System Sound manual

manual version 1.3.1 for System Sound version 3.0

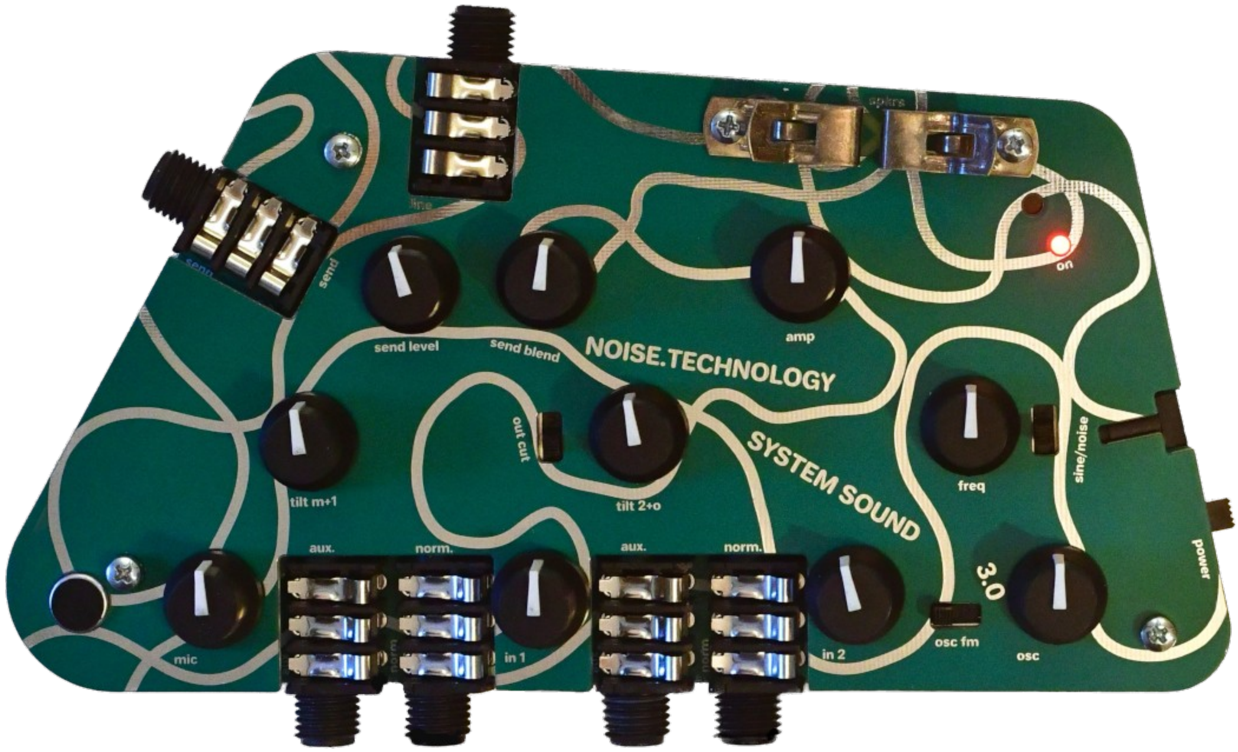

Figure 1: System Sound 3.0

Firstly, let me tell you that you don't have to understand how the System Sound works in order to use it. If you just want to make noise, turn it on and move knobs and switches. But sometimes a knob may seem inactive, or not do what you expect, at which point you may want to read this to make sense of it. It's not that difficult, even though this manual may seem wordy. I'm just trying to explain things plainly, rather than have lots of diagrams and complex stuff.

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Signal flow

But, lets begin with a drawing of the signal flow. It's what makes the System Sound be what it is—in essence, it consists of very simple subcircuits connected in a way designed to explore acoustic and electronic feedback sounds.

