The PAiA MS Stereo Microphone

This smooth and accurate stereo microphone will be a great addition to your audio arsenal. Frequency response is 20hz to 20Khz and noise specs are comparable to most pro audio condenser microphones.

The PAiA MS Stereo Microphone produces a spacious, airy stereo image that's ideal for recording acoustic instruments. And if you initially record "too much" stereo you can reduce spread, or even take the mix to mono, with no danger of frequency dependant phase cancellations changing the timbre..

Mid-Side (MS) microphone placement for stereo recording is a favorite trick in high end studios. It's used less often in project studios because it requires expensive dual element condenser microphones and an elaborate mixing board. The PAiA MS Stereo Mic uses three high quality electret microphone elements and analog mixing and matrixing circuits to produce a compact and inexpensive Mid-Side (MS) Stereo Microphone.

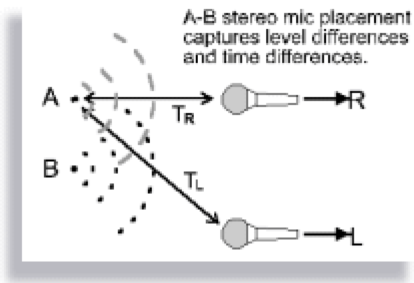
Side and Mid level controls set stereo spread at the mic's Left and Right outputs. Outputs are adjustable from 4dB line level down to low level and are balanced on 1/4" TRS jacks. Mono plugs can be used for unbalanced connections and cables.

How MS Stereo Works

by John Simonton

Usually the most intuitive way to perform a task is the best ... but not always.

Since we have Left and Right ears, the most intuitive way to encode stereo sound information is as Left and Right channels. And stereo delivery media such as CD, tape, vinyl and older film do it just this way.

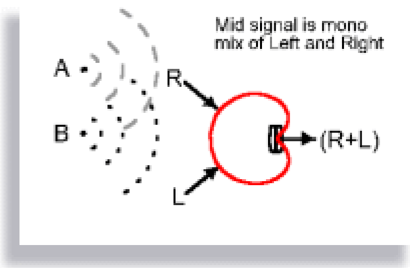


Similarly, when placing microphones to record a stereo signal the most intuitive way to go about it is what's known as A-B mic'ing. Two mics are placed some distance apart and pointed in the direction of the source to be recorded. The mic on the left is recorded as the L channel and the one on right as R. On the face of it this looks pretty fool-proof and when you listen to a ping pong game recorded this way, sure enough - it sounds like stereo.

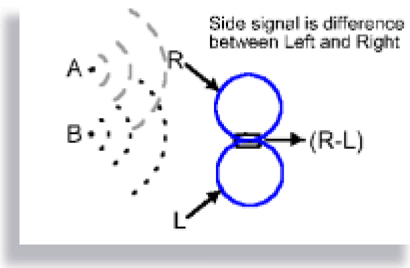
But when the novelty of ping pong wears off and you progress to more sonically complex sound sources (like musical instruments) some subtle problems begin showing up that have to do with the fact that the mics are encoding the L and R information not only as level differences but also as time differences. In the figure, the distance from the "A" source to the "L" mic is greater than to the "R" mic so the "L" signal is slightly delayed relative to the "R" signal.

On playback, there can be listener positions where delays from the speakers interact with the recorded delay to produce comb-filter effects. In other words, the timbre may change depending on the listener's location. There may be several randomly located "sweet spot" locations in a room where everything sounds great and other places where bass is weak and others where it's overbearing and other places where high end and mid range are problems. Also, when an A-B mic'd stereo signal is converted to mono - like when heard on most AM radios - some notes, even whole instruments, can go away because of phase cancellation of recording articfacts.

There are alternatives to A-B Mic'ing that prevent these problems, MS recording for instance. MS is an acronym for Mid-Side, named for the two signals that encode the location of sound sources and reverbrant ambiance of the stereo image. MS Stereo principles are used in the broadcast delivery of Mono/Stereo compatible sound and as part of Surround system but it is how these principles apply to microphone selection and placement during recording that is of concern here.

