Linux Sound Subsystem Documentation

Release

The kernel development community

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CHAPTER

ONE

ALSA KERNEL API DOCUMENTATION

The ALSA Driver API

Management of Cards and Devices

Card Management

void **snd_device_initialize**(struct device * *dev*, struct snd_card * *card*)
Initialize struct device for sound devices

Parameters

struct device * dev device to initialize

struct snd_card * card card to assign, optional

int **snd_card_new**(struct device * *parent*, int *idx*, const char * *xid*, struct module * *module*, int *extra_size*, struct snd_card ** *card_ret*)
create and initialize a soundcard structure

Parameters

struct device * parent the parent device object

int idx card index (address) [0 ... (SNDRV_CARDS-1)]

const char * xid card identification (ASCII string)

struct module * module top level module for locking

int extra_size allocate this extra size after the main soundcard structure

struct snd_card ** card_ret the pointer to store the created card instance

Description

Creates and initializes a soundcard structure.

The function allocates snd_card instance via kzalloc with the given space for the driver to use freely. The allocated struct is stored in the given card_ret pointer.

Return

Zero if successful or a negative error code.

int **snd_card_disconnect**(struct snd_card * *card*)
disconnect all APIs from the file-operations (user space)

Parameters

struct snd_card * card soundcard structure

Description

Disconnects all APIs from the file-operations (user space).

Return

Zero, otherwise a negative error code.

Note

The current implementation replaces all active file->f_op with special dummy file operations (they do nothing except release).

void **snd_card_disconnect_sync**(struct snd_card * *card*)
disconnect card and wait until files get closed

Parameters

struct snd_card * card card object to disconnect

Description

This calls *snd_card_disconnect()* for disconnecting all belonging components and waits until all pending files get closed. It assures that all accesses from user-space finished so that the driver can release its resources gracefully.

int **snd_card_free_when_closed**(struct snd_card * *card*)
Disconnect the card, free it later eventually

Parameters

struct snd_card * card soundcard structure

Description

Unlike *snd_card_free()*, this function doesn't try to release the card resource immediately, but tries to disconnect at first. When the card is still in use, the function returns before freeing the resources. The card resources will be freed when the refcount gets to zero.

int **snd_card_free**(struct snd_card * *card*)
frees given soundcard structure

Parameters

struct snd_card * card soundcard structure

Description

This function releases the soundcard structure and the all assigned devices automatically. That is, you don't have to release the devices by yourself.

This function waits until the all resources are properly released.

Return

Zero. Frees all associated devices and frees the control interface associated to given soundcard.

void **snd_card_set_id**(struct snd_card * *card*, const char * *nid*)
set card identification name

Parameters

struct snd_card * card soundcard structure

const char * nid new identification string

Description

This function sets the card identification and checks for name collisions.

int **snd_card_add_dev_attr**(struct snd_card * *card*, const struct attribute_group * *group*)
Append a new sysfs attribute group to card

Parameters

struct snd_card * card card instance

const struct attribute_group * group attribute group to append

int **snd_card_register**(struct snd_card * *card*)
register the soundcard

Parameters

struct snd_card * card soundcard structure

Description

This function registers all the devices assigned to the soundcard. Until calling this, the ALSA control interface is blocked from the external accesses. Thus, you should call this function at the end of the initialization of the card.

Return

Zero otherwise a negative error code if the registration failed.

int **snd_component_add**(struct snd_card * *card*, const char * *component*)
add a component string

Parameters

struct snd_card * card soundcard structure

const char * component the component id string

Description

This function adds the component id string to the supported list. The component can be referred from the alsa-lib.

Return

Zero otherwise a negative error code.

int **snd_card_file_add**(struct snd_card * *card*, struct file * *file*)
add the file to the file list of the card

Parameters

struct snd_card * card soundcard structure

struct file * file file pointer

Description

This function adds the file to the file linked-list of the card. This linked-list is used to keep tracking the connection state, and to avoid the release of busy resources by hotplug.

Return

zero or a negative error code.

int **snd_card_file_remove**(struct snd_card * *card*, struct file * *file*)
remove the file from the file list

Parameters

struct snd_card * card soundcard structure

struct file * file file pointer

Description

This function removes the file formerly added to the card via *snd_card_file_add()* function. If all files are removed and *snd_card_free_when_closed()* was called beforehand, it processes the pending release of resources.

Return

Zero or a negative error code.

int **snd_power_wait**(struct snd_card * *card*, unsigned int *power_state*)
wait until the power-state is changed.

Parameters

struct snd_card * card soundcard structure

unsigned int power_state expected power state

Description

Waits until the power-state is changed.

Return

Zero if successful, or a negative error code.

Device Components

int **snd_device_new**(struct snd_card * *card*, enum snd_device_type *type*, void * *device_data*, struct snd_device_ops * *ops*)
create an ALSA device component

Parameters

struct snd_card * card the card instance

enum snd_device_type type the device type, SNDRV_DEV_XXX

void * device_data the data pointer of this device

struct snd_device_ops * ops the operator table

Description

Creates a new device component for the given data pointer. The device will be assigned to the card and managed together by the card.

The data pointer plays a role as the identifier, too, so the pointer address must be unique and unchanged.

Return

Zero if successful, or a negative error code on failure.

void **snd_device_disconnect**(struct snd_card * *card*, void * *device_data*)
disconnect the device

Parameters

struct snd_card * card the card instance

void * device_data the data pointer to disconnect

Description

Turns the device into the disconnection state, invoking dev_disconnect callback, if the device was already registered.

Usually called from *snd_card_disconnect()*.

Return

Zero if successful, or a negative error code on failure or if the device not found.

void **snd_device_free**(struct snd_card * *card*, void * *device_data*)
release the device from the card

Parameters

struct snd_card * card the card instance

void * device_data the data pointer to release

Description

Removes the device from the list on the card and invokes the callbacks, dev_disconnect and dev_free, corresponding to the state. Then release the device.

int **snd_device_register**(struct snd_card * *card*, void * *device_data*)
register the device

Parameters

struct snd_card * card the card instance

void * device_data the data pointer to register

Description

Registers the device which was already created via *snd_device_new()*. Usually this is called from *snd_card_register()*, but it can be called later if any new devices are created after invocation of *snd_card_register()*.

Return

Zero if successful, or a negative error code on failure or if the device not found.

Module requests and Device File Entries

void **snd_request_card**(int *card*)
try to load the card module

Parameters

int card the card number

Description

Tries to load the module "snd-card-X" for the given card number via request_module. Returns immediately if already loaded.

void * **snd_lookup_minor_data**(unsigned int *minor*, int *type*)
get user data of a registered device

Parameters

unsigned int minor the minor number

int type device type (SNDRV_DEVICE_TYPE_XXX)

Description

Checks that a minor device with the specified type is registered, and returns its user data pointer.

This function increments the reference counter of the card instance if an associated instance with the given minor number and type is found. The caller must call `snd_card_unref()` appropriately later.

Return

The user data pointer if the specified device is found. `NULL` otherwise.

int **snd_register_device**(int *type*, struct snd_card * *card*, int *dev*, const struct file_operations * *f_ops*, void * *private_data*, struct device * *device*)
Register the ALSA device file for the card

Parameters

int type the device type, SNDRV_DEVICE_TYPE_XXX

struct snd_card * card the card instance

int dev the device index

const struct file_operations * f_ops the file operations

void * private_data user pointer for f_ops->:c:func:*open()*

struct device * device the device to register

Description

Registers an ALSA device file for the given card. The operators have to be set in reg parameter.

Return

Zero if successful, or a negative error code on failure.

int **snd_unregister_device**(struct device * *dev*)
unregister the device on the given card

Parameters

struct device * dev the device instance

Description

Unregisters the device file already registered via *snd_register_device()*.

Return

Zero if successful, or a negative error code on failure.

Memory Management Helpers

int **copy_to_user_fromio**(void __user * *dst*, const volatile void __iomem * *src*, size_t *count*)
copy data from mmio-space to user-space

Parameters

void __user * dst the destination pointer on user-space

const volatile void __iomem * src the source pointer on mmio

size_t count the data size to copy in bytes

Description

Copies the data from mmio-space to user-space.

Return

Zero if successful, or non-zero on failure.

int **copy_from_user_toio**(volatile void __iomem * *dst*, const void __user * *src*, size_t *count*)
copy data from user-space to mmio-space

Parameters

volatile void __iomem * dst the destination pointer on mmio-space

const void __user * src the source pointer on user-space

size_t count the data size to copy in bytes

Description

Copies the data from user-space to mmio-space.

Return

Zero if successful, or non-zero on failure.

void * **snd_malloc_pages**(size_t *size*, gfp_t *gfp_flags*)
allocate pages with the given size

Parameters

size_t size the size to allocate in bytes

gfp_t gfp_flags the allocation conditions, GFP_XXX

Description

Allocates the physically contiguous pages with the given size.

Return

The pointer of the buffer, or NULL if no enough memory.

void **snd_free_pages**(void * *ptr*, size_t *size*)
release the pages

Parameters

void * ptr the buffer pointer to release

size_t size the allocated buffer size

Description

Releases the buffer allocated via *snd_malloc_pages()*.

void **snd_malloc_dev_iram**(struct snd_dma_buffer * *dmab*, size_t *size*)
allocate memory from on-chip internal ram

Parameters

struct snd_dma_buffer * dmab buffer allocation record to store the allocated data

size_t size number of bytes to allocate from the iram

Description

This function requires iram phandle provided via of_node

void **snd_free_dev_iram**(struct snd_dma_buffer * *dmab*)
free allocated specific memory from on-chip internal ram

Parameters

struct snd_dma_buffer * dmab buffer allocation record to store the allocated data

int **snd_dma_alloc_pages**(int *type*, struct device * *device*, size_t *size*, struct snd_dma_buffer * *dmab*)
allocate the buffer area according to the given type

Parameters

int type the DMA buffer type

struct device * device the device pointer

size_t size the buffer size to allocate

struct snd_dma_buffer * dmab buffer allocation record to store the allocated data

Description

Calls the memory-allocator function for the corresponding buffer type.

Return

Zero if the buffer with the given size is allocated successfully, otherwise a negative value on error.

int **snd_dma_alloc_pages_fallback**(int *type*, struct device * *device*, size_t *size*, struct snd_dma_buffer * *dmab*)
allocate the buffer area according to the given type with fallback

Parameters

int type the DMA buffer type

struct device * device the device pointer

size_t size the buffer size to allocate

struct snd_dma_buffer * dmab buffer allocation record to store the allocated data

Description

Calls the memory-allocator function for the corresponding buffer type. When no space is left, this function reduces the size and tries to allocate again. The size actually allocated is stored in res_size argument.

Return

Zero if the buffer with the given size is allocated successfully, otherwise a negative value on error.

void **snd_dma_free_pages**(struct snd_dma_buffer * *dmab*)
release the allocated buffer

Parameters

struct snd_dma_buffer * dmab the buffer allocation record to release

Description

Releases the allocated buffer via *snd_dma_alloc_pages()*.

PCM API

PCM Core

const char * **snd_pcm_format_name**(snd_pcm_format_t *format*)
Return a name string for the given PCM format

Parameters

snd_pcm_format_t format PCM format

int **snd_pcm_new_stream**(struct snd_pcm * *pcm*, int *stream*, int *substream_count*)
create a new PCM stream

Parameters

struct snd_pcm * pcm the pcm instance

int stream the stream direction, SNDRV_PCM_STREAM_XXX

int substream_count the number of substreams

Description

Creates a new stream for the pcm. The corresponding stream on the pcm must have been empty before calling this, i.e. zero must be given to the argument of *snd_pcm_new()*.

Return

Zero if successful, or a negative error code on failure.

int **snd_pcm_new**(struct snd_card * *card*, const char * *id*, int *device*, int *playback_count*, int *capture_count*, struct snd_pcm ** *rpcm*)
create a new PCM instance

Parameters

struct snd_card * card the card instance

const char * id the id string

int device the device index (zero based)

int playback_count the number of substreams for playback

int capture_count the number of substreams for capture

struct snd_pcm ** rpcm the pointer to store the new pcm instance

Description

Creates a new PCM instance.

The pcm operators have to be set afterwards to the new instance via *snd_pcm_set_ops()*.

Return

Zero if successful, or a negative error code on failure.

int **snd_pcm_new_internal**(struct snd_card * *card*, const char * *id*, int *device*, int *playback_count*, int *capture_count*, struct snd_pcm ** *rpcm*)
create a new internal PCM instance

Parameters

struct snd_card * card the card instance

const char * id the id string

int device the device index (zero based - shared with normal PCMs)

int playback_count the number of substreams for playback

int capture_count the number of substreams for capture

struct snd_pcm ** rpcm the pointer to store the new pcm instance

Description

Creates a new internal PCM instance with no userspace device or procfs entries. This is used by ASoC Back End PCMs in order to create a PCM that will only be used internally by kernel drivers. i.e. it cannot be opened by userspace. It provides existing ASoC components drivers with a substream and access to any private data.

The pcm operators have to be set afterwards to the new instance via *snd_pcm_set_ops()*.

Return

Zero if successful, or a negative error code on failure.

int **snd_pcm_notify**(struct *snd_pcm_notify* * *notify*, int *nfree*)
Add/remove the notify list

Parameters

struct snd_pcm_notify * notify PCM notify list

int nfree 0 = register, 1 = unregister

Description

This adds the given notifier to the global list so that the callback is called for each registered PCM devices. This exists only for PCM OSS emulation, so far.

void **snd_pcm_set_ops**(struct snd_pcm * *pcm*, int *direction*, const struct snd_pcm_ops * *ops*)
set the PCM operators

Parameters

struct snd_pcm * pcm the pcm instance

int direction stream direction, SNDRV_PCM_STREAM_XXX

const struct snd_pcm_ops * ops the operator table

Description

Sets the given PCM operators to the pcm instance.

void **snd_pcm_set_sync**(struct snd_pcm_substream * *substream*)
set the PCM sync id

Parameters

struct snd_pcm_substream * substream the pcm substream

Description

Sets the PCM sync identifier for the card.

int **snd_interval_refine**(struct snd_interval * *i*, const struct snd_interval * *v*)
refine the interval value of configurator

Parameters

struct snd_interval * i the interval value to refine

const struct snd_interval * v the interval value to refer to

Description

Refines the interval value with the reference value. The interval is changed to the range satisfying both intervals. The interval status (min, max, integer, etc.) are evaluated.

Return

Positive if the value is changed, zero if it's not changed, or a negative error code.

void **snd_interval_div**(const struct snd_interval * *a*, const struct snd_interval * *b*, struct snd_interval * *c*)
refine the interval value with division

Parameters

const struct snd_interval * a dividend

const struct snd_interval * b divisor

struct snd_interval * c quotient

Description

c = a / b

Returns non-zero if the value is changed, zero if not changed.

void **snd_interval_muldivk**(const struct snd_interval * *a*, const struct snd_interval * *b*, unsigned int *k*, struct snd_interval * *c*)
refine the interval value

Parameters

const struct snd_interval * a dividend 1

const struct snd_interval * b dividend 2

unsigned int k divisor (as integer)

struct snd_interval * c result

Description

c = a * b / k

Returns non-zero if the value is changed, zero if not changed.

void **snd_interval_mulkdiv**(const struct snd_interval * *a*, unsigned int *k*, const struct snd_interval * *b*, struct snd_interval * *c*)
refine the interval value

Parameters

const struct snd_interval * a dividend 1

unsigned int k dividend 2 (as integer)

const struct snd_interval * b divisor

struct snd_interval * c result

Description

c = a * k / b

Returns non-zero if the value is changed, zero if not changed.

int **snd_interval_ratnum**(struct snd_interval * *i*, unsigned int *rats_count*, const struct snd_ratnum * *rats*, unsigned int * *nump*, unsigned int * *denp*)
refine the interval value

Parameters

struct snd_interval * i interval to refine

unsigned int rats_count number of ratnum_t

const struct snd_ratnum * rats ratnum_t array

unsigned int * nump pointer to store the resultant numerator

unsigned int * denp pointer to store the resultant denominator

Return

Positive if the value is changed, zero if it's not changed, or a negative error code.

int **snd_interval_ratden**(struct snd_interval * *i*, unsigned int *rats_count*, const struct snd_ratden * *rats*, unsigned int * *nump*, unsigned int * *denp*)
refine the interval value

Parameters

struct snd_interval * i interval to refine

unsigned int rats_count number of struct ratden

const struct snd_ratden * rats struct ratden array

unsigned int * nump pointer to store the resultant numerator

unsigned int * denp pointer to store the resultant denominator

Return

Positive if the value is changed, zero if it's not changed, or a negative error code.

int **snd_interval_list**(struct snd_interval * *i*, unsigned int *count*, const unsigned int * *list*, unsigned int *mask*)
refine the interval value from the list

Parameters

struct snd_interval * i the interval value to refine

unsigned int count the number of elements in the list

const unsigned int * list the value list

unsigned int mask the bit-mask to evaluate

Description

Refines the interval value from the list. When mask is non-zero, only the elements corresponding to bit 1 are evaluated.