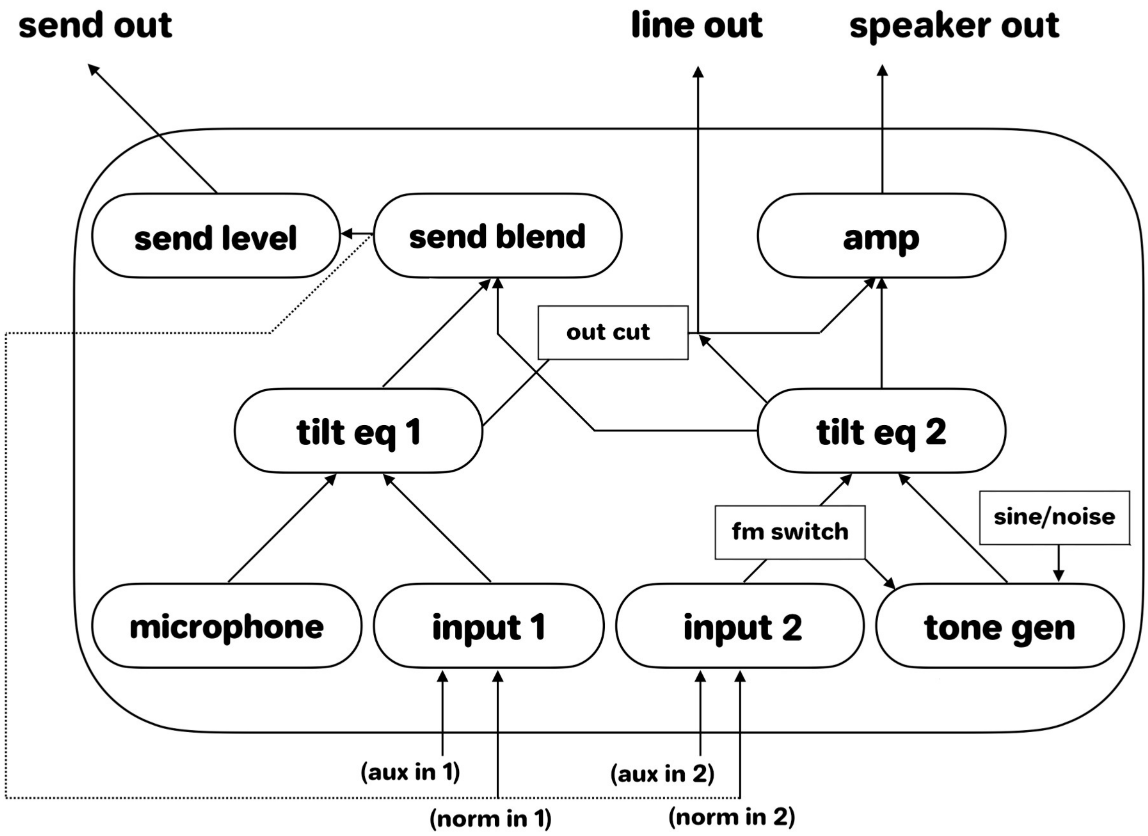

Figure 2: signal flow of the System Sound 3.0

In short, in writing:

- **The microphone and aux+norm inputs 1 go to tilt eq 1.**
- **Aux+norm inputs 2 and the oscillator go to tilt eq 2.**
- **Both tilt eqs go straight to the line output.**
- **Both tilt eqs go to the amplifier level control.**
- **If nothing is plugged into norm input 1 or 2, the send signal (e.g. the mix chosen with the send blend control) is instead presented to the inputs. Feedback occurs.**
- **The send blend control selects which of the tilt eqs goes to the send output.**
- **The send level control sets the output level for the send jack (not the normalised feedback return).**
- **The amplifier receives the signal from tilt eq 1 and 2 and sends to the speaker terminals (after a level control)**
- **The line out is an unattenuated mix of tilt eq 1 and 2, it has no level control.**

- **A switch can reroute input 2 from eq2, to the frequency control of the oscillator.**

(as this diagram only shows the flow of the signal, the other controls for the oscillator aren't mentioned. In the drawing, an oval corresponds to a potentiometer, and arrows entering or leaving the large oval are connectors (jack or speaker terminals).

The controls

Controls—potentiometers

I've laid out the pots on the System Sound in a way that is easy to memorise, so you can see (mostly) to which thing (e.g. microphone, output connectors, send) they correspond. The pots are nearby the thing they interact with.