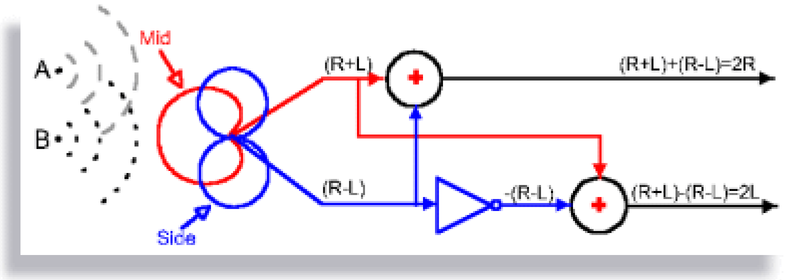


The Mid signal comes from a mic with a Cardioid polar sensitivity oriented so that the "front" faces the center of the stage. This signal is essentially a mono mix of the Right and Left sides of the stereo field and encodes very little information on placement of sound sources. For example, if the source "A" in the illustration is moved from the Right side of the stage to the equivalent position on the Left, the Mid signal will not change significantly. Mathematically the Mid signal can be expressed as R+L.



The Side signal comes from a mic with a Figure 8 polar sensitivity oriented so the lobes are facing Right and Left. Two characteristics of this map are important. First, the mic is deaf to sound sources located between the two lobes; for example, source "B" in the illustration will not be present in the Side signal at all. Also, there is a phase inversion between the Right and Left lobes - if source "A" is moved from the Right side of the stage to the equivalent postion on the Left, its relative phase will be inverted in the output. Math shorthand for this is R-L.

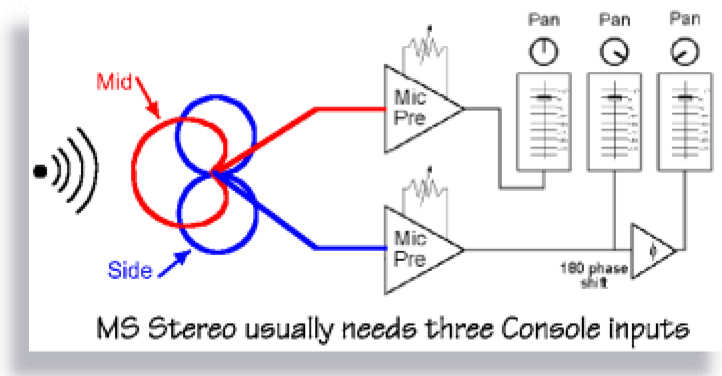
The MS Decoder performs sum and difference operations to extract Right and Left level and phase information from the Mid and Side signals. In the illustration the circles with "+" signs represent summing functions. The triangle represents a 180º phase inversion, the equivalent of changing the sign of the output relative to the input or multiplying the input by -1.



In an actual circuit OpAmps compute sums and phase inversion.

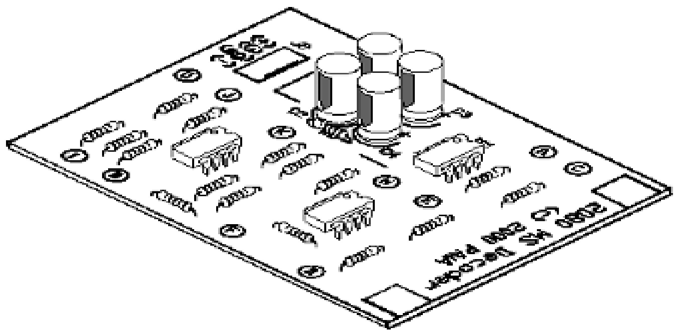
FM Stereo, Stereo TV and even some quad and surround sound systems use MS encoded signals because it is an easy path to sending a signal that is mono/stereo compatible. In these systems, the Mid signal modulates the main sound carrier while the Side signal modulates a stereo subcarrier. In simple monophonic receiving equipment only the main carrier is demodulated and the Mid signal is used for a mono output. When stereo outputs are desired, the stereo subcarrier is also demodulated and the Side signal combined with the Mid signal in a decoder to extract Left and Right Stereo channels.

MS Stereo Decoder / MS Mic



MS (Mid-Side) stereo is a classic recording method that produces a spacious, airy stereo ambiance ideal for acoustic instruments, vocals or small ensembles. Unlike A-B and even XY co-incident mic'ing, Mid-Side stereo does not produce the frequency dependent phase cancellations that can muddy sound. If you initially record "too much" stereo you can reduce spread, or even take the mix to mono, with no fear that comb filtering artifacts will wipe out that incredible timbre you captured in the original recording.

