REALTIME CONTROL OF AUDIO EFFECTS

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ABSTRACT

Many sound processing effects need or can benefit from realtime control of one or more parameters. During interaction with the processor optimum settings can be achieved, or settings can be signal dependent. Time variance is a keyword here. For some effects the routing and mapping of the controller signals is of higher complexity than the audio signal routing itself.

A concept for controlling a collection of effect blocks in real time with direct user interaction, and including controllers derived from the signal itself, is presented.

This consists of a hierarchy of physical controllers, logical controllers, connection matrix, mapping functions, and effect parameters. And below that an effect parameter often controls several parameters in the signal processing algorithm. One controller can control more than one parameter and the mapping function from controller to the parameter can be set individually.

Making this very flexible system easy to use presents some interesting challenges for the user interface design. Furthermore, as control signals can be derived from the audio signal itself, the signal processor has to calculate parameters from controller settings to a much higher extent than typically done, when a host processor takes care of this. The concept has been implemented in commercially available stand-alone effects processors.

1. INTRODUCTION

In general, audio effects are not linear time-invariant systems. On the contrary: An audio effect typically has one or more user controls which may be used to change the sound during performance. Very simple is a gain control, which is hardly an effect, but known to all. A modulated gain control is an effect, though: Tremolo. Automatic gain control such as compression or expansion are examples of a relatively simple signal controlled effects. Also a large variety of filters is used in effects processors: Equalisers, wah-wah filters, simple high and low pass filters etc. Dynamic delay and reverb, where the effects level depends on the input level, is quite popular. Below the surface a pitch shifter is also highly signal dependent, in order to split up the signal into periodic segments, and maybe to add the musically correct hamony intervals. A vocoder is very heavily based on signal analysis, and spectral resynthesis.

Important for a realtime controller is that it operates smoothly, without audible artefacts such as clicks or zipper noise. This calls for special care in the implementation.

2. A HIERARCHY OF CONTROLLERS

On analog equipment there is often a quite straightforward mapping of physical controller to signal processing parameter. A gain or filter frequency control potentiometer, for instance. Trying to turn three knobs simultaneously in order to acheive a desired effect can be quite challenging, however, so other means have been found. By using voltage controlled amplifers in clever circuit designs, a quite flexible mapping from controller to parameter is possible. The modular analog synthesizer is a good example of this.

A very simple mapping from controller to parameter is sometimes used in digital effects processors: Each user parameter is assigned to a MIDI controller number, and the MIDI range from 0 to 127 (or 0 to 16383) is mapped into the full parameter range. Very similar to most analog equipment apart from the remote control capability. Any scaling or coupling of controllers is left to the external controller. This is sufficient for simple setups but for complex multi-effect processors more control is needed.

Apart from physical controllers and signal processing parameters, a third element is well known: The modulation generator, often called an LFO, low frequency oscillator. Used for making tremolo, chorus or vibrato effects. A modulation generator does not necessarily deliver a periodic signal but can also be a one-shot type such a an envelope generator started by a key press event.

Due to the many controller and parameter types involved some clarification of the hierarchy is appropriate:

1. A person who wants to control an audio effect using some physical means. There are still some problems with telepathic control :-)

2. A physical controller like a MIDI source, an expression pedal connected to a DC input or a front panel knob.

3. This is mapped to any of a number of logical controllers, or modifiers. The term modifier is used instead of the more traditional modulator because both periodic and non-periodic variations are covered.

