Advanced Linux Sound Architecture - Driver Configuration guide

Kernel Configuration

To enable ALSA support you need at least to build the kernel with primary sound card support (`CONFIG_SOUND`). Since ALSA can emulate OSS, you don't have to choose any of the OSS modules.

Enable "OSS API emulation" (`CONFIG_SND_OSSEMUL`) and both OSS mixer and PCM supports if you want to run OSS applications with ALSA.

If you want to support the WaveTable functionality on cards such as SB Live! then you need to enable "Sequencer support" (`CONFIG_SND_SEQUENCER`).

To make ALSA debug messages more verbose, enable the "Verbose printk" and "Debug" options. To check for memory leaks, turn on "Debug memory" too. "Debug detection" will add checks for the detection of cards.

Please note that all the ALSA ISA drivers support the Linux isapnp API (if the card supports ISA PnP). You don't need to configure the cards using isapnptools.

Module parameters

The user can load modules with options. If the module supports more than one card and you have more than one card of the same type then you can specify multiple values for the option separated by commas.

Module snd

The core ALSA module. It is used by all ALSA card drivers. It takes the following options which have global effects.

major
: major number for sound driver; Default: 116

cards_limit
: limiting card index for auto-loading (1-8); Default: 1; For auto-loading more than one card, specify this option together with snd-card-X aliases.

slots
: Reserve the slot index for the given driver; This option takes multiple strings. See Module Autoloading Support section for details.

debug
: Specifies the debug message level; (0 = disable debug prints, 1 = normal debug messages, 2 = verbose debug messages); This option appears only when `CONFIG_SND_DEBUG=y`. This option can be dynamically changed via sysfs /sys/modules/snd/parameters/debug file.

Module snd-pcm-oss

The PCM OSS emulation module. This module takes options which change the mapping of devices.

dsp_map
: PCM device number maps assigned to the 1st OSS device; Default: 0

adsp_map
: PCM device number maps assigned to the 2st OSS device; Default: 1

nonblock_open
: Don't block opening busy PCM devices; Default: 1

For example, when `dsp_map=2`, /dev/dsp will be mapped to PCM #2 of the card #0. Similarly, when `adsp_map=0`, /dev/adsp will be mapped to PCM #0 of the card #0. For changing the second or later card, specify the option with commas, such like `dsp_map=0,1`.

`nonblock_open` option is used to change the behavior of the PCM regarding opening the device. When this option is non-zero, opening a busy OSS PCM device won't be blocked but return immediately with EAGAIN (just like O_NONBLOCK flag).

Module snd-rawmidi

This module takes options which change the mapping of devices. similar to those of the snd-pcm-oss module.

midi_map
: MIDI device number maps assigned to the 1st OSS device; Default: 0

amidi_map
: MIDI device number maps assigned to the 2st OSS device; Default: 1

Module snd-soc-core

The soc core module. It is used by all ALSA card drivers. It takes the following options which have global effects.

prealloc_buffer_size_kbytes
: Specify prealloc buffer size in kbytes (default: 512).

Common parameters for top sound card modules

Each of top level sound card module takes the following options.

index
: index (slot #) of sound card; Values: 0 through 31 or negative; If nonnegative, assign that index number; if negative, interpret as a bitmask of permissible indices; the first free permitted index is assigned; Default: -1

id
: card ID (identifier or name); Can be up to 15 characters long; Default: the card type; A directory by this name is created under /proc/asound/ containing information about the card; This ID can be used instead of the index number in identifying the card

enable
: enable card; Default: enabled, for PCI and ISA PnP cards

Module snd-adlib

Module for AdLib FM cards.

port
: port # for OPL chip

This module supports multiple cards. It does not support autoprobe, so the port must be specified. For actual AdLib FM cards it will be 0x388. Note that this card does not have PCM support and no mixer; only FM synthesis.

Make sure you have `sbiload` from the alsa-tools package available and, after loading the module, find out the assigned ALSA sequencer port number through `sbiload -l`.

Example output:

```
Port     Client name                       Port name
64:0     OPL2 FM synth                     OPL2 FM Port
```

Load the `std.sb` and `drums.sb` patches also supplied by `sbiload`:

```
sbiload -p 64:0 std.sb drums.sb
```

If you use this driver to drive an OPL3, you can use `std.o3` and `drums.o3` instead. To have the card produce sound, use `aplaymidi` from alsa-utils:

```
aplaymidi -p 64:0 foo.mid
```

Module snd-ad1816a

Module for sound cards based on Analog Devices AD1816A/AD1815 ISA chips.

clockfreq
: Clock frequency for AD1816A chip (default = 0, 33000Hz)

This module supports multiple cards, autoprobe and PnP.

Module snd-ad1848

Module for sound cards based on AD1848/AD1847/CS4248 ISA chips.

port
: port # for AD1848 chip

irq
: IRQ # for AD1848 chip

dma1
: DMA # for AD1848 chip (0,1,3)

This module supports multiple cards. It does not support autoprobe thus main port must be specified!!! Other ports are optional.

The power-management is supported.

Module snd-ad1889

Module for Analog Devices AD1889 chips.

ac97_quirk

AC'97 workaround for strange hardware; See the description of intel8x0 module for details.

This module supports multiple cards.

Module snd-ali5451

Module for ALi M5451 PCI chip.

pcm_channels
: Number of hardware channels assigned for PCM

spdif
: Support SPDIF I/O; Default: disabled

This module supports one chip and autoprobe.

The power-management is supported.

Module snd-als100

Module for sound cards based on Avance Logic ALS100/ALS120 ISA chips.

This module supports multiple cards, autoprobe and PnP.

The power-management is supported.

Module snd-als300

Module for Avance Logic ALS300 and ALS300+

This module supports multiple cards.

The power-management is supported.

Module snd-als4000

Module for sound cards based on Avance Logic ALS4000 PCI chip.

joystick_port
: port # for legacy joystick support; 0 = disabled (default), 1 = auto-detect

This module supports multiple cards, autoprobe and PnP.

The power-management is supported.

Module snd-asihpi

Module for AudioScience ASI soundcards

enable_hpi_hwdep
: enable HPI hwdep for AudioScience soundcard

This module supports multiple cards. The driver requires the firmware loader support on kernel.

Module snd-atiixp

Module for ATI IXP 150/200/250/400 AC97 controllers.

ac97_clock
: AC'97 clock (default = 48000)

ac97_quirk
: AC'97 workaround for strange hardware; See AC97 Quirk Option section below.

ac97_codec
: Workaround to specify which AC'97 codec instead of probing. If this works for you file a bug with your *lspci -vn* output. (-2 = Force probing, -1 = Default behavior, 0-2 = Use the specified codec.)

spdif_aclink
: S/PDIF transfer over AC-link (default = 1)

This module supports one card and autoprobe.

ATI IXP has two different methods to control SPDIF output. One is over AC-link and another is over the "direct" SPDIF output. The implementation depends on the motherboard, and you'll need to choose the correct one via spdif_aclink module option.

The power-management is supported.

Module snd-atiixp-modem

Module for ATI IXP 150/200/250 AC97 modem controllers.

This module supports one card and autoprobe.

Note: The default index value of this module is -2, i.e. the first slot is excluded.

The power-management is supported.

Module snd-au8810, snd-au8820, snd-au8830

Module for Aureal Vortex, Vortex2 and Advantage device.

pcifix
: Control PCI workarounds; 0 = Disable all workarounds, 1 = Force the PCI latency of the Aureal card to 0xff, 2 = Force the Extend PCI#2 Internal Master for Efficient Handling of Dummy Requests on the VIA KT133 AGP Bridge, 3 = Force both settings, 255 = Autodetect what is required (default)

This module supports all ADB PCM channels, ac97 mixer, SPDIF, hardware EQ, mpu401, gameport. A3D and wavetable support are still in development. Development and reverse engineering work is being coordinated at https://savannah.nongnu.org/projects/openvortex/ SPDIF output has a copy of the AC97 codec output, unless you use the `spdif` pcm device, which allows raw data passthru. The hardware EQ hardware and SPDIF is only present in the Vortex2 and Advantage.

Note: Some ALSA mixer applications don't handle the SPDIF sample rate control correctly. If you have problems regarding this, try another ALSA compliant mixer (alsamixer works).

Module snd-azt1605

Module for Aztech Sound Galaxy soundcards based on the Aztech AZT1605 chipset.

port
: port # for BASE (0x220,0x240,0x260,0x280)

wss_port
: port # for WSS (0x530,0x604,0xe80,0xf40)

irq
: IRQ # for WSS (7,9,10,11)

dma1
: DMA # for WSS playback (0,1,3)

dma2
: DMA # for WSS capture (0,1), -1 = disabled (default)

mpu_port
: port # for MPU-401 UART (0x300,0x330), -1 = disabled (default)

mpu_irq
: IRQ # for MPU-401 UART (3,5,7,9), -1 = disabled (default)

fm_port
: port # for OPL3 (0x388), -1 = disabled (default)

This module supports multiple cards. It does not support autoprobe: `port`, `wss_port`, `irq` and `dma1` have to be specified. The other values are optional.

`port` needs to match the BASE ADDRESS jumper on the card (0x220 or 0x240) or the value stored in the card's EEPROM for cards that have an EEPROM and their "CONFIG MODE" jumper set to "EEPROM SETTING". The other values can be chosen freely from the options enumerated above.

If `dma2` is specified and different from `dma1`, the card will operate in full-duplex mode. When `dma1=3`, only `dma2=0` is valid and the only way to enable capture since only channels 0 and 1 are available for capture.

Generic settings are `port=0x220 wss_port=0x530 irq=10 dma1=1 dma2=0 mpu_port=0x330 mpu_irq=9 fm_port=0x388`.

Whatever IRQ and DMA channels you pick, be sure to reserve them for legacy ISA in your BIOS.

