# **TI81XX Audio Driver User Guide**



# DM81xx AM38xx Audio Driver User Guide

Linux PSP

This work is licensed under a Creative Commons Attribution-ShareAlike 3.0 Unported License [1].

## **IMPORTANT**

TI81xx refers to DM816x, DM814x and DM813X.

# Introduction

The AIC31 audio module contains audio analog inputs and outputs. It is connected to the main TI81xx processor through the TDM/I2S interface (audio interface) and used to transmit and receive audio data. The AIC31 audio codec is connected via Multi-Channel Audio Serial Port (McASP) interface, a communication peripheral, to the main processor.

McASP provides a full-duplex direct serial interface between the host device (DM1xx) and other audio devices in the system such as the AIC31 codec. It provides a direct interface to industry standard codecs, analog interface chips (AICs) and other serially connected A/D and D/A devices:

- Inter-IC Sound (I2S) compliant devices
- Pulse Code Modulation (PCM) devices
- Time Division Multiplexed (TDM) bus devices.

The AIC31 audio module is controlled by internal registers that can be accessed by the high speed I2C control interface.

This user manual defines and describes the usage of user level and platform level interfaces of the ALSA SoC Audio driver.

## References

- 1. ALSA SoC Project Homepage [http://www.alsa-project.org/main/index.php/ASoC [2]]
- 2. ALSA Project Homepage [http://www.alsa-project.org/main/index.php/Main\_Page [3]]
- 3. ALSA User Space Library [http://www.alsa-project.org/alsa-doc/alsa-lib/ [4]]
- 4. Using ALSA Audio API [http://www.equalarea.com/paul/alsa-audio.html <sup>[5]</sup>] Author: Paul Davis
- 5. TLV320AIC31 Low-Power Stereo CODEC with HP Amplifier [http://focus. ti. com/lit/ds/symlink/tlv320aic31.pdf <sup>[6]</sup>]

# **Acronyms & Definitions**

# **Audio Driver: Acronyms**

Acronym	Definition
ALSA	Advanced Linux Sound Architecture
ALSA SoC	ALSA System on Chip
DMA	Direct Memory Access
I2C	Inter-Integrated Circuit
McASP	Multi-channel Audio Serial Port
PCM	Pulse Code Modulation
TDM	Time Division Multiplexing
OSS	Open Sound System
I2S	Inter-IC Sound

# **Features**

This section describes the features supported by ALSA SoC Audio driver.

- Supports AIC31 audio codec on DM81xx in ALSA SoC framework.
- Multiple sample rates support (8KHz, 11.025KHz, 16KHz, 22.05KHz, 32KHz, 44.1KHz, 48KHz, 64KHz, 88.2KHz and 96KHz) for both capture and playback.
- · Supports audio in stereo mode.
- Supports simultaneous playback and record (full-duplex mode).
- Start, stop, pause and resume feature.
- Supports mixer interface for audio codecs.
- Supports MMAP mode for both playback and capture.
- McASP is configured as slave and AIC31 Codec is configured as master.

# **ALSA SoC Architecture**

# Introduction

The overall project goal of the ALSA System on Chip (ASoC) layer is to provide better ALSA support for embedded system on chip processors and portable audio codecs. Currently there is some support in the kernel for SoC audio, however it has some limitations:

- Currently, codec drivers are often tightly coupled to the underlying SoC cpu. This is not really ideal and leads to code duplication.
- There is no standard method to signal user initiated audio events e.g. Headphone/Mic insertion, Headphone/Mic detection after an insertion event.
- Current drivers tend to power up the entire codec when playing (or recording) audio. This is fine for a PC, but tends to waste a lot of power on portable devices. There is also no support for saving power via changing codec oversampling rates, bias currents, etc.

# **Design**

The ASoC layer is designed to address these issues and provide the following features:

- Codec independence: Allows reuse of codec drivers on other platforms and machines.
- Easy I2S/PCM audio interface setup between codec and SoC. Each SoC interface and codec registers it's audio interface capabilities with the core and are subsequently matched and configured when the application hw params are known.
- Dynamic Audio Power Management (DAPM): DAPM automatically sets the codec to it's minimum power state
  at all times. This includes powering up/down internal power blocks depending on the internal codec audio routing
  and any active streams.
- **Pop and click reduction**: Pops and clicks can be reduced by powering the codec up/down in the correct sequence (including using digital mute). ASoC signals the codec when to change power states.