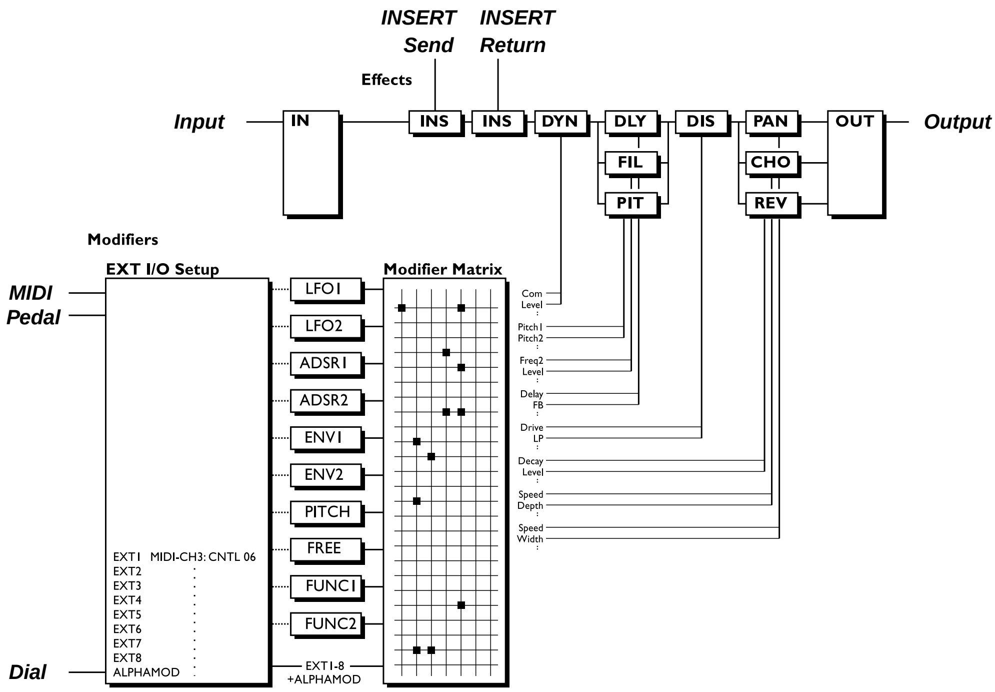

Figure 1. *An example of the control (and audio) signal flow in a multi-effects processor [1].*

4. The modifiers are mapped to effect parameters through a matrix consisting of connecting points with individual transfer functions.

5. The effect parameters are those visible and setable by the user such as filter frequencies in Hz and gains in dB.

6. Below that there is typically a large number of invisible filter coefficients, gain factors etc.

While this hierarchy may seem complex at first it is actually needed in order to make a complex system of effects maneagable. An example of the control signal flow in a multi-effects processor is shown in Figure 1.

The format of modifier signals is chosen to be continuous (24 bit resolution) and to lie between 0 and 1 (presented to the user as 0 to 100%). 0 and 1 corresponds to the endpoints in the parameter range visible by the user and with the same mapping function: Linear, logarithmic etc. This means that all controllable parameters must have a transformation function applied in order to convert the 0 to 1 range into an appropriate internal representation.

3. EXTERNAL MODIFIERS

Most modern effects units use presets, which are either made by the user or by the manufacturer. A combination of effects with their individual parameters is stored in a preset. So is the routing of the effects. This enables the user to change complex setups completely by just a few keystrokes which is fine for the audio signal flow, but how about the control signals? Basically the routing of control signals is also stored in the preset but there is a catch:

Consider the situation where a parameter or group of parameters is controlled by an analog pedal connected to a DC input. If another physical controller, such as a MIDI expression pedal, should be used with the same preset an extra layer of mapping is needed. Alternatively, users with different physical controllers cannot use the same presets. By separating the pure effects with their coupling of modifiers from the physical layer a large degree of flexibility and reuseability is achieved. The mapping of physical to logical controllers is stored separately from the audio preset.

The format of external controller signals needs to be adapted to the internal 0 to 1 format. Expression pedals deliver a somewhat arbitrary (quasi-) continuous signal. MIDI note messages, on the other hand, have a well defined meaning.

4. INTERNAL MODIFIERS

Apart from the external modifiers there are normally some internal such as LFO’s, envelope follower, envelope generators and a pitch detector. There may also be functions for combining modifier signals.

As some of these modifier signals are derived from the audio signal they must be implemented on the DSP level.