A disadvantage of MS Stereo is that normal Right and Left signals are not directly available. Instead, Mid and Side signals encode the phase and level information of the stereo stage and require some processing to produce Right and Left. Most large studio consoles can do this if you're willing to dedicate three inputs to a "stereo pair", as shown in the illustration above. Smaller home studio consoles don't have all the bells and whistles needed to do it all. So what are the options? One is this simple and cost-effective MS Decoder.



In this design by Jules Ryckebusch, low-noise 5532 op-amps are used to realize the inversion and summing matrices required to decode the L and R information from the Mid and Side Mic signals. MS stereo uses microphones with different polar sensitivities: A Cardioid response mic for the "Mid" and a Figure 8 response mic for the "Side". At the pro level, a pair of Neumann U87 mics, one set for Cardioid and the other for Figure 8, are typically used for both.

A Practical MS Decoder Circuit

by Jules Ryckebusch

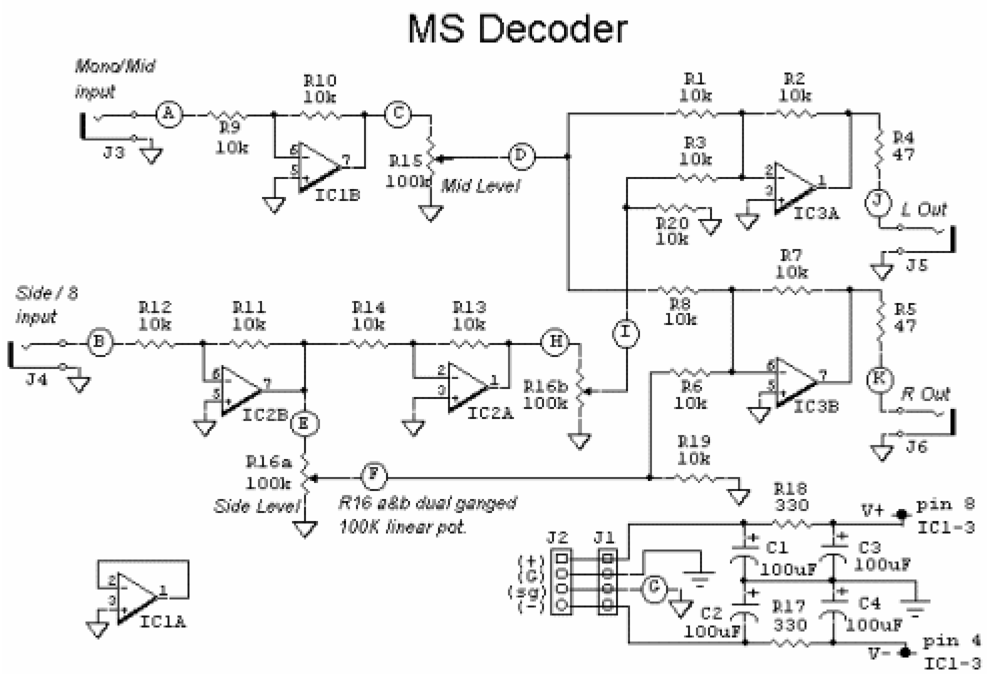
The schematic diagram of the MS Decoder may look complicated but is actually quite simple. Both the Mid and Side signals are initially buffered by unity gain inverting buffers formed around IC1b and IC2 b. This is necessary for two reasons, first to ensure enough drive current for the following sections. Second, the final summing sections invert the signal, to achieve zero phase shift through the unit, one more stage of phase inversion is required.

The Mid signal goes to level control potentiometer R15. It is then fed equally to the left and right summing amplifiers, which are formed around the two sections of IC3. The use of R15 (and R16) is to allow adjustment of the relative levels of the mid/side levels independent of the mic-pre gain setting. This is useful for directly feeding a recording device.