To achieve all this, ASoC splits an embedded audio system into three components:

- Codec driver: The codec driver is platform independent and contains audio controls, audio interface capabilities, codec dapm definition and codec IO functions.
- **Platform driver**: The platform driver contains the audio dma engine and audio interface drivers (e.g. I2S, AC97, PCM) for that platform.
- Machine driver: The machine driver handles any machine specific controls and audio events i.e. turning on an
  amp at start of playback.

Following architecture diagram shows all the components and the interactions among them:

# **Configuration**

To enable/disable audio support, start the *Linux Kernel Configuration* tool:

```
$ make menuconfig
```

Select Device Drivers from the main menu.

```
Power management options --->
[] Networking support --->
Device Drivers --->
File systems --->
Kernel hacking --->
...
```

Select Sound card support as shown here:

```
...

Multimedia devices --->
Graphics support --->

<*> Sound card support --->

[*] HID Devices --->

[*] USB support --->

...

...
```

Select Advanced Linux Sound Architecture as shown here:

```
--- Sound card support
<*> Advanced Linux Sound Architecture --->
< > Open Sound System (DEPRECATED) --->
```

Select ALSA for SoC audio support as shown here:

```
...
[*] ARM sound devices --->
[*] SPI sound devices --->
<*> ALSA for SoC audio support --->
```

Select SoC Audio for TI81XX chip as shown here:

```
--- ALSA for SoC audio support

<*> SoC Audio for the TI81XX chip

<*> SoC Audio support for TI81XX EVM

< > Build all ASoC CODEC drivers (NEW)
```

Note: Soc Audio support for TI81XX EVM option appears iff SoC Audio for the TI81XX chips is selected

# **Module Build**

Module build for the audio driver is supported. To do this, at all the places mentioned in the section above select module build (short-cut key M).

After the modules are built, the following sound related kernel modules (\*.ko) will be generated

```
sound/soundcore.ko
sound/core/snd.ko
sound/core/snd-hwdep.ko
sound/core/snd-timer.ko
sound/core/snd-page-alloc.ko
sound/core/snd-pcm.ko
sound/core/snd-rawmidi.ko
sound/core/snd-usb-audio.ko
sound/core/snd-usb-lib.ko
sound/soc/snd-soc-core.ko
sound/soc/codecs/snd-soc-tlv320aic3x.ko
sound/soc/davinci/snd-soc-davinci.ko
sound/soc/davinci/snd-soc-davinci-mcasp.ko
sound/soc/davinci/snd-soc-evm.ko
```

The kernel modules need to be inserted in the order listed above.

# Amixer commands for AIC3x audio codec

To unmute/mute the microphone input

```
$ amixer sset 'Right PGA Mixer Mic3L' on/off
$ amixer sset 'Right PGA Mixer Mic3R' on/off
$ amixer sset 'Left PGA Mixer Mic3L' on/off
$ amixer sset 'Left PGA Mixer Mic3R' on/off
```

## To unmute/mute the line input

```
$ amixer sset 'Right PGA Mixer Line1R' on/off
$ amixer sset 'Right PGA Mixer Line1L' on/off
$ amixer sset 'Left PGA Mixer Line1R' on/off
$ amixer sset 'Left PGA Mixer Line1L' on/off
```

## To select differential configuration for the line input

```
$ amixer sset 'Left Line1R Mux' differential
$ amixer sset 'Right Line1L Mux' differential
```

## To increase the volume for capture and playback

```
$ amixer cset name='PCM Playback Volume' x%,x% (x between 0-100)
$ amixer cset name='PGA Capture Volume' x%,x%
```

Note: When using one input port for audio capture (microphone or line) please ensure that the other port is muted. If both the input ports are left enabled or configured incorrectly the captured audio stream may be noisy.

# Amixer commands for AIC26 audio codec

## Not applicable for DM8168/DM8148 EVMs

To enable audio Capture:

```
$ amixer cset numid=9 on
$ amixer cset numid=10 7
$ amixer cset numid=19 2
$ amixer cset numid=11 15
$ amixer cset numid=12 0
$ amixer cset numid=15 100
$ amixer cset numid=13 0
$ amixer cset numid=14 0
$ amixer cset numid=16 0
```

## To enable audio Playback:

```
$ amixer cset numid=2 on
$ amixer cset numid=1 63
$ amixer cset numid=18 on
```

# Using mic input on the EVM

Make sure that the following steps have been completed with the help of appropriate amixer commands

- 1. Mic input unmuted
- 2. Line input muted
- 3. Capture volume set to a reasonable level of around 50

# Using line-in input on the EVM

Make sure that the following steps have been completed with the help of appropriate amixer commands

- 1. Line-in input unmuted and set for differential configuration
- 2. Mic input muted
- 3. Capture volume set to a reasonable level of around 50%

**Note:** Please adjust volume level of the audio source used to ensure the capture quality. If the source signal level increased beyond a limit, some part of the input signals will get clipped and the noise will be amplified along with the input signal and will result in noisy output.