Module snd-azt2316

Module for Aztech Sound Galaxy soundcards based on the Aztech AZT2316 chipset.

port
: port # for BASE (0x220,0x240,0x260,0x280)

wss_port
: port # for WSS (0x530,0x604,0xe80,0xf40)

irq
: IRQ # for WSS (7,9,10,11)

dma1
: DMA # for WSS playback (0,1,3)

dma2
: DMA # for WSS capture (0,1), -1 = disabled (default)

mpu_port
: port # for MPU-401 UART (0x300,0x330), -1 = disabled (default)

mpu_irq
: IRQ # for MPU-401 UART (5,7,9,10), -1 = disabled (default)

fm_port
: port # for OPL3 (0x388), -1 = disabled (default)

This module supports multiple cards. It does not support autoprobe: `port`, `wss_port`, `irq` and `dma1` have to be specified. The other values are optional.

`port` needs to match the BASE ADDRESS jumper on the card (0x220 or 0x240) or the value stored in the card's EEPROM for cards that have an EEPROM and their "CONFIG MODE" jumper set to "EEPROM SETTING". The other values can be chosen freely from the options enumerated above.

If `dma2` is specified and different from `dma1`, the card will operate in full-duplex mode. When `dma1=3`, only `dma2=0` is valid and the only way to enable capture since only channels 0 and 1 are available for capture.

Generic settings are `port=0x220 wss_port=0x530 irq=10 dma1=1 dma2=0 mpu_port=0x330 mpu_irq=9 fm_port=0x388`.

Whatever IRQ and DMA channels you pick, be sure to reserve them for legacy ISA in your BIOS.

Module snd-aw2

Module for Audiowerk2 sound card

This module supports multiple cards.

Module snd-azt2320

Module for sound cards based on Aztech System AZT2320 ISA chip (PnP only).

This module supports multiple cards, PnP and autoprobe.

The power-management is supported.

Module snd-azt3328

Module for sound cards based on Aztech AZF3328 PCI chip.

joystick
: Enable joystick (default off)

This module supports multiple cards.

Module snd-bt87x

Module for video cards based on Bt87x chips.

digital_rate
: Override the default digital rate (Hz)

load_all
: Load the driver even if the card model isn't known

This module supports multiple cards.

Note: The default index value of this module is -2, i.e. the first slot is excluded.

Module snd-ca0106

Module for Creative Audigy LS and SB Live 24bit

This module supports multiple cards.

Module snd-cmi8330

Module for sound cards based on C-Media CMI8330 ISA chips.

isapnp
: ISA PnP detection - 0 = disable, 1 = enable (default)

with `isapnp=0`, the following options are available:

wssport
: port # for CMI8330 chip (WSS)

wssirq
: IRQ # for CMI8330 chip (WSS)

wssdma
: first DMA # for CMI8330 chip (WSS)

sbport

port # for CMI8330 chip (SB16)

sbirq
: IRQ # for CMI8330 chip (SB16)

sbdma8
: 8bit DMA # for CMI8330 chip (SB16)

sbdma16
: 16bit DMA # for CMI8330 chip (SB16)

fmport
: (optional) OPL3 I/O port

mpuport
: (optional) MPU401 I/O port

mpuirq
: (optional) MPU401 irq #

This module supports multiple cards and autoprobe.

The power-management is supported.

Module snd-cmipci

Module for C-Media CMI8338/8738/8768/8770 PCI sound cards.

mpu_port
: port address of MIDI interface (8338 only): 0x300,0x310,0x320,0x330 = legacy port, 1 = integrated PCI port (default on 8738), 0 = disable

fm_port
: port address of OPL-3 FM synthesizer (8x38 only): 0x388 = legacy port, 1 = integrated PCI port (default on 8738), 0 = disable

soft_ac3
: Software-conversion of raw SPDIF packets (model 033 only) (default = 1)

joystick_port
: Joystick port address (0 = disable, 1 = auto-detect)

This module supports autoprobe and multiple cards.

The power-management is supported.

Module snd-cs4231

Module for sound cards based on CS4231 ISA chips.

port
: port # for CS4231 chip

mpu_port
: port # for MPU-401 UART (optional), -1 = disable

irq
: IRQ # for CS4231 chip

mpu_irq
: IRQ # for MPU-401 UART

dma1
: first DMA # for CS4231 chip

dma2
: second DMA # for CS4231 chip

This module supports multiple cards. This module does not support autoprobe thus main port must be specified!!! Other ports are optional.

The power-management is supported.

Module snd-cs4236

Module for sound cards based on CS4232/CS4232A, CS4235/CS4236/CS4236B/CS4237B/CS4238B/CS4239 ISA chips.

isapnp
: ISA PnP detection - 0 = disable, 1 = enable (default)

with `isapnp=0`, the following options are available:

port
: port # for CS4236 chip (PnP setup - 0x534)

cport
: control port # for CS4236 chip (PnP setup - 0x120,0x210,0xf00)

mpu_port

port # for MPU-401 UART (PnP setup - 0x300), -1 = disable

fm_port
: FM port # for CS4236 chip (PnP setup - 0x388), -1 = disable

irq
: IRQ # for CS4236 chip (5,7,9,11,12,15)

mpu_irq
: IRQ # for MPU-401 UART (9,11,12,15)

dma1
: first DMA # for CS4236 chip (0,1,3)

dma2
: second DMA # for CS4236 chip (0,1,3), -1 = disable

This module supports multiple cards. This module does not support autoprobe (if ISA PnP is not used) thus main port and control port must be specified!!! Other ports are optional.

The power-management is supported.

This module is aliased as snd-cs4232 since it provides the old snd-cs4232 functionality, too.

Module snd-cs4281

Module for Cirrus Logic CS4281 soundchip.

dual_codec
: Secondary codec ID (0 = disable, default)

This module supports multiple cards.

The power-management is supported.

Module snd-cs46xx

Module for PCI sound cards based on CS4610/CS4612/CS4614/CS4615/CS4622/ CS4624/CS4630/CS4280 PCI chips.

external_amp
: Force to enable external amplifier.

thinkpad
: Force to enable Thinkpad's CLKRUN control.

mmap_valid
: Support OSS mmap mode (default = 0).

This module supports multiple cards and autoprobe. Usually external amp and CLKRUN controls are detected automatically from PCI sub vendor/device ids. If they don't work, give the options above explicitly.

The power-management is supported.

Module snd-cs5530

Module for Cyrix/NatSemi Geode 5530 chip.

Module snd-cs5535audio

Module for multifunction CS5535 companion PCI device

The power-management is supported.

Module snd-ctxfi

Module for Creative Sound Blaster X-Fi boards (20k1 / 20k2 chips)

- Creative Sound Blaster X-Fi Titanium Fatal1ty Champion Series
- Creative Sound Blaster X-Fi Titanium Fatal1ty Professional Series
- Creative Sound Blaster X-Fi Titanium Professional Audio
- Creative Sound Blaster X-Fi Titanium
- Creative Sound Blaster X-Fi Elite Pro
- Creative Sound Blaster X-Fi Platinum
- Creative Sound Blaster X-Fi Fatal1ty
- Creative Sound Blaster X-Fi XtremeGamer
- Creative Sound Blaster X-Fi XtremeMusic

reference_rate
: reference sample rate, 44100 or 48000 (default)

multiple
: multiple to ref. sample rate, 1 or 2 (default)

subsystem
: override the PCI SSID for probing; the value consists of SSVID << 16 | SSDID. The default is zero, which means no override.

This module supports multiple cards.

Module snd-darla20

Module for Echoaudio Darla20

This module supports multiple cards. The driver requires the firmware loader support on kernel.

Module snd-darla24

Module for Echoaudio Darla24

This module supports multiple cards. The driver requires the firmware loader support on kernel.

Module snd-dt019x

Module for Diamond Technologies DT-019X / Avance Logic ALS-007 (PnP only)

This module supports multiple cards. This module is enabled only with ISA PnP support.

The power-management is supported.

Module snd-dummy

Module for the dummy sound card. This "card" doesn't do any output or input, but you may use this module for any application which requires a sound card (like RealPlayer).

pcm_devs
: Number of PCM devices assigned to each card (default = 1, up to 4)

pcm_substreams
: Number of PCM substreams assigned to each PCM (default = 8, up to 128)

hrtimer
: Use hrtimer (=1, default) or system timer (=0)

fake_buffer
: Fake buffer allocations (default = 1)

When multiple PCM devices are created, snd-dummy gives different behavior to each PCM device: * 0 = interleaved with mmap support * 1 = non-interleaved with mmap support * 2 = interleaved without mmap * 3 = non-interleaved without mmap

As default, snd-dummy drivers doesn't allocate the real buffers but either ignores read/write or mmap a single dummy page to all buffer pages, in order to save the resources. If your apps need the read/ written buffer data to be consistent, pass fake_buffer=0 option.

The power-management is supported.

Module snd-echo3g

Module for Echoaudio 3G cards (Gina3G/Layla3G)

This module supports multiple cards. The driver requires the firmware loader support on kernel.

Module snd-emu10k1

Module for EMU10K1/EMU10k2 based PCI sound cards.

- Sound Blaster Live!
- Sound Blaster PCI 512
- Emu APS (partially supported)
- Sound Blaster Audigy

extin
: bitmap of available external inputs for FX8010 (see bellow)

extout
: bitmap of available external outputs for FX8010 (see bellow)

seq_ports
: allocated sequencer ports (4 by default)

max_synth_voices
: limit of voices used for wavetable (64 by default)

max_buffer_size
: specifies the maximum size of wavetable/pcm buffers given in MB unit. Default value is 128.

enable_ir

enable IR

This module supports multiple cards and autoprobe.

