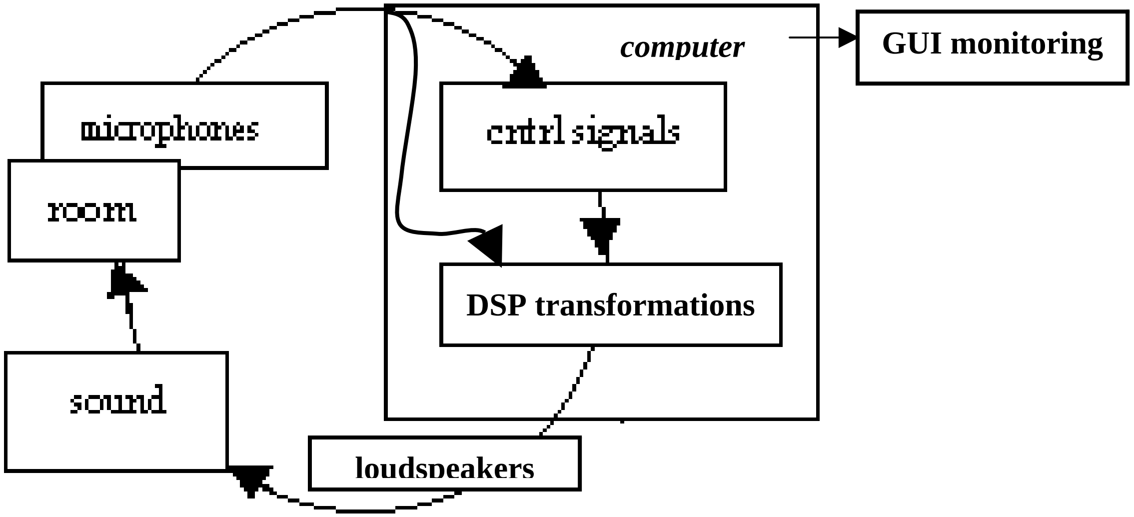
basic ecosystemic electroacoustic infrastructure



feedback is structural

the sound-generating process and the space (room) are tightly coupled via electroacoustic transduction terminals (microphones and speakers)

- space (room acoustics) heavily affects the system dynamics
- the sound-generating process adapts to factual circumstances relative to the technical implementation (real-time) and room acoustics (real-space)

specially designed DSP program (signal patch) implementing

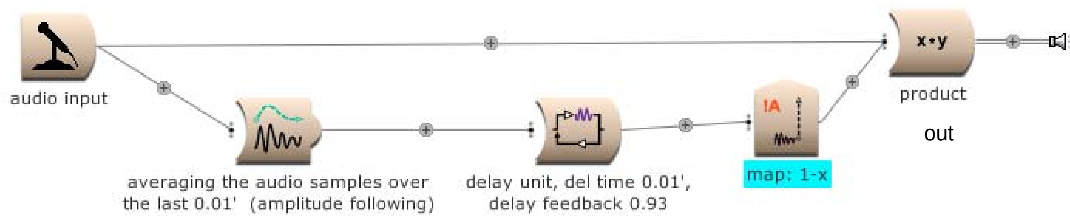
- control-signal synthesis and processing
feature-extraction methods (analysis of incoming and outgoing sounds)
processing of feature-extraction data, turning them into controls
over variables in the audio processing transformation(s)

- audio-signal processing transformations
(driven by the above control-signals)

example: **automated self-dampening ("self-gating") circuit**

sound source → feature-extaction → processing → audio signal operation

sound source:	any
feature-extraction:	amplitude following (averaging over a time-window)
control-signal processing:	delay unit, mapping function
sonic transformation:	downscaling (amplitude modulation)



