JAX ESSENTIAL SERIES

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The JAX ESSENTIAL SERIES will cover audio units, that usually are needed all times with music production. But these releases will by no means be any common effects that are “mass produced” or just copied from the basics to deliver average qualities on iOS.

So our first releases in the series are several kinds of pitch and frequency shifters, a highly specialized area of audio processing, that obviously has not been saturated yet, because such effects with any good quality are usually very difficult to achieve.

The pitch and frequency shifters for instance are strictly separated into single units because of their fundamental different DSP methods and also the sonic results, making clear how complex this area of advanced signal processing is. It very much depends on the kind of audio material, which effect processor to use for optimal results.

The JAX P2
Time Domain Pitch Shifter

A time domain pitch shifter will virtually modify the speed of audio playback. But there is a certain problem with this. A continuous audio stream (like realtime processing) is usually absolutely constant in its sample rate and buffer size. And also nobody can shift the time in the real world (yet!?).

This is very different when using just a sampler. A sampler will have access to a complete sound sample in memory and can play back at any sample rate, thus making it longer or shorter in time, raising or lowering the pitch this way.

A continuous stream is always fixed in the sample rate and the playback time cannot be modified without discontinuity.

Some tricks can be used, for modifying the playback speed virtually. We use a delay based approach, where 2 slightly time shifted overlapping delay buffers are filled with continuous data streams and then a windowing function is used to merge these two buffers together again without any crackles. The buffers are constant in length and a short variable latency will be introduced, that is pitch dependent.

This way, the speed of the reading pointer can be adjusted in realtime to a fractional part of the current sample rate. This all sounds quite simple, but in reality it is complex coding and also will produce some unavoidable side effects. Extreme values tend to become grainy.

The good news are, this takes polyphonic (in fact any) material and shifts its pitch without a problem. All transients and the formants are shifted equally with the pitch, which produces the typical “chipmonkeyfication”.

The time domain approach is better situated for usage with transient material, because it will not smear so much as the frequency domain approach of JAX ESSENTIAL SERIES P3.

The JAX P3
Frequency Domain Pitch Shifter

Our frequency domain approach for pitch shifting will use the “phase vocoder” method, that effectively is an FFT/inverse FFT. The term “vocoder” is somewhat misleading here, because it has nothing to do with a real “vocoder”.

FFT will introduce latency too, but the buffers can be kept relatively small and the input can be shifted by exactly this fixed amount of latency for the final mix internally.

Complex mathematical formulas will be used to extract the frequency components from the signal and their corresponding magnitudes. After the frequency is known, it can be modified in controlled manner, thus making pitch and frequency shifting possible in realtime.

If the frequency was modified across all magnitudes equally, the inverse FFT is applied, leaving the audio with a changed pitch. Like the time domain approach of JAX ESSENTIAL SERIES P2, this will produce “chipmonkeyfication” effects, because all transients and formants will be shifted this way too, proportional with the fundamental frequency.

The frequency domain approach has some side effects too. The audio will be smeared slightly but equally across the entire range. Grainy results are less audible than with the time domain approach. It is not well situated for using with strong transient audio material. Excellent results can be achieved with ambient audio material.

The JAX P5
Solo Voice Pitch & Formant Shifter

(release scheduled)

The P5 algorithm is very different to the P2 and P3 algorithms in the series, because it will not work with polyphonic material, but has great results with monophonic solo voices, even if these have a complex formant and transient structure.

(Drums are usually atonal and polyphonic and therefore will not be usable with this kind of pitch shifter.)

This approach is using a pitch synchronous delay buffering scheme and a pitch detection method for adjusting the delay time of the buffers in realtime. In some extent it is very similar to the time domain pitch shifter P2, but the difference here is the variable buffer size of the delay. The variable buffer playback speed hereby will effectively alter the formants of a voice.

Normally, with P5 the formants are preserved and also transients could be separated with this method. But more as this, the formants can be adjusted freely and generate virtual gender and age changes to a voice. The formant shifting feature can be used more decently for adjusting the tonal expression in realtime too.

The human voice and some expressive solo instruments are well predestinated for this kind of pitch/formant shifter and voices with a clearly defined fundamental frequency can be modified in their tonal expression by altering the formants in time dependency.