Return

Positive if the value is changed, zero if it's not changed, or a negative error code.

int **snd_interval_ranges**(struct snd_interval * *i*, unsigned int *count*, const struct snd_interval * *ranges*, unsigned int *mask*)
refine the interval value from the list of ranges

Parameters

struct snd_interval * i the interval value to refine

unsigned int count the number of elements in the list of ranges

const struct snd_interval * ranges the ranges list

unsigned int mask the bit-mask to evaluate

Description

Refines the interval value from the list of ranges. When mask is non-zero, only the elements corresponding to bit 1 are evaluated.

Return

Positive if the value is changed, zero if it's not changed, or a negative error code.

int **snd_pcm_hw_rule_add**(struct snd_pcm_runtime * *runtime*, unsigned int *cond*, int *var*, snd_pcm_hw_rule_func_t *func*, void * *private*, int *dep*, ...)

add the hw-constraint rule

Parameters

struct snd_pcm_runtime * runtime the pcm runtime instance

unsigned int cond condition bits

int var the variable to evaluate

snd_pcm_hw_rule_func_t func the evaluation function

void * private the private data pointer passed to function

int dep the dependent variables

... variable arguments

Return

Zero if successful, or a negative error code on failure.

int **snd_pcm_hw_constraint_mask**(struct snd_pcm_runtime * *runtime*, snd_pcm_hw_param_t *var*, u_int32_t *mask*)

apply the given bitmap mask constraint

Parameters

struct snd_pcm_runtime * runtime PCM runtime instance

snd_pcm_hw_param_t var hw_params variable to apply the mask

u_int32_t mask the bitmap mask

Description

Apply the constraint of the given bitmap mask to a 32-bit mask parameter.

Return

Zero if successful, or a negative error code on failure.

int **snd_pcm_hw_constraint_mask64**(struct snd_pcm_runtime * *runtime*, snd_pcm_hw_param_t *var*, u_int64_t *mask*)

apply the given bitmap mask constraint

Parameters

struct snd_pcm_runtime * runtime PCM runtime instance

snd_pcm_hw_param_t var hw_params variable to apply the mask

u_int64_t mask the 64bit bitmap mask

Description

Apply the constraint of the given bitmap mask to a 64-bit mask parameter.

Return

Zero if successful, or a negative error code on failure.

int **snd_pcm_hw_constraint_integer**(struct snd_pcm_runtime * *runtime*, snd_pcm_hw_param_t *var*)
apply an integer constraint to an interval

Parameters

struct snd_pcm_runtime * runtime PCM runtime instance

snd_pcm_hw_param_t var hw_params variable to apply the integer constraint

Description

Apply the constraint of integer to an interval parameter.

Return

Positive if the value is changed, zero if it's not changed, or a negative error code.

int **snd_pcm_hw_constraint_minmax**(struct snd_pcm_runtime * *runtime*, snd_pcm_hw_param_t *var*, unsigned int *min*, unsigned int *max*)
apply a min/max range constraint to an interval

Parameters

struct snd_pcm_runtime * runtime PCM runtime instance

snd_pcm_hw_param_t var hw_params variable to apply the range

unsigned int min the minimal value

unsigned int max the maximal value

Description

Apply the min/max range constraint to an interval parameter.

Return

Positive if the value is changed, zero if it's not changed, or a negative error code.

int **snd_pcm_hw_constraint_list**(struct snd_pcm_runtime * *runtime*, unsigned int *cond*, snd_pcm_hw_param_t *var*, const struct *snd_pcm_hw_constraint_list* * *l*)
apply a list of constraints to a parameter

Parameters

struct snd_pcm_runtime * runtime PCM runtime instance

unsigned int cond condition bits

snd_pcm_hw_param_t var hw_params variable to apply the list constraint

const struct snd_pcm_hw_constraint_list * l list

Description

Apply the list of constraints to an interval parameter.

Return

Zero if successful, or a negative error code on failure.

int **snd_pcm_hw_constraint_ranges**(struct snd_pcm_runtime * *runtime*, unsigned int *cond*, snd_pcm_hw_param_t *var*, const struct *snd_pcm_hw_constraint_ranges* * *r*)

apply list of range constraints to a parameter

Parameters

struct snd_pcm_runtime * runtime PCM runtime instance

unsigned int cond condition bits

snd_pcm_hw_param_t var hw_params variable to apply the list of range constraints

const struct snd_pcm_hw_constraint_ranges * r ranges

Description

Apply the list of range constraints to an interval parameter.

Return

Zero if successful, or a negative error code on failure.

int **snd_pcm_hw_constraint_ratnums**(struct snd_pcm_runtime * *runtime*, unsigned int *cond*, snd_pcm_hw_param_t *var*, const struct *snd_pcm_hw_constraint_ratnums* * *r*)

apply ratnums constraint to a parameter

Parameters

struct snd_pcm_runtime * runtime PCM runtime instance

unsigned int cond condition bits

snd_pcm_hw_param_t var hw_params variable to apply the ratnums constraint

const struct snd_pcm_hw_constraint_ratnums * r struct snd_ratnums constriants

Return

Zero if successful, or a negative error code on failure.

int **snd_pcm_hw_constraint_ratdens**(struct snd_pcm_runtime * *runtime*, unsigned int *cond*, snd_pcm_hw_param_t *var*, const struct *snd_pcm_hw_constraint_ratdens* * *r*)

apply ratdens constraint to a parameter

Parameters

struct snd_pcm_runtime * runtime PCM runtime instance

unsigned int cond condition bits

snd_pcm_hw_param_t var hw_params variable to apply the ratdens constraint

const struct snd_pcm_hw_constraint_ratdens * r struct snd_ratdens constriants

Return

Zero if successful, or a negative error code on failure.

int **snd_pcm_hw_constraint_msbits**(struct snd_pcm_runtime * *runtime*, unsigned int *cond*, unsigned int *width*, unsigned int *msbits*)

add a hw constraint msbits rule

Parameters

struct snd_pcm_runtime * runtime PCM runtime instance

unsigned int cond condition bits

unsigned int width sample bits width

unsigned int msbits msbits width

Description

This constraint will set the number of most significant bits (msbits) if a sample format with the specified width has been select. If width is set to 0 the msbits will be set for any sample format with a width larger than the specified msbits.

Return

Zero if successful, or a negative error code on failure.

int **snd_pcm_hw_constraint_step**(struct snd_pcm_runtime * *runtime*, unsigned int *cond*, snd_pcm_hw_param_t *var*, unsigned long *step*)
add a hw constraint step rule

Parameters

struct snd_pcm_runtime * runtime PCM runtime instance

unsigned int cond condition bits

snd_pcm_hw_param_t var hw_params variable to apply the step constraint

unsigned long step step size

Return

Zero if successful, or a negative error code on failure.

int **snd_pcm_hw_constraint_pow2**(struct snd_pcm_runtime * *runtime*, unsigned int *cond*, snd_pcm_hw_param_t *var*)
add a hw constraint power-of-2 rule

Parameters

struct snd_pcm_runtime * runtime PCM runtime instance

unsigned int cond condition bits

snd_pcm_hw_param_t var hw_params variable to apply the power-of-2 constraint

Return

Zero if successful, or a negative error code on failure.

int **snd_pcm_hw_rule_noresample**(struct snd_pcm_runtime * *runtime*, unsigned int *base_rate*)
add a rule to allow disabling hw resampling

Parameters

struct snd_pcm_runtime * runtime PCM runtime instance

unsigned int base_rate the rate at which the hardware does not resample

Return

Zero if successful, or a negative error code on failure.

int **snd_pcm_hw_param_value**(const struct snd_pcm_hw_params * *params*, snd_pcm_hw_param_t *var*, int * *dir*)
return **params** field **var** value

Parameters

const struct snd_pcm_hw_params * params the hw_params instance

snd_pcm_hw_param_t var parameter to retrieve

int * dir pointer to the direction (-1,0,1) or NULL

Return

The value for field **var** if it's fixed in configuration space defined by **params**. -EINVAL otherwise.

int **snd_pcm_hw_param_first**(struct snd_pcm_substream * *pcm*, struct snd_pcm_hw_params * *params*, snd_pcm_hw_param_t *var*, int * *dir*)
refine config space and return minimum value

Parameters

struct snd_pcm_substream * pcm PCM instance

struct snd_pcm_hw_params * params the hw_params instance

snd_pcm_hw_param_t var parameter to retrieve

int * dir pointer to the direction (-1,0,1) or NULL

Description

Inside configuration space defined by **params** remove from **var** all values > minimum. Reduce configuration space accordingly.

Return

The minimum, or a negative error code on failure.

int **snd_pcm_hw_param_last**(struct snd_pcm_substream * *pcm*, struct snd_pcm_hw_params * *params*, snd_pcm_hw_param_t *var*, int * *dir*)
refine config space and return maximum value

Parameters

struct snd_pcm_substream * pcm PCM instance

struct snd_pcm_hw_params * params the hw_params instance

snd_pcm_hw_param_t var parameter to retrieve

int * dir pointer to the direction (-1,0,1) or NULL

Description

Inside configuration space defined by **params** remove from **var** all values < maximum. Reduce configuration space accordingly.

Return

The maximum, or a negative error code on failure.

int **snd_pcm_lib_ioctl**(struct snd_pcm_substream * *substream*, unsigned int *cmd*, void * *arg*)
a generic PCM ioctl callback

Parameters

struct snd_pcm_substream * substream the pcm substream instance

unsigned int cmd ioctl command

void * arg ioctl argument

Description

Processes the generic ioctl commands for PCM. Can be passed as the ioctl callback for PCM ops.

Return

Zero if successful, or a negative error code on failure.

void **snd_pcm_period_elapsed**(struct snd_pcm_substream * *substream*)
update the pcm status for the next period

Parameters

struct snd_pcm_substream * substream the pcm substream instance

Description

This function is called from the interrupt handler when the PCM has processed the period size. It will update the current pointer, wake up sleepers, etc.

Even if more than one periods have elapsed since the last call, you have to call this only once.

int **snd_pcm_add_chmap_ctls**(struct snd_pcm * *pcm*, int *stream*, const struct snd_pcm_chmap_elem * *chmap*, int *max_channels*, unsigned long *private_value*, struct snd_pcm_chmap ** *info_ret*)
create channel-mapping control elements

Parameters

struct snd_pcm * pcm the assigned PCM instance

int stream stream direction

const struct snd_pcm_chmap_elem * chmap channel map elements (for query)

int max_channels the max number of channels for the stream

unsigned long private_value the value passed to each kcontrol's private_value field

struct snd_pcm_chmap ** info_ret store struct snd_pcm_chmap instance if non-NULL

Description

Create channel-mapping control elements assigned to the given PCM stream(s).

Return

Zero if successful, or a negative error value.

void **snd_pcm_stream_lock**(struct snd_pcm_substream * *substream*)
Lock the PCM stream

Parameters

struct snd_pcm_substream * substream PCM substream

Description

This locks the PCM stream's spinlock or mutex depending on the nonatomic flag of the given substream. This also takes the global link rw lock (or rw sem), too, for avoiding the race with linked streams.

void **snd_pcm_stream_unlock**(struct snd_pcm_substream * *substream*)
Unlock the PCM stream

Parameters

struct snd_pcm_substream * substream PCM substream

Description

This unlocks the PCM stream that has been locked via *snd_pcm_stream_lock()*.

void **snd_pcm_stream_lock_irq**(struct snd_pcm_substream * *substream*)
Lock the PCM stream

Parameters

struct snd_pcm_substream * substream PCM substream

Description

This locks the PCM stream like *snd_pcm_stream_lock()* and disables the local IRQ (only when nonatomic is false). In nonatomic case, this is identical as *snd_pcm_stream_lock()*.

void **snd_pcm_stream_unlock_irq**(struct snd_pcm_substream * *substream*)
Unlock the PCM stream

Parameters

struct snd_pcm_substream * substream PCM substream

Description

This is a counter-part of *snd_pcm_stream_lock_irq()*.

void **snd_pcm_stream_unlock_irqrestore**(struct snd_pcm_substream * *substream*, unsigned long *flags*)
Unlock the PCM stream

Parameters

struct snd_pcm_substream * substream PCM substream

unsigned long flags irq flags

Description

This is a counter-part of *snd_pcm_stream_lock_irqsave()*.

int **snd_pcm_hw_params_choose**(struct snd_pcm_substream * *pcm*, struct snd_pcm_hw_params * *params*)
choose a configuration defined by **params**

Parameters

struct snd_pcm_substream * pcm PCM instance

struct snd_pcm_hw_params * params the hw_params instance

Description

Choose one configuration from configuration space defined by **params**. The configuration chosen is that obtained fixing in this order: first access, first format, first subformat, min channels, min rate, min period time, max buffer size, min tick time

Return

Zero if successful, or a negative error code on failure.

int **snd_pcm_start**(struct snd_pcm_substream * *substream*)
start all linked streams

Parameters

struct snd_pcm_substream * substream the PCM substream instance

Return

Zero if successful, or a negative error code. The stream lock must be acquired before calling this function.

int **snd_pcm_stop**(struct snd_pcm_substream * *substream*, snd_pcm_state_t *state*)
try to stop all running streams in the substream group

Parameters

struct snd_pcm_substream * substream the PCM substream instance

snd_pcm_state_t state PCM state after stopping the stream

Description

The state of each stream is then changed to the given state unconditionally.

Return

Zero if successful, or a negative error code.

int **snd_pcm_drain_done**(struct snd_pcm_substream * *substream*)
stop the DMA only when the given stream is playback

Parameters

struct snd_pcm_substream * substream the PCM substream

Description

After stopping, the state is changed to SETUP. Unlike *snd_pcm_stop()*, this affects only the given stream.

Return

Zero if succesful, or a negative error code.

int **snd_pcm_stop_xrun**(struct snd_pcm_substream * *substream*)
stop the running streams as XRUN

Parameters

struct snd_pcm_substream * substream the PCM substream instance

Description

This stops the given running substream (and all linked substreams) as XRUN. Unlike *snd_pcm_stop()*, this function takes the substream lock by itself.

Return

Zero if successful, or a negative error code.

int **snd_pcm_suspend**(struct snd_pcm_substream * *substream*)
trigger SUSPEND to all linked streams

Parameters

struct snd_pcm_substream * substream the PCM substream

Description

After this call, all streams are changed to SUSPENDED state.

Return

Zero if successful (or **substream** is NULL), or a negative error code.

int **snd_pcm_suspend_all**(struct snd_pcm * *pcm*)
trigger SUSPEND to all substreams in the given pcm

Parameters

struct snd_pcm * pcm the PCM instance

Description

After this call, all streams are changed to SUSPENDED state.