A source volume level of 10% is advisable while using PC as the Audio source to the Line In for better Capture quality.

# **Application Interface**

This section provides the details of the Application Interface for the ALSA Audio driver.

Application developer uses ALSA-lib, a user space library, rather than the kernel API. The library offers 100% of the functionality of the kernel API, but adds major improvements in usability, making the application code simpler and better looking.

The online-documentation for the same is available at:

http://www.alsa-project.org/alsa-doc/alsa-lib/ [4]

# **Device Interface**

The operational interface in /dev/ contains three main types of devices:

- PCM devices for recording or playing digitized sound samples,
- CTL devices that allow manipulating the internal mixer and routing of the card, and,
- MIDI devices to control the MIDI port of the card, if any.

## **Device Interface**

Name	Description
/dev/snd/controlC0	Control devices (i.e. mixer, etc)
/dev/snd/pcmC0D0c	PCM Card 0 Device 0 Capture device
/dev/snd/pcmC0D0p	PCM Card 0 Device 0 Playback device

# **Proc Interface**

The /proc/asound kernel interface is a status and configuration interface. A lot of useful information about the sound system can be found in the /proc/asound subdirectory.

See the table below for different proc entries in /proc/asound:

# **Proc Interface**

Name	Description
cards	List of registered cards
version	Version and date the driver was built on
devices	List of registered ALSA devices
pcm	The list of allocated PCM streams
cardX/(X = 0-7)	The card specific directory
cardX/pcm0p	The directory of the given PCM playback stream
cardX/pcm0c	The directory of the given PCM capture stream

# **Commonly Used APIs**

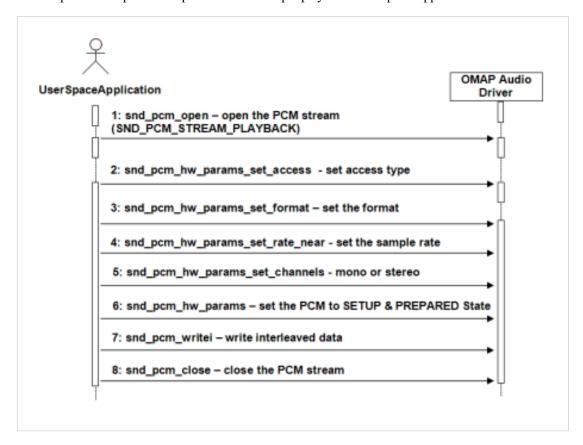
Some of the commonly used APIs to write an ALSA based application are:

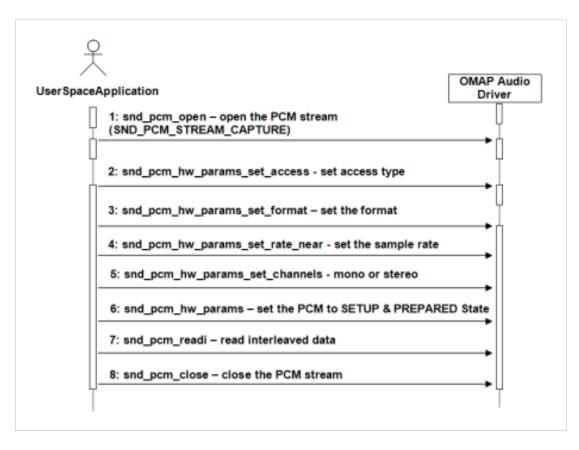
# **Commonly Used APIs**

Name	Description
snd_pcm_open	Opens a PCM stream
snd_pcm_close	Closes a previously opened PCM stream
snd_pcm_hw_params_any	Fill params with a full configuration space for a PCM
snd_pcm_hw_params_test_ < <pre>&lt;<pre>&lt;<pre>parameter&gt;&gt;</pre></pre></pre>	Test the availability of important parameters like number of channels, sample rate etc. For e.g. snd_pcm_hw_params_test_format, snd_pcm_hw_params_test_rate etc.
snd_pcm_hw_params_set_ < <pre>&lt;<pre>&lt;<pre>parameter&gt;&gt;</pre></pre></pre>	Set the different configuration parameters. For e.g. snd_pcm_hw_params_set_format, snd_pcm_hw_params_set_rate etc.
snd_pcm_hw_params	Install one PCM hardware configuration chosen from a configuration space
snd_pcm_writei	Write interleaved frames to a PCM
snd_pcm_readi	Read interleaved frames from a PCM
snd_pcm_prepare	Prepare PCM for use
snd_pcm_drop	Stop a PCM dropping pending frames
snd_pcm_drain	Stop a PCM preserving pending frames

# **User Space Interactions**

This section depicts the sequence of operations for a simple playback and capture application.