Input & Output configurations [extin/extout] * Creative Card wo/Digital out [0x0003/0x1f03] * Creative Card w/Digital out [0x0003/0x1f0f] * Creative Card w/Digital CD in [0x000f/0x1f0f] * Creative Card wo/Digital out + LiveDrive [0x3fc3/0x1fc3] * Creative Card w/Digital out + LiveDrive [0x3fc3/0x1fcf] * Creative Card w/Digital CD in + LiveDrive [0x3fcf/0x1fcf] * Creative Card wo/Digital out + Digital I/O 2 [0x0fc3/0x1f0f] * Creative Card w/Digital out + Digital I/O 2 [0x0fc3/0x1f0f] * Creative Card w/Digital CD in + Digital I/O 2 [0x0fcf/0x1f0f] * Creative Card 5.1/w Digital out + LiveDrive [0x3fc3/0x1fff] * Creative Card 5.1 (c) 2003 [0x3fc3/0x7cff] * Creative Card all ins and outs [0x3fff/0x7fff]

The power-management is supported.

Module snd-emu10k1x

Module for Creative Emu10k1X (SB Live Dell OEM version)

This module supports multiple cards.

Module snd-ens1370

Module for Ensoniq AudioPCI ES1370 PCI sound cards.

- SoundBlaster PCI 64
- SoundBlaster PCI 128

joystick
: Enable joystick (default off)

This module supports multiple cards and autoprobe.

The power-management is supported.

Module snd-ens1371

Module for Ensoniq AudioPCI ES1371 PCI sound cards.

- SoundBlaster PCI 64
- SoundBlaster PCI 128
- SoundBlaster Vibra PCI

joystick_port
: port # for joystick (0x200,0x208,0x210,0x218), 0 = disable (default), 1 = auto-detect

This module supports multiple cards and autoprobe.

The power-management is supported.

Module snd-es1688

Module for ESS AudioDrive ES-1688 and ES-688 sound cards.

isapnp
: ISA PnP detection - 0 = disable, 1 = enable (default)

mpu_port
: port # for MPU-401 port (0x300,0x310,0x320,0x330), -1 = disable (default)

mpu_irq
: IRQ # for MPU-401 port (5,7,9,10)

fm_port
: port # for OPL3 (option; share the same port as default)

with `isapnp=0`, the following additional options are available:

port
: port # for ES-1688 chip (0x220,0x240,0x260)

irq
: IRQ # for ES-1688 chip (5,7,9,10)

dma8
: DMA # for ES-1688 chip (0,1,3)

This module supports multiple cards and autoprobe (without MPU-401 port) and PnP with the ES968 chip.

Module snd-es18xx

Module for ESS AudioDrive ES-18xx sound cards.

isapnp

ISA PnP detection - 0 = disable, 1 = enable (default)

with `isapnp=0`, the following options are available:

port
: port # for ES-18xx chip (0x220,0x240,0x260)

mpu_port
: port # for MPU-401 port (0x300,0x310,0x320,0x330), -1 = disable (default)

fm_port
: port # for FM (optional, not used)

irq
: IRQ # for ES-18xx chip (5,7,9,10)

dma1
: first DMA # for ES-18xx chip (0,1,3)

dma2
: first DMA # for ES-18xx chip (0,1,3)

This module supports multiple cards, ISA PnP and autoprobe (without MPU-401 port if native ISA PnP routines are not used). When `dma2` is equal with `dma1`, the driver works as half-duplex.

The power-management is supported.

Module snd-es1938

Module for sound cards based on ESS Solo-1 (ES1938,ES1946) chips.

This module supports multiple cards and autoprobe.

The power-management is supported.

Module snd-es1968

Module for sound cards based on ESS Maestro-1/2/2E (ES1968/ES1978) chips.

total_bufsize
: total buffer size in kB (1-4096kB)

pcm_substreams_p
: playback channels (1-8, default=2)

pcm_substreams_c
: capture channels (1-8, default=0)

clock
: clock (0 = auto-detection)

use_pm
: support the power-management (0 = off, 1 = on, 2 = auto (default))

enable_mpu
: enable MPU401 (0 = off, 1 = on, 2 = auto (default))

joystick
: enable joystick (default off)

This module supports multiple cards and autoprobe.

The power-management is supported.

Module snd-fm801

Module for ForteMedia FM801 based PCI sound cards.

tea575x_tuner
: Enable TEA575x tuner; 1 = MediaForte 256-PCS, 2 = MediaForte 256-PCPR, 3 = MediaForte 64-PCR High 16-bits are video (radio) device number + 1; example: 0x10002 (MediaForte 256-PCPR, device 1)

This module supports multiple cards and autoprobe.

The power-management is supported.

Module snd-gina20

Module for Echoaudio Gina20

This module supports multiple cards. The driver requires the firmware loader support on kernel.

Module snd-gina24

Module for Echoaudio Gina24

This module supports multiple cards. The driver requires the firmware loader support on kernel.

Module snd-gusclassic

Module for Gravis UltraSound Classic sound card.

port
: port # for GF1 chip (0x220,0x230,0x240,0x250,0x260)

irq
: IRQ # for GF1 chip (3,5,9,11,12,15)

dma1
: DMA # for GF1 chip (1,3,5,6,7)

dma2
: DMA # for GF1 chip (1,3,5,6,7,-1=disable)

joystick_dac
: 0 to 31, (0.59V-4.52V or 0.389V-2.98V)

voices
: GF1 voices limit (14-32)

pcm_voices
: reserved PCM voices

This module supports multiple cards and autoprobe.

Module snd-gusextreme

Module for Gravis UltraSound Extreme (Synergy ViperMax) sound card.

port
: port # for ES-1688 chip (0x220,0x230,0x240,0x250,0x260)

gf1_port
: port # for GF1 chip (0x210,0x220,0x230,0x240,0x250,0x260,0x270)

mpu_port
: port # for MPU-401 port (0x300,0x310,0x320,0x330), -1 = disable

irq
: IRQ # for ES-1688 chip (5,7,9,10)

gf1_irq
: IRQ # for GF1 chip (3,5,9,11,12,15)

mpu_irq
: IRQ # for MPU-401 port (5,7,9,10)

dma8
: DMA # for ES-1688 chip (0,1,3)

dma1
: DMA # for GF1 chip (1,3,5,6,7)

joystick_dac
: 0 to 31, (0.59V-4.52V or 0.389V-2.98V)

voices
: GF1 voices limit (14-32)

pcm_voices
: reserved PCM voices

This module supports multiple cards and autoprobe (without MPU-401 port).

Module snd-gusmax

Module for Gravis UltraSound MAX sound card.

port
: port # for GF1 chip (0x220,0x230,0x240,0x250,0x260)

irq
: IRQ # for GF1 chip (3,5,9,11,12,15)

dma1
: DMA # for GF1 chip (1,3,5,6,7)

dma2
: DMA # for GF1 chip (1,3,5,6,7,-1=disable)

joystick_dac
: 0 to 31, (0.59V-4.52V or 0.389V-2.98V)

voices
: GF1 voices limit (14-32)

pcm_voices
: reserved PCM voices

This module supports multiple cards and autoprobe.

Module snd-hda-intel

Module for Intel HD Audio (ICH6, ICH6M, ESB2, ICH7, ICH8, ICH9, ICH10, PCH, SCH), ATI SB450, SB600, R600, RS600, RS690, RS780, RV610, RV620, RV630, RV635, RV670, RV770, VIA VT8251/VT8237A, SIS966, ULI M5461

[Multiple options for each card instance]

model
: force the model name

position_fix
: Fix DMA pointer; -1 = system default: choose appropriate one per controller hardware, 0 = auto: falls back to LPIB when POSBUF doesn't work, 1 = use LPIB, 2 = POSBUF: use position buffer, 3 = VIACOMBO: VIA-specific workaround for capture, 4 = COMBO: use LPIB for playback, auto for capture stream 5 = SKL+: apply the delay calculation available on recent Intel chips 6 = FIFO: correct the position with the fixed FIFO size, for recent AMD chips

probe_mask
: Bitmask to probe codecs (default = -1, meaning all slots); When the bit 8 (0x100) is set, the lower 8 bits are used as the "fixed" codec slots; i.e. the driver probes the slots regardless what hardware reports back

probe_only
: Only probing and no codec initialization (default=off); Useful to check the initial codec status for debugging

bdl_pos_adj
: Specifies the DMA IRQ timing delay in samples. Passing -1 will make the driver to choose the appropriate value based on the controller chip.

patch
: Specifies the early "patch" files to modify the HD-audio setup before initializing the codecs. This option is available only when `CONFIG_SND_HDA_PATCH_LOADER=y` is set. See hd-audio/notes.rst for details.

beep_mode
: Selects the beep registration mode (0=off, 1=on); default value is set via `CONFIG_SND_HDA_INPUT_BEEP_MODE` kconfig.

[Single (global) options]

single_cmd
: Use single immediate commands to communicate with codecs (for debugging only)

enable_msi
: Enable Message Signaled Interrupt (MSI) (default = off)

power_save
: Automatic power-saving timeout (in second, 0 = disable)

power_save_controller
: Reset HD-audio controller in power-saving mode (default = on)

align_buffer_size
: Force rounding of buffer/period sizes to multiples of 128 bytes. This is more efficient in terms of memory access but isn't required by the HDA spec and prevents users from specifying exact period/buffer sizes. (default = on)

snoop
: Enable/disable snooping (default = on)

This module supports multiple cards and autoprobe.

See hd-audio/notes.rst for more details about HD-audio driver.

Each codec may have a model table for different configurations. If your machine isn't listed there, the default (usually minimal) configuration is set up. You can pass `model=<name>` option to specify a certain model in such a case. There are different models depending on the codec chip. The list of available models is found in hd-audio/models.rst.

The model name `generic` is treated as a special case. When this model is given, the driver uses the generic codec parser without "codec-patch". It's sometimes good for testing and debugging.