To ease understanding, I've laid out the controls in three "rows". The bottom row I call the input row. It controls how much of the various signals go into the circuit.

From left to right, we have the levels for microphone, input 1, input 2, and the oscillator.

As you can see, the oscillator level control is near the oscillator chip, and other controls for the oscillator (explained further on). The two input level controls are next to their jacks, and the microphone level is next to the microphone. So it's easy to see which one controls what even without reading the text.

All of these level controls have a lot of gain, and can easily distort.

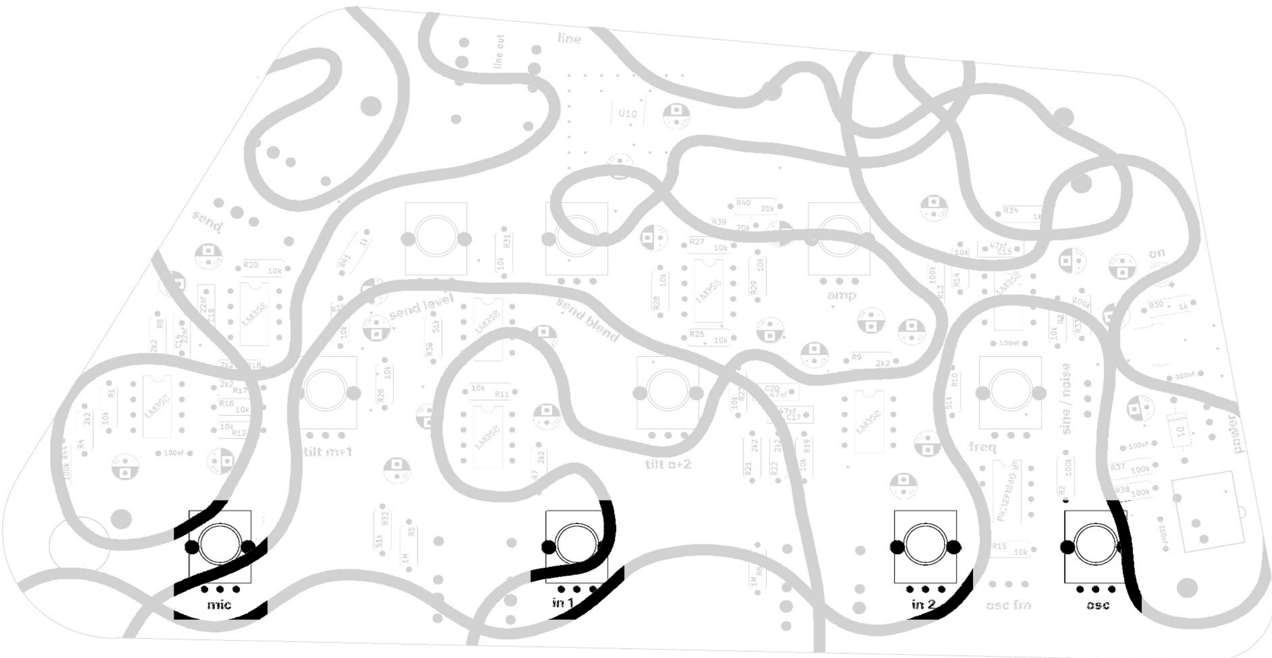

Figure 3: input row

The second row I call the frequency row. From left to right, we have first t

tilt eq 1, tilt eq 2, and the oscillator frequency. The tilt equalisers both cut bass and boost treble as you turn clockwise, and cut treble and boost bass as you turn counter clockwise. With the knob facing straight up, they let all frequencies pass equally.

The tilt-point of the two equalisers (visualise it as the midpoint of a see-saw) are slightly different. If you send the same sound into both inputs you can thus get all kinds of new sound shapes, if you boost bass on one and cut on the other. Notches, bands, extreme bass, etc.

The oscillator frequency goes up as you turn the know clockwise.

