Sound Blaster Audigy mixer / default DSP code

This is based on sb-live-mixer.rst.

The EMU10K2 chips have a DSP part which can be programmed to support various ways of sample processing, which is described here. (This article does not deal with the overall functionality of the EMU10K2 chips. See the manuals section for further details.)

The ALSA driver programs this portion of chip by default code (can be altered later) which offers the following functionality:

Digital mixer controls

These controls are built using the DSP instructions. They offer extended functionality. Only the default build-in code in the ALSA driver is described here. Note that the controls work as attenuators: the maximum value is the neutral position leaving the signal unchanged. Note that if the same destination is mentioned in multiple controls, the signal is accumulated and can be wrapped (set to maximal or minimal value without checking of overflow).

Explanation of used abbreviations:

DAC
: digital to analog converter

ADC
: analog to digital converter

I2S
: one-way three wire serial bus for digital sound by Philips Semiconductors (this standard is used for connecting standalone DAC and ADC converters)

LFE
: low frequency effects (subwoofer signal)

AC97
: a chip containing an analog mixer, DAC and ADC converters

IEC958
: S/PDIF

FX-bus
: the EMU10K2 chip has an effect bus containing 64 accumulators. Each of the synthesizer voices can feed its output to these accumulators and the DSP microcontroller can operate with the resulting sum.

name='PCM Front Playback Volume',index=0

This control is used to attenuate samples for left and right front PCM FX-bus accumulators. ALSA uses accumulators 8 and 9 for left and right front PCM samples for 5.1 playback. The result samples are forwarded to the front DAC PCM slots of the Philips DAC.

name='PCM Surround Playback Volume',index=0

This control is used to attenuate samples for left and right surround PCM FX-bus accumulators. ALSA uses accumulators 2 and 3 for left and right surround PCM samples for 5.1 playback. The result samples are forwarded to the surround DAC PCM slots of the Philips DAC.

name='PCM Center Playback Volume',index=0

This control is used to attenuate samples for center PCM FX-bus accumulator. ALSA uses accumulator 6 for center PCM sample for 5.1 playback. The result sample is forwarded to the center DAC PCM slot of the Philips DAC.

name='PCM LFE Playback Volume',index=0

This control is used to attenuate sample for LFE PCM FX-bus accumulator. ALSA uses accumulator 7 for LFE PCM sample for 5.1 playback. The result sample is forwarded to the LFE DAC PCM slot of the Philips DAC.

name='PCM Playback Volume',index=0

This control is used to attenuate samples for left and right PCM FX-bus accumulators. ALSA uses accumulators 0 and 1 for left and right PCM samples for stereo playback. The result samples are forwarded to the front DAC PCM slots of the Philips DAC.

name='PCM Capture Volume',index=0

This control is used to attenuate samples for left and right PCM FX-bus accumulator. ALSA uses accumulators 0 and 1 for left and right PCM. The result is forwarded to the ADC capture FIFO (thus to the standard capture PCM device).

name='Music Playback Volume',index=0

This control is used to attenuate samples for left and right MIDI FX-bus accumulators. ALSA uses accumulators 4 and 5 for left and right MIDI samples. The result samples are forwarded to the front DAC PCM slots of the AC97 codec.

name='Music Capture Volume',index=0

These controls are used to attenuate samples for left and right MIDI FX-bus accumulator. ALSA uses accumulators 4 and 5 for left and right PCM. The result is forwarded to the ADC capture FIFO (thus to the standard capture PCM device).

name='Mic Playback Volume',index=0

This control is used to attenuate samples for left and right Mic input. For Mic input is used AC97 codec. The result samples are forwarded to the front DAC PCM slots of the Philips DAC. Samples are forwarded to Mic capture FIFO (device 1 - 16bit/8KHz mono) too without volume control.

name='Mic Capture Volume',index=0

This control is used to attenuate samples for left and right Mic input. The result is forwarded to the ADC capture FIFO (thus to the standard capture PCM device).

name='Audigy CD Playback Volume',index=0

This control is used to attenuate samples from left and right IEC958 TTL digital inputs (usually used by a CDROM drive). The result samples are forwarded to the front DAC PCM slots of the Philips DAC.

name='Audigy CD Capture Volume',index=0

This control is used to attenuate samples from left and right IEC958 TTL digital inputs (usually used by a CDROM drive). The result samples are forwarded to the ADC capture FIFO (thus to the standard capture PCM device).