# **Sample Applications**

This chapter describes the audio sample applications provided along with the package. The source for these sample applications are available in the Examples directory of the Release Package folder.

## Introduction

Writing an audio application involves the following steps:

- Opening the audio device
- · Set the parameters of the device
- · Receive audio data from the device or deliver audio data to the device
- · Close the device

These steps are explained in detail in this section.

#### Note

User space ALSA libraries can be downloaded from this link http://www.alsa-project.org/main/index.php/Download <sup>[7]</sup>. User needs to build and install them before he starts using the ALSA based applications. Follow the INSTALL guide in the example code package to build the sample apps

# A minimal playback application

This program opens an audio interface for playback, configures it for stereo, 16 bit, 44.1kHz, interleaved conventional read/write access. Then its delivers a chunk of random data to it, and exits. It represents about the simplest possible use of the ALSA Audio API, and isn't meant to be a real program.

# Opening the audio device

To write a simple PCM application for ALSA, we first need a handle for the PCM device. Then we have to specify the direction of the PCM stream, which can be either playback or capture. We also have to provide some information about the configuration we would like to use, like buffer size, sample rate, pcm data format. So, first we declare:

```
#include <stdio.h>
#include <stdlib h>
#include <alsa/asoundlib.h>
#define BUFF SIZE 4096
int main (int argc, char *argv[])
 int err;
 short buf[BUFF_SIZE];
 int rate = 44100; /* Sample rate */
 unsigned int exact_rate; /* Sample rate returned by */
 /* Handle for the PCM device */
 snd_pcm_t *playback_handle;
  /* Playback stream */
 snd pcm stream t stream = SND PCM STREAM PLAYBACK;
  /* This structure contains information about the hardware and can be used to specify the configuration to be used for */
  /* the PCM stream. */
 snd_pcm_hw_params_t *hw_params;
```

The most important ALSA interfaces to the PCM devices are the "plughw" and the "hw" interface. If you use the "plughw" interface, you need not care much about the sound hardware. If your sound card does not support the sample rate or sample format you specify, your data will be automatically converted. This also applies to the access type and the number of channels. With the "hw" interface, you have to check whether your hardware supports the configuration you would like to use. Otherwise, user can use the default interface for playback by:

```
/* Name of the PCM device, like plughw:0,0 */
  /* The first number is the number of the soundcard, the second number is the number of the device. */
  static char *device = "default"; /* playback device */
```

Now we can open the PCM device:

```
/* Open PCM. The last parameter of this function is the mode. */
if ((err = snd_pcm_open (&playback_handle, device, stream, 0)) < 0) {
   fprintf (stderr, "cannot open audio device (%s)\n", snd_strerror (err));
   exit (1);
}</pre>
```

# Setting the parameters of the device

Now we initialize the variables and allocate the hwparams structure:

```
/* Allocate the snd_pcm_hw_params_t structure on the stack. */
if ((err = snd_pcm_hw_params_malloc (&hw_params)) < 0) {
   fprintf (stderr, "cannot allocate hardware parameters (%s)\n", snd_strerror (err));
   exit (1);
}</pre>
```

Before we can write PCM data to the soundcard, we have to specify access type, sample format, sample rate, number of channels, number of periods and period size. First, we initialize the hwparams structure with the full configuration space of the soundcard:

```
/* Init hwparams with full configuration space */
if ((err = snd_pcm_hw_params_any (playback_handle, hw_params)) < 0) {
   fprintf (stderr, "cannot initialize hardware parameter structure (%s)\n", snd_strerror (err));
   exit (1);
}</pre>
```

