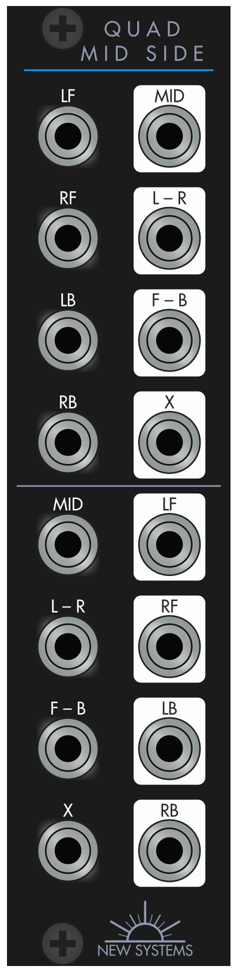
QUAD MID SIDE



Quadraphonic Mid–Side Processing

Manual Revision 1.0.2

NEW SYSTEMS INSTRUMENTS

SPECIFICATIONS

Size	6HP
Depth	22mm
Power Consumption	+12V 75mA −12V 75mA
Output Swing	−10V–+10V peak
Input Range	−10V–+10V peak
Input Impedance	94kΩ
Output Impedance	150Ω
Output Drive	600Ω (min), 2kΩ+ (ideal)

INSTALLATION

Before installing the module, make sure the power is off. Attach the power cable to the module and to the bus. Double check the alignment of the red stripe (or the brown wire for a multicolor cable) with the markings on the module and the bus. The red stripe should correspond with −12V, as is standard in Eurorack. Check the documentation of your bus and power solution if you are unsure. Screw the module to the rails of the case using the provided screws. (M2.5 and M3 size screws are provided.)

New Systems Instruments modules all have keyed headers and properly wired cables. But please remember to double check the other side of the cable for proper installation with the bus. Additionally, if using a different power cable, note that not every company wires modular power cables such that the red stripe will align properly with a keyed header. While our modules are reverse polarity protected as much as is practical, it is still possible that you could damage the module, your power supply, or another module by installing the power cable improperly.

Lastly, please fully screw down the module before powering on your case. The electronics are potentially sensitive to shorts, and if the module is not properly attached to a case, there is a risk of contact with conductive or flammable matter.

BASIS

Mid–side processing characterizes a set of sound sources by what is common in each of them, and by the ways they differ from one another. With a quadraphonic sound pallete with the sources RF, LF, RB, and LB, the middle and sides are:

$$M = (RF + LF + RB + LB)/2$$

$$LR = (LF + LB)/2 - (RF + RB)/2$$

$$FB = (LF + RF)/2 - (LB + RB)/2$$

$$X = (LF + RB)/2 - (RF + LB)/2$$

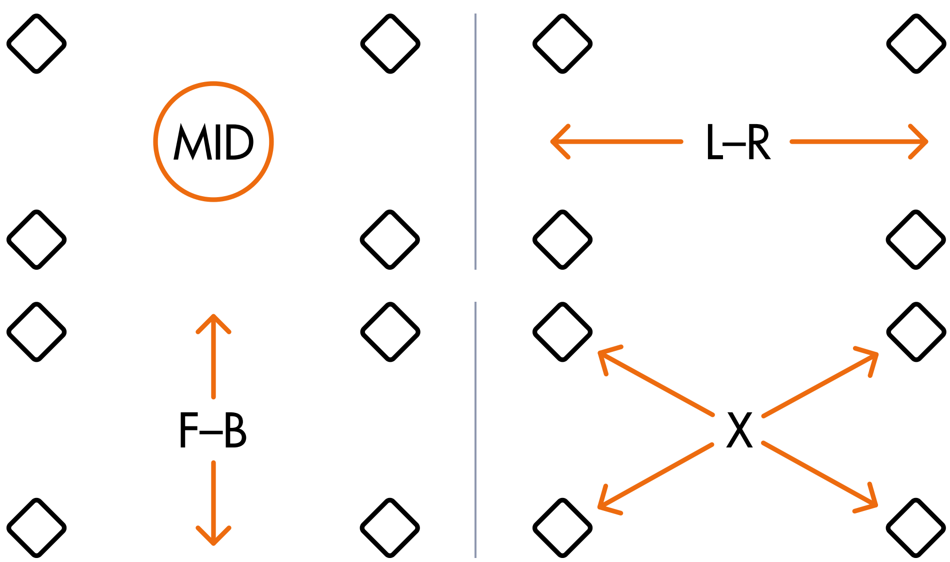
Using these values, one can recover the final sound sources with the following equations:

$$LF = (M + LR + FB + X)/2$$

$$RF = (M - LR + FB - X)/2$$

$$LB = (M + LR - FB - X)/2$$

$$RB = (M - LR - FB + X)/2$$



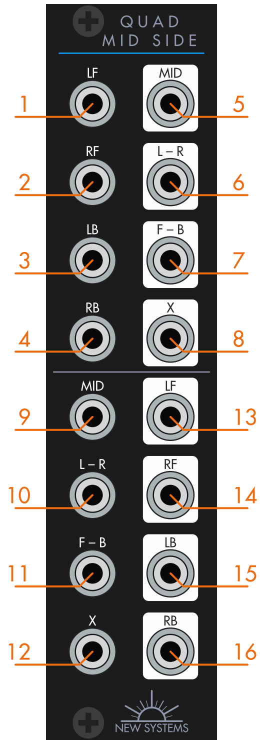
The correlation and difference signals between four sound sources.

EXPLANATION

Any set of sound sources have some amount of correlation and difference between them. Instead of focusing on the sound sources individually, mid–side processing focuses on this correlation and difference. The mid signal is the correlation between all the signals. In addition, Quad Mid Side gives you three characterizations of the difference: LR, the difference between the left pair and right pair of sound sources; FB, the difference between the front pair and back pair of sound sources; and X, the difference between the diagonal pairs of sound sources. This preserves the same information encoded in the four sources without loss or redundancy.

By processing these signals, or by directly injecting signals into the middle and side signals, the correlation and difference of all four sound sources is affected. This provides a powerful tool for altering and creating the spatiality of a signal.

INTERFACE



1. Left front input, for transformation to mid–side. Normalled to 0V.

2. Right front input, for transformation to mid–side. Normalled to 0V.

3. Left back input, for transformation to mid–side. Normalled to 0V.

4. Right back input, for transformation to mid–side. Normalled to 0V.

5. Middle signal output.

6. Left–right side signal output.

7. Front–back side signal output.

8. Diagonal or cross side signal output.

9. Middle signal input, for transformation to individual source signals. Normalled to MID output.

10. Left–right side signal input, for transformation to individual source signals. Normalled to L–R output.

11. Front–back side signal input, for transformation to individual source signals. Normalled to F–B output.

12. Diagonal or cross side signal input, for transformation to individual source signals. Normalled to X output.

13. Left front output.

14. Right front output.

15. Left back output.

16. Right back output.

USING QUAD MID SIDE

Route the four lower outputs of Quad Mid Side to four sound sources positioned in a space. Although some types of quadraphonic effects will work better if these are four identical speakers in a well treated room, for plenty of other types of effects this is not necessary at all. Use whatever you have.

If you are creating a quadraphonic signal from scratch, simply route signals to the MID input, where you want all sound sources to correlate, or to any of the side signal inputs where you want to create a difference between multiple channels. By adjusting the amplitude of any side signal, you adjust the magnitude of the difference you are creating.

If you already have a quadraphonic signal that you want to process, route the four signal sources to the four upper inputs. Because the upper outputs are normalled to the lower inputs, you should immediately hear sound. To alter either the correlation or any of the difference components of the signal, route the upper output to your processing chain, and route the processed output back to the corresponding lower input (or to a different input, if desired).

Mid–side processing responds well to any effect, but particularly well to phase-related signals, such as the outputs of phasers, reverb, chorus, delays, wavefolders, the three forms of modulation in Babel, and the quadrature outputs of the Harmonic Shift Oscillator.

Quad Mid Side can create complex, rich sound textures in whatever room you're in. In addition to exploring this spatial creation with the modular, explore the sonic space that results. Move around the room and listen to how the sound changes depending on where you are.