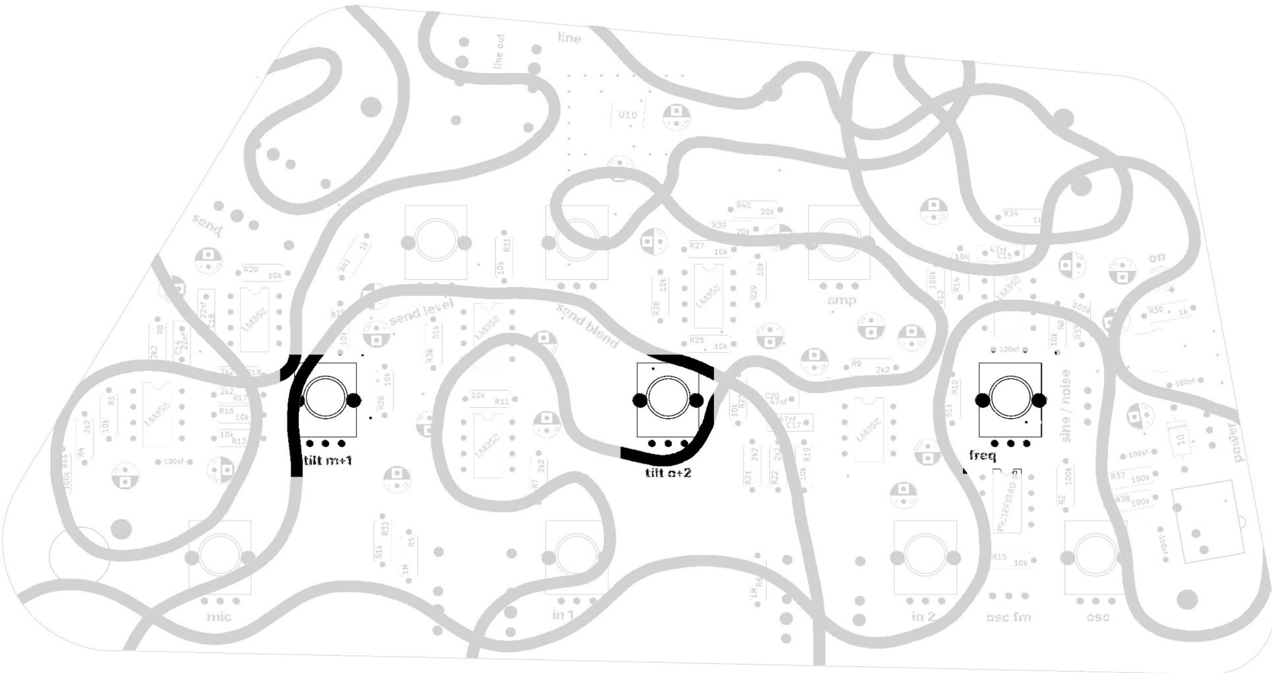

Figure 4: frequency row

The third row is the output row. Here we find the controls for the amplifier and the send output (also the normalised send return).

From the left, we have first the send level. It controls the level of the signal going to the send jack, but, note, NOT the level in the normalised feedback return. The send level goes from zero (no signal) to lots of gain. There is plenty of gain here.

Next comes the send blend. It selects which inputs go to the send output, turned counter-clockwise the two leftmost sources (microphone and input 1) are chosen, turned clockwise, the two right (input 2 and oscillator). In the middle, you get an equal mix of both. There is some “leakage” in this passive blend, so you never get a complete cut-out of the un-selected signal.

The send blend is before the normalised feedback return, and thus does (greatly) affect how the feedback sounds and behaves.

Next, the control labelled amp is used to control the level going to the power amplifier, from zero to lots. It only controls the level of signal going to the speaker connectors and has no other effect on send, feedback, or line out.