Return

Zero if successful (or **pcm** is NULL), or a negative error code.

int **snd_pcm_prepare**(struct snd_pcm_substream * *substream*, struct file * *file*)
prepare the PCM substream to be triggerable

Parameters

struct snd_pcm_substream * substream the PCM substream instance

struct file * file file to refer f_flags

Return

Zero if successful, or a negative error code.

int **snd_pcm_kernel_ioctl**(struct snd_pcm_substream * *substream*, unsigned int *cmd*, void * *arg*)
Execute PCM ioctl in the kernel-space

Parameters

struct snd_pcm_substream * substream PCM substream

unsigned int cmd IOCTL cmd

void * arg IOCTL argument

Description

The function is provided primarily for OSS layer and USB gadget drivers, and it allows only the limited set of ioctls (hw_params, sw_params, prepare, start, drain, drop, forward).

int **snd_pcm_lib_default_mmap**(struct snd_pcm_substream * *substream*, struct vm_area_struct * *area*)
Default PCM data mmap function

Parameters

struct snd_pcm_substream * substream PCM substream

struct vm_area_struct * area VMA

Description

This is the default mmap handler for PCM data. When mmap pcm_ops is NULL, this function is invoked implicitly.

int **snd_pcm_lib_mmap_iomem**(struct snd_pcm_substream * *substream*, struct vm_area_struct * *area*)
Default PCM data mmap function for I/O mem

Parameters

struct snd_pcm_substream * substream PCM substream

struct vm_area_struct * area VMA

Description

When your hardware uses the iomapped pages as the hardware buffer and wants to mmap it, pass this function as mmap pcm_ops. Note that this is supposed to work only on limited architectures.

int **snd_pcm_stream_linked**(struct snd_pcm_substream * *substream*)
Check whether the substream is linked with others

Parameters

struct snd_pcm_substream * substream substream to check

Description

Returns true if the given substream is being linked with others.

snd_pcm_stream_lock_irqsave(*substream*, *flags*)
Lock the PCM stream

Parameters

substream PCM substream

flags irq flags

Description

This locks the PCM stream like *snd_pcm_stream_lock()* but with the local IRQ (only when nonatomic is false). In nonatomic case, this is identical as *snd_pcm_stream_lock()*.

snd_pcm_group_for_each_entry(*s*, *substream*)
iterate over the linked substreams

Parameters

s the iterator

substream the substream

Description

Iterate over the all linked substreams to the given **substream**. When **substream** isn't linked with any others, this gives returns **substream** itself once.

int **snd_pcm_running**(struct snd_pcm_substream * *substream*)
Check whether the substream is in a running state

Parameters

struct snd_pcm_substream * substream substream to check

Description

Returns true if the given substream is in the state RUNNING, or in the state DRAINING for playback.

ssize_t **bytes_to_samples**(struct snd_pcm_runtime * *runtime*, ssize_t *size*)
Unit conversion of the size from bytes to samples

Parameters

struct snd_pcm_runtime * runtime PCM runtime instance

ssize_t size size in bytes

snd_pcm_sframes_t **bytes_to_frames**(struct snd_pcm_runtime * *runtime*, ssize_t *size*)
Unit conversion of the size from bytes to frames

Parameters

struct snd_pcm_runtime * runtime PCM runtime instance

ssize_t size size in bytes

ssize_t **samples_to_bytes**(struct snd_pcm_runtime * *runtime*, ssize_t *size*)
Unit conversion of the size from samples to bytes

Parameters

struct snd_pcm_runtime * runtime PCM runtime instance

ssize_t size size in samples

ssize_t **frames_to_bytes**(struct snd_pcm_runtime * *runtime*, snd_pcm_sframes_t *size*)
Unit conversion of the size from frames to bytes

Parameters

struct snd_pcm_runtime * runtime PCM runtime instance

snd_pcm_sframes_t size size in frames

int **frame_aligned**(struct snd_pcm_runtime * *runtime*, ssize_t *bytes*)
Check whether the byte size is aligned to frames

Parameters

struct snd_pcm_runtime * runtime PCM runtime instance

ssize_t bytes size in bytes

size_t **snd_pcm_lib_buffer_bytes**(struct snd_pcm_substream * *substream*)
Get the buffer size of the current PCM in bytes

Parameters

struct snd_pcm_substream * substream PCM substream

size_t **snd_pcm_lib_period_bytes**(struct snd_pcm_substream * *substream*)
Get the period size of the current PCM in bytes

Parameters

struct snd_pcm_substream * substream PCM substream

snd_pcm_uframes_t **snd_pcm_playback_avail**(struct snd_pcm_runtime * *runtime*)
Get the available (writable) space for playback

Parameters

struct snd_pcm_runtime * runtime PCM runtime instance

Description

Result is between 0 ... (boundary - 1)

snd_pcm_uframes_t **snd_pcm_capture_avail**(struct snd_pcm_runtime * *runtime*)
Get the available (readable) space for capture

Parameters

struct snd_pcm_runtime * runtime PCM runtime instance

Description

Result is between 0 ... (boundary - 1)

snd_pcm_sframes_t **snd_pcm_playback_hw_avail**(struct snd_pcm_runtime * *runtime*)
Get the queued space for playback

Parameters

struct snd_pcm_runtime * runtime PCM runtime instance

snd_pcm_sframes_t **snd_pcm_capture_hw_avail**(struct snd_pcm_runtime * *runtime*)
Get the free space for capture

Parameters

struct snd_pcm_runtime * runtime PCM runtime instance

int **snd_pcm_playback_ready**(struct snd_pcm_substream * *substream*)
check whether the playback buffer is available

Parameters

struct snd_pcm_substream * substream the pcm substream instance

Description

Checks whether enough free space is available on the playback buffer.

Return

Non-zero if available, or zero if not.

int **snd_pcm_capture_ready**(struct snd_pcm_substream * *substream*)
check whether the capture buffer is available

Parameters

struct snd_pcm_substream * substream the pcm substream instance

Description

Checks whether enough capture data is available on the capture buffer.

Return

Non-zero if available, or zero if not.

int **snd_pcm_playback_data**(struct snd_pcm_substream * *substream*)
check whether any data exists on the playback buffer

Parameters

struct snd_pcm_substream * substream the pcm substream instance

Description

Checks whether any data exists on the playback buffer.

Return

Non-zero if any data exists, or zero if not. If stop_threshold is bigger or equal to boundary, then this function returns always non-zero.

int **snd_pcm_playback_empty**(struct snd_pcm_substream * *substream*)
check whether the playback buffer is empty

Parameters

struct snd_pcm_substream * substream the pcm substream instance

Description

Checks whether the playback buffer is empty.

Return

Non-zero if empty, or zero if not.

int **snd_pcm_capture_empty**(struct snd_pcm_substream * *substream*)
check whether the capture buffer is empty

Parameters

struct snd_pcm_substream * substream the pcm substream instance

Description

Checks whether the capture buffer is empty.

Return

Non-zero if empty, or zero if not.

void **snd_pcm_trigger_done**(struct snd_pcm_substream * *substream*, struct snd_pcm_substream * *master*)
Mark the master substream

Parameters

struct snd_pcm_substream * substream the pcm substream instance

struct snd_pcm_substream * master the linked master substream

Description

When multiple substreams of the same card are linked and the hardware supports the single-shot operation, the driver calls this in the loop in *snd_pcm_group_for_each_entry()* for marking the substream as “done”. Then most of trigger operations are performed only to the given master substream.

The trigger_master mark is cleared at timestamp updates at the end of trigger operations.

unsigned int **params_channels**(const struct snd_pcm_hw_params * *p*)
Get the number of channels from the hw params

Parameters

const struct snd_pcm_hw_params * p hw params

unsigned int **params_rate**(const struct snd_pcm_hw_params * *p*)
Get the sample rate from the hw params

Parameters

const struct snd_pcm_hw_params * p hw params

unsigned int **params_period_size**(const struct snd_pcm_hw_params * *p*)
Get the period size (in frames) from the hw params

Parameters

const struct snd_pcm_hw_params * p hw params

unsigned int **params_periods**(const struct snd_pcm_hw_params * *p*)
Get the number of periods from the hw params

Parameters

const struct snd_pcm_hw_params * p hw params

unsigned int **params_buffer_size**(const struct snd_pcm_hw_params * *p*)
Get the buffer size (in frames) from the hw params

Parameters

const struct snd_pcm_hw_params * p hw params

unsigned int **params_buffer_bytes**(const struct snd_pcm_hw_params * *p*)
Get the buffer size (in bytes) from the hw params

Parameters

const struct snd_pcm_hw_params * p hw params

int **snd_pcm_hw_constraint_single**(struct snd_pcm_runtime * *runtime*, snd_pcm_hw_param_t *var*, unsigned int *val*)
Constrain parameter to a single value

Parameters

struct snd_pcm_runtime * runtime PCM runtime instance

snd_pcm_hw_param_t var The hw_params variable to constrain

unsigned int val The value to constrain to

Return

Positive if the value is changed, zero if it's not changed, or a negative error code.

int **snd_pcm_format_cpu_endian**(snd_pcm_format_t *format*)
Check the PCM format is CPU-endian

Parameters

snd_pcm_format_t format the format to check

Return

1 if the given PCM format is CPU-endian, 0 if opposite, or a negative error code if endian not specified.

void **snd_pcm_set_runtime_buffer**(struct snd_pcm_substream * *substream*, struct snd_dma_buffer * *bufp*)
Set the PCM runtime buffer

Parameters

struct snd_pcm_substream * substream PCM substream to set

struct snd_dma_buffer * bufp the buffer information, NULL to clear

Description

Copy the buffer information to runtime->dma_buffer when **bufp** is non-NULL. Otherwise it clears the current buffer information.

void **snd_pcm_gettime**(struct snd_pcm_runtime * *runtime*, struct timespec * *tv*)
Fill the timespec depending on the timestamp mode

Parameters

struct snd_pcm_runtime * runtime PCM runtime instance

struct timespec * tv timespec to fill

int **snd_pcm_lib_alloc_vmalloc_buffer**(struct snd_pcm_substream * *substream*, size_t *size*)
allocate virtual DMA buffer

Parameters

struct snd_pcm_substream * substream the substream to allocate the buffer to

size_t size the requested buffer size, in bytes

Description

Allocates the PCM substream buffer using `vmalloc()`, i.e., the memory is contiguous in kernel virtual space, but not in physical memory. Use this if the buffer is accessed by kernel code but not by device DMA.

Return

1 if the buffer was changed, 0 if not changed, or a negative error code.

int **snd_pcm_lib_alloc_vmalloc_32_buffer**(struct snd_pcm_substream * *substream*, size_t *size*)
allocate 32-bit-addressable buffer

Parameters

struct snd_pcm_substream * substream the substream to allocate the buffer to

size_t size the requested buffer size, in bytes

Description

This function works like *snd_pcm_lib_alloc_vmalloc_buffer()*, but uses `vmalloc_32()`, i.e., the pages are allocated from 32-bit-addressable memory.

Return

1 if the buffer was changed, 0 if not changed, or a negative error code.

dma_addr_t **snd_pcm_sgbuf_get_addr**(struct snd_pcm_substream * *substream*, unsigned int *ofs*)
Get the DMA address at the corresponding offset

Parameters

struct snd_pcm_substream * substream PCM substream

unsigned int ofs byte offset

void * **snd_pcm_sgbuf_get_ptr**(struct snd_pcm_substream * *substream*, unsigned int *ofs*)
Get the virtual address at the corresponding offset

Parameters

struct snd_pcm_substream * substream PCM substream

unsigned int ofs byte offset

unsigned int **snd_pcm_sgbuf_get_chunk_size**(struct snd_pcm_substream * *substream*, unsigned int *ofs*, unsigned int *size*)
Compute the max size that fits within the contig. page from the given size

Parameters

struct snd_pcm_substream * substream PCM substream

unsigned int ofs byte offset

unsigned int size byte size to examine

void **snd_pcm_mmap_data_open**(struct vm_area_struct * *area*)
increase the mmap counter

Parameters

struct vm_area_struct * area VMA

Description

PCM mmap callback should handle this counter properly

void **snd_pcm_mmap_data_close**(struct vm_area_struct * *area*)
decrease the mmap counter

Parameters

struct vm_area_struct * area VMA

Description

PCM mmap callback should handle this counter properly

void **snd_pcm_limit_isa_dma_size**(int *dma*, size_t * *max*)
Get the max size fitting with ISA DMA transfer

Parameters

int dma DMA number

size_t * max pointer to store the max size

const char * **snd_pcm_stream_str**(struct snd_pcm_substream * *substream*)
Get a string naming the direction of a stream

Parameters

struct snd_pcm_substream * substream the pcm substream instance

Return

A string naming the direction of the stream.

struct snd_pcm_substream * **snd_pcm_chmap_substream**(struct snd_pcm_chmap * *info*, unsigned int *idx*)
get the PCM substream assigned to the given chmap info

Parameters

struct snd_pcm_chmap * info chmap information

unsigned int idx the substream number index

u64 **pcm_format_to_bits**(snd_pcm_format_t *pcm_format*)
Strong-typed conversion of pcm_format to bitwise

Parameters

snd_pcm_format_t pcm_format PCM format

PCM Format Helpers

int **snd_pcm_format_signed**(snd_pcm_format_t *format*)
Check the PCM format is signed linear

Parameters

snd_pcm_format_t format the format to check

Return

1 if the given PCM format is signed linear, 0 if unsigned linear, and a negative error code for non-linear formats.

int **snd_pcm_format_unsigned**(snd_pcm_format_t *format*)
Check the PCM format is unsigned linear

Parameters

snd_pcm_format_t format the format to check

Return

1 if the given PCM format is unsigned linear, 0 if signed linear, and a negative error code for non-linear formats.

int **snd_pcm_format_linear**(snd_pcm_format_t *format*)
Check the PCM format is linear

Parameters

snd_pcm_format_t format the format to check

Return

1 if the given PCM format is linear, 0 if not.

int **snd_pcm_format_little_endian**(snd_pcm_format_t *format*)
Check the PCM format is little-endian

Parameters

snd_pcm_format_t format the format to check

Return

1 if the given PCM format is little-endian, 0 if big-endian, or a negative error code if endian not specified.

int **snd_pcm_format_big_endian**(snd_pcm_format_t *format*)
Check the PCM format is big-endian

Parameters

snd_pcm_format_t format the format to check

Return

1 if the given PCM format is big-endian, 0 if little-endian, or a negative error code if endian not specified.

int **snd_pcm_format_width**(snd_pcm_format_t *format*)
return the bit-width of the format

Parameters

snd_pcm_format_t format the format to check

Return

The bit-width of the format, or a negative error code if unknown format.

int **snd_pcm_format_physical_width**(snd_pcm_format_t *format*)
return the physical bit-width of the format

Parameters

snd_pcm_format_t format the format to check

Return

The physical bit-width of the format, or a negative error code if unknown format.

ssize_t **snd_pcm_format_size**(snd_pcm_format_t *format*, size_t *samples*)
return the byte size of samples on the given format

Parameters

snd_pcm_format_t format the format to check

size_t samples sampling rate

Return

The byte size of the given samples for the format, or a negative error code if unknown format.

const unsigned char * **snd_pcm_format_silence_64**(snd_pcm_format_t *format*)
return the silent data in 8 bytes array

Parameters

snd_pcm_format_t format the format to check

Return

The format pattern to fill or NULL if error.

int **snd_pcm_format_set_silence**(snd_pcm_format_t *format*, void * *data*, unsigned int *samples*)
set the silence data on the buffer

Parameters

snd_pcm_format_t format the PCM format

void * data the buffer pointer

unsigned int samples the number of samples to set silence

Description

Sets the silence data on the buffer for the given samples.