Now configure the desired parameters. For this example, we assume that the soundcard can be configured for stereo playback of 16 Bit Little Endian data, sampled at 44100 Hz. Therefore, we restrict the configuration space to match this configuration only. The access type specifies the way in which multi-channel data is stored in the buffer. For INTERLEAVED access, each frame in the buffer contains the consecutive sample data for the channels. For 16 Bit stereo data, this means that the buffer contains alternating words of sample data for the left and right channel.

```
/* Set access type. */
if ((err = snd_pcm_hw_params_set_access (playback_handle, hw_params, SND_PCM_ACCESS_RW_INTERLEAVED)) < 0) {
   fprintf (stderr, "cannot set access type (%s)\n", snd_strerror (err));
   exit (1);
}

/* Set sample format */</pre>
```

```
if ((err = snd_pcm_hw_params_set_format (playback_handle, hw_params, SND_PCM_FORMAT_S16_LE)) < 0) {
    fprintf (stderr, "cannot set sample format (%s)\n", snd_strerror (err));
    exit (1);
}

/* Set sample rate. If the exact rate is not supported by the hardware, use nearest possible rate. */
exact_rate = rate;
if ((err = snd_pcm_hw_params_set_rate_near (playback_handle, hw_params, &exact_rate, 0)) < 0) {
    fprintf (stderr, "cannot set sample rate (%s)\n", snd_strerror (err));
    exit (1);
}

if (rate != exact_rate) {
    fprintf(stderr, "The rate %d Hz is not supported by your hardware.\n ==> Using %d Hz instead.\n", rate, exact_rate);
}

/* Set number of channels */
if ((err = snd_pcm_hw_params_set_channels (playback_handle, hw_params, 2)) < 0) {
    fprintf (stderr, "cannot set channel count (%s)\n", snd_strerror (err));
    exit (1);
}</pre>
```

Now we apply the configuration to the PCM device pointed to by pcm\_handle and prepare the PCM device.

```
/* Apply HW parameter settings to PCM device and prepare device. */
if ((err = snd_pcm_hw_params (playback_handle, hw_params)) < 0) {
   fprintf (stderr, "cannot set parameters (%s)\n", snd_strerror (err));
   exit (1);
}

snd_pcm_hw_params_free (hw_params);

if ((err = snd_pcm_prepare (playback_handle)) < 0) {
   fprintf (stderr, "cannot prepare audio interface for use (%s)\n", snd_strerror (err));
   exit (1);
}</pre>
```

# Writing data to the device

After the PCM device is configured, we can start writing PCM data to it. The first write access will start the PCM playback. For interleaved write access, we use the function:

```
/* Write some junk data to produce sound. */
if ((err = snd_pcm_writei (playback_handle, buf, BUFF_SIZE/2)) != BUFF_SIZE/2) {
   fprintf (stderr, "write to audio interface failed (%s)\n", snd_strerror (err));
   exit (1);
} else {
   fprintf (stdout, "snd_pcm_writei successful\n");
}
```

After the PCM playback is started, we have to make sure that our application sends enough data to the soundcard buffer. Otherwise, a buffer under-run will occur. After such an under-run has occurred, snd\_pcm\_prepare should be called.

# Closing the device

After the data has been transferred, the device needs to be closed by calling:

```
snd_pcm_close (playback_handle);
exit (0);
}
```

# A minimal record application

This program opens an audio interface for capture, configures it for stereo, 16 bit, 44.1kHz, interleaved conventional read/write access. Then its reads a chunk of random data from it, and exits. It isn't meant to be a real program.

