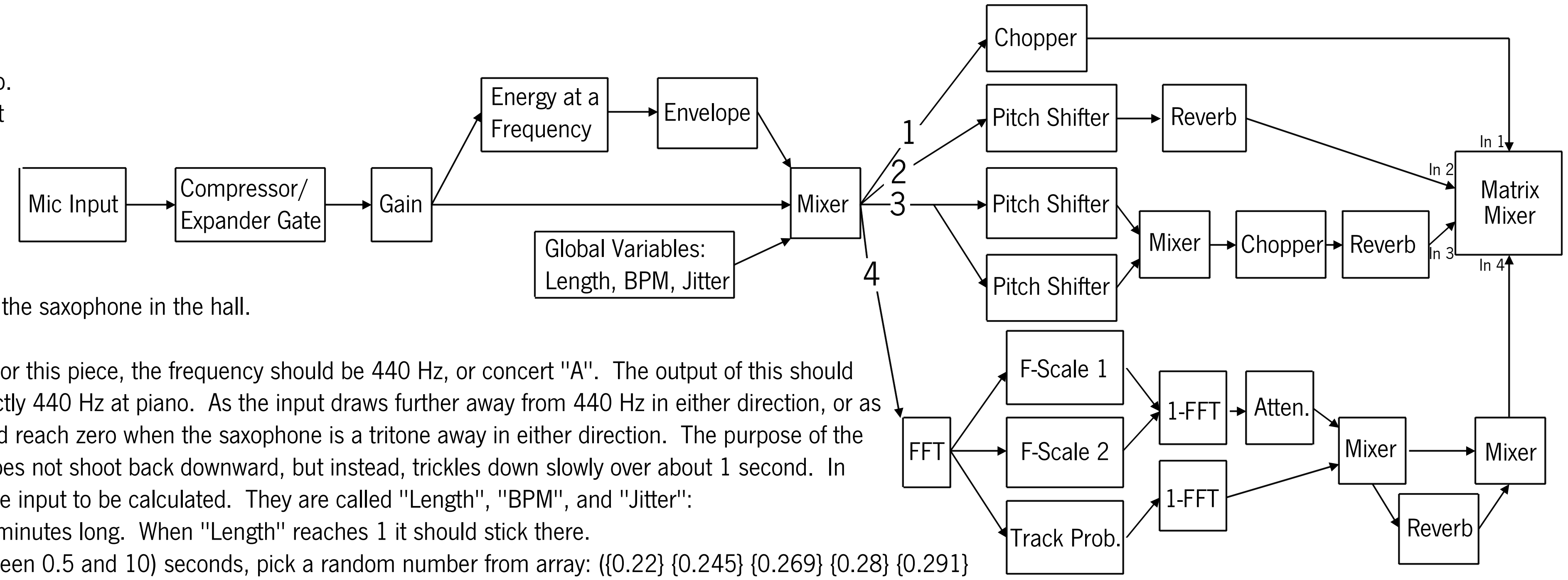
Construction of the Electronic Components:



The *Compressor/Expander Gate* should attenuate inputs lower than an extremely low threshold to zero. This serves as a method of making sure sounds like coughing and such before the piece begins do not get processed and sent out the speakers. It should also attenuate very high inputs to a managable intensity in order to avoid any feedback in the hall.

The *Gain* is to be used as a scaling factor, as described in note 6 to the left. Here, the inputs should be scaled upward so that a piano input, which is the loudest the saxophone will play, is perceived as a much louder sound. The overall goal is for the outputs of the speakers to envelop the actual sound of the saxophone in the hall.

Energy at a Frequency is an tool that measures the intensity of an input at any particular frequency. For this piece, the frequency should be 440 Hz, or concert "A". The output of this should be a variable called "Energy" that is somewhere between 0 and 1 (1 being the saxophone playing exactly 440 Hz at piano. As the input draws further away from 440 Hz in either direction, or as the input's intensity diminishes, the "Energy" variable's value should lessen towards zero, and it should reach zero when the saxophone is a tritone away in either direction. The purpose of the *Envelope* is to provide a decay to the value of "Energy" so that if the value spikes upward, the value does not shoot back downward, but instead, trickles down slowly over about 1 second. In addition to this global variable, there are three others that must be implemented, but do not require the input to be calculated. They are called "Length", "BPM", and "Jitter":

"Length" is a variable that should move slowly from 0 to 1 over the course of the piece, which is 8 minutes long. When "Length" reaches 1 it should stick there.

