Sound Blaster Live mixer / default DSP code

The EMU10K1 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 EMU10K1 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:

1) IEC958 (S/PDIF) raw PCM

This PCM device (it's the 4th PCM device (index 3!) and first subdevice (index 0) for a given card) allows to forward 48kHz, stereo, 16-bit little endian streams without any modifications to the digital output (coaxial or optical). The universal interface allows the creation of up to 8 raw PCM devices operating at 48kHz, 16-bit little endian. It would be easy to add support for multichannel devices to the current code, but the conversion routines exist only for stereo (2-channel streams) at the time.

Look to tram_poke routines in lowlevel/emu10k1/emufx.c for more details.

2) 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 EMU10K1 chip has an effect bus containing 16 accumulators.
         Each of the synthesizer voices can feed its output to these accumulators
         and the DSP microcontroller can operate with the resulting sum.
```

name='Wave 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. The result samples are forwarded to the front DAC PCM slots of the AC97 codec.

name='Wave Surround 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. The result samples are forwarded to the rear I2S DACs. These DACs operates separately (they are not inside the AC97 codec).

name='Wave Center 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. The result is mixed to mono signal (single channel) and forwarded to the ??rear?? right DAC PCM slot of the AC97 codec.

name='Wave LFE 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. The result is mixed to mono signal (single channel) and forwarded to the ??rear?? left DAC PCM slot of the AC97 codec.

name='Wave Capture Volume',index=0
name='Wave Capture Switch',index=0

These controls are 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
name='Music Capture Switch',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='Surround Playback Volume',index=0

This control is used to attenuate samples for left and right rear PCM FX-bus accumulators. ALSA uses accumulators 2 and 3 for left and right rear PCM samples.

The result samples are forwarded to the rear I2S DACs. These DACs operate separately (they are not inside the AC97 codec).

name='Surround Capture Volume',index=0
name='Surround Capture Switch',index=0

These controls are used to attenuate samples for left and right rear PCM FX-bus accumulators. ALSA uses accumulators 2 and 3 for left and right rear PCM samples.
The result is forwarded to the ADC capture FIFO (thus to the standard capture PCM device).

name='Center Playback Volume',index=0

This control is used to attenuate sample for center PCM FX-bus accumulator. ALSA uses accumulator 6 for center PCM sample. The result sample is forwarded to the ??rear?? right DAC PCM slot of the AC97 codec.

name='LFE Playback Volume',index=0

This control is used to attenuate sample for center PCM FX-bus accumulator. ALSA uses accumulator 6 for center PCM sample. The result sample is forwarded to the ??rear?? left DAC PCM slot of the AC97 codec.

name='AC97 Playback Volume',index=0

This control is used to attenuate samples for left and right front ADC PCM slots of the AC97 codec. The result samples are forwarded to the front DAC PCM slots of the AC97 codec.

```
********************************************************************************
*** Note: This control should be zero for the standard operations, otherwise ***
*** a digital loopback is activated.                                         ***
********************************************************************************
```

name='AC97 Capture Volume',index=0

This control is used to attenuate samples for left and right front ADC PCM slots of the AC97 codec. The result is forwarded to the ADC capture FIFO (thus to the standard capture PCM device).

```
********************************************************************************
*** Note: This control should be 100 (maximal value), otherwise no analog    ***
*** inputs of the AC97 codec can be captured (recorded).                     ***
********************************************************************************
```

name='IEC958 TTL 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 AC97 codec.

name='IEC958 TTL 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='Zoom Video Playback Volume',index=0

This control is used to attenuate samples from left and right zoom video digital inputs (usually used by a CDROM drive). The result samples are forwarded to the front DAC PCM slots of the AC97 codec.

name='Zoom Video Capture Volume',index=0

This control is used to attenuate samples from left and right zoom video 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 LiveDrive 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 AC97 codec.

name='IEC958 LiveDrive 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='IEC958 Coaxial Playback Volume',index=0

This control is used to attenuate samples from left and right IEC958 coaxial digital inputs. The result samples are forwarded to the front DAC PCM slots of the AC97 codec.

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

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

name='Line LiveDrive Playback Volume',index=0
name='Line LiveDrive Playback Volume',index=1

This control is used to attenuate samples from left and right I2S ADC inputs (on the LiveDrive). The result samples are forwarded to the front DAC PCM slots of the AC97 codec.

name='Line LiveDrive Capture Volume',index=1
name='Line LiveDrive Capture Volume',index=1

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

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='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.

name='Headphone Playback Volume',index=1

This control attenuates the samples for the headphone output.

name='Headphone Center Playback Switch',index=1

If this switch is on, then the sample for the center PCM is put to the
left headphone output (useful for SB Live cards without separate center/LFE
output).

name='Headphone LFE Playback Switch',index=1

If this switch is on, then the sample for the center PCM is put to the
right headphone output (useful for SB Live cards without separate center/LFE
output).

3) 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 are
twelve values with this mapping:

```
         0 -  mono, A destination (FX-bus 0-15), default 0
         1 -  mono, B destination (FX-bus 0-15), default 1
         2 -  mono, C destination (FX-bus 0-15), default 2
         3 -  mono, D destination (FX-bus 0-15), default 3
```

```
         4 -  left, A destination (FX-bus 0-15), default 0
         5 -  left, B destination (FX-bus 0-15), default 1
         6 -  left, C destination (FX-bus 0-15), default 2
         7 -  left, D destination (FX-bus 0-15), default 3
         8 - right, A destination (FX-bus 0-15), default 0
         9 - right, B destination (FX-bus 0-15), default 1
        10 - right, C destination (FX-bus 0-15), default 2
        11 - right, D destination (FX-bus 0-15), default 3
```

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 -  left, A destination attn, default 255 (no attenuation)
         5 -  left, B destination attn, default 0 (mute)
         6 -  left, C destination attn, default 0 (mute)
         7 -  left, D destination attn, default 0 (mute)
         8 - right, A destination attn, default 0 (mute)
         9 - right, B destination attn, default 255 (no attenuation)
        10 - right, C destination attn, default 0 (mute)
        11 - right, D destination attn, default 0 (mute)
```

4) MANUALS/PATENTS:

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

Files:

LM4545.pdf	AC97 Codec
m2049.pdf	The EMU10K1 Digital Audio Processor
hog63.ps	FX8010 - A DSP Chip Architecture for Audio Effects

WIPO Patents

Patent numbers:

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

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 (http://www.uspto.gov/)

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

US 5928342 Audio Effects Processor integrated on a single chip
(Jul. 27, 1999)
with a multiport memory onto which multiple asynchronous
digital sound samples can be concurrently loaded

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 (Oct. 24, 2000)
system bus with prioritization and modification of bus
transfers
in accordance with loop ends and minimum block sizes

US 6151670 Method for conserving memory storage using a (Nov. 21,
2000)
pool of short term memory registers

US 6195715 Interrupt control for multiple programs communicating
with (Feb. 27, 2001)
a common interrupt by associating programs to GP
registers,
defining interrupt register, polling GP registers, and
invoking
callback routine associated with defined interrupt
register