LFO's are normally built into the individual effects which rely on them, such as chorus and tremolo, and we think it is the most sensible way to keep it so. Ease of use is important for a complex audio processor. The LFO's provided as modifiers are extra, and they may control any of the modifiable parameters in the effects. And they may even be controlled to some extent from other modifiers.

Two modifier signals may be combined in various ways to generate quite complex control signal patterns. Among the combination functions are $y=x1+x2+c$, $y=x1-x2+c$ and $y=x1*x2+c$, where $x1$ and $x2$ are the two inputs, c is a constant and y is the output. Also the conversion from note information (MIDI and pitch detector) and pitch bend into the continuous modifier format can be done.

The internal modifiers may be useful for e.g. dynamically changing delay level, pitch dependent panner speed or just an extra gate.

5. CONNECTION POINTS

Mapping a modifer to a parameter may be as simple as 1:1, but almost immediately this presents problems, for how is the modifer range mapped into the parameter range? Without a transfer function no detailed control is possible. The parameter range chosen at algorithm design time may not be what the user wants the expression pedal to cover. Furthermore, the control law should not necessarily be linear. And how about the reaction time?

One approach to making the matrix connections is to use a piecewise linear transfer function but we have chosen to use a third order polynomial which gives a high degree of flexibility using only 4 parameters, se Figure 2. The small triangle below the curve shows the current value of the input to the connection point.

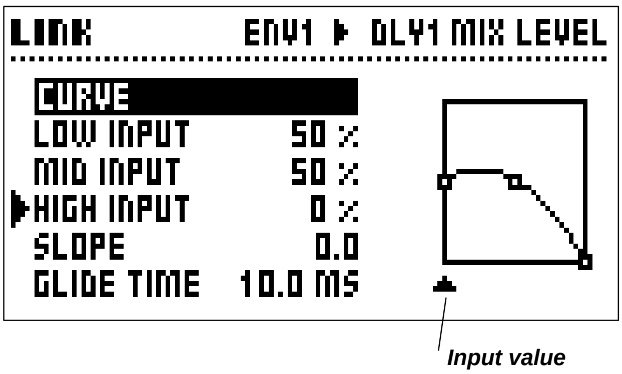

Figure 2. *The mapping function in a modifier connection point [1].*

The parameters are:

- Low, Mid, High: Output value at 0, 50 and 100% input.
- Slope: When set to 0 the tangent in the middle point has the same slope as defined by the two endpoints. Relative slopes between –10 and 10 are possible.
- Glide time. Described further below.

These parameters are transformed by the host processor to the polynomial coefficients a, b, c, and d. The modifier input is x, in the range from 0 to 1:

$$y = ax^3 + bx^2 + cx + d \tag{1}$$

Due to the timing sensitive nature of the modifiers the DSP performs the polynomial calculation of y. The output y is saturated to lie between at 0 and 1.

An adjustable smoothing filter is applied to the output of each mapping function. If one physical controller is used to control several parameters, these may therefore be updated at different speeds.

6. THE USER INTERFACE CHALLENGE

Constructing a complex system may be a challenge in itself but in general it is actually easy to make such a system impossible to use without a day-to-day practice - or a thick handbook. For comparison, take a look at the development of video recorders. Not the playback and fast winding functions but the programmed recording. Some early video recorders needed the user to sit or lie down at the unit, tapping times and dates using "morse" codes on two buttons. And shame on the person which tapped one minute too far. Then there was only one way: 59 more minutes forward. Needless to say that these units actually were equipped with wireless remote controls.

Modern video recorders allow the user to use the remote control to directly key in times and dates in a table shown on the TV screen. And it's even possible to enter recording data from "secret" codes in programme magazines.

Similarly, the control of a multi-effects processor should exploit the possiblilities of modern graphic displays. Due to the limited space on the front panel of a typical 19" rack-mount unit it is impossible to show a complete status and allow access to every parameter with just a single turn of a dedicated knob for each. A workaround could be to reduce functionality drastically but this is not considered here.