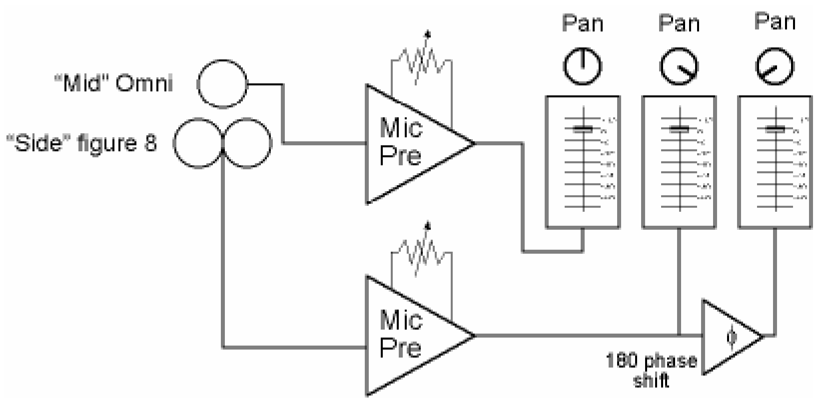
The Side signal has a little different path. After initial buffering, it is fed to the right summing amplifier via one section of dual potentiometer R16. It is also fed to a unity gain inverter formed by IC2a and its associated resistors. This inverted signal goes to the other half of the dual ganged potentiometer section. Then it is summed into the left channel via IC3a. One section of IC1 is not used. Both of its inputs are tied to ground to keep any thing strange from happening.

R17, R18, and C1-4 are for power supply filtering. The circuit will run on +/-9 to +/-18 VDC.

OK, so what are R19 and R20 doing? That is an interesting question. These are there to load down R16 so that when the levels of the Side signal are adjusted the potentiometer gives the same feel as the Mid level adjustment potentiometer R15. The op-amp summing sections are virtual grounds. This means that signals entering the op-amp see a load equal to the input resistor. All of these are 10K resistors. The Mid potentiometer feeds two of these so it is presented with a 5K load. To make the load on the Side potentiometer the same one additional 10K resistor tied to ground is added to each wiper so it "sees" 5K also. Here is the other interesting thing that all of this causes. Because all the potentiometers are linear taper, loading them down with a load much smaller than the resistance of the potentiometer causes a change in the characteristics of the potentiometer. This in essence, makes them respond more logarithmically than linearly, which is the way we hear anyway which is a good thing!



Alternatives:



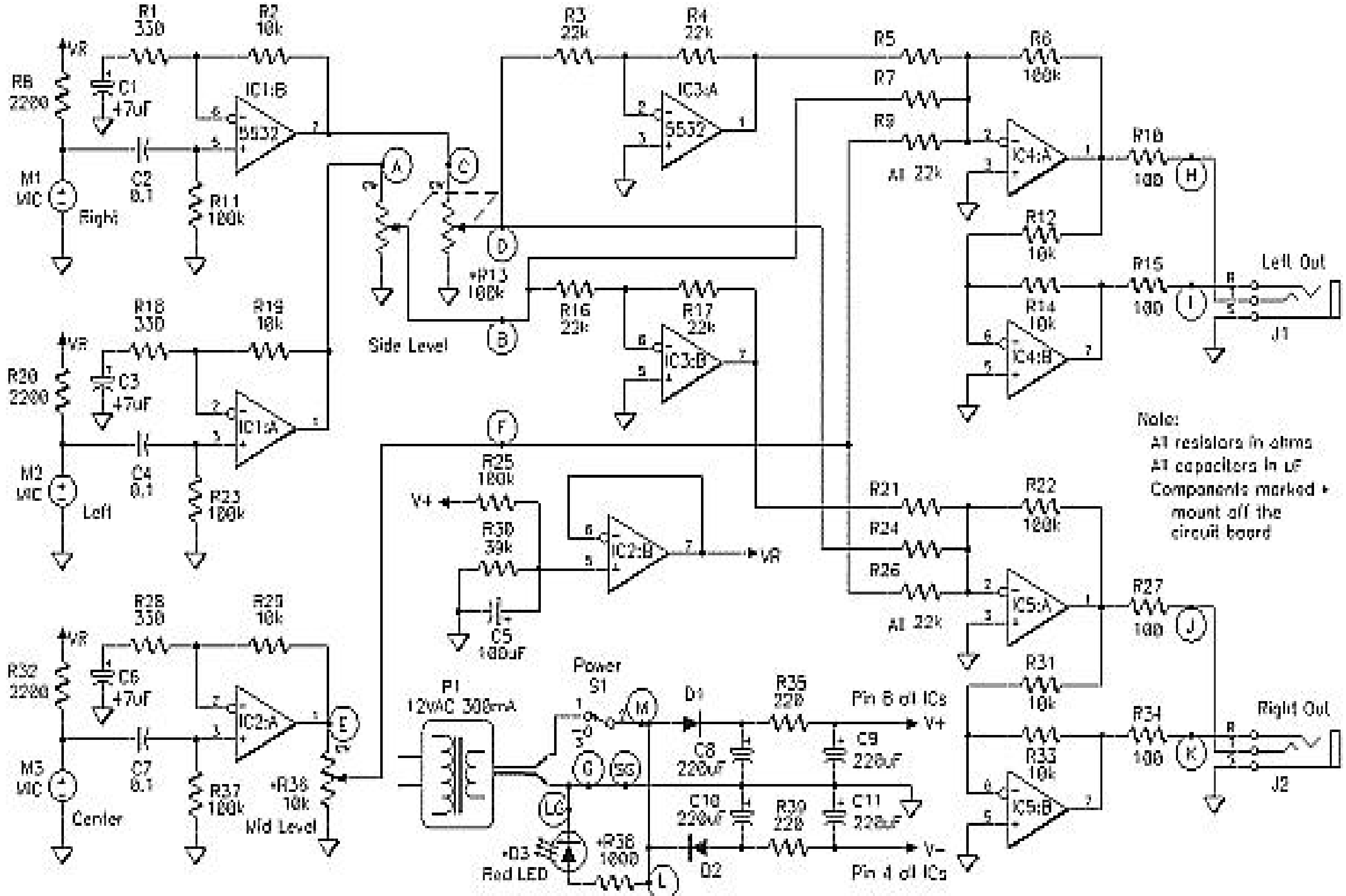
Well, you can always break out the patch cords if your console supports phase switches. Be careful, some only have the phase switch on the channel microphone pre-amp, not the channel itself. Simply patch out of the preamp on the first channel and into the line input on the next channel. Depending on the patch bay, you may need to use a half patch technique, which is beyond the scope of this article. Before panning hard left and right, center both and adjust the level on the second channel until the side signal is completely cancelled. This means that you have the signal levels matched. Now pan one hard right and the other hard left. Most multi-track audio programs allow you to do this too. I use Studio Vision Pro and Pro Tools but I know most others have similar features. Take the side signal pre fader and send it to an effects buss. On the original channel pan it hard right. On the effect buss, phase invert the signal and pan it to the opposite channel as the original. Voila! You are done. There are usually many ways to skin a cat.