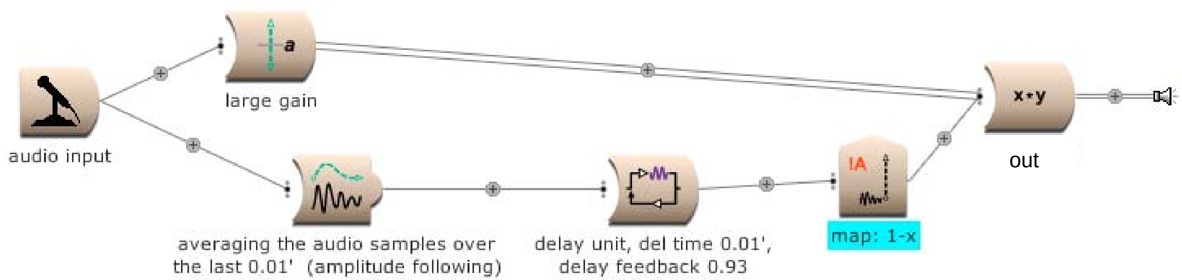
amplitude following is implemented by averaging the absolute values of individual input samples, over signal segments of given duration ("time-window", 0.01' in this example). The longer the time-window duration, the smoother the output and the less quickly it will respond to transients in the input. Can also be done by taking the RMS (requires more computation). An other implementation would be a FIR (finite-impulse-response) low-pass filter of higher order (indeed a 1st order low pass would be the averaging function out(t) = [out(t)+out(t-1)]/2). In which case the cutoff freq should be extremely low, say 1 Hz (the cutoff freq equals the expected rate of change in the input envelope).

delay unit with feedback is a typical digital signal processing filter equation: out(t) = in(t) + A*out(t-n). That is, a IIR (infinite-impulse-response) filter design, where t-n is the delay time (a small memory buffer is needed), and A is a scaling factor for the signal recirculating in the feedback loop. In the present context, rhe use of a short time delay unit with a rather large feedback factor is that it creates a kind of *hysteresis* and *latency* in the signal, allowing our control signals to move in a way closer to mechanical and analog devices. Upon special circumstances (feedback close to 1, very short delay times), this acts as a kind of *accumulator*, which can also be of use in the context of sub-audio signals (such as control signals are).

map is here a simplest function that turns the input range upside-down: it maps the numeric interval in the delayed amp-follower signal [0,1] into the reversed range [1,0].

example: **self-dampening ("self-gating") audio feedback circuit**

sound source:	audio feedback tones (Larsen effect), due to high gain in the microphone-loudspeaker loop
feature-extraction:	amplitude following
control-signal processing:	delay unit, mapping function
sonic transformation:	scaling (amplitude modulation)



same concept, but signal gets largely amplified, in order to cause the feedback loop to ring (accumulation of background noise, until a peak resonance takes over). Sine tones or clusters of sine tones are created, they are sustained by the circuit but preventing saturation. (The circuit has a double systemic function: **generation** and **self-regulation**).

Attributes of sound (freq, amp, onset time) depend on **mic-loudspeaker distance**, on **room acoustics**, as well as on tech specifics (frequency- and impulse-response) of involved **equipment**.

The circuit may manifest a variety of dynamical behaviours, depending both on external conditions (room acoustics, equipment) and total amp gain (“total” as it depends, too, on the room resonance properties). The behaviour will also depend on particulars in the circuit itself, such as the latency induced by delay unit. Typical behaviours include:

- audio oscillator at constant amplitude (typically with a rather slow onset),
- audio oscillator with amplitude increasing and decreasing at some rate – in which case, the system exhibits a second-order development, that could be either a stable one (equally-spaced amp peaks, i.e. a built-in “rhythm”) or a more random one, with non-patterned amp peaks. The latter circumstance often occur as a consequence of a quicker onset in the audio feedback tones. That shows that the overall system dynamics considered here (mic, speakers, self-gating circuit) is heavily affected by the **initial conditions** at which it is put to work.

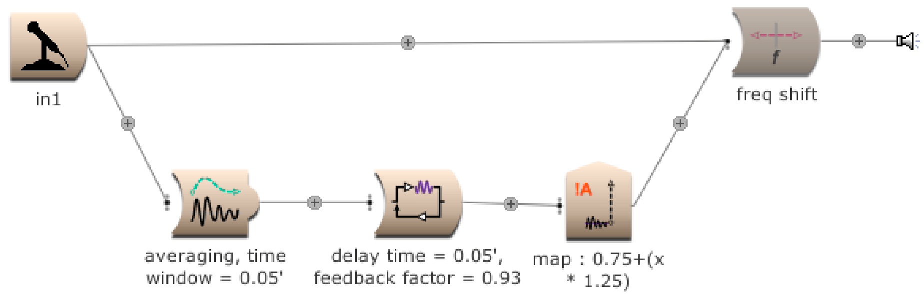
Other system behaviours may appear, that include frequency changes in feedback tones.