The model option can be used also for aliasing to another PCI or codec SSID. When it's passed in the form of `model=XXXX:YYYY` where XXXX and YYYY are the sub-vendor and sub-device IDs in hex numbers, respectively, the driver will refer to that SSID as a reference to the quirk table.

If the default configuration doesn't work and one of the above matches with your device, report it together with alsa-info.sh output (with `--no-upload` option) to kernel bugzilla or alsa-devel ML (see the section Links and Addresses).

`power_save` and `power_save_controller` options are for power-saving mode. See powersave.rst for details.

Note 2: If you get click noises on output, try the module option `position_fix=1` or `2`. `position_fix=1` will use the SD_LPIB register value without FIFO size correction as the current DMA pointer. `position_fix=2` will make the driver to use the position buffer instead of reading SD_LPIB register. (Usually SD_LPIB register is more accurate than the position buffer.)

`position_fix=3` is specific to VIA devices. The position of the capture stream is checked from both LPIB and POSBUF values. `position_fix=4` is a combination mode, using LPIB for playback and POSBUF for capture.

NB: If you get many `azx_get_response timeout` messages at loading, it's likely a problem of interrupts (e.g. ACPI irq routing). Try to boot with options like `pci=noacpi`. Also, you can try `single_cmd=1` module option. This will switch the communication method between HDA controller and codecs to the single immediate commands instead of CORB/RIRB. Basically, the single command mode is provided only for BIOS, and you won't get unsolicited events, too. But, at least, this works independently from the irq. Remember this is a last resort, and should be avoided as much as possible...

MORE NOTES ON `azx_get_response timeout` PROBLEMS: On some hardware, you may need to add a proper probe_mask option to avoid the `azx_get_response timeout` problem above, instead. This occurs when the access to non-existing or non-working codec slot (likely a modem one) causes a stall of the communication via HD-audio bus. You can see which codec slots are probed by enabling `CONFIG_SND_DEBUG_VERBOSE`, or simply from the file name of the codec proc files. Then limit the slots to probe by probe_mask option. For example, `probe_mask=1` means to probe only the first slot, and `probe_mask=4` means only the third slot.

The power-management is supported.

Module snd-hdsp

Module for RME Hammerfall DSP audio interface(s)

This module supports multiple cards.

Note: The firmware data can be automatically loaded via hotplug when `CONFIG_FW_LOADER` is set. Otherwise, you need to load the firmware via hdsploader utility included in alsa-tools package. The firmware data is found in alsa-firmware package.

Note: snd-page-alloc module does the job which snd-hammerfall-mem module did formerly. It will allocate the buffers in advance when any HDSP cards are found. To make the buffer allocation sure, load snd-page-alloc module in the early stage of boot sequence. See Early Buffer Allocation section.

Module snd-hdspm

Module for RME HDSP MADI board.

precise_ptr
: Enable precise pointer, or disable.

line_outs_monitor
: Send playback streams to analog outs by default.

enable_monitor
: Enable Analog Out on Channel 63/64 by default.

See hdspm.rst for details.

Module snd-ice1712

Module for Envy24 (ICE1712) based PCI sound cards.

- MidiMan M Audio Delta 1010
- MidiMan M Audio Delta 1010LT
- MidiMan M Audio Delta DiO 2496
- MidiMan M Audio Delta 66
- MidiMan M Audio Delta 44
- MidiMan M Audio Delta 410
- MidiMan M Audio Audiophile 2496
- TerraTec EWS 88MT
- TerraTec EWS 88D
- TerraTec EWX 24/96
- TerraTec DMX 6Fire
- TerraTec Phase 88
- Hoontech SoundTrack DSP 24
- Hoontech SoundTrack DSP 24 Value
- Hoontech SoundTrack DSP 24 Media 7.1
- Event Electronics, EZ8
- Digigram VX442
- Lionstracs, Mediastaton
- Terrasoniq TS 88

model
: Use the given board model, one of the following: delta1010, dio2496, delta66, delta44, audiophile, delta410, delta1010lt, vx442, ewx2496, ews88mt, ews88mt_new, ews88d, dmx6fire, dsp24, dsp24_value, dsp24_71, ez8, phase88, mediastation

omni
: Omni I/O support for MidiMan M-Audio Delta44/66

cs8427_timeout
: reset timeout for the CS8427 chip (S/PDIF transceiver) in msec resolution, default value is 500 (0.5 sec)

This module supports multiple cards and autoprobe. Note: The consumer part is not used with all Envy24 based cards (for example in the MidiMan Delta siree).

Note: The supported board is detected by reading EEPROM or PCI SSID (if EEPROM isn't available). You can override the model by passing `model` module option in case that the driver isn't configured properly or you want to try another type for testing.

Module snd-ice1724

Module for Envy24HT (VT/ICE1724), Envy24PT (VT1720) based PCI sound cards.

- MidiMan M Audio Revolution 5.1
- MidiMan M Audio Revolution 7.1
- MidiMan M Audio Audiophile 192
- AMP Ltd AUDIO2000
- TerraTec Aureon 5.1 Sky
- TerraTec Aureon 7.1 Space
- TerraTec Aureon 7.1 Universe
- TerraTec Phase 22
- TerraTec Phase 28
- AudioTrak Prodigy 7.1
- AudioTrak Prodigy 7.1 LT
- AudioTrak Prodigy 7.1 XT
- AudioTrak Prodigy 7.1 HIFI
- AudioTrak Prodigy 7.1 HD2
- AudioTrak Prodigy 192
- Pontis MS300
- Albatron K8X800 Pro II
- Chaintech ZNF3-150
- Chaintech ZNF3-250
- Chaintech 9CJS
- Chaintech AV-710
- Shuttle SN25P
- Onkyo SE-90PCI
- Onkyo SE-200PCI
- ESI Juli@
- ESI Maya44
- Hercules Fortissimo IV
- EGO-SYS WaveTerminal 192M

model
: Use the given board model, one of the following: revo51, revo71, amp2000, prodigy71, prodigy71lt, prodigy71xt, prodigy71hifi, prodigyhd2, prodigy192, juli, aureon51, aureon71, universe, ap192, k8x800, phase22, phase28, ms300, av710, se200pci, se90pci, fortissimo4, sn25p, WT192M, maya44

This module supports multiple cards and autoprobe.

Note: The supported board is detected by reading EEPROM or PCI SSID (if EEPROM isn't available). You can override the model by passing `model` module option in case that the driver isn't configured properly or you want to try another type for testing.

Module snd-indigo

Module for Echoaudio Indigo

This module supports multiple cards. The driver requires the firmware loader support on kernel.

Module snd-indigodj

Module for Echoaudio Indigo DJ

This module supports multiple cards. The driver requires the firmware loader support on kernel.

Module snd-indigoio

Module for Echoaudio Indigo IO

This module supports multiple cards. The driver requires the firmware loader support on kernel.

Module snd-intel8x0

Module for AC'97 motherboards from Intel and compatibles.

- Intel i810/810E, i815, i820, i830, i84x, MX440 ICH5, ICH6, ICH7, 6300ESB, ESB2
- SiS 7012 (SiS 735)
- NVidia NForce, NForce2, NForce3, MCP04, CK804 CK8, CK8S, MCP501

- AMD AMD768, AMD8111
- ALi m5455

ac97_clock
: AC'97 codec clock base (0 = auto-detect)

ac97_quirk
: AC'97 workaround for strange hardware; See AC97 Quirk Option section below.

buggy_irq
: Enable workaround for buggy interrupts on some motherboards (default yes on nForce chips, otherwise off)

buggy_semaphore
: Enable workaround for hardware with buggy semaphores (e.g. on some ASUS laptops) (default off)

spdif_aclink
: Use S/PDIF over AC-link instead of direct connection from the controller chip (0 = off, 1 = on, -1 = default)

This module supports one chip and autoprobe.

Note: the latest driver supports auto-detection of chip clock. if you still encounter too fast playback, specify the clock explicitly via the module option `ac97_clock=41194`.

Joystick/MIDI ports are not supported by this driver. If your motherboard has these devices, use the ns558 or snd-mpu401 modules, respectively.

The power-management is supported.

Module snd-intel8x0m

Module for Intel ICH (i8x0) chipset MC97 modems.

- Intel i810/810E, i815, i820, i830, i84x, MX440 ICH5, ICH6, ICH7
- SiS 7013 (SiS 735)
- NVidia NForce, NForce2, NForce2s, NForce3
- AMD AMD8111
- ALi m5455

ac97_clock
: AC'97 codec clock base (0 = auto-detect)

This module supports one card and autoprobe.

Note: The default index value of this module is -2, i.e. the first slot is excluded.

The power-management is supported.

Module snd-interwave

Module for Gravis UltraSound PnP, Dynasonic 3-D/Pro, STB Sound Rage 32 and other sound cards based on AMD InterWave (tm) chip.

joystick_dac
: 0 to 31, (0.59V-4.52V or 0.389V-2.98V)

midi
: 1 = MIDI UART enable, 0 = MIDI UART disable (default)

pcm_voices
: reserved PCM voices for the synthesizer (default 2)

effect
: 1 = InterWave effects enable (default 0); requires 8 voices

isapnp
: ISA PnP detection - 0 = disable, 1 = enable (default)

with `isapnp=0`, the following options are available:

port
: port # for InterWave chip (0x210,0x220,0x230,0x240,0x250,0x260)

irq
: IRQ # for InterWave chip (3,5,9,11,12,15)

dma1
: DMA # for InterWave chip (0,1,3,5,6,7)

dma2
: DMA # for InterWave chip (0,1,3,5,6,7,-1=disable)

This module supports multiple cards, autoprobe and ISA PnP.