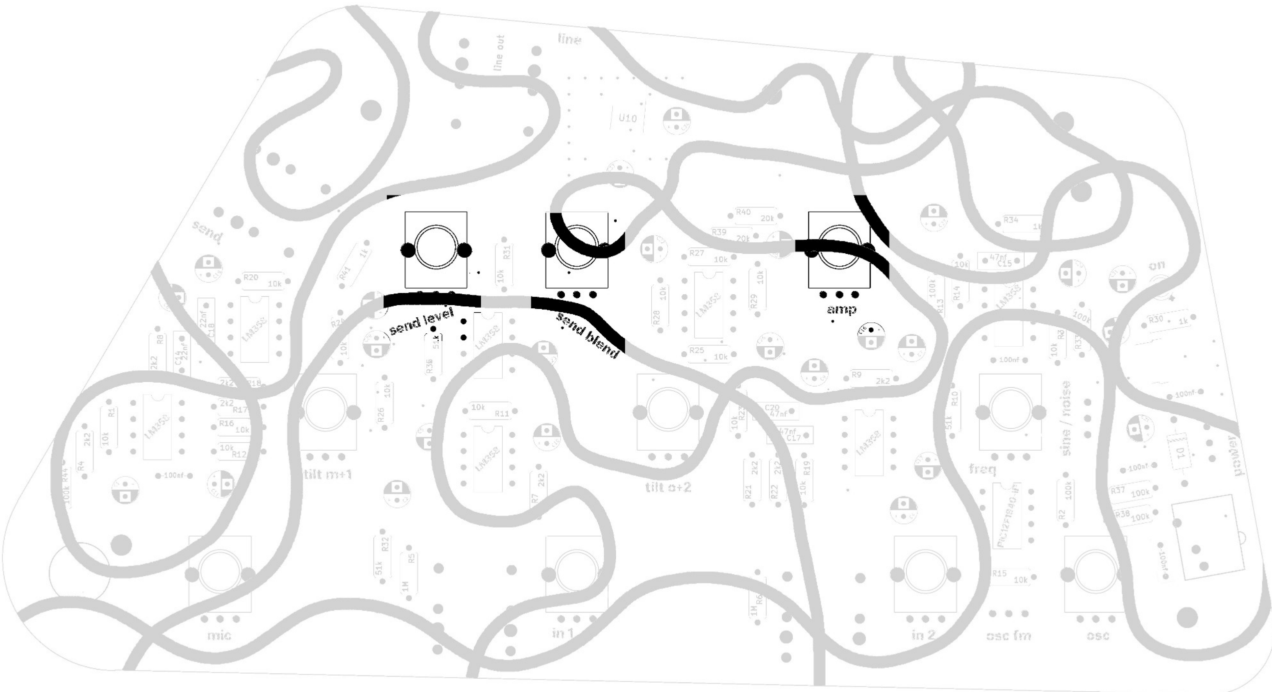

Figure 5: output row

Controls—Switches

There are four switches on the System Sound. To the right, near the power connector, is a switch labelled Power. It turns the device on. Up is on, down is off.

Between the in 2 and osc potentiometers is a horizontal switch labelled osc fm. Set to the left, it does “nothing”, and input 2 is sent to tilt eq 2, as described above. If you set it to its right position, input 2 is instead sent to the oscillators control input.

This allows you to make frequency modulated sounds, where any sound presented to input 2 will control the frequency of the oscillator. Use the in 2 knob to control how much signal is sent, as you would the gain knob in normal use.

This works equally with the normalised feedback, so any signal being fed back (selected with send blend) can control the oscillator—including the oscillator itself.

When oscillator fm is on, the signal from input 2 does not arrive at tilt eq 2, and is thus not available for the send, line out or speaker output.

If you aren't using the oscillator (the level is turned down), you can use osc fm as a mute switch for input 2.

Note that the mixing of signal and control voltage from the freq knob is pretty simple, and at the extremes (maximum or minimum frequency), the fm signal from input 2 is basically removed entirely. In the middle range, however, it works fine.

Also, note that you can only use sound here, as the input is ac coupled. See mods below if you really want to send in sequences or LFOs (but then you should probably just use an external oscillator.

The third switch is labelled sine/noise, and chooses if the oscillator creates a digital sine wave or a digital random signal (noise). The freq knob works the same way for both, but for the digital noise it changes the pitch of the noise. Combined with the tilt eq and distortion, many flavours of noise can be gotten. Mixing in small amounts of feedback gives even more options.

