic x2
R1, R2 22K red-red-black-red-brown x4
R3 953 ohm white-green-orange-black-brown x2
R4, R5, R11 11K brown-brown-black-red-brown x6
R6, R7 3.6K orange-blue-black-brown-brown x4
R8, R9 1.8K brown-grey-black-brown-brown x4
R13 270 ohm red-purple-black-black-brown x2
R 10K, brown-black-black-red-brown alternative values for R1/R2 x2
R 47K, yellow-purple-black-red-brown alternative values for R1/R2 x2
R 100K, brown-black-black-orange-brown alternative values for R1/R2 x2
R 953 ohm white-green-orange-black-brown alternative values for R1/R2 x2
IC1, IC2 LME49720AN x2
IC Socket-8 pin x2
Terminals, screw - 3 position x3
PCB x1
P1, P2 Potentiometer, stereo 100K linear x2
P3 Potentiometer, stereo 500K linear x1

IMPORTANT:
Some resistor color codes may be difficult to read. They may include extra color bands for added precision and/or temperature coefficient. This can make it difficult to understand which end to start reading from. As a precaution, measure the resistance with an ohmmeter to confirm values.

Note that values and appearance of parts supplied with the kit may change, as updates and improvements are made. The parts supplied with a future kit may not be identical to the parts supplied with this one. If cosmetically identical builds are required, it is best to purchase all kits at the same time.

For easy identification of circuit position, all parts belonging to the second channel use same part designations as the first channel, with "A" added after it. "CP" designated capacitors are part of the power supply.