Module snd-interwave-stb

Module for UltraSound 32-Pro (sound card from STB used by Compaq) and other sound cards based on AMD InterWave (tm) chip with TEA6330T circuit for extended control of bass, treble and master volume.

joystick_dac
: 0 to 31, (0.59V-4.52V or 0.389V-2.98V)

midi
: 1 = MIDI UART enable, 0 = MIDI UART disable (default)

pcm_voices
: reserved PCM voices for the synthesizer (default 2)

effect
: 1 = InterWave effects enable (default 0); requires 8 voices

isapnp
: ISA PnP detection - 0 = disable, 1 = enable (default)

with `isapnp=0`, the following options are available:

port
: port # for InterWave chip (0x210,0x220,0x230,0x240,0x250,0x260)

port_tc
: tone control (i2c bus) port # for TEA6330T chip (0x350,0x360,0x370,0x380)

irq
: IRQ # for InterWave chip (3,5,9,11,12,15)

dma1
: DMA # for InterWave chip (0,1,3,5,6,7)

dma2
: DMA # for InterWave chip (0,1,3,5,6,7,-1=disable)

This module supports multiple cards, autoprobe and ISA PnP.

Module snd-jazz16

Module for Media Vision Jazz16 chipset. The chipset consists of 3 chips: MVD1216 + MVA416 + MVA514.

port
: port # for SB DSP chip (0x210,0x220,0x230,0x240,0x250,0x260)

irq
: IRQ # for SB DSP chip (3,5,7,9,10,15)

dma8
: DMA # for SB DSP chip (1,3)

dma16
: DMA # for SB DSP chip (5,7)

mpu_port
: MPU-401 port # (0x300,0x310,0x320,0x330)

mpu_irq
: MPU-401 irq # (2,3,5,7)

This module supports multiple cards.

Module snd-korg1212

Module for Korg 1212 IO PCI card

This module supports multiple cards.

Module snd-layla20

Module for Echoaudio Layla20

This module supports multiple cards. The driver requires the firmware loader support on kernel.

Module snd-layla24

Module for Echoaudio Layla24

This module supports multiple cards. The driver requires the firmware loader support on kernel.

Module snd-lola

Module for Digigram Lola PCI-e boards

This module supports multiple cards.

Module snd-lx6464es

Module for Digigram LX6464ES boards

This module supports multiple cards.

Module snd-maestro3

Module for Allegro/Maestro3 chips

external_amp
: enable external amp (enabled by default)

amp_gpio
: GPIO pin number for external amp (0-15) or -1 for default pin (8 for allegro, 1 for others)

This module supports autoprobe and multiple chips.

Note: the binding of amplifier is dependent on hardware. If there is no sound even though all channels are unmuted, try to specify other gpio connection via amp_gpio option. For example, a Panasonic notebook might need `amp_gpio=0x0d` option.

The power-management is supported.

Module snd-mia

Module for Echoaudio Mia

This module supports multiple cards. The driver requires the firmware loader support on kernel.

Module snd-miro

Module for Miro soundcards: miroSOUND PCM 1 pro, miroSOUND PCM 12, miroSOUND PCM 20 Radio.

port
: Port # (0x530,0x604,0xe80,0xf40)

irq
: IRQ # (5,7,9,10,11)

dma1
: 1st dma # (0,1,3)

dma2
: 2nd dma # (0,1)

mpu_port
: MPU-401 port # (0x300,0x310,0x320,0x330)

mpu_irq
: MPU-401 irq # (5,7,9,10)

fm_port
: FM Port # (0x388)

wss
: enable WSS mode

ide
: enable onboard ide support

Module snd-mixart

Module for Digigram miXart8 sound cards.

This module supports multiple cards. Note: One miXart8 board will be represented as 4 alsa cards. See Documentation/sound/cards/mixart.rst for details.

When the driver is compiled as a module and the hotplug firmware is supported, the firmware data is loaded via hotplug automatically. Install the necessary firmware files in alsa-firmware package. When no hotplug fw loader is available, you need to load the firmware via mixartloader utility in alsa-tools package.

Module snd-mona

Module for Echoaudio Mona

This module supports multiple cards. The driver requires the firmware loader support on kernel.

Module snd-mpu401

Module for MPU-401 UART devices.

port
: port number or -1 (disable)

irq
: IRQ number or -1 (disable)

pnp
: PnP detection - 0 = disable, 1 = enable (default)

This module supports multiple devices and PnP.

Module snd-msnd-classic

Module for Turtle Beach MultiSound Classic, Tahiti or Monterey soundcards.

io
: Port # for msnd-classic card

irq
: IRQ # for msnd-classic card

mem
: Memory address (0xb0000, 0xc8000, 0xd0000, 0xd8000, 0xe0000 or 0xe8000)

write_ndelay
: enable write ndelay (default = 1)

calibrate_signal
: calibrate signal (default = 0)

isapnp
: ISA PnP detection - 0 = disable, 1 = enable (default)

digital
: Digital daughterboard present (default = 0)

cfg
: Config port (0x250, 0x260 or 0x270) default = PnP

reset
: Reset all devices

mpu_io
: MPU401 I/O port

mpu_irq
: MPU401 irq#

ide_io0
: IDE port #0

ide_io1
: IDE port #1

ide_irq
: IDE irq#

joystick_io
: Joystick I/O port

The driver requires firmware files `turtlebeach/msndinit.bin` and `turtlebeach/msndperm.bin` in the proper firmware directory.

See Documentation/sound/cards/multisound.sh for important information about this driver. Note that it has been discontinued, but the Voyetra Turtle Beach knowledge base entry for it is still available at https://www.turtlebeach.com

Module snd-msnd-pinnacle

Module for Turtle Beach MultiSound Pinnacle/Fiji soundcards.

io
: Port # for pinnacle/fiji card

irq
: IRQ # for pinnalce/fiji card

mem
: Memory address (0xb0000, 0xc8000, 0xd0000, 0xd8000, 0xe0000 or 0xe8000)

write_ndelay
: enable write ndelay (default = 1)

calibrate_signal
: calibrate signal (default = 0)

isapnp
: ISA PnP detection - 0 = disable, 1 = enable (default)

The driver requires firmware files `turtlebeach/pndspini.bin` and `turtlebeach/pndsperm.bin` in the proper firmware directory.

Module snd-mtpav

Module for MOTU MidiTimePiece AV multiport MIDI (on the parallel port).

port
: I/O port # for MTPAV (0x378,0x278, default=0x378)

irq

IRQ # for MTPAV (7,5, default=7)

hwports
: number of supported hardware ports, default=8.

Module supports only 1 card. This module has no enable option.

Module snd-mts64

Module for Ego Systems (ESI) Miditerminal 4140

This module supports multiple devices. Requires parport (`CONFIG_PARPORT`).

Module snd-nm256

Module for NeoMagic NM256AV/ZX chips

playback_bufsize
: max playback frame size in kB (4-128kB)

capture_bufsize
: max capture frame size in kB (4-128kB)

force_ac97
: 0 or 1 (disabled by default)

buffer_top
: specify buffer top address

use_cache
: 0 or 1 (disabled by default)

vaio_hack
: alias buffer_top=0x25a800

reset_workaround
: enable AC97 RESET workaround for some laptops

reset_workaround2
: enable extended AC97 RESET workaround for some other laptops

This module supports one chip and autoprobe.

The power-management is supported.

Note: on some notebooks the buffer address cannot be detected automatically, or causes hang-up during initialization. In such a case, specify the buffer top address explicitly via the buffer_top option. For example, Sony F250: buffer_top=0x25a800 Sony F270: buffer_top=0x272800 The driver supports only ac97 codec. It's possible to force to initialize/use ac97 although it's not detected. In such a case, use `force_ac97=1` option - but *NO* guarantee whether it works!

Note: The NM256 chip can be linked internally with non-AC97 codecs. This driver supports only the AC97 codec, and won't work with machines with other (most likely CS423x or OPL3SAx) chips, even though the device is detected in lspci. In such a case, try other drivers, e.g. snd-cs4232 or snd-opl3sa2. Some has ISA-PnP but some doesn't have ISA PnP. You'll need to specify `isapnp=0` and proper hardware parameters in the case without ISA PnP.

Note: some laptops need a workaround for AC97 RESET. For the known hardware like Dell Latitude LS and Sony PCG-F305, this workaround is enabled automatically. For other laptops with a hard freeze, you can try `reset_workaround=1` option.

Note: Dell Latitude CSx laptops have another problem regarding AC97 RESET. On these laptops, reset_workaround2 option is turned on as default. This option is worth to try if the previous reset_workaround option doesn't help.

Note: This driver is really crappy. It's a porting from the OSS driver, which is a result of black-magic reverse engineering. The detection of codec will fail if the driver is loaded *after* X-server as described above. You might be able to force to load the module, but it may result in hang-up. Hence, make sure that you load this module *before* X if you encounter this kind of problem.

Module snd-opl3sa2

Module for Yamaha OPL3-SA2/SA3 sound cards.

isapnp
: ISA PnP detection - 0 = disable, 1 = enable (default)

with `isapnp=0`, the following options are available:

port
: control port # for OPL3-SA chip (0x370)

sb_port
: SB port # for OPL3-SA chip (0x220,0x240)

wss_port
: WSS port # for OPL3-SA chip (0x530,0xe80,0xf40,0x604)

midi_port
: port # for MPU-401 UART (0x300,0x330), -1 = disable

fm_port
: FM port # for OPL3-SA chip (0x388), -1 = disable

irq
: IRQ # for OPL3-SA chip (5,7,9,10)

dma1
: first DMA # for Yamaha OPL3-SA chip (0,1,3)

dma2
: second DMA # for Yamaha OPL3-SA chip (0,1,3), -1 = disable

This module supports multiple cards and ISA PnP. It does not support autoprobe (if ISA PnP is not used) thus all ports must be specified!!!

The power-management is supported.