Note that it is not possible to use one pcm handle for both playback and capture. So you have to configure two handles if you want to access the PCM device in both directions.

```
#include <stdio.h>
#include <stdlib.h>
#include <alsa/asoundlib.h>
#define BUFF_SIZE 4096
int main (int argc, char *argv[])
 int err:
 short buf[BUFF_SIZE];
 int rate = 44100; /* Sample rate */
 int exact_rate; /* Sample rate returned by */ \,
 snd_pcm_t *capture_handle;
 /* This structure contains information about the hardware and can be used to specify the configuration to be used for */
 /* the PCM stream. */
 snd_pcm_hw_params_t *hw_params;
 /* Name of the PCM device, like hw:0,0 */
 /\star The first number is the number of the soundcard, the second number is the number of the device. \star/
 static char *device = "hw:0,0"; /* capture device */
 /* Open PCM. The last parameter of this function is the mode. */
 fprintf (stderr, "cannot open audio device (%s)\n", snd_strerror (err));
   exit (1);
 memset (buf, 0, BUFF SIZE);
```

```
/\!\!\!\!\!^\star Allocate the snd_pcm_hw_params_t structure on the stack. \!\!\!\!^\star/\!\!\!\!
if ((err = snd_pcm_hw_params_malloc (&hw_params)) < 0) {</pre>
 fprintf (stderr, "cannot allocate hardware parameter structure (%s)\n", snd_strerror (err));
 exit (1);
}
/\star Init hwparams with full configuration space \star/
if ((err = snd pcm hw params any (capture handle, hw params)) < 0) {
 fprintf (stderr, "cannot initialize hardware parameter structure (%s) \n", snd_strerror (err));
 exit (1);
}
/* Set access type. */
if ((err = snd_pcm_hw_params_set_access (capture_handle, hw_params, SND_PCM_ACCESS_RW_INTERLEAVED)) < 0) {
 fprintf (stderr, "cannot set access type (%s)\n", snd_strerror (err));
}
/* Set sample format */
if ((err = snd_pcm_hw_params_set_format (capture_handle, hw_params, SND_PCM_FORMAT_S16_LE)) < 0) {
 fprintf (stderr, "cannot set sample format (%s)\n", snd_strerror (err));
 exit (1);
}
/\star Set sample rate. If the exact rate is not supported by the hardware, use nearest possible rate. \star/
exact_rate = rate;
 \  \  \text{if ((err = snd\_pcm\_hw\_params\_set\_rate\_near (capture\_handle, hw\_params, \&exact\_rate, 0))} < 0) \  \  \{ \  \  \  \  \} 
 fprintf (stderr, "cannot set sample rate (%s)\n", snd_strerror (err));
 exit (1);
if (rate != exact_rate) {
  fprintf(stderr, "The rate %d Hz is not supported by your hardware.\n ==> Using %d Hz instead.\n", rate, exact_rate);
/* Set number of channels */
if ((err = snd pcm hw params set channels(capture handle, hw params, 2)) < 0) {
  fprintf (stderr, "cannot set channel count (%s)\n", snd_strerror (err));
  exit (1);
/* Apply HW parameter settings to PCM device and prepare device. */
if ((err = snd_pcm_hw_params (capture_handle, hw_params)) < 0) {</pre>
  fprintf (stderr, "cannot set parameters (%s)\n", snd_strerror (err));
  exit (1);
```

```
snd_pcm_hw_params_free (hw_params);

if ((err = snd_pcm_prepare (capture_handle)) < 0) {
    fprintf (stderr, "cannot prepare audio interface for use (%s)\n", snd_strerror (err));
    exit (1);
}

/* Read data into the buffer. */
if ((err = snd_pcm_readi (capture_handle, buf, 128)) != 128) {
    fprintf (stderr, "read from audio interface failed (%s)\n", snd_strerror (err));
    exit (1);
} else {
    fprintf (stdout, "snd_pcm_readi successful\n");
}

snd_pcm_close (capture_handle);
exit (0);
}</pre>
```

# References

- [1] http://creativecommons.org/licenses/by-sa/3.0/
- [2] http://www.alsa-project.org/main/index.php/ASoC
- [3] http://www.alsa-project.org/main/index.php/Main\_Page
- [4] http://www.alsa-project.org/alsa-doc/alsa-lib/
- [5] http://www.equalarea.com/paul/alsa-audio.html
- $[6] \ http://focus.ti.com/lit/ds/symlink/tlv320aic31.pdf$
- [7] http://www.alsa-project.org/main/index.php/Download

# **Article Sources and Contributors**

TI81XX Audio Driver User Guide Source: http://processors.wiki.ti.com/index.php?oldid=96268 Contributors: Deepu.raj, Parth.saxena

# **Image Sources, Licenses and Contributors**

Image:TIBanner.png Source: http://processors.wiki.ti.com/index.php?title=File:TIBanner.png License: unknown Contributors: Nsnehaprabha

Image:Alsa-usecase1.png Source: http://processors.wiki.ti.com/index.php?title=File:Alsa-usecase1.png License: unknown Contributors: SriramAG

Image: Alsa-usecase 2.png Source: http://processors.wiki.ti.com/index.php?title=File: Alsa-usecase 2.png License: unknown Contributors: SriramAG

# License

THE WORK (AS DEFINED BELOW) IS PROVIDED UNDER THE TERMS OF THIS CREATIVE COMMONS PUBLIC LICENSE ("CCPL" OR "LICENSE"). THE WORK IS PROTECTED BY COPYRIGHT AND/OR OTHER APPLICABLE LAW. ANY USE OF THE WORK OTHER THAN AS AUTHORIZED UNDER THIS LICENSE OR COPYRIGHT LAW IS PROHIBITED.