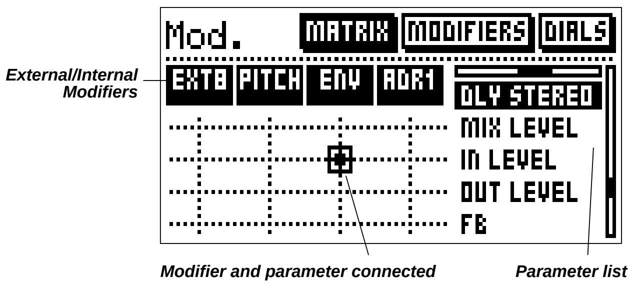

Figure 3. *The modifier matrix page in the G-Force [2].*

As an example look at Figure 3. This is a the display page for editing the modifier matrix. Two encoder wheels on the front panel are used to navigate in horizontal and vertical direction, respectively. A connection point is set with the Enter key, and removed with the Exit key. Pressing Enter on an already set connection point opens the modifier mapping page, as shown in Figure 2. Notice the use of scrollbars in both directions.

If the modifier matrix is very large it is difficult to get an overview of the connections. A slightly different approach to showing the connections can therefore be taken, see Figure 4. In this case all modifiers are shown at the screen without the need for scrolling, but the price is that only one name at a time (the selected modifier) is visible.

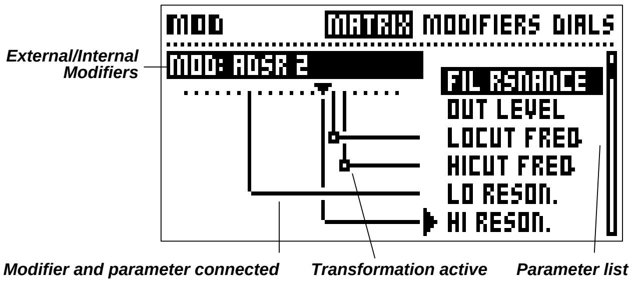

Figure 4. *The modifier matrix page in the FireworX [1].*

7. IMPLEMENTATION ISSUES

Many effects processors consist of a signal processor supported by a host processor for handling presets, reading the user interface and calculating coefficients based on the user parameters. The host processor normally also handles external controllers. When a controller or parameter has been changed new coefficients must be calculated.

The bandwidth of a controller signal is often several orders of magnitude lower than for the audio signal itself, so the necessary sampling rate is much lower. However, in order to get smooth transisions between changing parameters the coefficients must be interpolated between the slowly changing fixpoints. The relatively low sampling rate of the controller signals is helpful in making it possible to implement a complex routing structure for the control signals on a single DSP chip together with the audio signal processing.

Having the transformation from the modifier range of 0 to 1 to algorithm parameters done by the DSP requires relatively sophisticated mathematic functions to be implemented in the DSP: Conversion from gain expressed in dB to a gain factor. Calculation of first and second order filter coefficients based on a modifier value on a logarithmic frequency scale. Filter bandwidth expressed in octaves. LFO speed, also on a logarithmic scale. Optimised DSP code is needed here (also).

Some parameters are not controllable via the modifiers. These include most of the reverb parameters which are quite complicated and not necessarily relevant for realtime modification. Such parameters are calculated exclusively by the host processor.

8. CONCLUSIONS

A system for flexible realtime control of complex setups of audio effects has been presented. The controllers are arranged in a hierarchy ranging from the user to algorithm coefficients with the controller signals routed though a matrix. By having the effects and the control integrated in a single stand-alone unit handling is simplified due to the use of presets for storing audio as well as controller settings. The user interface design becomes increasingly critical with the increased complexity of effects processors. And the complexity will probably increase with increasing computing power being available.

9. REFERENCES

[1] TC Electronic, *FireworX User's Manual*, TC Electronic A/S, Risskov, 1998, http://www.tcelectronic.com.

[2] TC Electronic, *G-Force User's Manual,* TC Electronic A/S, Risskov, 1997, http://www.tcelectronic.com.