Module snd-opti92x-ad1848

Module for sound cards based on OPTi 82c92x and Analog Devices AD1848 chips. Module works with OAK Mozart cards as well.

isapnp
: ISA PnP detection - 0 = disable, 1 = enable (default)

with `isapnp=0`, the following options are available:

port
: port # for WSS chip (0x530,0xe80,0xf40,0x604)

mpu_port
: port # for MPU-401 UART (0x300,0x310,0x320,0x330)

fm_port
: port # for OPL3 device (0x388)

irq
: IRQ # for WSS chip (5,7,9,10,11)

mpu_irq
: IRQ # for MPU-401 UART (5,7,9,10)

dma1
: first DMA # for WSS chip (0,1,3)

This module supports only one card, autoprobe and PnP.

Module snd-opti92x-cs4231

Module for sound cards based on OPTi 82c92x and Crystal CS4231 chips.

isapnp
: ISA PnP detection - 0 = disable, 1 = enable (default)

with `isapnp=0`, the following options are available:

port
: port # for WSS chip (0x530,0xe80,0xf40,0x604)

mpu_port
: port # for MPU-401 UART (0x300,0x310,0x320,0x330)

fm_port
: port # for OPL3 device (0x388)

irq
: IRQ # for WSS chip (5,7,9,10,11)

mpu_irq
: IRQ # for MPU-401 UART (5,7,9,10)

dma1
: first DMA # for WSS chip (0,1,3)

dma2
: second DMA # for WSS chip (0,1,3)

This module supports only one card, autoprobe and PnP.

Module snd-opti93x

Module for sound cards based on OPTi 82c93x chips.

isapnp
: ISA PnP detection - 0 = disable, 1 = enable (default)

with `isapnp=0`, the following options are available:

port
: port # for WSS chip (0x530,0xe80,0xf40,0x604)

mpu_port
: port # for MPU-401 UART (0x300,0x310,0x320,0x330)

fm_port
: port # for OPL3 device (0x388)

irq
: IRQ # for WSS chip (5,7,9,10,11)

mpu_irq
: IRQ # for MPU-401 UART (5,7,9,10)

dma1
: first DMA # for WSS chip (0,1,3)

dma2
: second DMA # for WSS chip (0,1,3)

This module supports only one card, autoprobe and PnP.

Module snd-oxygen

Module for sound cards based on the C-Media CMI8786/8787/8788 chip:

- Asound A-8788
- Asus Xonar DG/DGX
- AuzenTech X-Meridian
- AuzenTech X-Meridian 2G
- Bgears b-Enspirer
- Club3D Theatron DTS
- HT-Omega Claro (plus)
- HT-Omega Claro halo (XT)
- Kuroutoshikou CMI8787-HG2PCI
- Razer Barracuda AC-1
- Sondigo Inferno
- TempoTec HiFier Fantasia
- TempoTec HiFier Serenade

This module supports autoprobe and multiple cards.

Module snd-pcsp

Module for internal PC-Speaker.

nopcm
: Disable PC-Speaker PCM sound. Only beeps remain.

nforce_wa
: enable NForce chipset workaround. Expect bad sound.

This module supports system beeps, some kind of PCM playback and even a few mixer controls.

Module snd-pcxhr

Module for Digigram PCXHR boards

This module supports multiple cards.

Module snd-portman2x4

Module for Midiman Portman 2x4 parallel port MIDI interface

This module supports multiple cards.

Module snd-powermac (on ppc only)

Module for PowerMac, iMac and iBook on-board soundchips

enable_beep
: enable beep using PCM (enabled as default)

Module supports autoprobe a chip.

Note: the driver may have problems regarding endianness.

The power-management is supported.

Module snd-pxa2xx-ac97 (on arm only)

Module for AC97 driver for the Intel PXA2xx chip

For ARM architecture only.

The power-management is supported.

Module snd-riptide

Module for Conexant Riptide chip

joystick_port
: Joystick port # (default: 0x200)

mpu_port
: MPU401 port # (default: 0x330)

opl3_port
: OPL3 port # (default: 0x388)

This module supports multiple cards. The driver requires the firmware loader support on kernel. You need to install the firmware file `riptide.hex` to the standard firmware path (e.g. /lib/firmware).

Module snd-rme32

Module for RME Digi32, Digi32 Pro and Digi32/8 (Sek'd Prodif32, Prodif96 and Prodif Gold) sound cards.

This module supports multiple cards.

Module snd-rme96

Module for RME Digi96, Digi96/8 and Digi96/8 PRO/PAD/PST sound cards.

This module supports multiple cards.

Module snd-rme9652

Module for RME Digi9652 (Hammerfall, Hammerfall-Light) sound cards.

precise_ptr
: Enable precise pointer (doesn't work reliably). (default = 0)

This module supports multiple cards.

Note: snd-page-alloc module does the job which snd-hammerfall-mem module did formerly. It will allocate the buffers in advance when any RME9652 cards are found. To make the buffer allocation sure, load snd-page-alloc module in the early stage of boot sequence. See Early Buffer Allocation section.

Module snd-sa11xx-uda1341 (on arm only)

Module for Philips UDA1341TS on Compaq iPAQ H3600 sound card.

Module supports only one card. Module has no enable and index options.

The power-management is supported.

Module snd-sb8

Module for 8-bit SoundBlaster cards: SoundBlaster 1.0, SoundBlaster 2.0, SoundBlaster Pro

port
: port # for SB DSP chip (0x220,0x240,0x260)

irq
: IRQ # for SB DSP chip (5,7,9,10)

dma8
: DMA # for SB DSP chip (1,3)

This module supports multiple cards and autoprobe.

The power-management is supported.

Module snd-sb16 and snd-sbawe

Module for 16-bit SoundBlaster cards: SoundBlaster 16 (PnP), SoundBlaster AWE 32 (PnP), SoundBlaster AWE 64 PnP

mic_agc
: Mic Auto-Gain-Control - 0 = disable, 1 = enable (default)

csp
: ASP/CSP chip support - 0 = disable (default), 1 = enable

isapnp

ISA PnP detection - 0 = disable, 1 = enable (default)

with isapnp=0, the following options are available:

port
: port # for SB DSP 4.x chip (0x220,0x240,0x260)

mpu_port
: port # for MPU-401 UART (0x300,0x330), -1 = disable

awe_port
: base port # for EMU8000 synthesizer (0x620,0x640,0x660) (snd-sbawe module only)

irq
: IRQ # for SB DSP 4.x chip (5,7,9,10)

dma8
: 8-bit DMA # for SB DSP 4.x chip (0,1,3)

dma16
: 16-bit DMA # for SB DSP 4.x chip (5,6,7)

This module supports multiple cards, autoprobe and ISA PnP.

Note: To use Vibra16X cards in 16-bit half duplex mode, you must disable 16bit DMA with dma16 = -1 module parameter. Also, all Sound Blaster 16 type cards can operate in 16-bit half duplex mode through 8-bit DMA channel by disabling their 16-bit DMA channel.

The power-management is supported.

Module snd-sc6000

Module for Gallant SC-6000 soundcard and later models: SC-6600 and SC-7000.

port
: Port # (0x220 or 0x240)

mss_port
: MSS Port # (0x530 or 0xe80)

irq
: IRQ # (5,7,9,10,11)

mpu_irq
: MPU-401 IRQ # (5,7,9,10) ,0 - no MPU-401 irq

dma
: DMA # (1,3,0)

joystick
: Enable gameport - 0 = disable (default), 1 = enable

This module supports multiple cards.

This card is also known as Audio Excel DSP 16 or Zoltrix AV302.

Module snd-sscape

Module for ENSONIQ SoundScape cards.

port
: Port # (PnP setup)

wss_port
: WSS Port # (PnP setup)

irq
: IRQ # (PnP setup)

mpu_irq
: MPU-401 IRQ # (PnP setup)

dma
: DMA # (PnP setup)

dma2
: 2nd DMA # (PnP setup, -1 to disable)

joystick
: Enable gameport - 0 = disable (default), 1 = enable

This module supports multiple cards.

The driver requires the firmware loader support on kernel.

Module snd-sun-amd7930 (on sparc only)

Module for AMD7930 sound chips found on Sparcs.

This module supports multiple cards.

Module snd-sun-cs4231 (on sparc only)

Module for CS4231 sound chips found on Sparcs.

This module supports multiple cards.

Module snd-sun-dbri (on sparc only)

Module for DBRI sound chips found on Sparcs.

This module supports multiple cards.

Module snd-wavefront

Module for Turtle Beach Maui, Tropez and Tropez+ sound cards.

use_cs4232_midi
: Use CS4232 MPU-401 interface (inaccessibly located inside your computer)

isapnp
: ISA PnP detection - 0 = disable, 1 = enable (default)

with isapnp=0, the following options are available:

cs4232_pcm_port
: Port # for CS4232 PCM interface.

cs4232_pcm_irq
: IRQ # for CS4232 PCM interface (5,7,9,11,12,15).

cs4232_mpu_port
: Port # for CS4232 MPU-401 interface.

cs4232_mpu_irq
: IRQ # for CS4232 MPU-401 interface (9,11,12,15).

ics2115_port
: Port # for ICS2115

ics2115_irq
: IRQ # for ICS2115

fm_port
: FM OPL-3 Port #

dma1
: DMA1 # for CS4232 PCM interface.

dma2
: DMA2 # for CS4232 PCM interface.