Return

Zero if successful, or a negative error code on failure.

int **snd_pcm_limit_hw_rates**(struct snd_pcm_runtime * *runtime*)
determine rate_min/rate_max fields

Parameters

struct snd_pcm_runtime * runtime the runtime instance

Description

Determines the rate_min and rate_max fields from the rates bits of the given runtime->hw.

Return

Zero if successful.

unsigned int **snd_pcm_rate_to_rate_bit**(unsigned int *rate*)
converts sample rate to SNDRV_PCM_RATE_xxx bit

Parameters

unsigned int rate the sample rate to convert

Return

The SNDRV_PCM_RATE_xxx flag that corresponds to the given rate, or SNDRV_PCM_RATE_KNOT for an unknown rate.

unsigned int **snd_pcm_rate_bit_to_rate**(unsigned int *rate_bit*)
converts SNDRV_PCM_RATE_xxx bit to sample rate

Parameters

unsigned int rate_bit the rate bit to convert

Return

The sample rate that corresponds to the given SNDRV_PCM_RATE_xxx flag or 0 for an unknown rate bit.

unsigned int **snd_pcm_rate_mask_intersect**(unsigned int *rates_a*, unsigned int *rates_b*)
computes the intersection between two rate masks

Parameters

unsigned int rates_a The first rate mask

unsigned int rates_b The second rate mask

Description

This function computes the rates that are supported by both rate masks passed to the function. It will take care of the special handling of SNDRV_PCM_RATE_CONTINUOUS and SNDRV_PCM_RATE_KNOT.

Return

A rate mask containing the rates that are supported by both rates_a and rates_b.

unsigned int **snd_pcm_rate_range_to_bits**(unsigned int *rate_min*, unsigned int *rate_max*)
converts rate range to SNDRV_PCM_RATE_xxx bit

Parameters

unsigned int rate_min the minimum sample rate

unsigned int rate_max the maximum sample rate

Description

This function has an implicit assumption: the rates in the given range have only the pre-defined rates like 44100 or 16000.

Return

The SNDRV_PCM_RATE_xxx flag that corresponds to the given rate range, or SNDRV_PCM_RATE_KNOT for an unknown range.

PCM Memory Management

int **snd_pcm_lib_preallocate_free**(struct snd_pcm_substream * *substream*)
release the preallocated buffer of the specified substream.

Parameters

struct snd_pcm_substream * substream the pcm substream instance

Description

Releases the pre-allocated buffer of the given substream.

Return

Zero if successful, or a negative error code on failure.

int **snd_pcm_lib_preallocate_free_for_all**(struct snd_pcm * *pcm*)
release all pre-allocated buffers on the pcm

Parameters

struct snd_pcm * pcm the pcm instance

Description

Releases all the pre-allocated buffers on the given pcm.

Return

Zero if successful, or a negative error code on failure.

int **snd_pcm_lib_preallocate_pages**(struct snd_pcm_substream * *substream*, int *type*, struct device * *data*, size_t *size*, size_t *max*)
pre-allocation for the given DMA type

Parameters

struct snd_pcm_substream * substream the pcm substream instance

int type DMA type (SNDRV_DMA_TYPE_*)

struct device * data DMA type dependent data

size_t size the requested pre-allocation size in bytes

size_t max the max. allowed pre-allocation size

Description

Do pre-allocation for the given DMA buffer type.

Return

Zero if successful, or a negative error code on failure.

int **snd_pcm_lib_preallocate_pages_for_all**(struct snd_pcm * *pcm*, int *type*, void * *data*, size_t *size*, size_t *max*)
pre-allocation for continuous memory type (all substreams)

Parameters

struct snd_pcm * pcm the pcm instance

int type DMA type (SNDRV_DMA_TYPE_*)

void * data DMA type dependent data

size_t size the requested pre-allocation size in bytes

size_t max the max. allowed pre-allocation size

Description

Do pre-allocation to all substreams of the given pcm for the specified DMA type.

Return

Zero if successful, or a negative error code on failure.

struct page * **snd_pcm_sgbuf_ops_page**(struct snd_pcm_substream * *substream*, unsigned long *offset*)
get the page struct at the given offset

Parameters

struct snd_pcm_substream * substream the pcm substream instance

unsigned long offset the buffer offset

Description

Used as the page callback of PCM ops.

Return

The page struct at the given buffer offset. `NULL` on failure.

int **snd_pcm_lib_malloc_pages**(struct snd_pcm_substream * *substream*, size_t *size*)
allocate the DMA buffer

Parameters

struct snd_pcm_substream * substream the substream to allocate the DMA buffer to

size_t size the requested buffer size in bytes

Description

Allocates the DMA buffer on the BUS type given earlier to `snd_pcm_lib_preallocate_xxx_pages()`.

Return

1 if the buffer is changed, 0 if not changed, or a negative code on failure.

int **snd_pcm_lib_free_pages**(struct snd_pcm_substream * *substream*)
release the allocated DMA buffer.

Parameters

struct snd_pcm_substream * substream the substream to release the DMA buffer

Description

Releases the DMA buffer allocated via *snd_pcm_lib_malloc_pages()*.

Return

Zero if successful, or a negative error code on failure.

int **snd_pcm_lib_free_vmalloc_buffer**(struct snd_pcm_substream * *substream*)
free vmalloc buffer

Parameters

struct snd_pcm_substream * substream the substream with a buffer allocated by *snd_pcm_lib_alloc_vmalloc_buffer()*

Return

Zero if successful, or a negative error code on failure.

struct page * **snd_pcm_lib_get_vmalloc_page**(struct snd_pcm_substream * *substream*, unsigned long *offset*)
map vmalloc buffer offset to page struct

Parameters

struct snd_pcm_substream * substream the substream with a buffer allocated by *snd_pcm_lib_alloc_vmalloc_buffer()*

unsigned long offset offset in the buffer

Description

This function is to be used as the page callback in the PCM ops.

Return

The page struct, or NULL on failure.

PCM DMA Engine API

int **snd_hwparams_to_dma_slave_config**(const struct snd_pcm_substream * *substream*, const struct snd_pcm_hw_params * *params*, struct dma_slave_config * *slave_config*)
Convert hw_params to dma_slave_config

Parameters

const struct snd_pcm_substream * substream PCM substream

const struct snd_pcm_hw_params * params hw_params

struct dma_slave_config * slave_config DMA slave config

Description

This function can be used to initialize a dma_slave_config from a substream and hw_params in a dmaengine based PCM driver implementation.

void **snd_dmaengine_pcm_set_config_from_dai_data**(const struct snd_pcm_substream * *substream*, const struct *snd_dmaengine_dai_dma_data* * *dma_data*, struct dma_slave_config * *slave_config*)
Initializes a dma slave config using DAI DMA data.

Parameters

const struct snd_pcm_substream * substream PCM substream

const struct snd_dmaengine_dai_dma_data * dma_data DAI DMA data

struct dma_slave_config * slave_config DMA slave configuration

Description

Initializes the {dst,src}_addr, {dst,src}_maxburst, {dst,src}_addr_width and slave_id fields of the DMA slave config from the same fields of the DAI DMA data struct. The src and dst fields will be initialized depending on the direction of the substream. If the substream is a playback stream the dst fields will be initialized, if it is a capture stream the src fields will be initialized. The {dst,src}_addr_width field will only be initialized if the SND_DMAENGINE_PCM_DAI_FLAG_PACK flag is set or if the addr_width field of the DAI DMA data struct is not equal to DMA_SLAVE_BUSWIDTH_UNDEFINED. If both conditions are met the latter takes priority.

int **snd_dmaengine_pcm_trigger**(struct snd_pcm_substream * *substream*, int *cmd*)
dmaengine based PCM trigger implementation

Parameters

struct snd_pcm_substream * substream PCM substream

int cmd Trigger command

Description

Returns 0 on success, a negative error code otherwise.

This function can be used as the PCM trigger callback for dmaengine based PCM driver implementations.

snd_pcm_uframes_t **snd_dmaengine_pcm_pointer_no_residue**(struct snd_pcm_substream * *substream*)
dmaengine based PCM pointer implementation

Parameters

struct snd_pcm_substream * substream PCM substream

Description

This function is deprecated and should not be used by new drivers, as its results may be unreliable.

snd_pcm_uframes_t **snd_dmaengine_pcm_pointer**(struct snd_pcm_substream * *substream*)
dmaengine based PCM pointer implementation

Parameters

struct snd_pcm_substream * substream PCM substream

Description

This function can be used as the PCM pointer callback for dmaengine based PCM driver implementations.

struct dma_chan * **snd_dmaengine_pcm_request_channel**(dma_filter_fn *filter_fn*, void * *filter_data*)
Request channel for the dmaengine PCM

Parameters

dma_filter_fn filter_fn Filter function used to request the DMA channel

void * filter_data Data passed to the DMA filter function

Description

Returns NULL or the requested DMA channel.

This function request a DMA channel for usage with dmaengine PCM.

int **snd_dmaengine_pcm_open**(struct snd_pcm_substream * *substream*, struct dma_chan * *chan*)
Open a dmaengine based PCM substream

Parameters

struct snd_pcm_substream * substream PCM substream

struct dma_chan * chan DMA channel to use for data transfers

Description

Returns 0 on success, a negative error code otherwise.

The function should usually be called from the pcm open callback. Note that this function will use private_data field of the substream's runtime. So it is not available to your pcm driver implementation.

int **snd_dmaengine_pcm_open_request_chan**(struct snd_pcm_substream * *substream*, dma_filter_fn *filter_fn*, void * *filter_data*)
Open a dmaengine based PCM substream and request channel

Parameters

struct snd_pcm_substream * substream PCM substream

dma_filter_fn filter_fn Filter function used to request the DMA channel

void * filter_data Data passed to the DMA filter function

Description

Returns 0 on success, a negative error code otherwise.

This function will request a DMA channel using the passed filter function and data. The function should usually be called from the pcm open callback. Note that this function will use private_data field of the substream's runtime. So it is not available to your pcm driver implementation.

int **snd_dmaengine_pcm_close**(struct snd_pcm_substream * *substream*)
Close a dmaengine based PCM substream

Parameters

struct snd_pcm_substream * substream PCM substream

int **snd_dmaengine_pcm_close_release_chan**(struct snd_pcm_substream * *substream*)
Close a dmaengine based PCM substream and release channel

Parameters

struct snd_pcm_substream * substream PCM substream

Description

Releases the DMA channel associated with the PCM substream.

enum dma_transfer_direction **snd_pcm_substream_to_dma_direction**(const struct snd_pcm_substream * *substream*)
Get dma_transfer_direction for a PCM substream

Parameters

const struct snd_pcm_substream * substream PCM substream

struct **snd_dmaengine_dai_dma_data**
DAI DMA configuration data

Definition

```
struct snd_dmaengine_dai_dma_data {
  dma_addr_t addr;
  enum dma_slave_buswidth addr_width;
  u32 maxburst;
  unsigned int slave_id;
  void *filter_data;
```

```
  const char *chan_name;
  unsigned int fifo_size;
  unsigned int flags;
};
```

Members

addr Address of the DAI data source or destination register.

addr_width Width of the DAI data source or destination register.

maxburst Maximum number of words(note: words, as in units of the src_addr_width member, not bytes) that can be send to or received from the DAI in one burst.

slave_id Slave requester id for the DMA channel.

filter_data Custom DMA channel filter data, this will usually be used when requesting the DMA channel.

chan_name Custom channel name to use when requesting DMA channel.

fifo_size FIFO size of the DAI controller in bytes

flags PCM_DAI flags, only SND_DMAENGINE_PCM_DAI_FLAG_PACK for now

struct **snd_dmaengine_pcm_config**
Configuration data for dmaengine based PCM

Definition

```
struct snd_dmaengine_pcm_config {
  int (*prepare_slave_config)(struct snd_pcm_substream *substream,struct snd_pcm_hw_params *params, stru
  struct dma_chan *(*compat_request_channel)(struct snd_soc_pcm_runtime *rtd, struct snd_pcm_substream *
  int (*process)(struct snd_pcm_substream *substream,int channel, unsigned long hwoff, void *buf, unsigr
  dma_filter_fn compat_filter_fn;
  struct device *dma_dev;
  const char *chan_names[SNDRV_PCM_STREAM_LAST + 1];
  const struct snd_pcm_hardware *pcm_hardware;
  unsigned int prealloc_buffer_size;
};
```

Members

prepare_slave_config Callback used to fill in the DMA slave_config for a PCM substream. Will be called from the PCM drivers hwparams callback.

compat_request_channel Callback to request a DMA channel for platforms which do not use devicetree.

process Callback used to apply processing on samples transferred from/to user space.

compat_filter_fn Will be used as the filter function when requesting a channel for platforms which do not use devicetree. The filter parameter will be the DAI's DMA data.

dma_dev If set, request DMA channel on this device rather than the DAI device.

chan_names If set, these custom DMA channel names will be requested at registration time.

pcm_hardware snd_pcm_hardware struct to be used for the PCM.

prealloc_buffer_size Size of the preallocated audio buffer.

Note

If both compat_request_channel and compat_filter_fn are set compat_request_channel will be used to request the channel and compat_filter_fn will be ignored. Otherwise the channel will be requested using dma_request_channel with compat_filter_fn as the filter function.

Control/Mixer API

General Control Interface

void **snd_ctl_notify**(struct snd_card * *card*, unsigned int *mask*, struct snd_ctl_elem_id * *id*)
Send notification to user-space for a control change

Parameters

struct snd_card * card the card to send notification

unsigned int mask the event mask, SNDRV_CTL_EVENT_*

struct snd_ctl_elem_id * id the ctl element id to send notification

Description

This function adds an event record with the given id and mask, appends to the list and wakes up the user-space for notification. This can be called in the atomic context.

int **snd_ctl_new**(struct snd_kcontrol ** *kctl*, unsigned int *count*, unsigned int *access*, struct snd_ctl_file * *file*)
create a new control instance with some elements

Parameters

struct snd_kcontrol ** kctl the pointer to store new control instance

unsigned int count the number of elements in this control

unsigned int access the default access flags for elements in this control

struct snd_ctl_file * file given when locking these elements

Description

Allocates a memory object for a new control instance. The instance has elements as many as the given number (**count**). Each element has given access permissions (**access**). Each element is locked when **file** is given.

Return

0 on success, error code on failure

struct snd_kcontrol * **snd_ctl_new1**(const struct snd_kcontrol_new * *ncontrol*, void * *private_data*)
create a control instance from the template

Parameters

const struct snd_kcontrol_new * ncontrol the initialization record

void * private_data the private data to set

Description

Allocates a new struct snd_kcontrol instance and initialize from the given template. When the access field of ncontrol is 0, it's assumed as READWRITE access. When the count field is 0, it's assumes as one.

Return

The pointer of the newly generated instance, or NULL on failure.

void **snd_ctl_free_one**(struct snd_kcontrol * *kcontrol*)
release the control instance

Parameters

struct snd_kcontrol * kcontrol the control instance

Description

Releases the control instance created via *snd_ctl_new()* or *snd_ctl_new1()*. Don't call this after the control was added to the card.

int **snd_ctl_add**(struct snd_card * *card*, struct snd_kcontrol * *kcontrol*)
add the control instance to the card

Parameters

struct snd_card * card the card instance

struct snd_kcontrol * kcontrol the control instance to add

Description

Adds the control instance created via *snd_ctl_new()* or *snd_ctl_new1()* to the given card. Assigns also an unique numid used for fast search.

It frees automatically the control which cannot be added.

Return

Zero if successful, or a negative error code on failure.

int **snd_ctl_replace**(struct snd_card * *card*, struct snd_kcontrol * *kcontrol*, bool *add_on_replace*)
replace the control instance of the card

Parameters

struct snd_card * card the card instance

struct snd_kcontrol * kcontrol the control instance to replace

bool add_on_replace add the control if not already added

Description

Replaces the given control. If the given control does not exist and the add_on_replace flag is set, the control is added. If the control exists, it is destroyed first.