name='IEC958 Optical Playback Volume',index=0

This control is used to attenuate samples from left and right IEC958 optical digital input. The result samples are forwarded to the front DAC PCM slots of the Philips DAC.

name='IEC958 Optical Capture Volume',index=0

This control is used to attenuate samples from left and right IEC958 optical digital inputs. The result samples are forwarded to the ADC capture FIFO (thus to the standard capture PCM device).

name='Line2 Playback Volume',index=0

This control is used to attenuate samples from left and right I2S ADC inputs (on the AudigyDrive). The result samples are forwarded to the front DAC PCM slots of the Philips DAC.

name='Line2 Capture Volume',index=1

This control is used to attenuate samples from left and right I2S ADC inputs (on the AudigyDrive). The result samples are forwarded to the ADC capture FIFO (thus to the standard capture PCM device).

name='Analog Mix Playback Volume',index=0

This control is used to attenuate samples from left and right I2S ADC inputs from Philips ADC. The result samples are forwarded to the front DAC PCM slots of the Philips DAC. This contains mix from analog sources like CD, Line In, Aux, ...

name='Analog Mix Capture Volume',index=1

This control is used to attenuate samples from left and right I2S ADC inputs Philips ADC. The result samples are forwarded to the ADC capture FIFO (thus to the standard capture PCM device).

name='Aux2 Playback Volume',index=0

This control is used to attenuate samples from left and right I2S ADC inputs (on the AudigyDrive). The result samples are forwarded to the front DAC PCM slots of the Philips DAC.

name='Aux2 Capture Volume',index=1

This control is used to attenuate samples from left and right I2S ADC inputs (on the AudigyDrive). The result samples are forwarded to the ADC capture FIFO (thus to the standard capture PCM device).

name='Front Playback Volume',index=0

All stereo signals are mixed together and mirrored to surround, center and LFE. This control is used to attenuate samples for left and right front speakers of this mix.

name='Surround Playback Volume',index=0

All stereo signals are mixed together and mirrored to surround, center and LFE. This control is used to attenuate samples for left and right surround speakers of this mix.

name='Center Playback Volume',index=0

All stereo signals are mixed together and mirrored to surround, center and LFE. This control is used to attenuate sample for center speaker of this mix.

name='LFE Playback Volume',index=0

All stereo signals are mixed together and mirrored to surround, center and LFE. This control is used to attenuate sample for LFE speaker of this mix.

name='Tone Control - Switch',index=0

This control turns the tone control on or off. The samples for front, rear and center / LFE outputs are affected.

name='Tone Control - Bass',index=0

This control sets the bass intensity. There is no neutral value!! When the tone control code is activated, the samples are always modified. The closest value to pure signal is 20.

name='Tone Control - Treble',index=0

This control sets the treble intensity. There is no neutral value!! When the tone control code is activated, the samples are always modified. The closest value to pure signal is 20.

name='Master Playback Volume',index=0

This control is used to attenuate samples for front, surround, center and LFE outputs.

name='IEC958 Optical Raw Playback Switch',index=0

If this switch is on, then the samples for the IEC958 (S/PDIF) digital output are taken only from the raw FX8010 PCM, otherwise standard front PCM samples are taken.

PCM stream related controls

name='EMU10K1 PCM Volume',index 0-31

Channel volume attenuation in range 0-0xffff. The maximum value (no attenuation) is default. The channel mapping for three values is as follows:

- 0 - mono, default 0xffff (no attenuation)
- 1 - left, default 0xffff (no attenuation)
- 2 - right, default 0xffff (no attenuation)

name='EMU10K1 PCM Send Routing',index 0-31

This control specifies the destination - FX-bus accumulators. There 24 values with this mapping:

- 0 - mono, A destination (FX-bus 0-63), default 0
- 1 - mono, B destination (FX-bus 0-63), default 1
- 2 - mono, C destination (FX-bus 0-63), default 2
- 3 - mono, D destination (FX-bus 0-63), default 3
- 4 - mono, E destination (FX-bus 0-63), default 0
- 5 - mono, F destination (FX-bus 0-63), default 0
- 6 - mono, G destination (FX-bus 0-63), default 0
- 7 - mono, H destination (FX-bus 0-63), default 0
- 8 - left, A destination (FX-bus 0-63), default 0
- 9 - left, B destination (FX-bus 0-63), default 1
- 10 - left, C destination (FX-bus 0-63), default 2
- 11 - left, D destination (FX-bus 0-63), default 3
- 12 - left, E destination (FX-bus 0-63), default 0
- 13 - left, F destination (FX-bus 0-63), default 0