The below are options for wavefront_synth features:

wf_raw
: Assume that we need to boot the OS (default:no); If yes, then during driver loading, the state of the board is ignored, and we reset the board and load the firmware anyway.

fx_raw
: Assume that the FX process needs help (default:yes); If false, we'll leave the FX processor in whatever state it is when the driver is loaded. The default is to download the microprogram and associated coefficients to set it up for "default" operation, whatever that means.

debug_default
: Debug parameters for card initialization

wait_usecs
: How long to wait without sleeping, usecs (default:150); This magic number seems to give pretty optimal throughput based on my limited experimentation. If you want to play around with it and find a better value, be my guest. Remember, the idea is to get a number that causes us to just busy wait for as many WaveFront commands as possible, without coming up with a number so large that we hog the whole CPU. Specifically, with this number, out of about 134,000 status waits, only about 250 result in a sleep.

sleep_interval
: How long to sleep when waiting for reply (default: 100)

sleep_tries
: How many times to try sleeping during a wait (default: 50)

ospath
: Pathname to processed ICS2115 OS firmware (default:wavefront.os); The path name of the ISC2115 OS firmware. In the recent version, it's handled via firmware loader framework, so it must be installed in the proper path, typically, /lib/firmware.

reset_time
: How long to wait for a reset to take effect (default:2)

ramcheck_time

How many seconds to wait for the RAM test (default:20)

osrun_time
: How many seconds to wait for the ICS2115 OS (default:10)

This module supports multiple cards and ISA PnP.

Note: the firmware file `wavefront.os` was located in the earlier version in /etc. Now it's loaded via firmware loader, and must be in the proper firmware path, such as /lib/firmware. Copy (or symlink) the file appropriately if you get an error regarding firmware downloading after upgrading the kernel.

Module snd-sonicvibes

Module for S3 SonicVibes PCI sound cards. * PINE Schubert 32 PCI

reverb
: Reverb Enable - 1 = enable, 0 = disable (default); SoundCard must have onboard SRAM for this.

mge
: Mic Gain Enable - 1 = enable, 0 = disable (default)

This module supports multiple cards and autoprobe.

Module snd-serial-u16550

Module for UART16550A serial MIDI ports.

port
: port # for UART16550A chip

irq
: IRQ # for UART16550A chip, -1 = poll mode

speed
: speed in bauds (9600,19200,38400,57600,115200) 38400 = default

base
: base for divisor in bauds (57600,115200,230400,460800) 115200 = default

outs
: number of MIDI ports in a serial port (1-4) 1 = default

adaptor
: Type of adaptor.
  0 = Soundcanvas, 1 = MS-124T, 2 = MS-124W S/A, 3 = MS-124W M/B, 4 = Generic

This module supports multiple cards. This module does not support autoprobe thus the main port must be specified!!! Other options are optional.

Module snd-trident

Module for Trident 4DWave DX/NX sound cards. * Best Union Miss Melody 4DWave PCI * HIS 4DWave PCI * Warpspeed ONSpeed 4DWave PCI * AzTech PCI 64-Q3D * Addonics SV 750 * CHIC True Sound 4Dwave * Shark Predator4D-PCI * Jaton SonicWave 4D * SiS SI7018 PCI Audio * Hoontech SoundTrack Digital 4DWave NX

pcm_channels
: max channels (voices) reserved for PCM

wavetable_size
: max wavetable size in kB (4-?kb)

This module supports multiple cards and autoprobe.

The power-management is supported.

Module snd-ua101

Module for the Edirol UA-101/UA-1000 audio/MIDI interfaces.

This module supports multiple devices, autoprobe and hotplugging.

Module snd-usb-audio

Module for USB audio and USB MIDI devices.

vid
: Vendor ID for the device (optional)

pid
: Product ID for the device (optional)

nrpacks

Max. number of packets per URB (default: 8)

device_setup

Device specific magic number (optional); Influence depends on the device Default: 0x0000

ignore_ctl_error

Ignore any USB-controller regarding mixer interface (default: no)

autoclock

Enable auto-clock selection for UAC2 devices (default: yes)

quirk_alias

Quirk alias list, pass strings like `0123abcd:5678beef`, which applies the existing quirk for the device 5678:beef to a new device 0123:abcd.

implicit_fb

Apply the generic implicit feedback sync mode. When this is set and the playback stream sync mode is ASYNC, the driver tries to tie an adjacent ASYNC capture stream as the implicit feedback source.

use_vmalloc

Use vmalloc() for allocations of the PCM buffers (default: yes). For architectures with non-coherent memory like ARM or MIPS, the mmap access may give inconsistent results with vmalloc'ed buffers. If mmap is used on such architectures, turn off this option, so that the DMA-coherent buffers are allocated and used instead.

delayed_register

The option is needed for devices that have multiple streams defined in multiple USB interfaces. The driver may invoke registrations multiple times (once per interface) and this may lead to the insufficient device enumeration. This option receives an array of strings, and you can pass ID:INTERFACE like `0123abcd:4` for performing the delayed registration to the given device. In this example, when a USB device 0123:abcd is probed, the driver waits the registration until the USB interface 4 gets probed. The driver prints a message like "Found post-registration device assignment: 1234abcd:04" for such a device, so that user can notice the need.

quirk_flags

Contains the bit flags for various device specific workarounds. Applied to the corresponding card index.

- bit 0: Skip reading sample rate for devices
- bit 1: Create Media Controller API entries
- bit 2: Allow alignment on audio sub-slot at transfer
- bit 3: Add length specifier to transfers
- bit 4: Start playback stream at first in implement feedback mode
- bit 5: Skip clock selector setup
- bit 6: Ignore errors from clock source search
- bit 7: Indicates ITF-USB DSD based DACs
- bit 8: Add a delay of 20ms at each control message handling
- bit 9: Add a delay of 1-2ms at each control message handling
- bit 10: Add a delay of 5-6ms at each control message handling
- bit 11: Add a delay of 50ms at each interface setup
- bit 12: Perform sample rate validations at probe
- bit 13: Disable runtime PM autosuspend
- bit 14: Ignore errors for mixer access
- bit 15: Support generic DSD raw U32_BE format
- bit 16: Set up the interface at first like UAC1

This module supports multiple devices, autoprobe and hotplugging.

NB: `nrpacks` parameter can be modified dynamically via sysfs. Don't put the value over 20. Changing via sysfs has no sanity check.

NB: `ignore_ctl_error=1` may help when you get an error at accessing the mixer element such as URB error -22. This happens on some buggy USB device or the controller. This workaround corresponds to the `quirk_flags` bit 14, too.

NB: `quirk_alias` option is provided only for testing / development. If you want to have a proper support, contact to upstream for adding the matching quirk in the driver code statically. Ditto for `quirk_flags`. If a device is known to require specific workarounds, please report to the upstream.

Module snd-usb-caiaq

Module for caiaq UB audio interfaces,

- Native Instruments RigKontrol2
- Native Instruments Kore Controller

- Native Instruments Audio Kontrol 1
- Native Instruments Audio 8 DJ

This module supports multiple devices, autoprobe and hotplugging.

Module snd-usb-usx2y

Module for Tascam USB US-122, US-224 and US-428 devices.

This module supports multiple devices, autoprobe and hotplugging.

Note: you need to load the firmware via `usx2yloader` utility included in alsa-tools and alsa-firmware packages.

Module snd-via82xx

Module for AC'97 motherboards based on VIA 82C686A/686B, 8233, 8233A, 8233C, 8235, 8237 (south) bridge.

mpu_port
: 0x300,0x310,0x320,0x330, otherwise obtain BIOS setup [VIA686A/686B only]

joystick
: Enable joystick (default off) [VIA686A/686B only]

ac97_clock
: AC'97 codec clock base (default 48000Hz)

dxs_support
: support DXS channels, 0 = auto (default), 1 = enable, 2 = disable, 3 = 48k only, 4 = no VRA, 5 = enable any sample rate and different sample rates on different channels [VIA8233/C, 8235, 8237 only]

ac97_quirk
: AC'97 workaround for strange hardware; See AC97 Quirk Option section below.

This module supports one chip and autoprobe.

Note: on some SMP motherboards like MSI 694D the interrupts might not be generated properly. In such a case, please try to set the SMP (or MPS) version on BIOS to 1.1 instead of default value 1.4. Then the interrupt number will be assigned under 15. You might also upgrade your BIOS.

Note: VIA8233/5/7 (not VIA8233A) can support DXS (direct sound) channels as the first PCM. On these channels, up to 4 streams can be played at the same time, and the controller can perform sample rate conversion with separate rates for each channel. As default (`dxs_support = 0`), 48k fixed rate is chosen except for the known devices since the output is often noisy except for 48k on some mother boards due to the bug of BIOS. Please try once `dxs_support=5` and if it works on other sample rates (e.g. 44.1kHz of mp3 playback), please let us know the PCI subsystem vendor/device id's (output of `lspci -nv`). If `dxs_support=5` does not work, try `dxs_support=4`; if it doesn't work too, try dxs_support=1. (dxs_support=1 is usually for old motherboards. The correct implemented board should work with 4 or 5.) If it still doesn't work and the default setting is ok, `dxs_support=3` is the right choice. If the default setting doesn't work at all, try `dxs_support=2` to disable the DXS channels. In any cases, please let us know the result and the subsystem vendor/device ids. See Links and Addresses below.

Note: for the MPU401 on VIA823x, use snd-mpu401 driver additionally. The mpu_port option is for VIA686 chips only.

The power-management is supported.

Module snd-via82xx-modem

Module for VIA82xx AC97 modem

ac97_clock
: AC'97 codec clock base (default 48000Hz)

This module supports one card and autoprobe.

Note: The default index value of this module is -2, i.e. the first slot is excluded.

The power-management is supported.

Module snd-virmidi

Module for virtual rawmidi devices. This module creates virtual rawmidi devices which communicate to the corresponding ALSA sequencer ports.

midi_devs
: MIDI devices # (1-4, default=4)

This module supports multiple cards.

Module snd-virtuoso

Module for sound cards based on the Asus AV66/AV100/AV200 chips, i.e., Xonar D1, DX, D2, D2X, DS, DSX, Essence ST (Deluxe), Essence STX (II), HDAV1.3 (Deluxe), and HDAV1.3 Slim.

This module supports autoprobe and multiple cards.

Module snd-vx222

Module for Digigram VX-Pocket VX222, V222 v2 and Mic cards.

mic
: Enable Microphone on V222 Mic (NYI)

ibl
: Capture IBL size. (default = 0, minimum size)

This module supports multiple cards.