It frees automatically the control which cannot be added or replaced.

Return

Zero if successful, or a negative error code on failure.

int **snd_ctl_remove**(struct snd_card * *card*, struct snd_kcontrol * *kcontrol*)
remove the control from the card and release it

Parameters

struct snd_card * card the card instance

struct snd_kcontrol * kcontrol the control instance to remove

Description

Removes the control from the card and then releases the instance. You don't need to call *snd_ctl_free_one()*. You must be in the write lock - down_write(`card->controls_rwsem`).

Return

0 if successful, or a negative error code on failure.

int **snd_ctl_remove_id**(struct snd_card * *card*, struct snd_ctl_elem_id * *id*)
remove the control of the given id and release it

Parameters

struct snd_card * card the card instance

struct snd_ctl_elem_id * id the control id to remove

Description

Finds the control instance with the given id, removes it from the card list and releases it.

Return

0 if successful, or a negative error code on failure.

int **snd_ctl_remove_user_ctl**(struct snd_ctl_file * *file*, struct snd_ctl_elem_id * *id*)
remove and release the unlocked user control

Parameters

struct snd_ctl_file * file active control handle

struct snd_ctl_elem_id * id the control id to remove

Description

Finds the control instance with the given id, removes it from the card list and releases it.

Return

0 if successful, or a negative error code on failure.

int **snd_ctl_activate_id**(struct snd_card * *card*, struct snd_ctl_elem_id * *id*, int *active*)
activate/inactivate the control of the given id

Parameters

struct snd_card * card the card instance

struct snd_ctl_elem_id * id the control id to activate/inactivate

int active non-zero to activate

Description

Finds the control instance with the given id, and activate or inactivate the control together with notification, if changed. The given ID data is filled with full information.

Return

0 if unchanged, 1 if changed, or a negative error code on failure.

int **snd_ctl_rename_id**(struct snd_card * *card*, struct snd_ctl_elem_id * *src_id*, struct snd_ctl_elem_id * *dst_id*)
replace the id of a control on the card

Parameters

struct snd_card * card the card instance

struct snd_ctl_elem_id * src_id the old id

struct snd_ctl_elem_id * dst_id the new id

Description

Finds the control with the old id from the card, and replaces the id with the new one.

Return

Zero if successful, or a negative error code on failure.

struct snd_kcontrol * **snd_ctl_find_numid**(struct snd_card * *card*, unsigned int *numid*)
find the control instance with the given number-id

Parameters

struct snd_card * card the card instance

unsigned int numid the number-id to search

Description

Finds the control instance with the given number-id from the card.

The caller must down card->controls_rwsem before calling this function (if the race condition can happen).

Return

The pointer of the instance if found, or NULL if not.

struct snd_kcontrol * **snd_ctl_find_id**(struct snd_card * *card*, struct snd_ctl_elem_id * *id*)
find the control instance with the given id

Parameters

struct snd_card * card the card instance

struct snd_ctl_elem_id * id the id to search

Description

Finds the control instance with the given id from the card.

The caller must down card->controls_rwsem before calling this function (if the race condition can happen).

Return

The pointer of the instance if found, or NULL if not.

int **snd_ctl_register_ioctl**(snd_kctl_ioctl_func_t *fcn*)
register the device-specific control-ioctls

Parameters

snd_kctl_ioctl_func_t fcn ioctl callback function

Description

called from each device manager like pcm.c, hwdep.c, etc.

int **snd_ctl_register_ioctl_compat**(snd_kctl_ioctl_func_t *fcn*)
register the device-specific 32bit compat control-ioctls

Parameters

snd_kctl_ioctl_func_t fcn ioctl callback function

int **snd_ctl_unregister_ioctl**(snd_kctl_ioctl_func_t *fcn*)
de-register the device-specific control-ioctls

Parameters

snd_kctl_ioctl_func_t fcn ioctl callback function to unregister

int **snd_ctl_unregister_ioctl_compat**(snd_kctl_ioctl_func_t *fcn*)
de-register the device-specific compat 32bit control-ioctls

Parameters

snd_kctl_ioctl_func_t fcn ioctl callback function to unregister

int **snd_ctl_boolean_mono_info**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_info * *uinfo*)
Helper function for a standard boolean info callback with a mono channel

Parameters

struct snd_kcontrol * kcontrol the kcontrol instance

struct snd_ctl_elem_info * uinfo info to store

Description

This is a function that can be used as info callback for a standard boolean control with a single mono channel.

int **snd_ctl_boolean_stereo_info**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_info * *uinfo*)
Helper function for a standard boolean info callback with stereo two channels

Parameters

struct snd_kcontrol * kcontrol the kcontrol instance

struct snd_ctl_elem_info * uinfo info to store

Description

This is a function that can be used as info callback for a standard boolean control with stereo two channels.

int **snd_ctl_enum_info**(struct snd_ctl_elem_info * *info*, unsigned int *channels*, unsigned int *items*, const char *const *names*)
fills the info structure for an enumerated control

Parameters

struct snd_ctl_elem_info * info the structure to be filled

unsigned int channels the number of the control's channels; often one

unsigned int items the number of control values; also the size of **names**

const char *const names an array containing the names of all control values

Description

Sets all required fields in **info** to their appropriate values. If the control's accessibility is not the default (readable and writable), the caller has to fill **info**->access.

Return

Zero.

AC97 Codec API

void **snd_ac97_write**(struct snd_ac97 * *ac97*, unsigned short *reg*, unsigned short *value*)
write a value on the given register

Parameters

struct snd_ac97 * ac97 the ac97 instance

unsigned short reg the register to change

unsigned short value the value to set

Description

Writes a value on the given register. This will invoke the write callback directly after the register check. This function doesn't change the register cache unlike #:c:func:*snd_ca97_write_cache()*, so use this only when you don't want to reflect the change to the suspend/resume state.

unsigned short **snd_ac97_read**(struct snd_ac97 * *ac97*, unsigned short *reg*)
read a value from the given register

Parameters

struct snd_ac97 * ac97 the ac97 instance

unsigned short reg the register to read

Description

Reads a value from the given register. This will invoke the read callback directly after the register check.

Return

The read value.

void **snd_ac97_write_cache**(struct snd_ac97 * *ac97*, unsigned short *reg*, unsigned short *value*)
write a value on the given register and update the cache

Parameters

struct snd_ac97 * ac97 the ac97 instance

unsigned short reg the register to change

unsigned short value the value to set

Description

Writes a value on the given register and updates the register cache. The cached values are used for the cached-read and the suspend/resume.

int **snd_ac97_update**(struct snd_ac97 * *ac97*, unsigned short *reg*, unsigned short *value*)
update the value on the given register

Parameters

struct snd_ac97 * ac97 the ac97 instance

unsigned short reg the register to change

unsigned short value the value to set

Description

Compares the value with the register cache and updates the value only when the value is changed.

Return

1 if the value is changed, 0 if no change, or a negative code on failure.

int **snd_ac97_update_bits**(struct snd_ac97 * *ac97*, unsigned short *reg*, unsigned short *mask*, unsigned short *value*)
update the bits on the given register

Parameters

struct snd_ac97 * ac97 the ac97 instance

unsigned short reg the register to change

unsigned short mask the bit-mask to change

unsigned short value the value to set

Description

Updates the masked-bits on the given register only when the value is changed.

Return

1 if the bits are changed, 0 if no change, or a negative code on failure.

const char * **snd_ac97_get_short_name**(struct snd_ac97 * *ac97*)
retrieve codec name

Parameters

struct snd_ac97 * ac97 the codec instance

Return

The short identifying name of the codec.

int **snd_ac97_bus**(struct snd_card * *card*, int *num*, struct snd_ac97_bus_ops * *ops*, void * *private_data*, struct *snd_ac97_bus* ** *rbus*)
create an AC97 bus component

Parameters

struct snd_card * card the card instance

int num the bus number

struct snd_ac97_bus_ops * ops the bus callbacks table

void * private_data private data pointer for the new instance

struct snd_ac97_bus ** rbus the pointer to store the new AC97 bus instance.

Description

Creates an AC97 bus component. An struct snd_ac97_bus instance is newly allocated and initialized.

The ops table must include valid callbacks (at least read and write). The other callbacks, wait and reset, are not mandatory.

The clock is set to 48000. If another clock is needed, set `(*rbus)->clock` manually.

The AC97 bus instance is registered as a low-level device, so you don't have to release it manually.

Return

Zero if successful, or a negative error code on failure.

int **snd_ac97_mixer**(struct *snd_ac97_bus* * *bus*, struct snd_ac97_template * *template*, struct snd_ac97 ** *rac97*)
create an Codec97 component

Parameters

struct snd_ac97_bus * bus the AC97 bus which codec is attached to

struct snd_ac97_template * template the template of ac97, including index, callbacks and the private data.

struct snd_ac97 ** rac97 the pointer to store the new ac97 instance.

Description

Creates an Codec97 component. An struct snd_ac97 instance is newly allocated and initialized from the template. The codec is then initialized by the standard procedure.

The template must include the codec number (num) and address (addr), and the private data (private_data).

The ac97 instance is registered as a low-level device, so you don't have to release it manually.

Return

Zero if successful, or a negative error code on failure.

int **snd_ac97_update_power**(struct snd_ac97 * *ac97*, int *reg*, int *powerup*)
update the powerdown register

Parameters

struct snd_ac97 * ac97 the codec instance

int reg the rate register, e.g. AC97_PCM_FRONT_DAC_RATE

int powerup non-zero when power up the part

Description

Update the AC97 powerdown register bits of the given part.

Return

Zero.

void **snd_ac97_suspend**(struct snd_ac97 * *ac97*)
General suspend function for AC97 codec

Parameters

struct snd_ac97 * ac97 the ac97 instance

Description

Suspends the codec, power down the chip.

void **snd_ac97_resume**(struct snd_ac97 * *ac97*)
General resume function for AC97 codec

Parameters

struct snd_ac97 * ac97 the ac97 instance

Description

Do the standard resume procedure, power up and restoring the old register values.

int **snd_ac97_tune_hardware**(struct snd_ac97 * *ac97*, const struct ac97_quirk * *quirk*, const char * *override*)
tune up the hardware

Parameters

struct snd_ac97 * ac97 the ac97 instance

const struct ac97_quirk * quirk quirk list

const char * override explicit quirk value (overrides the list if non-NULL)

Description

Do some workaround for each pci device, such as renaming of the headphone (true line-out) control as “Master”. The quirk-list must be terminated with a zero-filled entry.

Return

Zero if successful, or a negative error code on failure.

int **snd_ac97_set_rate**(struct snd_ac97 * *ac97*, int *reg*, unsigned int *rate*)
change the rate of the given input/output.

Parameters

struct snd_ac97 * ac97 the ac97 instance

int reg the register to change

unsigned int rate the sample rate to set

Description

Changes the rate of the given input/output on the codec. If the codec doesn’t support VAR, the rate must be 48000 (except for SPDIF).

The valid registers are AC97_PMC_MIC_ADC_RATE, AC97_PCM_FRONT_DAC_RATE, AC97_PCM_LR_ADC_RATE. AC97_PCM_SURR_DAC_RATE and AC97_PCM_LFE_DAC_RATE are accepted if the codec supports them. AC97_SPDIF is accepted as a pseudo register to modify the SPDIF status bits.

Return

Zero if successful, or a negative error code on failure.

int **snd_ac97_pcm_assign**(struct *snd_ac97_bus* * *bus*, unsigned short *pcms_count*, const struct ac97_pcm * *pcms*)
assign AC97 slots to given PCM streams

Parameters

struct snd_ac97_bus * bus the ac97 bus instance

unsigned short pcms_count count of PCMs to be assigned

const struct ac97_pcm * pcms PCMs to be assigned

Description

It assigns available AC97 slots for given PCMs. If none or only some slots are available, pcm->xxx.slots and pcm->xxx.rslots[] members are reduced and might be zero.

Return

Zero if successful, or a negative error code on failure.

int **snd_ac97_pcm_open**(struct ac97_pcm * *pcm*, unsigned int *rate*, enum ac97_pcm_cfg *cfg*, unsigned short *slots*)
opens the given AC97 pcm

Parameters

struct ac97_pcm * pcm the ac97 pcm instance

unsigned int rate rate in Hz, if codec does not support VRA, this value must be 48000Hz

enum ac97_pcm_cfg cfg output stream characteristics

unsigned short slots a subset of allocated slots (snd_ac97_pcm_assign) for this pcm

Description

It locks the specified slots and sets the given rate to AC97 registers.

Return

Zero if successful, or a negative error code on failure.

int **snd_ac97_pcm_close**(struct ac97_pcm * *pcm*)
closes the given AC97 pcm

Parameters

struct ac97_pcm * pcm the ac97 pcm instance

Description

It frees the locked AC97 slots.

Return

Zero.

int **snd_ac97_pcm_double_rate_rules**(struct snd_pcm_runtime * *runtime*)
set double rate constraints

Parameters

struct snd_pcm_runtime * runtime the runtime of the ac97 front playback pcm

Description

Installs the hardware constraint rules to prevent using double rates and more than two channels at the same time.

Return

Zero if successful, or a negative error code on failure.

Virtual Master Control API

struct snd_kcontrol * **snd_ctl_make_virtual_master**(char * *name*, const unsigned int * *tlv*)
Create a virtual master control

Parameters

char * name name string of the control element to create

const unsigned int * tlv optional TLV int array for dB information

Description

Creates a virtual master control with the given name string.

After creating a vmaster element, you can add the slave controls via *snd_ctl_add_slave()* or *snd_ctl_add_slave_uncached()*.

The optional argument **tlv** can be used to specify the TLV information for dB scale of the master control. It should be a single element with #SNDRV_CTL_TLVT_DB_SCALE, #SNDRV_CTL_TLV_DB_MINMAX or #SNDRV_CTL_TLVT_DB_MINMAX_MUTE type, and should be the max 0dB.

Return

The created control element, or NULL for errors (ENOMEM).

int **snd_ctl_add_vmaster_hook**(struct snd_kcontrol * *kcontrol*, void (*hook) (void **private_data*, int, void * *private_data*)
Add a hook to a vmaster control

Parameters

struct snd_kcontrol * kcontrol vmaster kctl element

void (*)(void *private_data, int) hook the hook function

void * private_data the private_data pointer to be saved

Description

Adds the given hook to the vmaster control element so that it's called at each time when the value is changed.

Return

Zero.

void **snd_ctl_sync_vmaster**(struct snd_kcontrol * *kcontrol*, bool *hook_only*)
Sync the vmaster slaves and hook

Parameters

struct snd_kcontrol * kcontrol vmaster kctl element

bool hook_only sync only the hook

Description

Forcibly call the put callback of each slave and call the hook function to synchronize with the current value of the given vmaster element. NOP when NULL is passed to **kcontrol**.

int **snd_ctl_apply_vmaster_slaves**(struct snd_kcontrol * *kctl*, int (*func) (struct snd_kcontrol *vslave, struct snd_kcontrol *slave, void *arg, void * *arg*)
Apply function to each vmaster slave

Parameters

struct snd_kcontrol * kctl vmaster kctl element

int (*)(struct snd_kcontrol *vslave, struct snd_kcontrol *slave, void *arg) func
function to apply

void * arg optional function argument

Description

Apply the function **func** to each slave kctl of the given vmaster kctl. Returns 0 if successful, or a negative error code.

int **snd_ctl_add_slave**(struct snd_kcontrol * *master*, struct snd_kcontrol * *slave*)
Add a virtual slave control

Parameters

struct snd_kcontrol * master vmaster element

struct snd_kcontrol * slave slave element to add

Description

Add a virtual slave control to the given master element created via `snd_ctl_create_virtual_master()` beforehand.

All slaves must be the same type (returning the same information via info callback). The function doesn't check it, so it's your responsibility.