- 14 - left, G destination (FX-bus 0-63), default 0
- 15 - left, H destination (FX-bus 0-63), default 0
- 16 - right, A destination (FX-bus 0-63), default 0
- 17 - right, B destination (FX-bus 0-63), default 1
- 18 - right, C destination (FX-bus 0-63), default 2
- 19 - right, D destination (FX-bus 0-63), default 3
- 20 - right, E destination (FX-bus 0-63), default 0
- 21 - right, F destination (FX-bus 0-63), default 0
- 22 - right, G destination (FX-bus 0-63), default 0
- 23 - right, H destination (FX-bus 0-63), default 0

Don't forget that it's illegal to assign a channel to the same FX-bus accumulator more than once (it means 0=0 && 1=0 is an invalid combination).

name='EMU10K1 PCM Send Volume',index 0-31

It specifies the attenuation (amount) for given destination in range 0-255. The channel mapping is following:

- 0 - mono, A destination attn, default 255 (no attenuation)
- 1 - mono, B destination attn, default 255 (no attenuation)
- 2 - mono, C destination attn, default 0 (mute)
- 3 - mono, D destination attn, default 0 (mute)
- 4 - mono, E destination attn, default 0 (mute)
- 5 - mono, F destination attn, default 0 (mute)
- 6 - mono, G destination attn, default 0 (mute)
- 7 - mono, H destination attn, default 0 (mute)
- 8 - left, A destination attn, default 255 (no attenuation)
- 9 - left, B destination attn, default 0 (mute)
- 10 - left, C destination attn, default 0 (mute)
- 11 - left, D destination attn, default 0 (mute)
- 12 - left, E destination attn, default 0 (mute)
- 13 - left, F destination attn, default 0 (mute)
- 14 - left, G destination attn, default 0 (mute)
- 15 - left, H destination attn, default 0 (mute)
- 16 - right, A destination attn, default 0 (mute)
- 17 - right, B destination attn, default 255 (no attenuation)
- 18 - right, C destination attn, default 0 (mute)
- 19 - right, D destination attn, default 0 (mute)
- 20 - right, E destination attn, default 0 (mute)
- 21 - right, F destination attn, default 0 (mute)
- 22 - right, G destination attn, default 0 (mute)
- 23 - right, H destination attn, default 0 (mute)

MANUALS/PATENTS

ftp://opensource.creative.com/pub/doc

LM4545.pdf
: AC97 Codec

m2049.pdf
: The EMU10K1 Digital Audio Processor

hog63.ps
: FX8010 - A DSP Chip Architecture for Audio Effects

WIPO Patents

WO 9901813 (A1)
: Audio Effects Processor with multiple asynchronous streams (Jan. 14, 1999)

WO 9901814 (A1)
: Processor with Instruction Set for Audio Effects (Jan. 14, 1999)

WO 9901953 (A1)
: Audio Effects Processor having Decoupled Instruction Execution and Audio Data Sequencing (Jan. 14, 1999)

US Patents (https://www.uspto.gov/)

US 5925841
: Digital Sampling Instrument employing cache memory (Jul. 20, 1999)

US 5928342

Audio Effects Processor integrated on a single chip with a multiport memory onto which multiple asynchronous digital sound samples can be concurrently loaded (Jul. 27, 1999)

US 5930158

Processor with Instruction Set for Audio Effects (Jul. 27, 1999)

US 6032235

Memory initialization circuit (Tram) (Feb. 29, 2000)

US 6138207

Interpolation looping of audio samples in cache connected to system bus with prioritization and modification of bus transfers in accordance with loop ends and minimum block sizes (Oct. 24, 2000)

US 6151670

Method for conserving memory storage using a pool of short term memory registers (Nov. 21, 2000)

US 6195715

Interrupt control for multiple programs communicating with a common interrupt by associating programs to GP registers, defining interrupt register, polling GP registers, and invoking callback routine associated with defined interrupt register (Feb. 27, 2001)