"BPM" is a variable that can be described in the following expression: (Every (random number between 0.5 and 10) seconds, pick a random number from array: ({0.22} {0.245} {0.269} {0.28} {0.291} {0.45} {0.66} {0.74} {0.77} {0.79} {0.81} {0.829} {0.845} {0.871}) x ("Length"^3)) x 0.66 (smooth changes)

"Jitter" is another variable that is applied as a scaling factor to many of the fields to come. Henceforth, when something is said to be "jittered", this expression should be applied to the object, variable, or expression that is to be jittered: (Every (random number between 0.1 and 5) seconds, pick a random number from array: ({0.7} {0.8} {0.9} {1} {1.1} {1.2} {1.3}) (smooth changes)

At this point, the patch should split off into four separate processes, which are mixed and channeled at the final *Matrix Mixer*. Process 1 is simply the input sent through a *Chopper,* which should have variables for "length of chopped sections" and "space between length of chopped sections", where the chopped sections are left untouched save for quick fades in and out, and the spaces are sections of the sound attenuated to 0. For this chopper in process 1, the chopper's chopped section length should be ("Length"^10 minutes) with smoothed changes, so the chops get longer as "Length" increases over the course of the piece. The space between the chopped sections should be ("BPM" x 2) milliseconds, jittered.

Process 2 involves a *Pitch Shifter* and *Reverb.* The *Pitch Shifter* simply changes the pitch of a particular interval in real time, without changing the speed. This *Pitch Shifter* should always shift upwards in semitones, and its function can be expressed thusly: (Shift the input upwards by (Every (random number between 0.5 and 5) seconds, pick a random number between 0.7 and 1.3) semitones, jittered. The *Reverb* should have a medium-high decay length but very little reverb "wetness"; do not use a Low Pass Filter. These figures are for all *Reverbs* found in this piece.

Process 3 is two *Pitch Shifters*, a *Chopper,* and *Reverb.* The *Pitch Shifters* work in a slightly different way than that which is depicted in Process 2. Here, the formulas for the two are: ((1-"Energy") x 13) semitones up, jittered; ((1-"Energy") x 13) semitones down, jittered. This means that "Energy", which again, is between 0 and 1, is scaled by 13 and converted as semitones, so when the saxophone is playing exactly "A" at *piano,* the "Energy" field will be 1,meaning (1-"Energy") x 13 = 0, or no change, and the farther away from "A" or the quieter the input gets, the larger the shift. It is also important to note that the shift occurs in both directions. After this, the two shifted signals should be mixed and fed into a *Chopper.* This time, the *Chopper's* "length of chopped sections" should be ("BPM" x 45) milliseconds with smoothed changes, and the "space between chopped sections" should be the same, but jittered. The *Reverb* should be the same as in Process 2.

Process 4 involves live spectral analysis, or the Fast Fourier Transform algorithm, here abbreviated as *FFT.* The input should be sent through the *FFT,* or analyzed, then sent in three different directions. It is sent to two *Frequency Scalers,* which are effectively the same as the *Pitch Shifters,* only they are processing Fourier-Transformed data that must be resynthesized. The algorithms for the first *Frequency Scaler* is: (Every (random number between 0.5 and 5) seconds, multiply the frequency by a random number between 3.7 and 4.4). The second algorithm for the other *Frequency Scaler* is the same, except that the random number should be between 3.9 and 4.7. From here, this data should go through a reverse Fourier process, or resynthesis, here called *1-FFT*, and sent to an *Attenuator* to be scaled down considerably. These sounds, which will be quite pitched, should sound only like echoing harmonics above the rest of the patch. They should not be in the foreground. The other half of this process sends the original Fourier-Transformed data not to a *Frequency Scaler*, but instead an object called *Track Probability.* In FFT, a sound is reduced to many simultaneous tracks; the *Track Probability* should pick randomly which tracks should be allowed and which should not according to the value of "Probability", between 0 and 1, where 0 is no sound allowed, and 1 is the entire analyzed sound allowed. The value "Probability" should be : (Every (random number between 0.3 and 1) seconds, pick a random number between 0.05 and 0.33, and multiply that by "Length"), smoothing changes. Therefore, if the chosen value is 0.1 and "Length" is currently at 0.4, the value for "Probability" will be 0.04, meaning that 4% of the tracks will be selected, randomly. The sound should be unpitched, quite percussive, and bubbly. This process should also go through a resynthesis (*1-FFT*) and be mixed with the other two signals, the output of which should be sent to a *Reverb* and be mixed with itself without reverb.

These four processes are then to be sent to a *Matrix Mixer* with four inputs and four outputs, the four inputs being Processes 1-4 and the four outputs being 4 loudspeakers, distributed from left to right on the front of the stage. The amplitude of Process 1 should be ("Length"^8), meaning that as "Length" approaches 1, this process will get louder. Process 2 should fade in from 0-1 when "Length" is moving upwards between 0.2 and 0.8, and as "Length" moves from 0.8 to 1, the process should fade out, so that it is completely gone by the time that "Length" is 1. Process 3 should fade in from 0 to 1 when "Length" is moving upwards between 0.1 and 0.65. At the point where "Length" is 0.65, the amplitude of Process 3 should be 1, and it should lessen and lessen as "Length" continues upwards. By the time that "Length" is 0.85, the amplitude of Process 3 should be 0. The amplitude of Process 4 should be (1-"Length")^2, meaning that it will lessen over the course of the piece. All these amplitude envelopes should be jittered slightly from speaker to speaker, so that none are exactly equal.


60''
7.5''
N.B.: Never play above the dynamic p until the very end of the piece.
poco pppp
pppp
poco ppp
ppp
poco pp
pp
poco p
p
sfffz