Also, some additional limitations: at most two channels, logarithmic volume control (dB level) thus no linear volume, master can only attenuate the volume without gain

Return

Zero if successful or a negative error code.

int **snd_ctl_add_slave_uncached**(struct snd_kcontrol * *master*, struct snd_kcontrol * *slave*)
Add a virtual slave control

Parameters

struct snd_kcontrol * master vmaster element

struct snd_kcontrol * slave slave element to add

Description

Add a virtual slave control to the given master. Unlike *snd_ctl_add_slave()*, the element added via this function is supposed to have volatile values, and get callback is called at each time queried from the master.

When the control peeks the hardware values directly and the value can be changed by other means than the put callback of the element, this function should be used to keep the value always up-to-date.

Return

Zero if successful or a negative error code.

MIDI API

Raw MIDI API

int **snd_rawmidi_receive**(struct snd_rawmidi_substream * *substream*, const unsigned char * *buffer*, int *count*)
receive the input data from the device

Parameters

struct snd_rawmidi_substream * substream the rawmidi substream

const unsigned char * buffer the buffer pointer

int count the data size to read

Description

Reads the data from the internal buffer.

Return

The size of read data, or a negative error code on failure.

int **snd_rawmidi_transmit_empty**(struct snd_rawmidi_substream * *substream*)
check whether the output buffer is empty

Parameters

struct snd_rawmidi_substream * substream the rawmidi substream

Return

1 if the internal output buffer is empty, 0 if not.

int **__snd_rawmidi_transmit_peek**(struct snd_rawmidi_substream * *substream*, unsigned char * *buffer*, int *count*)
copy data from the internal buffer

Parameters

struct snd_rawmidi_substream * substream the rawmidi substream

unsigned char * buffer the buffer pointer

int count data size to transfer

Description

This is a variant of *snd_rawmidi_transmit_peek()* without spinlock.

int **snd_rawmidi_transmit_peek**(struct snd_rawmidi_substream * *substream*, unsigned char * *buffer*, int *count*)
copy data from the internal buffer

Parameters

struct snd_rawmidi_substream * substream the rawmidi substream

unsigned char * buffer the buffer pointer

int count data size to transfer

Description

Copies data from the internal output buffer to the given buffer.

Call this in the interrupt handler when the midi output is ready, and call *snd_rawmidi_transmit_ack()* after the transmission is finished.

Return

The size of copied data, or a negative error code on failure.

int **__snd_rawmidi_transmit_ack**(struct snd_rawmidi_substream * *substream*, int *count*)
acknowledge the transmission

Parameters

struct snd_rawmidi_substream * substream the rawmidi substream

int count the transferred count

Description

This is a variant of *__snd_rawmidi_transmit_ack()* without spinlock.

int **snd_rawmidi_transmit_ack**(struct snd_rawmidi_substream * *substream*, int *count*)
acknowledge the transmission

Parameters

struct snd_rawmidi_substream * substream the rawmidi substream

int count the transferred count

Description

Advances the hardware pointer for the internal output buffer with the given size and updates the condition. Call after the transmission is finished.

Return

The advanced size if successful, or a negative error code on failure.

int **snd_rawmidi_transmit**(struct snd_rawmidi_substream * *substream*, unsigned char * *buffer*, int *count*)
copy from the buffer to the device

Parameters

struct snd_rawmidi_substream * substream the rawmidi substream

unsigned char * buffer the buffer pointer

int count the data size to transfer

Description

Copies data from the buffer to the device and advances the pointer.

Return

The copied size if successful, or a negative error code on failure.

int **snd_rawmidi_new**(struct snd_card * *card*, char * *id*, int *device*, int *output_count*, int *input_count*, struct snd_rawmidi ** *rrawmidi*)
create a rawmidi instance

Parameters

struct snd_card * card the card instance

char * id the id string

int device the device index

int output_count the number of output streams

int input_count the number of input streams

struct snd_rawmidi ** rrawmidi the pointer to store the new rawmidi instance

Description

Creates a new rawmidi instance. Use *snd_rawmidi_set_ops()* to set the operators to the new instance.

Return

Zero if successful, or a negative error code on failure.

void **snd_rawmidi_set_ops**(struct snd_rawmidi * *rmidi*, int *stream*, const struct snd_rawmidi_ops * *ops*)
set the rawmidi operators

Parameters

struct snd_rawmidi * rmidi the rawmidi instance

int stream the stream direction, SNDRV_RAWMIDI_STREAM_XXX

const struct snd_rawmidi_ops * ops the operator table

Description

Sets the rawmidi operators for the given stream direction.

MPU401-UART API

irqreturn_t **snd_mpu401_uart_interrupt**(int *irq*, void * *dev_id*)
generic MPU401-UART interrupt handler

Parameters

int irq the irq number

void * dev_id mpu401 instance

Description

Processes the interrupt for MPU401-UART i/o.

Return

IRQ_HANDLED if the interrupt was handled. IRQ_NONE otherwise.

irqreturn_t **snd_mpu401_uart_interrupt_tx**(int *irq*, void * *dev_id*)
generic MPU401-UART transmit irq handler

Parameters

int irq the irq number

void * dev_id mpu401 instance

Description

Processes the interrupt for MPU401-UART output.

Return

IRQ_HANDLED if the interrupt was handled. IRQ_NONE otherwise.

int **snd_mpu401_uart_new**(struct snd_card * *card*, int *device*, unsigned short *hardware*, unsigned long *port*, unsigned int *info_flags*, int *irq*, struct snd_rawmidi ** *rrawmidi*)
create an MPU401-UART instance

Parameters

struct snd_card * card the card instance

int device the device index, zero-based

unsigned short hardware the hardware type, MPU401_HW_XXXX

unsigned long port the base address of MPU401 port

unsigned int info_flags bitflags MPU401_INFO_XXX

int irq the ISA irq number, -1 if not to be allocated

struct snd_rawmidi ** rrawmidi the pointer to store the new rawmidi instance

Description

Creates a new MPU-401 instance.

Note that the rawmidi instance is returned on the rrawmidi argument, not the mpu401 instance itself. To access to the mpu401 instance, cast from rawmidi->private_data (with struct snd_mpu401 magic-cast).

Return

Zero if successful, or a negative error code.

Proc Info API

Proc Info Interface

int **snd_info_get_line**(struct snd_info_buffer * *buffer*, char * *line*, int *len*)
read one line from the procfs buffer

Parameters

struct snd_info_buffer * buffer the procfs buffer

char * line the buffer to store

int len the max. buffer size

Description

Reads one line from the buffer and stores the string.

Return

Zero if successful, or 1 if error or EOF.

const char * **snd_info_get_str**(char * *dest*, const char * *src*, int *len*)
parse a string token

Parameters

char * dest the buffer to store the string token

const char * src the original string

int len the max. length of token - 1

Description

Parses the original string and copy a token to the given string buffer.

Return

The updated pointer of the original string so that it can be used for the next call.

struct snd_info_entry * **snd_info_create_module_entry**(struct module * *module*, const char * *name*, struct snd_info_entry * *parent*)
create an info entry for the given module

Parameters

struct module * module the module pointer

const char * name the file name

struct snd_info_entry * parent the parent directory

Description

Creates a new info entry and assigns it to the given module.

Return

The pointer of the new instance, or NULL on failure.

struct snd_info_entry * **snd_info_create_card_entry**(struct snd_card * *card*, const char * *name*, struct snd_info_entry * *parent*)
create an info entry for the given card

Parameters

struct snd_card * card the card instance

const char * name the file name

struct snd_info_entry * parent the parent directory

Description

Creates a new info entry and assigns it to the given card.

Return

The pointer of the new instance, or NULL on failure.

void **snd_info_free_entry**(struct snd_info_entry * *entry*)
release the info entry

Parameters

struct snd_info_entry * entry the info entry

Description

Releases the info entry.

int **snd_info_register**(struct snd_info_entry * *entry*)
register the info entry

Parameters

struct snd_info_entry * entry the info entry

Description

Registers the proc info entry.

Return

Zero if successful, or a negative error code on failure.

Compress Offload

Compress Offload API

int **snd_compress_register**(struct *snd_compr* * *device*)
register compressed device

Parameters

struct snd_compr * device compressed device to register

struct **snd_compressed_buffer**
compressed buffer

Definition

```
struct snd_compressed_buffer {
  __u32 fragment_size;
  __u32 fragments;
};
```

Members

fragment_size size of buffer fragment in bytes

fragments number of such fragments

struct **snd_compr_params**
compressed stream params

Definition

```
struct snd_compr_params {
  struct snd_compressed_buffer buffer;
  struct snd_codec codec;
  __u8 no_wake_mode;
};
```

Members

buffer buffer description

codec codec parameters

no_wake_mode dont wake on fragment elapsed

struct **snd_compr_tstamp**
timestamp descriptor

Definition

```
struct snd_compr_tstamp {
  __u32 byte_offset;
  __u32 copied_total;
  __u32 pcm_frames;
  __u32 pcm_io_frames;
  __u32 sampling_rate;
};
```

Members

byte_offset Byte offset in ring buffer to DSP

copied_total Total number of bytes copied from/to ring buffer to/by DSP

pcm_frames Frames decoded or encoded by DSP. This field will evolve by large steps and should only be used to monitor encoding/decoding progress. It shall not be used for timing estimates.

pcm_io_frames Frames rendered or received by DSP into a mixer or an audio output/input. This field should be used for A/V sync or time estimates.

sampling_rate sampling rate of audio

struct **snd_compr_avail**
avail descriptor

Definition

```
struct snd_compr_avail {
  __u64 avail;
  struct snd_compr_tstamp tstamp;
};
```

Members

avail Number of bytes available in ring buffer for writing/reading

tstamp timestamp information

struct **snd_compr_caps**
caps descriptor

Definition

```
struct snd_compr_caps {
  __u32 num_codecs;
  __u32 direction;
  __u32 min_fragment_size;
  __u32 max_fragment_size;
  __u32 min_fragments;
  __u32 max_fragments;
  __u32 codecs[MAX_NUM_CODECS];
  __u32 reserved[11];
};
```

Members

num_codecs number of codecs supported

direction direction supported. Of type snd_compr_direction

min_fragment_size minimum fragment supported by DSP

max_fragment_size maximum fragment supported by DSP

min_fragments min fragments supported by DSP

max_fragments max fragments supported by DSP

codecs pointer to array of codecs

reserved reserved field

struct **snd_compr_codec_caps**
query capability of codec

Definition

```
struct snd_compr_codec_caps {
  __u32 codec;
  __u32 num_descriptors;
  struct snd_codec_desc descriptor[MAX_NUM_CODEC_DESCRIPTORS];
};
```

Members

codec codec for which capability is queried

num_descriptors number of codec descriptors

descriptor array of codec capability descriptor

enum **sndrv_compress_encoder**

Constants

SNDRV_COMPRESS_ENCODER_PADDING no of samples appended by the encoder at the end of the track

SNDRV_COMPRESS_ENCODER_DELAY no of samples inserted by the encoder at the beginning of the track

struct **snd_compr_metadata**
compressed stream metadata

Definition

```
struct snd_compr_metadata {
  __u32 key;
  __u32 value[8];
};
```

Members

key key id

value key value

SNDRV_COMPRESS_IOCTL_VERSION()

Parameters

Description

SNDRV_COMPRESS_GET_CAPS: Query capability of DSP SNDRV_COMPRESS_GET_CODEC_CAPS: Query capability of a codec SNDRV_COMPRESS_SET_PARAMS: Set codec and stream parameters

Note

only codec params can be changed runtime and stream params cant be SNDRV_COMPRESS_GET_PARAMS: Query codec params SNDRV_COMPRESS_TSTAMP: get the current timestamp value SNDRV_COMPRESS_AVAIL: get the current buffer avail value. This also queries the tstamp properties SNDRV_COMPRESS_PAUSE: Pause the running stream SNDRV_COMPRESS_RESUME: resume a paused stream SNDRV_COMPRESS_START: Start a stream SNDRV_COMPRESS_STOP: stop a running stream, discarding ring buffer content and the buffers currently with DSP SNDRV_COMPRESS_DRAIN: Play till end of buffers and stop after that SNDRV_COMPRESS_IOCTL_VERSION: Query the API version

struct **snd_enc_vorbis**

Definition

```
struct snd_enc_vorbis {
  __s32 quality;
  __u32 managed;
  __u32 max_bit_rate;
  __u32 min_bit_rate;
  __u32 downmix;
};
```

Members

quality Sets encoding quality to n, between -1 (low) and 10 (high). In the default mode of operation, the quality level is 3. Normal quality range is 0 - 10.

managed Boolean. Set bitrate management mode. This turns off the normal VBR encoding, but allows hard or soft bitrate constraints to be enforced by the encoder. This mode can be slower, and may also be lower quality. It is primarily useful for streaming.

max_bit_rate Enabled only if managed is TRUE

min_bit_rate Enabled only if managed is TRUE

downmix Boolean. Downmix input from stereo to mono (has no effect on non-stereo streams). Useful for lower-bitrate encoding.

Description

These options were extracted from the OpenMAX IL spec and Gstreamer vorbisenc properties

For best quality users should specify VBR mode and set quality levels.

struct **snd_enc_real**

Definition

```
struct snd_enc_real {
  __u32 quant_bits;
  __u32 start_region;
  __u32 num_regions;
};
```

Members

quant_bits number of coupling quantization bits in the stream

start_region coupling start region in the stream

num_regions number of regions value

Description

These options were extracted from the OpenMAX IL spec

struct **snd_enc_flac**

Definition

```
struct snd_enc_flac {
  __u32 num;
  __u32 gain;
};
```

Members

num serial number, valid only for OGG formats needs to be set by application

gain Add replay gain tags

Description

These options were extracted from the FLAC online documentation at http://flac.sourceforge.net/documentation_tools_flac.html

To make the API simpler, it is assumed that the user will select quality profiles. Additional options that affect encoding quality and speed can be added at a later stage if needed.

By default the Subset format is used by encoders.

