Resonance Strings Reverb

Resonance, sometimes annoying, can also be very pleasant in certain circumstances. Try the following. Open the lid of a piano or grand piano and press the sustain pedal. All felt pads that normally damp the strings are now lifted from the strings. In principle, each string can vibrate freely. However, that free vibration does not occur automatically. The string must be excited. Apart from being struck, the string can also be set in vibration through a excitating sound. Speak something in the piano while sustaining the pedal. Beautiful reverberation isn't? With this simple trick, the piano has been transformed into a beautiful mechanical-acoustical reverberation box.

Nowadays we think of reverberation simulation in terms of DSP algorithms, virtual space simulations, or even more modern, 'sampling' real spaces by means of convolution. However, this simple example with the piano already indicates that it can be done also in al low tech way. The test with sustained pedal resulted in a beautiful uncoloured reverberation, because all the strings could participate.

You can also do the same test reverberation with just one string. Press one key lightly without striking the string and hold that key. In the sympathetic sound the frequencies corresponding to the natural overtone frequencies of the string are excitated. This kind of reverberation of such has what you could call a tonal character.

Tonal reverberation
In Indian music culture we hear this principle of such resonance strings in instruments such as sitar, sarod, sarangi, dilruba and esraj.
These plucked and bowed instruments have, in addition to normal strings, resonance strings located under the playing board. Their number usually varies between 13 and 19. They are not tuned chromatically, but according to the scale of the piece to be played. In traditional Indian classical music there is no transposition from one key to the other. If the piece is in dorian it remains dorian. Such a dorian reverberation emphasizes the modal mood of the piece of music.

`5ResStrings.pch2`
Open the G2 demo patch above. You now have five resonance strings that spontaneously go along with the input signal. This input is formed by a random LFO that sends a noise generator in both pitch and bandwidth.

In addition, the output of the same LFO is fed back to the Rate input. This results in random noise tones whose lows last longer than the high ones. Frequencies in this noise input that correspond to one of the natural frequencies of the four virtual strings will actually sound them.

Listen to the eight different presets under 'variation'. You now hear various tonal colored reverb variants. The five strings have been implemented by means of five StringOsc.

This oscillator is an already ready-made (simple) physical model of a string, the switch-off time can be set with the Decay in combination with the Damp parameter. With the Semitone button, the strings can be tuned to each desired harmonic relationship.

How it works
Yes, you hear that work, but how? Well, the heart of this StringOsc is formed by a sound delay line, a TimeDelay module. Such a delay can best be regarded as the joker from the sound processing and sound synthesis tools. For example, the vast majority of the effects from an arbitrary FX processor is based on one or more such time delays.

For this episode, we limit ourselves to one of the many applications: the TimeDelay as harmonic resonator. Let's start at the beginning. The first echoes and delays were made with a tape recorder. If you feed the sound from the playback head back to the recording head via a mixer, repetitions occur, much or little, depending on how much signal is returned to the recording head.

The speed of the delay sequences depends on the tape speed and on the distance between the recording and reproducing head. The speed with which the delays could follow each other thus naturally had physical limits, determined by the dimensions of the tape recorder.

Delays in micro-time
In the seventies of the last century, the first electronic delay ICs came on the market, the so-called bucket bridge memories, or charge coupled devices (ccd) also named bucket bridge delays (bbd). These made it possible to realize much shorter delay times, even up to tenths of a millisecond.

When we return the output to the input via a mixer at these very short delay times, a completely new phenomenon is created for the observation. We no longer hear separate delays, but a kind of filter effect. This is called a comb filter.
Suppose that the dealy time is set to 1 millisecond, and a frequency of 1000 Hz is present at the input. This means that one period of this signal lasts exactly one millisecond. The returned signal of the output now runs exactly synchronously with the input. In other words, the two signals, input and output are in the same phase. This means that a gain occurs, the amplitudes are added, the amplitude is therefore doubled.

But, this story applies not only to a frequency of 1000 Hz, but also to all integer multiples of 1000 Hz. For all these frequencies the input signal is in phase with the output signal. Thus a large number of reinforcements occur in the end result of which the frequencies relate as the sequence of natural numbers: 1, 2, 3, ..., also called the harmonic series.

In this concrete example, these reinforcements - the teeth of the comb - are at 1000 Hz. If we now take noise as an input signal, plus the fact of a very large feedback of about 90% or more, for example, we will hear a decaying sound with a 'sawtooth-like' character with a momentary noise pulse at the entrance. If we add a low pass filter in the feedback loop - the feedback loop - the feedback value for frequencies higher than the set low pass frequency will decrease. This expresses itself in the decaying sound, but now the higher frequencies, the overtones, are decaying earlier than the lowest, the fundamental tone.

Open `BasicResonator.pch2` and you have implemented the situation just described in your G2 demo. Listen to the variations again and experiment with the delay time and feedback value. In this patch it is also possible to invert the feedback signal, to reverse it, or to put it another way, to multiply by -1. Because of this signal reversal the amplifiers form an odd harmonic sequence: 1, 3, 5, ..., a 'square wave character'. The result also sounds an octave lower.
With the two different possibilities of positive and negative feedback we now have an abstraction of a string and air column. Positive feedback simulates a model generating all harmonics. Negative feedback results in a model that only generates odd harmonics. More theory regarding time delays and comb filters can be found in BasicTD formulas.pch2 and BasicResOscString.pch2.

Attention! You can’t load `SympStringsRevFX.pch2` in the G2 demo software synth. It is a patch for 'the real thing'. A resonance string reverberation FX that works with external audio. With the normal G2, without extra dsp extension, this patch simulates seven resonance strings. You simpely choose these resonance string by hitting de desired keys on your keyboard.

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internet

Ustaad Ranbir Singh, Dilruba, Raag Tilang
https://www.youtube.com/watch?v=sDRRrSYieU8