The oscillator fm works the same way for both sine wave and noise.

Lastly, the "out cut" switch, in the middle of the panel, cuts the left pair of channels (mic and channel 1) from the outputs, i.e. the line out and speaker out. The sound from the channels can still be blended into the send out.

Connections

At the bottom there are two inputs, the left is input 1, the right is input 2. I think you understand what they do.

On the left side is the send output. It is described above. But do note that this signal, apart from being affected by the send level knob, is also out of phase with the inputs (and with the normalised feedback path). This is intentional, so you can get different kinds of feedback.

On the top left is the line output, which (as described above) is the mix of all inputs, or more correctly the mix of tilt eq 1 and tilt eq 2. Basically if a level control in the bottom row is turned up, that signal is sent to the output jack, unaffected by other level controls (but affected by the eq).

The line output is also out of phase with the inputs. My intention was to use it as a send to ta PA, for instance, but you can of course use it for feedback, via other devices or straight back into the System Sound.

Next are the funky Fahnclip or Fahnestock clip connectors. They are pretty strong! Push the "wing" down and poke a speaker cable in. The polarity doesn't

matter as far as I've noticed. But if you do care, the positive output (this output is in phase with the inputs), where you'd plug the red cable if you cared, is on the right.

And, on the right, is the power connector. It's not great, but due to cost limitations in board manufacturing it's what I went with. The plug doesn't go in so far, but it does seem solid enough, I've yet to have a cable disconnect by mistake in a performance or otherwise.

Power

The System Sound should be powered with a 9V power supply, centre negative. This connector is the standard used by Boss guitar pedals.
Actually, the voltage is then regulated to 5V (which is the operating voltage of the device), so a voltage down to ~7.V may work, and also upwards to around 12 should be fine. But higher voltages give more heat, and is not recommended, and lower voltages may cause power drops and strange behaviour.

The current requirements of the System Sound depend on what you are doing. If you aren't using the power amplifier, you can use a 9V battery and get good results. If you are using small speakers, or not pushing the amp to the limit with loud bass, you can equally use weaker power supplies or even batteries. If, however, you are plugged into large speakers, the current consumption does go up. The amp is the thing that determines power consumption, everything else is pretty low current. The chip is rated for around 1,2W (which I don't think it's actually creating in the System Sound circuit—but Texas Instruments are measuring clean (undistorted) amplification I think that measurement goes out of the window anyway.

Suffice to say that 1000mA and 9V should be a good power supply for most uses.

I've found that excessive bass sent to large speakers can cause the System Sound to begin switching off and on again rapidly, which is probably due to the amplifier drawing too much current, or possibly inductive kick-back from the speaker coil. It will sound on-and-off again and the LED will start blinking. If yours starts doing that, I'm sorry but you should probably turn the amp down a bit. You can always add a PA or a guitar amp for more low end. It doesn't seem to hurt the amp.

Mounting holes

There are four mounting holes on the System Sound. You can ignore them of course, but if you want to you can attach it to something using screws. The holes are for M3 screws, but will also work e.g. with 2.5mm wood screws. You should add a spacer behind the board if mounting to a surface, or it will flex and possibly become damaged!
The mounting holes are all connected to ground.

Modifications

DC coupled input 2
If you want to control the oscillator with an LFO or other low frequency signal via input 2, you can bridge the 47uF capacitor nearest to input 2. Now that input is DC coupled.

More FM!!!
If you really want to FM the chip, you can solder a wire straight to pin 7 on the pic chip. Find a suitable resistor that you can tack onto, or solder to the back of the socket. I've added a small 3.5mm jack and super glued it to the board for this.

V1.0 to V2.0
If you want your 1.0 board to be (mostly) like a V2.0 board, look at www.noise.technology/pdf/systemsound-2-0-mods.pdf

If you have suggestions for mods, questions, or so on—just write. info@noise.technology is my email.

Thanks for reading!
Max Wainwright
May 2023
—updated March 2024
—updated again April 2025