BY EXERCISING ANY RIGHTS TO THE WORK PROVIDED HERE, YOU ACCEPT AND AGREE TO BE BOUND BY THE TERMS OF THIS LICENSE. TO THE EXTENT THIS LICENSE MAY BE CONSIDERED TO BE A CONTRACT, THE LICENSOR GRANTS YOU THE RIGHTS CONTAINED HERE IN CONSIDERATION OF YOUR ACCEPTANCE OF SUCH TERMS AND CONDITIONS.

#### License

#### 1. Definitions

- Adaptation means a work based upon the Work, or upon the Work and other pre-existing works, such as a translation, adaptation, derivative work, arrangement of music or other alterations of a literary or artistic work, or phonogram or performance and includes cinematographic adaptations or any other form in which the Work may be recast, transformed, or adapted including in any form recognizably derived from the original, except that a work interestications with a moving image ("synching") will be considered an Adaptation for the purpose of this License.

  "Collection" means a collection of literary or artistic works, such as encyclopedias and anthologies, or performances, phonograms or broadcasts, or other works or subject matter other than works listed in Section 1(f) below, which, by reason of the selection and arrangement of their contents, constitute intellectual creations, in which the work is included in its entirety in unmodified form along with one or more other contributions, each constituting separate and independent works in themselves, which together are assembled into a collective whole. A work that constitutes a Collection will not be considered an Adaptation (as defined below) for the purpose of this License.

  "Creative Commons Compatible Licenses" means a license that is listed at http://creative/commons.org/coverommons.or

## 2. Fair Dealing Rights

ended to reduce, limit, or restrict any uses free from copyright or rights arising from limitations or exceptions that are provided for in connection with the copyright protection under copyright law or other Nothing in this Licer applicable laws.

## 3. License Grant

s and conditions of this License, Licensor hereby grants You a worldwide, royalty-free, non-exclusive, perpetual (for the duration of the applicable copyright) license to exercise the rights in the Work as stated

- to Reproduce the Work, to incorporate the Work into one or more Collections, and to Reproduce the Work as incorporated in the Collections; to create and Reproduce Adaptations provided that any such Adaptation, including any translation in any medium, takes reasonable steps to clearly label, demarcate or otherwise identify that changes were made to the original Work. For example, a translation could be marked "The original work was translated from English to Spanish," or a modification could indicate "The original work has been modified."; to Distribute and Publicly Perform the Work including as incorporated in Collections; and, to Distribute and Publicly Perform Adaptations.

  For the avoidance of doubt:

- i. Non-waivable Compulsory License Schemes. In those jurisdictions in which the right to collect royalties through any statutory or compulsory licensing scheme cannot be waived, the Licensor reserves the exclusive right to collect such royalties for any exercise by You of the rights granted under this License;
  ii. Waivable Compulsory License Schemes. In those jurisdictions in which the right to collect royalties through any statutory or compulsory licensing scheme can be waived, the Licensor waives the exclusive right to collect such royalties for any exercise by You of the rights granted under this License; and,
  iii. Voluntary License Schemes. The Licensor waives the right to collect only the collect such royalties for any exercise by You of the rights granted under this License; and,
  that society, from any exercise by You of the rights granted under this License.

  The above rights may be exercised in all media and formats whether now known or hereafter devised. The above rights include the right to make such modifications as are technically necessary to exercise the rights in other media and formats. Subject to Section 8(f), all rights not expressly granted by Licensor are hereby reserved.

nse granted in Section 3 above is expressly made subject to and limited by the following restrictions:

- Restrictions: license granted in Section 3 above is expressly made subject to and limited by the following restrictions:

  You may Distribute or Publicly Perform the Work only under the terms of this License. You must include a copy of, or the Uniform Resource Identifier (URI) for, this License with every copy of the Work You Distribute or Publicly Perform. You may not offer or impowe any terms on the Work that refer to this License and to the disclaimer of warranties with every copy of the Work You Distribute or Publicly Perform. When You must keep intact all notices that refer to this License and to the disclaimer of warranties with every copy of the Work You Distribute or Publicly Perform. When You impose any effective technological measures on the Work that restrict the ability of a recipient of the Work from You to exercise the rights granted to that recipient under the terms of the License. This Section 4(a) applies to the Work as incorporated in a Collection, but this does not require the Collection apart from the Work itself to be made subject to the terms of this License. If You create a Adaptation on you must, to the extent practicable, remove from the Adaptation any credit as required by Section 4(c), as requested.