TAGS such as pictures, etc, cannot be handled by an offloaded encoder and are not supported in this API.

struct **snd_compr_runtime**

Definition

```
struct snd_compr_runtime {
  snd_pcm_state_t state;
  struct snd_compr_ops *ops;
  void *buffer;
  u64 buffer_size;
  u32 fragment_size;
  u32 fragments;
  u64 total_bytes_available;
  u64 total_bytes_transferred;
  wait_queue_head_t sleep;
  void *private_data;
};
```

Members

state stream state

ops pointer to DSP callbacks

buffer pointer to kernel buffer, valid only when not in mmap mode or DSP doesn't implement copy

buffer_size size of the above buffer

fragment_size size of buffer fragment in bytes

fragments number of such fragments

total_bytes_available cumulative number of bytes made available in the ring buffer

total_bytes_transferred cumulative bytes transferred by offload DSP

sleep poll sleep

private_data driver private data pointer

struct **snd_compr_stream**

Definition

```
struct snd_compr_stream {
  const char *name;
  struct snd_compr_ops *ops;
  struct snd_compr_runtime *runtime;
  struct snd_compr *device;
  struct delayed_work error_work;
  enum snd_compr_direction direction;
  bool metadata_set;
  bool next_track;
  void *private_data;
};
```

Members

name device name

ops pointer to DSP callbacks

runtime pointer to runtime structure

device device pointer

error_work delayed work used when closing the stream due to an error

direction stream direction, playback/recording

metadata_set metadata set flag, true when set

next_track has userspace signal next track transition, true when set

private_data pointer to DSP private data

struct **snd_compr_ops**

Definition

```
struct snd_compr_ops {
  int (*open)(struct snd_compr_stream *stream);
  int (*free)(struct snd_compr_stream *stream);
  int (*set_params)(struct snd_compr_stream *stream, struct snd_compr_params *params);
  int (*get_params)(struct snd_compr_stream *stream, struct snd_codec *params);
  int (*set_metadata)(struct snd_compr_stream *stream, struct snd_compr_metadata *metadata);
  int (*get_metadata)(struct snd_compr_stream *stream, struct snd_compr_metadata *metadata);
  int (*trigger)(struct snd_compr_stream *stream, int cmd);
  int (*pointer)(struct snd_compr_stream *stream, struct snd_compr_tstamp *tstamp);
  int (*copy)(struct snd_compr_stream *stream, char __user *buf, size_t count);
  int (*mmap)(struct snd_compr_stream *stream, struct vm_area_struct *vma);
  int (*ack)(struct snd_compr_stream *stream, size_t bytes);
  int (*get_caps) (struct snd_compr_stream *stream, struct snd_compr_caps *caps);
  int (*get_codec_caps) (struct snd_compr_stream *stream, struct snd_compr_codec_caps *codec);
};
```

Members

open Open the compressed stream This callback is mandatory and shall keep dsp ready to receive the stream parameter

free Close the compressed stream, mandatory

set_params Sets the compressed stream parameters, mandatory This can be called in during stream creation only to set codec params and the stream properties

get_params retrieve the codec parameters, mandatory

set_metadata Set the metadata values for a stream

get_metadata retrieves the requested metadata values from stream

trigger Trigger operations like start, pause, resume, drain, stop. This callback is mandatory

pointer Retrieve current h/w pointer information. Mandatory

copy Copy the compressed data to/from userspace, Optional Can't be implemented if DSP supports mmap

mmap DSP mmap method to mmap DSP memory

ack Ack for DSP when data is written to audio buffer, Optional Not valid if copy is implemented

get_caps Retrieve DSP capabilities, mandatory

get_codec_caps Retrieve capabilities for a specific codec, mandatory

struct **snd_compr**

Definition

```
struct snd_compr {
  const char *name;
  struct device dev;
  struct snd_compr_ops *ops;
```

```
  void *private_data;
  struct snd_card *card;
  unsigned int direction;
  struct mutex lock;
  int device;
#ifdef CONFIG_SND_VERBOSE_PROCFS;
};
```

Members

name DSP device name

dev associated device instance

ops pointer to DSP callbacks

private_data pointer to DSP pvt data

card sound card pointer

direction Playback or capture direction

lock device lock

device device id

ASoC

ASoC Core API

struct **snd_soc_jack_pin**
Describes a pin to update based on jack detection

Definition

```
struct snd_soc_jack_pin {
  struct list_head list;
  const char *pin;
  int mask;
  bool invert;
};
```

Members

list internal list entry

pin name of the pin to update

mask bits to check for in reported jack status

invert if non-zero then pin is enabled when status is not reported

struct **snd_soc_jack_zone**
Describes voltage zones of jack detection

Definition

```
struct snd_soc_jack_zone {
  unsigned int min_mv;
  unsigned int max_mv;
  unsigned int jack_type;
  unsigned int debounce_time;
  struct list_head list;
};
```

Members

min_mv start voltage in mv

max_mv end voltage in mv

jack_type type of jack that is expected for this voltage

debounce_time debounce_time for jack, codec driver should wait for this duration before reading the adc for voltages

list internal list entry

struct **snd_soc_jack_gpio**
Describes a gpio pin for jack detection

Definition

```
struct snd_soc_jack_gpio {
  unsigned int gpio;
  unsigned int idx;
  struct device *gpiod_dev;
  const char *name;
  int report;
  int invert;
  int debounce_time;
  bool wake;
  int (*jack_status_check)(void *data);
};
```

Members

gpio legacy gpio number

idx gpio descriptor index within the function of the GPIO consumer device

gpiod_dev GPIO consumer device

name gpio name. Also as connection ID for the GPIO consumer device function name lookup

report value to report when jack detected

invert report presence in low state

debounce_time debounce time in ms

wake enable as wake source

jack_status_check callback function which overrides the detection to provide more complex checks (eg, reading an ADC).

struct snd_soc_component * **snd_soc_dapm_to_component**(struct snd_soc_dapm_context * *dapm*)
Casts a DAPM context to the component it is embedded in

Parameters

struct snd_soc_dapm_context * dapm The DAPM context to cast to the component

Description

This function must only be used on DAPM contexts that are known to be part of a component (e.g. in a component driver). Otherwise the behavior is undefined.

struct snd_soc_dapm_context * **snd_soc_component_get_dapm**(struct snd_soc_component * *component*)
Returns the DAPM context associated with a component

Parameters

struct snd_soc_component * component The component for which to get the DAPM context

void **snd_soc_component_init_bias_level**(struct snd_soc_component * *component*, enum snd_soc_bias_level *level*)
Initialize COMPONENT DAPM bias level

Parameters

struct snd_soc_component * component The COMPONENT for which to initialize the DAPM bias level

enum snd_soc_bias_level level The DAPM level to initialize to

Description

Initializes the COMPONENT DAPM bias level. See `snd_soc_dapm_init_bias_level()`.

enum snd_soc_bias_level **snd_soc_component_get_bias_level**(struct snd_soc_component * *component*)
Get current COMPONENT DAPM bias level

Parameters

struct snd_soc_component * component The COMPONENT for which to get the DAPM bias level

Return

The current DAPM bias level of the COMPONENT.

int **snd_soc_component_force_bias_level**(struct snd_soc_component * *component*, enum snd_soc_bias_level *level*)
Set the COMPONENT DAPM bias level

Parameters

struct snd_soc_component * component The COMPONENT for which to set the level

enum snd_soc_bias_level level The level to set to

Description

Forces the COMPONENT bias level to a specific state. See *snd_soc_dapm_force_bias_level()*.

struct snd_soc_component * **snd_soc_dapm_kcontrol_component**(struct snd_kcontrol * *kcontrol*)
Returns the component associated to a kcontrol

Parameters

struct snd_kcontrol * kcontrol The kcontrol

Description

This function must only be used on DAPM contexts that are known to be part of a COMPONENT (e.g. in a COMPONENT driver). Otherwise the behavior is undefined.

int **snd_soc_component_cache_sync**(struct snd_soc_component * *component*)
Sync the register cache with the hardware

Parameters

struct snd_soc_component * component COMPONENT to sync

Note

This function will call `regcache_sync()`

struct snd_soc_component * **snd_soc_kcontrol_component**(struct snd_kcontrol * *kcontrol*)
Returns the component that registered the control

Parameters

struct snd_kcontrol * kcontrol The control for which to get the component

Note

This function will work correctly if the control has been registered for a component. With `snd_soc_add_codec_controls()` or via table based setup for either a CODEC or component driver. Otherwise the behavior is undefined.

struct snd_soc_dai * **snd_soc_find_dai**(const struct snd_soc_dai_link_component * *dlc*)
Find a registered DAI

Parameters

const struct snd_soc_dai_link_component * dlc name of the DAI or the DAI driver and optional component info to match

Description

This function will search all registered components and their DAIs to find the DAI of the same name. The component's of_node and name should also match if being specified.

Return

pointer of DAI, or NULL if not found.

struct snd_soc_dai_link * **snd_soc_find_dai_link**(struct snd_soc_card * *card*, int *id*, const char * *name*, const char * *stream_name*)
Find a DAI link

Parameters

struct snd_soc_card * card soc card

int id DAI link ID to match

const char * name DAI link name to match, optional

const char * stream_name DAI link stream name to match, optional

Description

This function will search all existing DAI links of the soc card to find the link of the same ID. Since DAI links may not have their unique ID, so name and stream name should also match if being specified.

Return

pointer of DAI link, or NULL if not found.

int **snd_soc_add_dai_link**(struct snd_soc_card * *card*, struct snd_soc_dai_link * *dai_link*)
Add a DAI link dynamically

Parameters

struct snd_soc_card * card The ASoC card to which the DAI link is added

struct snd_soc_dai_link * dai_link The new DAI link to add

Description

This function adds a DAI link to the ASoC card's link list.

Note

Topology can use this API to add DAI links when probing the topology component. And machine drivers can still define static DAI links in dai_link array.

void **snd_soc_remove_dai_link**(struct snd_soc_card * *card*, struct snd_soc_dai_link * *dai_link*)
Remove a DAI link from the list

Parameters

struct snd_soc_card * card The ASoC card that owns the link

struct snd_soc_dai_link * dai_link The DAI link to remove

Description

This function removes a DAI link from the ASoC card's link list.

For DAI links previously added by topology, topology should remove them by using the dobj embedded in the link.

int **snd_soc_runtime_set_dai_fmt**(struct snd_soc_pcm_runtime * *rtd*, unsigned int *dai_fmt*)
Change DAI link format for a ASoC runtime

Parameters

struct snd_soc_pcm_runtime * rtd The runtime for which the DAI link format should be changed

unsigned int dai_fmt The new DAI link format

Description

This function updates the DAI link format for all DAIs connected to the DAI link for the specified runtime.

Note

For setups with a static format set the dai_fmt field in the corresponding snd_dai_link struct instead of using this function.

Returns 0 on success, otherwise a negative error code.

int **snd_soc_set_dmi_name**(struct snd_soc_card * *card*, const char * *flavour*)
Register DMI names to card

Parameters

struct snd_soc_card * card The card to register DMI names

const char * flavour The flavour "differentiator" for the card amongst its peers.

Description

An Intel machine driver may be used by many different devices but are difficult for userspace to differentiate, since machine drivers ususally use their own name as the card short name and leave the card long name blank. To differentiate such devices and fix bugs due to lack of device-specific configurations, this function allows DMI info to be used as the sound card long name, in the format of "vendor-product-version-board" (Character '-' is used to separate different DMI fields here). This will help the user space to load the device-specific Use Case Manager (UCM) configurations for the card.

Possible card long names may be: DellInc.-XPS139343-01-0310JH ASUSTeKCOMPUTERINC.-T100TA-1.0-T100TA Circuitco-MinnowboardMaxD0PLATFORM-D0-MinnowBoardMAX

This function also supports flavoring the card longname to provide the extra differentiation, like "vendor-product-version-board-flavor".

We only keep number and alphabet characters and a few separator characters in the card long name since UCM in the user space uses the card long names as card configuration directory names and AudoConf cannot support special charactors like SPACE.

Returns 0 on success, otherwise a negative error code.

struct snd_kcontrol * **snd_soc_cnew**(const struct snd_kcontrol_new * *_template*, void * *data*, const char * *long_name*, const char * *prefix*)
create new control

Parameters

const struct snd_kcontrol_new * _template control template

void * data control private data

const char * long_name control long name

const char * prefix control name prefix

Description

Create a new mixer control from a template control.

Returns 0 for success, else error.

int **snd_soc_add_component_controls**(struct snd_soc_component * *component*, const struct snd_kcontrol_new * *controls*, unsigned int *num_controls*)
Add an array of controls to a component.

Parameters

struct snd_soc_component * component Component to add controls to

const struct snd_kcontrol_new * controls Array of controls to add

unsigned int num_controls Number of elements in the array

Return

0 for success, else error.

int **snd_soc_add_card_controls**(struct snd_soc_card * *soc_card*, const struct snd_kcontrol_new * *controls*, int *num_controls*)
add an array of controls to a SoC card. Convenience function to add a list of controls.

Parameters

struct snd_soc_card * soc_card SoC card to add controls to

const struct snd_kcontrol_new * controls array of controls to add

int num_controls number of elements in the array

Description

Return 0 for success, else error.

int **snd_soc_add_dai_controls**(struct snd_soc_dai * *dai*, const struct snd_kcontrol_new * *controls*, int *num_controls*)
add an array of controls to a DAI. Convienience function to add a list of controls.

Parameters

struct snd_soc_dai * dai DAI to add controls to

const struct snd_kcontrol_new * controls array of controls to add

int num_controls number of elements in the array

Description

Return 0 for success, else error.

int **snd_soc_dai_set_sysclk**(struct snd_soc_dai * *dai*, int *clk_id*, unsigned int *freq*, int *dir*)
configure DAI system or master clock.

Parameters

struct snd_soc_dai * dai DAI

int clk_id DAI specific clock ID

unsigned int freq new clock frequency in Hz

int dir new clock direction - input/output.

Description

Configures the DAI master (MCLK) or system (SYSCLK) clocking.

int **snd_soc_component_set_sysclk**(struct snd_soc_component * *component*, int *clk_id*, int *source*, unsigned int *freq*, int *dir*)
configure COMPONENT system or master clock.

Parameters

struct snd_soc_component * component COMPONENT

int clk_id DAI specific clock ID

int source Source for the clock

unsigned int freq new clock frequency in Hz

int dir new clock direction - input/output.

Description

Configures the CODEC master (MCLK) or system (SYSCLK) clocking.

int **snd_soc_dai_set_clkdiv**(struct snd_soc_dai * *dai*, int *div_id*, int *div*)
configure DAI clock dividers.

Parameters

struct snd_soc_dai * dai DAI

int div_id DAI specific clock divider ID

int div new clock divisor.

Description

Configures the clock dividers. This is used to derive the best DAI bit and frame clocks from the system or master clock. It's best to set the DAI bit and frame clocks as low as possible to save system power.

int **snd_soc_dai_set_pll**(struct snd_soc_dai * *dai*, int *pll_id*, int *source*, unsigned int *freq_in*, unsigned int *freq_out*)
configure DAI PLL.

Parameters

struct snd_soc_dai * dai DAI

int pll_id DAI specific PLL ID

int source DAI specific source for the PLL

unsigned int freq_in PLL input clock frequency in Hz

unsigned int freq_out requested PLL output clock frequency in Hz

Description

Configures and enables PLL to generate output clock based on input clock.

int **snd_soc_dai_set_bclk_ratio**(struct snd_soc_dai * *dai*, unsigned int *ratio*)
configure BCLK to sample rate ratio.

Parameters

struct snd_soc_dai * dai DAI

unsigned int ratio Ratio of BCLK to Sample rate.

Description

Configures the DAI for a preset BCLK to sample rate ratio.

int **snd_soc_dai_set_fmt**(struct snd_soc_dai * *dai*, unsigned int *fmt*)
configure DAI hardware audio format.

Parameters

struct snd_soc_dai * dai DAI

unsigned int fmt SND_SOC_DAIFMT_* format value.

Description

Configures the DAI hardware format and clocking.

int **snd_soc_xlate_tdm_slot_mask**(unsigned int *slots*, unsigned int * *tx_mask*, unsigned int * *rx_mask*)
generate tx/rx slot mask.

Parameters

unsigned int slots Number of slots in use.

unsigned int * tx_mask bitmask representing active TX slots.

unsigned int * rx_mask bitmask representing active RX slots.

Description

Generates the TDM tx and rx slot default masks for DAI.

int **snd_soc_dai_set_tdm_slot**(struct snd_soc_dai * *dai*, unsigned int *tx_mask*, unsigned int *rx_mask*, int *slots*, int *slot_width*)
Configures a DAI for TDM operation

Parameters

struct snd_soc_dai * dai The DAI to configure

unsigned int tx_mask bitmask representing active TX slots.

unsigned int rx_mask bitmask representing active RX slots.

int slots Number of slots in use.

int slot_width Width in bits for each slot.

Description

This function configures the specified DAI for TDM operation. **slot** contains the total number of slots of the TDM stream and **slot_with** the width of each slot in bit clock cycles. **tx_mask** and **rx_mask** are bitmasks specifying the active slots of the TDM stream for the specified DAI, i.e. which slots the DAI should write to or read from. If a bit is set the corresponding slot is active, if a bit is cleared the corresponding slot is inactive. Bit 0 maps to the first slot, bit 1 to the second slot and so on. The first active slot maps to the first channel of the DAI, the second active slot to the second channel and so on.

TDM mode can be disabled by passing 0 for **slots**. In this case **tx_mask**, **rx_mask** and **slot_width** will be ignored.