WORKING WITH STEREO SIGNALS

Babel can work with stereo signals in two ways: (1) it can be used as a tool to upmix from stereo to quad, and (2) it can be used as a stereo mid–side processor.

To upmix stereo, first pick either the left–right, front–back, or diagonal axis and route the stereo signal into the upper four inputs. For example, route the existing left signal into LF and LB, and the right signal into RF and RB. This will result in a quadraphonic signal with no difference along the other axes. Continuing the example, if routing a stereo signal to the left–right axis, there will be no signal coming from the F-B or X outputs. What remains is to create that difference, either from another signal or by processing MID or the difference signal along the chosen axis, and to apply it via the remaining side inputs.

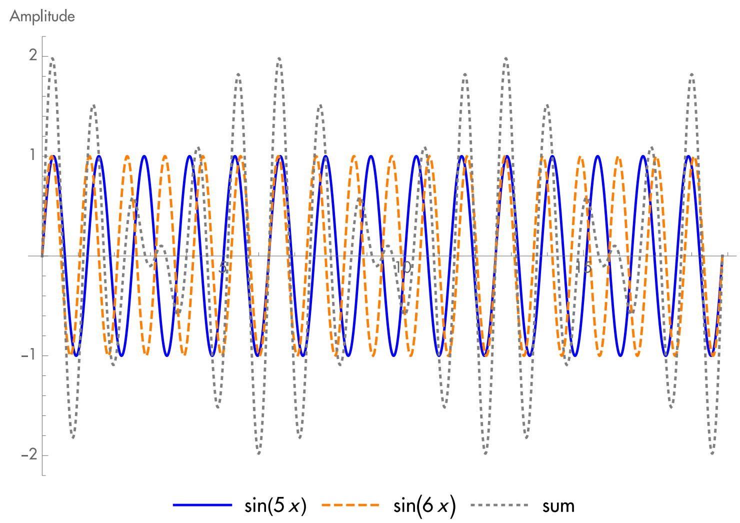
When using Quad Mid Side with stereo input and output, there are two options. Plugging the left and right signals into both the front and back inputs for each results in a MID signal, a L–R signal, and zero for the other side signals. The MID and L–R signals can be processed and returned as normal and the output taken from the LF and RF outputs. Alternately, plugging just the front inputs of left and right results in MID and F–B carrying the mid signal, L–R and X carrying the side signal.

With stereo output, the extra inputs of the Quad Mid Side act as a simple unity mixer. When taking the LF and RF outputs, the mid signal that creates them will be a mix of the signals at the MID and F–B inputs, and the side signal a mix of the signals at the L–R and X inputs. Be aware that if you plug nothing into these extra inputs, the normalled signals will mix when plugging just the LF and RF inputs.

ANTICORRELATION, PHASE, AND WAVELENGTH

Or, How to Make Sounds Spread Out in Space
Or, How to Upset Your Mastering Engineer

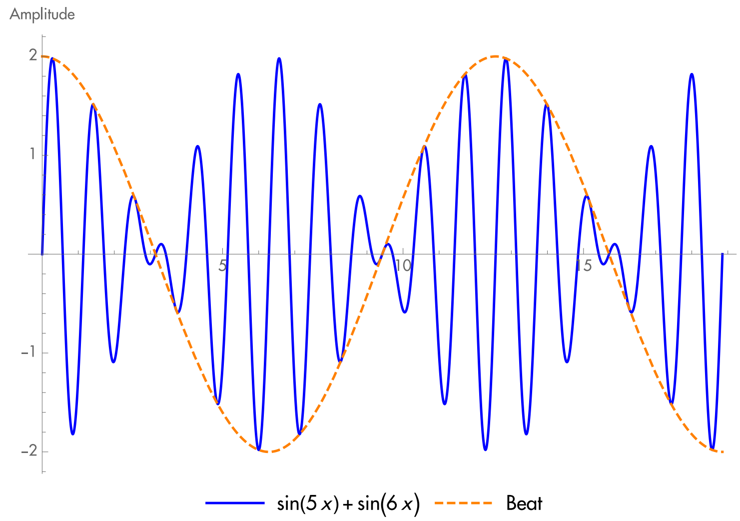
Although building complex soundscapes with Quad Mid Side is an intuitive process, it can be helpful to understand the way that sound waves interact with each other within a space.