So why build or use an outboard MS Decoder? The real benefit from having a dedicated MS Decoder is that you can listen to the effect in real time while you position the microphones and make adjustments before recording. I have in the past, recorded the Mid and Side signals directly only to find out they really didn't work together while decoding during playback and mix down. We were recording a Leslie and fortunately after taking my lumps, I was able to re-record it. I had to buy a round of beers over that one.

Use:

Just patch the Project r MS decoder between your microphone pre-amps and your mixing board. You can also run the signal straight into a DAT or you're A/D converters if recording into your computer. The key to good results is the ability to monitor the stereo signal while recording. Then you can hear exactly how the final MS recording will sound. This allows adjustments of the relative levels of mid/side and microphone positions. Normally the Mid level will be fully up and the Side level almost all the way up. Depending on the amount of ambient information, you may want to back off a bit. In a relatively dead space, you may want to do the opposite and back off on the Mid signal a bit.

Now here is something that is really different: Faux MS miking. So you don't have Neumann U87. In fact you don't have a figure 8 microphone at all. Well, fake it. Close mike your source with one microphone and send that to the Mid input. Then take a second microphone and place that back form your sound source to capture room ambience and send that to the Side input. Viola! Fake MS miking. It isn't the real thing but it sounds good. It is completely different than just panning the second microphone to a different position than the first. To hear what this sounds like just switch the Figure 8 microphone to cardioid or omni. Another interesting thing to do is to separate the two microphones used. Normally the two microphones are as close together as possible. I have recorded in a blues studio in Florida that uses a figure 8 mic as the room mike and mixes that in the same way as MS miking with the exception that there isn't even a Mid mic. This works great as a drum overhead too. Experimentation is the key. With that Leslie I mentioned earlier, we had the mid microphone in the bottom of the cabinet where the large speaker and rotating baffle are. The Side mic was placed about a foot from the upper rotating horns and about three feet from the Mid mic. Sounded great. Happy Soldering. Even if you don't build the Project r MS Decoder, hopefully you picked up a couple new tips for your audio arsenal.