  You may Distribute or Publicly Perform an Adaptation only under the terms of: (i) this License; (ii) a later version of this License with the same License Elements as this License. (iii) a Creative Commons Compatible License. If you clines the Adaptation under one of the licenses mentioned in (iv), you must comply with the terms of that License. If you must comply with the terms of the Adaptation of the licenses with the same License generally and the following provisions; (i) You must include a copy of, to the URI for, the Applicable Licenses, If you diverse the Adaptation and the very copy of each Adaptation You Distribute or Publicly Perform; (IV) when You Distribute or Impose any terms on the Adaptation that recipient under the terms of the Applicable License with the Applicable Licens

5. Representations, Warranties and Disclaimer
UNLESS OTHERWISE MUTUALLY AGREED TO BY THE PARTIES IN WRITING, LICENSOR OFFERS THE WORK AS-IS AND MAKES NO REPRESENTATIONS OR WARRANTIES OF ANY KIND CONCERNING
THE WORK, EXPRESS, IMPLIED, STATUTORY OR OTHERWISE, INCLUDING, WITHOUT LIMITATION, WARRANTIES OF TITLE, MERCHANTIBILITY, FITNESS FOR A PARTICULAR PURPOSE,
NONINFRINGEMENT, OR THE ABSENCE OF LATENT OR OTHER DEFECTS, ACCURACY, OR THE PRESENCE OF ABSENCE OF ERRORS, WHETHER OR NOT DISCOVERABLE. SOME JURISDICTIONS DO NOT
ALLOW THE EXCLUSION OF IMPLIED WARRANTIES, SO SUCH EXCLUSION MAY NOT APPLY TO YOU.

6. Limitation on Liability

EXCEPT TO THE EXTENT REQUIRED BY APPLICABLE LAW, IN NO EVENT WILL LICENSOR BE LIABLE TO YOU ON ANY LEGAL THEORY FOR ANY SPECIAL, INCIDENTAL, CONSEQUENTIAL, PUNITIVE OR EXEMPLARY DAMAGES ARISING OUT OF THIS LICENSE OR THE USE OF THE WORK, EVEN IF LICENSOR HAS BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES.

### 7. Termination

This License and the rights granted hereunder will terminate automatically upon any breach by You of the terms of this License. Individuals or entities who have received Adaptations or Collections from You under this License, however, will not have their licenses terminated provided such individuals or entities remain in full compliance with those licenses. Sections 1, 2, 5, 6, 7, and 8 will survive any termination of this License.

License 16

Subject to the above terms and conditions, the license granted here is perpetual (for the duration of the applicable copyright in the Work). Notwithstanding the above, Licensor reserves the right to release the Work under different license terms or to stop distributing the Work at any time; provided, however that any such election will not serve to withdraw this License (or any other license that has been, or is required to be, granted under the terms of this License, and this License will continue in full force and effect unless terminated as stated above.

#### 8. Miscellaneous

- Each time You Distribute or Publicly Perform the Work or a Collection, the Licensor offers to the recipient a license to the Work on the same terms and conditions as the license granted to You under this License. Each time You Distribute or Publicly Perform an Adaptation, Licensor offers to the recipient a license to the original Work on the same terms and conditions as the license granted to You under this License. If any provision of this License is invalid or unenforceable law, it shall not affect the voltage under applicable law, it shall not affect the voltage received in the remainder of the terms of this License, and without further action by the parties to this agreement, such provision shall be reformed to the minimum extent necessary to make such provision valid and enforceable.

  No term or provision of this License shall be deemed waived and no breach consented to unless such waiver or consent shall be in writing and signed by the party to be charged with such waiver or consent.

  This License constitutes the entire agreement between the parties with respect to the Work licensed here. There are no understandings, agreements or representations with respect to the Work not specified here. Licensor shall not be bound by any additional provisions that may appear in any communication from You. This License may not be modified without the mutual written agreement of the Licensor and You.

  The rights granted under, and the subject matter referenced, in this License were darfied utilizing the terminology of the Berne Convention for Iby Convention of 10461, the WIPO Copyright Treaty of 1996, the WIPO Copyright Treaty of 1996, the WIPO Performances and Phonograms Treaty of 1996 and the Universal Copyright Convention of as revised on July 24, 1971). These rights and subject matter take effect in the relevant jurisdiction in which the License terms are sought to be enforced according to the corresponding provisions of the implementation of those treaty provisions in the applicable analomal law. If the license of