Beside “self-gating”, other ways can be devised of implementing **self-dampening audio feedback**.

- topics for work and assignments

    feedback control by means of phase cancellation
        with frequency shifters (inducing beatings)
        with dynamical delay units
    feedback control by means of ring-modulation
    feedback control by heavy dynamic processing

Both the above examples involve **amp-to-amp** mapping (output amp is a function of input amp). Other control-signal generation designs may involve **amp-to-freq**, **freq-to-amp**, **amp-to-spectral** properties, **spectral-to-amp**, **amp-to-density** and **density-to-amp** (density = amount of overlapping sonic events in time) example : **input signal amp drives the frequency shift of input signal itself**

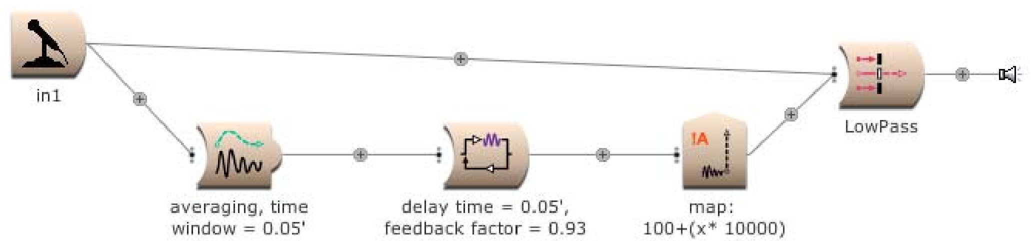


The frequency shiter (rightmost box in the graph) expects the ratio by which it should shift the signal frequency up or down. The tracked down amplitude drives that ratio. The **amp-to-freq** map in this example is : [0,1] => [0.75, 2]. That is, when the input signal is silent, frequency is shifted down by a ratio of 0.75 (musical interval of a lower 4th); when input signal is maximum loudness, frequency is shifted up by a ratio of 2 (an octave higher).

This particular way of mapping (louder = higher) is just an example. However, on a perceptual plan, it makes sense to create a connection between amplitude and frequency, in order to support the perception of a correlation of loudness and pitch (two psychoacoustical dimensions hardly independent of one another). In any case, in all mechanical devices frequency and amplitude can hardly be totally separated (i.e. human beings can hardly sing a very high pitch *ppp*). Mapping from one dimension to another may foster an ecologically relevant auditory perception.

physical phenomena are the outcome of exchange and interaction in a system of interconnected components (incl. the environment), and no single element works without affecting other elements in the system: the ear hears out the link that binds any one element to any other

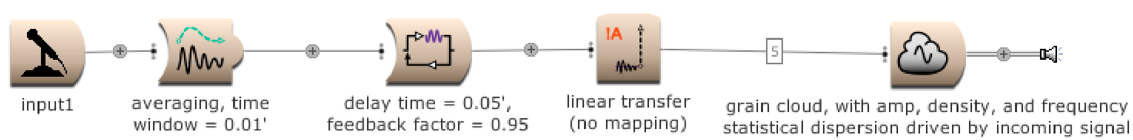
example : **input amp drives cutoff freq in a low-pass the input passes through**



The cutoff freq of the low-pass filter (rightmost box in the graph) is driven by the input signal amplitude. The **amp-to-freq** map is : [0,1] => [100, 1000]. That is, when the input signal amplitude is small, the spectrum width will be reduced; when the input is loud, instead, the spectrum width will be larger.

This example again points to an ecologically relevant link: in a mechanical device, stronger efforts (stronger energies put into the device) results into a larger frequency spectrum. A trumpet cannot play louder without also sounding brighter. Loudness and brightness are psychoacoustical correlates.

example : **input signal amp drives the amount of sonic droplets (density of sound grains) in a granular-texture generator (cloud generator)**



The max grain density, in the sound cloud generator (rightmost box), is 20 overlapping grain events at any given time. The grain amplitude, and the actual amount of overlapping grain (plus other variables in the grain cloud), are driven by the input signal amplitude. The amp-to-density map is a linear transfer: [0,1] => [0, 20]. This too points to an ecologically relevant link. If you smash a glass bottle on the floor, the more violent the impact, and the larger the amount of glass pieces that break apart, hitting one another in the process of bouncing on the floor : the micro-level structure of the sonic event is made of a higher amount of sonic droplets, a higher density of microevents in time.