When the driver is compiled as a module and the hotplug firmware is supported, the firmware data is loaded via hotplug automatically. Install the necessary firmware files in alsa-firmware package. When no hotplug fw loader is available, you need to load the firmware via vxloader utility in alsa-tools package. To invoke vxloader automatically, add the following to /etc/modprobe.d/alsa.conf

```
install snd-vx222 /sbin/modprobe --first-time -i snd-vx222\
  && /usr/bin/vxloader
```

(for 2.2/2.4 kernels, add `post-install /usr/bin/vxloader` to /etc/modules.conf, instead.) IBL size defines the interrupts period for PCM. The smaller size gives smaller latency but leads to more CPU consumption, too. The size is usually aligned to 126. As default (=0), the smallest size is chosen. The possible IBL values can be found in /proc/asound/cardX/vx-status proc file.

The power-management is supported.

Module snd-vxpocket

Module for Digigram VX-Pocket VX2 and 440 PCMCIA cards.

ibl
: Capture IBL size. (default = 0, minimum size)

This module supports multiple cards. The module is compiled only when PCMCIA is supported on kernel.

With the older 2.6.x kernel, to activate the driver via the card manager, you'll need to set up /etc/pcmcia/vxpocket.conf. See the sound/pcmcia/vx/vxpocket.c. 2.6.13 or later kernel requires no longer require a config file.

When the driver is compiled as a module and the hotplug firmware is supported, the firmware data is loaded via hotplug automatically. Install the necessary firmware files in alsa-firmware package. When no hotplug fw loader is available, you need to load the firmware via vxloader utility in alsa-tools package.

About capture IBL, see the description of snd-vx222 module.

Note: snd-vxp440 driver is merged to snd-vxpocket driver since ALSA 1.0.10.

The power-management is supported.

Module snd-ymfpci

Module for Yamaha PCI chips (YMF72x, YMF74x & YMF75x).

mpu_port
: 0x300,0x330,0x332,0x334, 0 (disable) by default, 1 (auto-detect for YMF744/754 only)

fm_port
: 0x388,0x398,0x3a0,0x3a8, 0 (disable) by default 1 (auto-detect for YMF744/754 only)

joystick_port
: 0x201,0x202,0x204,0x205, 0 (disable) by default, 1 (auto-detect)

rear_switch
: enable shared rear/line-in switch (bool)

This module supports autoprobe and multiple chips.

The power-management is supported.

Module snd-pdaudiocf

Module for Sound Core PDAudioCF sound card.

The power-management is supported.

AC97 Quirk Option

The ac97_quirk option is used to enable/override the workaround for specific devices on drivers for on-board AC'97 controllers like snd-intel8x0. Some hardware have swapped output pins between Master and Headphone, or Surround (thanks to confusion of AC'97 specifications from version to version :-)

The driver provides the auto-detection of known problematic devices, but some might be unknown or wrongly detected. In such a case, pass the proper value with this option.

The following strings are accepted:

default
: Don't override the default setting

none
: Disable the quirk

hp_only
: Bind Master and Headphone controls as a single control

swap_hp
: Swap headphone and master controls

swap_surround
: Swap master and surround controls

ad_sharing
: For AD1985, turn on OMS bit and use headphone

alc_jack
: For ALC65x, turn on the jack sense mode

inv_eapd
: Inverted EAPD implementation

mute_led
: Bind EAPD bit for turning on/off mute LED

For backward compatibility, the corresponding integer value -1, 0, ... are accepted, too.

For example, if `Master` volume control has no effect on your device but only `Headphone` does, pass ac97_quirk=hp_only module option.

Configuring Non-ISAPNP Cards

When the kernel is configured with ISA-PnP support, the modules supporting the isapnp cards will have module options `isapnp`. If this option is set, *only* the ISA-PnP devices will be probed. For probing the non ISA-PnP cards, you have to pass `isapnp=0` option together with the proper i/o and irq configuration.

When the kernel is configured without ISA-PnP support, isapnp option will be not built in.

Module Autoloading Support

The ALSA drivers can be loaded automatically on demand by defining module aliases. The string `snd-card-%1` is requested for ALSA native devices where `%i` is sound card number from zero to seven.

To auto-load an ALSA driver for OSS services, define the string `sound-slot-%i` where `%i` means the slot number for OSS, which corresponds to the card index of ALSA. Usually, define this as the same card module.

An example configuration for a single emu10k1 card is like below:

```
----- /etc/modprobe.d/alsa.conf
alias snd-card-0 snd-emu10k1
alias sound-slot-0 snd-emu10k1
----- /etc/modprobe.d/alsa.conf
```

The available number of auto-loaded sound cards depends on the module option `cards_limit` of snd module. As default it's set to 1. To enable the auto-loading of multiple cards, specify the number of sound cards in that option.

When multiple cards are available, it'd better to specify the index number for each card via module option, too, so that the order of cards is kept consistent.

An example configuration for two sound cards is like below:

```
----- /etc/modprobe.d/alsa.conf
# ALSA portion
options snd cards_limit=2
alias snd-card-0 snd-interwave
alias snd-card-1 snd-ens1371
options snd-interwave index=0
options snd-ens1371 index=1
# OSS/Free portion
alias sound-slot-0 snd-interwave
alias sound-slot-1 snd-ens1371
----- /etc/modprobe.d/alsa.conf
```

In this example, the interwave card is always loaded as the first card (index 0) and ens1371 as the second (index 1).

Alternative (and new) way to fixate the slot assignment is to use `slots` option of snd module. In the case above, specify like the following:

```
options snd slots=snd-interwave,snd-ens1371
```

Then, the first slot (#0) is reserved for snd-interwave driver, and the second (#1) for snd-ens1371. You can omit index option in each driver if slots option is used (although you can still have them at the same time as long as they don't conflict).

The slots option is especially useful for avoiding the possible hot-plugging and the resultant slot conflict. For example, in the case above again, the first two slots are already reserved. If any other driver (e.g. snd-usb-audio) is loaded before snd-interwave or snd-ens1371, it will be assigned to the third or later slot.

When a module name is given with '!', the slot will be given for any modules but that name. For example, `slots=!snd-pcsp` will reserve the first slot for any modules but snd-pcsp.

ALSA PCM devices to OSS devices mapping

```
/dev/snd/pcmC0D0[c|p]  -> /dev/audio0 (/dev/audio) -> minor 4
/dev/snd/pcmC0D0[c|p]  -> /dev/dsp0 (/dev/dsp)     -> minor 3
/dev/snd/pcmC0D1[c|p]  -> /dev/adsp0 (/dev/adsp)   -> minor 12
/dev/snd/pcmC1D0[c|p]  -> /dev/audio1              -> minor 4+16 = 20
/dev/snd/pcmC1D0[c|p]  -> /dev/dsp1                -> minor 3+16 = 19
/dev/snd/pcmC1D1[c|p]  -> /dev/adsp1               -> minor 12+16 = 28
/dev/snd/pcmC2D0[c|p]  -> /dev/audio2              -> minor 4+32 = 36
/dev/snd/pcmC2D0[c|p]  -> /dev/dsp2                -> minor 3+32 = 39
/dev/snd/pcmC2D1[c|p]  -> /dev/adsp2               -> minor 12+32 = 44
```

The first number from `/dev/snd/pcmC{X}D{Y}[c|p]` expression means sound card number and second means device number. The ALSA devices have either `c` or `p` suffix indicating the direction, capture and playback, respectively.

Please note that the device mapping above may be varied via the module options of snd-pcm-oss module.

Proc interfaces (/proc/asound)

/proc/asound/card#/pcm#[cp]/oss

erase

erase all additional information about OSS applications

<app_name> <fragments> <fragment_size> [<options>]

<app_name>

name of application with (higher priority) or without path

<fragments>

number of fragments or zero if auto

<fragment_size>

size of fragment in bytes or zero if auto

<options>

optional parameters

disable
: the application tries to open a pcm device for this channel but does not want to use it. (Cause a bug or mmap needs) It's good for Quake etc...

direct
: don't use plugins

block
: force block mode (rvplayer)

non-block
: force non-block mode

whole-frag
: write only whole fragments (optimization affecting playback only)

no-silence
: do not fill silence ahead to avoid clicks

buggy-ptr
: Returns the whitespace blocks in GETOPTR ioctl instead of filled blocks

Example:

```
echo "x11amp 128 16384" > /proc/asound/card0/pcm0p/oss
echo "squake 0 0 disable" > /proc/asound/card0/pcm0c/oss
echo "rvplayer 0 0 block" > /proc/asound/card0/pcm0p/oss
```

Early Buffer Allocation

Some drivers (e.g. hdsp) require the large contiguous buffers, and sometimes it's too late to find such spaces when the driver module is actually loaded due to memory fragmentation. You can pre-allocate the PCM buffers by loading snd-page-alloc module and write commands to its proc file in prior, for example, in the early boot stage like `/etc/init.d/*.local` scripts.

Reading the proc file /proc/drivers/snd-page-alloc shows the current usage of page allocation. In writing, you can send the following commands to the snd-page-alloc driver:

- add VENDOR DEVICE MASK SIZE BUFFERS

VENDOR and DEVICE are PCI vendor and device IDs. They take integer numbers (0x prefix is needed for the hex). MASK is the PCI DMA mask. Pass 0 if not restricted. SIZE is the size of each buffer to allocate. You can pass k and m suffix for KB and MB. The max number is 16MB. BUFFERS is the number of buffers to allocate. It must be greater than 0. The max number is 4.

- erase

This will erase the all pre-allocated buffers which are not in use.

Links and Addresses

ALSA project homepage
: http://www.alsa-project.org

Kernel Bugzilla
: http://bugzilla.kernel.org/

ALSA Developers ML
: mailto:alsa-devel@alsa-project.org

alsa-info.sh script
: https://www.alsa-project.org/alsa-info.sh