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Date 10/14/95 Drawn by js		Rev 4
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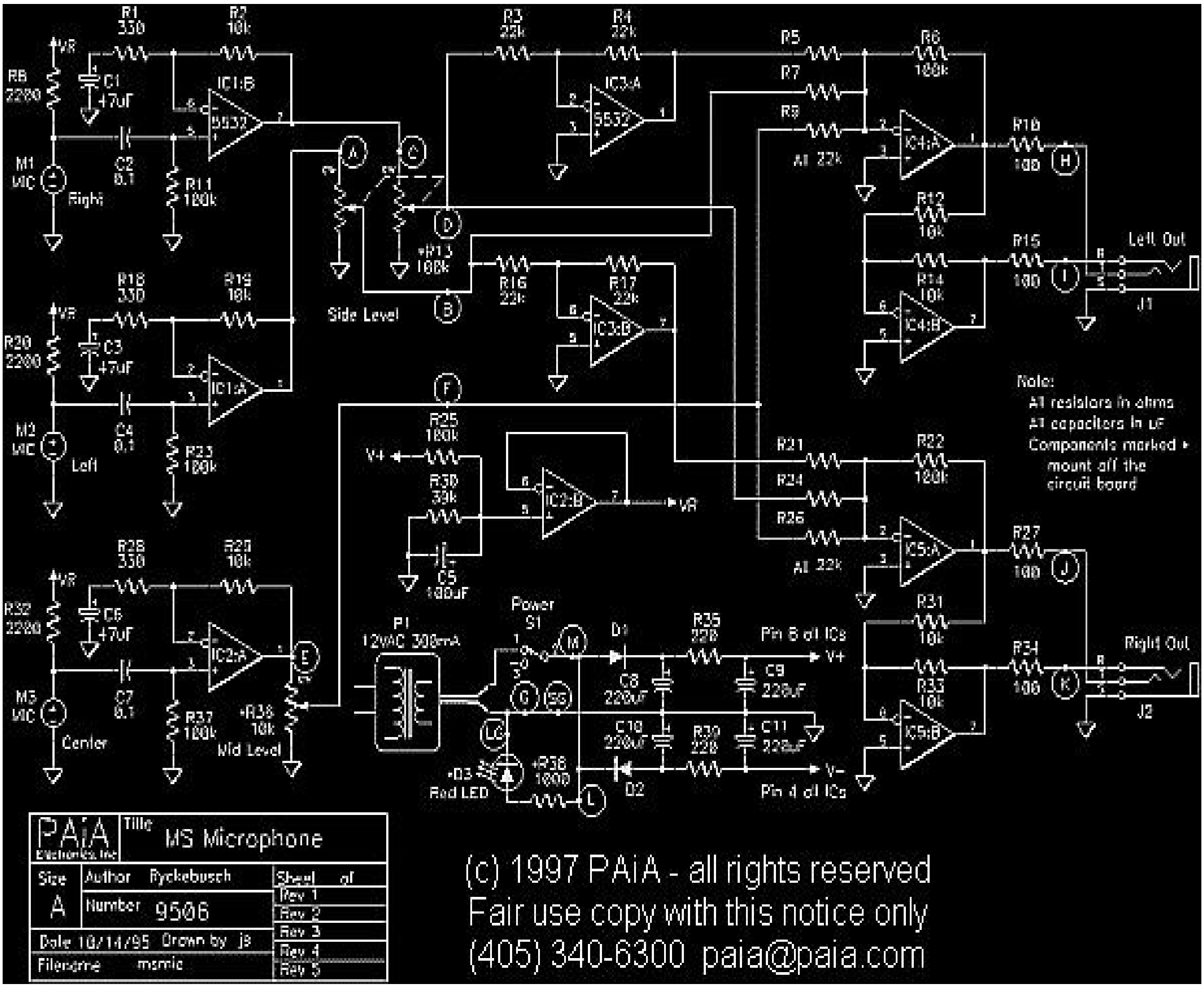
PAiA Electronics, Inc
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Size A
Author Ryckebusch
Number 9506
Date 10/14/95 Drawn by jb
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Rev 1
Rev 2
Rev 3
Rev 4
Rev 5
Note:
All resistors in ohms
All capacitors in uF
Components marked • mount off the circuit board
Side Level
Mid Level
Left Out
Right Out
Power S1
12VAC 300mA
Red LED
Pin 8 of ICs
Pin 4 of ICs
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