Constructive and Destructive Interference

When two waveforms combine, they produce constructive and destructive interference. That is, when the crests of one wave line up with the crests of another wave, the overall waveform gets louder. Alternately, when the crests of one wave line up with the troughs of another wave, the overall waveform gets quieter. The distance between the crests of two different waves is the *relative phase* of those waves. When two waves of two different frequencies interact with each other, that relative phase changes at a constant rate, as the slower wave keeps falling further behind the faster one. This continual change in phase produces a cyclic change in amplitude, known as a *beat frequency*. This frequency will always be equal to half the difference between the frequencies of the two waves being considered. If we ignore the sign of this waveform, focusing just on the change in amplitude itself, the beat frequency is just the difference in frequencies. We can express it like this:

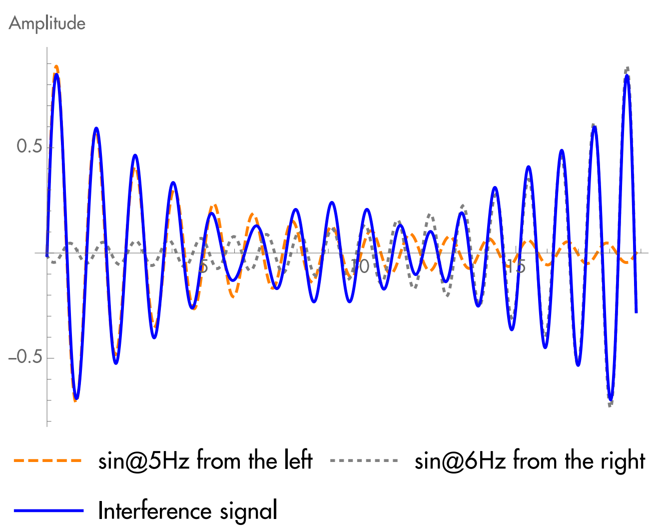
$$\sin(a) + \sin(b) = 2\cos((a-b)/2)\sin((a+b)/2)$$



Beat Frequencies

If sound moved instantaneously and without resistance directly to your ears, this interference would form from a simple combination of all the waveforms produced by all the sound sources, and the sound everywhere in the room would be the same. Two phenomena keep that from being the case. On the one hand, the volume of a sound decreases the further you get from its source, due to air resistance and directional dissipation. On the other hand, the sound takes time to get from one point to another.

It is easy to understand how changes in volume affect the interference pattern of a sound. Simply put, when one waveform is louder than another, not much interference occurs; it is only with similar amplitudes that waves fully interfere with each other. As both waves diminish when traveling from the sound sources, close to the sound sources there is very little interference pattern; it is only in the center, when the relative volumes approach each other, that an interference pattern occurs. By changing the relative volume of the waves that are interfering with each other, you can change the place where the maximum level of interference occurs.

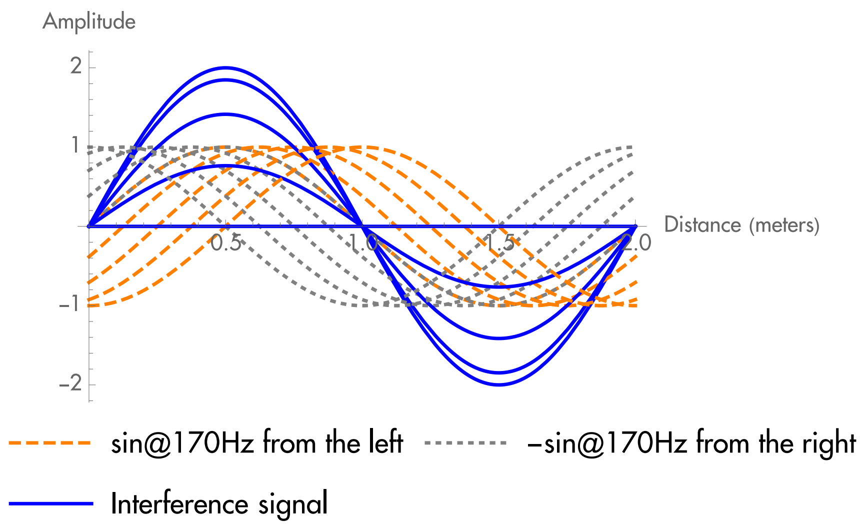


Interference Signal with Diminishing Amplitude

Because sound takes time to travel, a portion of each waveform, and therefore a portion of the interference pattern created by multiple waveforms, exists in the space between sound sources. To find out how much of a waveform exists in a given space, find the wavelength using the following formula:

$$\lambda = v/f$$

Where λ is wavelength, f is frequency, and v is 340 m/s, the speed of sound in air. 20Hz–20000Hz is 17m–17mm. We reach 1m at 340Hz, about F4.



The Way an Interference Signal in Space Develops over Time

As two sound waves interact through a space, a portion of their interference pattern is spread over the space, such that some places within that space are quieter (for a given pair of frequencies) and others louder. How much of that interference pattern is in space and how much in time depends on the relationship between the wavelength of each wave and the amount of space in which they are interfering. Two interfering waves with an average wavelength of λ will produce one quieter place and one louder place for each $\lambda/2$ which fits within a given space. Further, these quieter and louder places will move within the space at the rate of 1/2 the difference frequency of the two waveforms, towards the source with the lower frequency. That is, the beat frequency is not just a temporal phenomena, but determines the rate at which nodes (quieter places) move around the room.

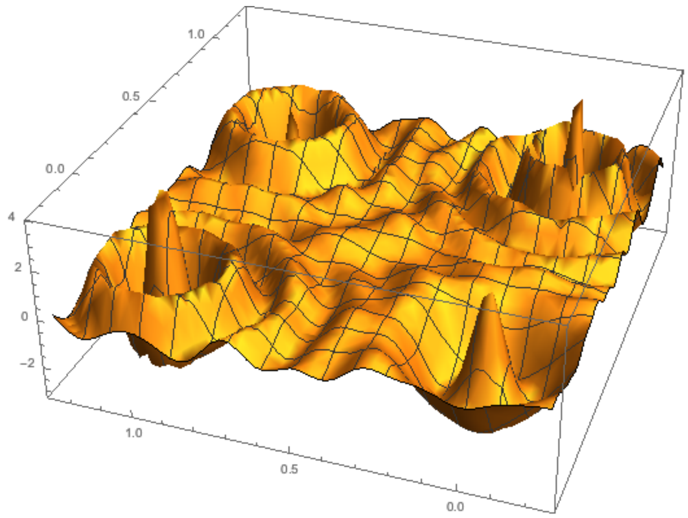
An easy way to calculate how many nodes a given frequency interference will produce is to begin with the frequency whose wavelength matches the distance between sound sources. The number of spatial nodes is then the highest multiple of this frequency that is less than the average frequency in question. This frequency is obtained by:

$$f = v/\lambda$$

For example, for a two meter distance, the frequency matching the wavelength between two sound sources is 170Hz. Any frequency interference with frequencies whose average frequency is above one half of that, 85Hz, will have at least one node in the space between. 170Hz and above, two nodes; 255Hz and above, 3; etc. While higher frequencies produce more nodes, those nodes are correspondingly smaller, and higher frequencies tend to have more various frequency content. For this reason, although high frequency content *can* create some rich textures, nodes are often a lower mid frequency phenomenon. The acoustics between the source and hearing also play a significant role. For very small nodes, they may disappear entirely within the acoustics of your ear.

While the number of nodes will vary with speaker placement, the propagation rate of those nodes through space only depends on the relative frequency of the two waves that are interfering. Specifically, when the difference in frequency is *zero*, those nodes become purely spatial rather than temporal.

This is the exact phenomenon that the Quad Mid Side creates. The difference signals are a measure of the anticorrelation between different sets of sound sources. By injecting a signal into these sides, an interference pattern based on that signal is distributed through space.



With four, rather than two, sound sources, and with a room whose reflections make up a significant portion of the sound, the situation is more complex. These abstractions are nevertheless useful for thinking through how to sculpt a sonic environment that is lush, various, and explorable.