Returns 0 on success, a negative error code otherwise.

int **snd_soc_dai_set_channel_map**(struct snd_soc_dai * *dai*, unsigned int *tx_num*, unsigned int * *tx_slot*, unsigned int *rx_num*, unsigned int * *rx_slot*)
configure DAI audio channel map

Parameters

struct snd_soc_dai * dai DAI

unsigned int tx_num how many TX channels

unsigned int * tx_slot pointer to an array which imply the TX slot number channel 0~num-1 uses

unsigned int rx_num how many RX channels

unsigned int * rx_slot pointer to an array which imply the RX slot number channel 0~num-1 uses

Description

configure the relationship between channel number and TDM slot number.

int **snd_soc_dai_set_tristate**(struct snd_soc_dai * *dai*, int *tristate*)
configure DAI system or master clock.

Parameters

struct snd_soc_dai * dai DAI

int tristate tristate enable

Description

Tristates the DAI so that others can use it.

int **snd_soc_dai_digital_mute**(struct snd_soc_dai * *dai*, int *mute*, int *direction*)
configure DAI system or master clock.

Parameters

struct snd_soc_dai * dai DAI

int mute mute enable

int direction stream to mute

Description

Mutes the DAI DAC.

int **snd_soc_register_card**(struct snd_soc_card * *card*)
Register a card with the ASoC core

Parameters

struct snd_soc_card * card Card to register

int **snd_soc_unregister_card**(struct snd_soc_card * *card*)
Unregister a card with the ASoC core

Parameters

struct snd_soc_card * card Card to unregister

void **snd_soc_unregister_dais**(struct snd_soc_component * *component*)
Unregister DAIs from the ASoC core

Parameters

struct snd_soc_component * component The component for which the DAIs should be unregistered

int **snd_soc_register_dais**(struct snd_soc_component * *component*, struct snd_soc_dai_driver * *dai_drv*, size_t *count*)
Register a DAI with the ASoC core

Parameters

struct snd_soc_component * component The component the DAIs are registered for

struct snd_soc_dai_driver * dai_drv DAI driver to use for the DAIs

size_t count Number of DAIs

int **snd_soc_register_dai**(struct snd_soc_component * *component*, struct snd_soc_dai_driver * *dai_drv*)
Register a DAI dynamically & create its widgets

Parameters

struct snd_soc_component * component The component the DAIs are registered for

struct snd_soc_dai_driver * dai_drv DAI driver to use for the DAI

Description

Topology can use this API to register DAIs when probing a component. These DAIs's widgets will be freed in the card cleanup and the DAIs will be freed in the component cleanup.

void **snd_soc_component_init_regmap**(struct snd_soc_component * *component*, struct regmap * *regmap*)
Initialize regmap instance for the component

Parameters

struct snd_soc_component * component The component for which to initialize the regmap instance

struct regmap * regmap The regmap instance that should be used by the component

Description

This function allows deferred assignment of the regmap instance that is associated with the component. Only use this if the regmap instance is not yet ready when the component is registered. The function must also be called before the first IO attempt of the component.

void **snd_soc_component_exit_regmap**(struct snd_soc_component * *component*)
De-initialize regmap instance for the component

Parameters

struct snd_soc_component * component The component for which to de-initialize the regmap instance

Description

Calls `regmap_exit()` on the regmap instance associated to the component and removes the regmap instance from the component.

This function should only be used if *snd_soc_component_init_regmap()* was used to initialize the regmap instance.

int **__snd_soc_unregister_component**(struct device * *dev*)
Unregister all related component from the ASoC core

Parameters

struct device * dev The device to unregister

int **devm_snd_soc_register_component**(struct device * *dev*, const struct snd_soc_component_driver * *cmpnt_drv*, struct snd_soc_dai_driver * *dai_drv*, int *num_dai*)
resource managed component registration

Parameters

struct device * dev Device used to manage component

const struct snd_soc_component_driver * cmpnt_drv Component driver

struct snd_soc_dai_driver * dai_drv DAI driver

int num_dai Number of DAIs to register

Description

Register a component with automatic unregistration when the device is unregistered.

int **devm_snd_soc_register_card**(struct device * *dev*, struct snd_soc_card * *card*)
resource managed card registration

Parameters

struct device * dev Device used to manage card

struct snd_soc_card * card Card to register

Description

Register a card with automatic unregistration when the device is unregistered.

int **devm_snd_dmaengine_pcm_register**(struct device * *dev*, const struct *snd_dmaengine_pcm_config* * *config*, unsigned int *flags*)
resource managed dmaengine PCM registration

Parameters

struct device * dev The parent device for the PCM device

const struct snd_dmaengine_pcm_config * config Platform specific PCM configuration

unsigned int flags Platform specific quirks

Description

Register a dmaengine based PCM device with automatic unregistration when the device is unregistered.

int **snd_soc_component_read**(struct snd_soc_component * *component*, unsigned int *reg*, unsigned int * *val*)
Read register value

Parameters

struct snd_soc_component * component Component to read from

unsigned int reg Register to read

unsigned int * val Pointer to where the read value is stored

Return

0 on success, a negative error code otherwise.

int **snd_soc_component_write**(struct snd_soc_component * *component*, unsigned int *reg*, unsigned int *val*)
Write register value

Parameters

struct snd_soc_component * component Component to write to

unsigned int reg Register to write

unsigned int val Value to write to the register

Return

0 on success, a negative error code otherwise.

int **snd_soc_component_update_bits**(struct snd_soc_component * *component*, unsigned int *reg*, unsigned int *mask*, unsigned int *val*)
Perform read/modify/write cycle

Parameters

struct snd_soc_component * component Component to update

unsigned int reg Register to update

unsigned int mask Mask that specifies which bits to update

unsigned int val New value for the bits specified by mask

Return

1 if the operation was successful and the value of the register changed, 0 if the operation was successful, but the value did not change. Returns a negative error code otherwise.

int **snd_soc_component_update_bits_async**(struct snd_soc_component * *component*, unsigned int *reg*, unsigned int *mask*, unsigned int *val*)
Perform asynchronous read/modify/write cycle

Parameters

struct snd_soc_component * component Component to update

unsigned int reg Register to update

unsigned int mask Mask that specifies which bits to update

unsigned int val New value for the bits specified by mask

Description

This function is similar to *snd_soc_component_update_bits()*, but the update operation is scheduled asynchronously. This means it may not be completed when the function returns. To make sure that all scheduled updates have been completed *snd_soc_component_async_complete()* must be called.

Return

1 if the operation was successful and the value of the register changed, 0 if the operation was successful, but the value did not change. Returns a negative error code otherwise.

void **snd_soc_component_async_complete**(struct snd_soc_component * *component*)
Ensure asynchronous I/O has completed

Parameters

struct snd_soc_component * component Component for which to wait

Description

This function blocks until all asynchronous I/O which has previously been scheduled using *snd_soc_component_update_bits_async()* has completed.

int **snd_soc_component_test_bits**(struct snd_soc_component * *component*, unsigned int *reg*, unsigned int *mask*, unsigned int *value*)
Test register for change

Parameters

struct snd_soc_component * component component

unsigned int reg Register to test

unsigned int mask Mask that specifies which bits to test

unsigned int value Value to test against

Description

Tests a register with a new value and checks if the new value is different from the old value.

Return

1 for change, otherwise 0.

void **snd_soc_runtime_activate**(struct snd_soc_pcm_runtime * *rtd*, int *stream*)
Increment active count for PCM runtime components

Parameters

struct snd_soc_pcm_runtime * rtd ASoC PCM runtime that is activated

int stream Direction of the PCM stream

Description

Increments the active count for all the DAIs and components attached to a PCM runtime. Should typically be called when a stream is opened.

Must be called with the rtd->pcm_mutex being held

void **snd_soc_runtime_deactivate**(struct snd_soc_pcm_runtime * *rtd*, int *stream*)
Decrement active count for PCM runtime components

Parameters

struct snd_soc_pcm_runtime * rtd ASoC PCM runtime that is deactivated

int stream Direction of the PCM stream

Description

Decrements the active count for all the DAIs and components attached to a PCM runtime. Should typically be called when a stream is closed.

Must be called with the rtd->pcm_mutex being held

bool **snd_soc_runtime_ignore_pmdown_time**(struct snd_soc_pcm_runtime * *rtd*)
Check whether to ignore the power down delay

Parameters

struct snd_soc_pcm_runtime * rtd The ASoC PCM runtime that should be checked.

Description

This function checks whether the power down delay should be ignored for a specific PCM runtime. Returns true if the delay is 0, if it the DAI link has been configured to ignore the delay, or if none of the components benefits from having the delay.

int **snd_soc_set_runtime_hwparams**(struct snd_pcm_substream * *substream*, const struct snd_pcm_hardware * *hw*)
set the runtime hardware parameters

Parameters

struct snd_pcm_substream * substream the pcm substream

const struct snd_pcm_hardware * hw the hardware parameters

Description

Sets the substream runtime hardware parameters.

int **snd_soc_info_enum_double**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_info * *uinfo*)
enumerated double mixer info callback

Parameters

struct snd_kcontrol * kcontrol mixer control

struct snd_ctl_elem_info * uinfo control element information

Description

Callback to provide information about a double enumerated mixer control.

Returns 0 for success.

int **snd_soc_get_enum_double**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_value * *ucontrol*)
enumerated double mixer get callback

Parameters

struct snd_kcontrol * kcontrol mixer control

struct snd_ctl_elem_value * ucontrol control element information

Description

Callback to get the value of a double enumerated mixer.

Returns 0 for success.

int **snd_soc_put_enum_double**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_value * *ucontrol*)
enumerated double mixer put callback

Parameters

struct snd_kcontrol * kcontrol mixer control

struct snd_ctl_elem_value * ucontrol control element information

Description

Callback to set the value of a double enumerated mixer.

Returns 0 for success.

int **snd_soc_read_signed**(struct snd_soc_component * *component*, unsigned int *reg*, unsigned int *mask*, unsigned int *shift*, unsigned int *sign_bit*, int * *signed_val*)
Read a codec register and interpret as signed value

Parameters

struct snd_soc_component * component component

unsigned int reg Register to read

unsigned int mask Mask to use after shifting the register value

unsigned int shift Right shift of register value

unsigned int sign_bit Bit that describes if a number is negative or not.

int * signed_val Pointer to where the read value should be stored

Description

This functions reads a codec register. The register value is shifted right by 'shift' bits and masked with the given 'mask'. Afterwards it translates the given registervalue into a signed integer if sign_bit is non-zero.

Returns 0 on sucess, otherwise an error value

int **snd_soc_info_volsw**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_info * *uinfo*)
single mixer info callback

Parameters

struct snd_kcontrol * kcontrol mixer control

struct snd_ctl_elem_info * uinfo control element information

Description

Callback to provide information about a single mixer control, or a double mixer control that spans 2 registers.

Returns 0 for success.

int **snd_soc_info_volsw_sx**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_info * *uinfo*)
Mixer info callback for SX TLV controls

Parameters

struct snd_kcontrol * kcontrol mixer control

struct snd_ctl_elem_info * uinfo control element information

Description

Callback to provide information about a single mixer control, or a double mixer control that spans 2 registers of the SX TLV type. SX TLV controls have a range that represents both positive and negative values either side of zero but without a sign bit.

Returns 0 for success.

int **snd_soc_get_volsw**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_value * *ucontrol*)
single mixer get callback

Parameters

struct snd_kcontrol * kcontrol mixer control

struct snd_ctl_elem_value * ucontrol control element information

Description

Callback to get the value of a single mixer control, or a double mixer control that spans 2 registers.

Returns 0 for success.

int **snd_soc_put_volsw**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_value * *ucontrol*)
single mixer put callback

Parameters

struct snd_kcontrol * kcontrol mixer control

struct snd_ctl_elem_value * ucontrol control element information

Description

Callback to set the value of a single mixer control, or a double mixer control that spans 2 registers.

Returns 0 for success.

int **snd_soc_get_volsw_sx**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_value * *ucontrol*)
single mixer get callback

Parameters

struct snd_kcontrol * kcontrol mixer control

struct snd_ctl_elem_value * ucontrol control element information

Description

Callback to get the value of a single mixer control, or a double mixer control that spans 2 registers.

Returns 0 for success.

int **snd_soc_put_volsw_sx**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_value * *ucontrol*)
double mixer set callback

Parameters

struct snd_kcontrol * kcontrol mixer control

struct snd_ctl_elem_value * ucontrol control element information

Description

Callback to set the value of a double mixer control that spans 2 registers.

Returns 0 for success.

int **snd_soc_info_volsw_range**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_info * *uinfo*)
single mixer info callback with range.

Parameters

struct snd_kcontrol * kcontrol mixer control

struct snd_ctl_elem_info * uinfo control element information

Description

Callback to provide information, within a range, about a single mixer control.

returns 0 for success.

int **snd_soc_put_volsw_range**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_value * *ucontrol*)
single mixer put value callback with range.

Parameters

struct snd_kcontrol * kcontrol mixer control

struct snd_ctl_elem_value * ucontrol control element information

Description

Callback to set the value, within a range, for a single mixer control.

Returns 0 for success.

int **snd_soc_get_volsw_range**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_value * *ucontrol*)
single mixer get callback with range

Parameters

struct snd_kcontrol * kcontrol mixer control

struct snd_ctl_elem_value * ucontrol control element information

Description

Callback to get the value, within a range, of a single mixer control.

Returns 0 for success.

int **snd_soc_limit_volume**(struct snd_soc_card * *card*, const char * *name*, int *max*)
Set new limit to an existing volume control.

Parameters

struct snd_soc_card * card where to look for the control

const char * name Name of the control

int max new maximum limit

Description

Return 0 for success, else error.

int **snd_soc_info_xr_sx**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_info * *uinfo*)
signed multi register info callback

Parameters

struct snd_kcontrol * kcontrol mreg control

struct snd_ctl_elem_info * uinfo control element information

Description

Callback to provide information of a control that can span multiple codec registers which together forms a single signed value in a MSB/LSB manner.

Returns 0 for success.

int **snd_soc_get_xr_sx**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_value * *ucontrol*)
signed multi register get callback

Parameters

struct snd_kcontrol * kcontrol mreg control

struct snd_ctl_elem_value * ucontrol control element information

Description

Callback to get the value of a control that can span multiple codec registers which together forms a single signed value in a MSB/LSB manner. The control supports specifying total no of bits used to allow for bitfields across the multiple codec registers.

Returns 0 for success.

int **snd_soc_put_xr_sx**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_value * *ucontrol*)
signed multi register get callback

Parameters

struct snd_kcontrol * kcontrol mreg control

struct snd_ctl_elem_value * ucontrol control element information

Description

Callback to set the value of a control that can span multiple codec registers which together forms a single signed value in a MSB/LSB manner. The control supports specifying total no of bits used to allow for bitfields across the multiple codec registers.

Returns 0 for success.

int **snd_soc_get_strobe**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_value * *ucontrol*)
strobe get callback

Parameters

struct snd_kcontrol * kcontrol mixer control

struct snd_ctl_elem_value * ucontrol control element information

Description

Callback get the value of a strobe mixer control.

Returns 0 for success.

int **snd_soc_put_strobe**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_value * *ucontrol*)
strobe put callback

Parameters

struct snd_kcontrol * kcontrol mixer control

struct snd_ctl_elem_value * ucontrol control element information

Description

Callback strobe a register bit to high then low (or the inverse) in one pass of a single mixer enum control.

Returns 1 for success.

int **snd_soc_new_compress**(struct snd_soc_pcm_runtime * *rtd*, int *num*)
create a new compress.

Parameters

struct snd_soc_pcm_runtime * rtd The runtime for which we will create compress

int num the device index number (zero based - shared with normal PCMs)

Return

0 for success, else error.

ASoC DAPM API

struct snd_soc_dapm_widget * **snd_soc_dapm_kcontrol_widget**(struct snd_kcontrol * *kcontrol*)
Returns the widget associated to a kcontrol

Parameters

struct snd_kcontrol * kcontrol The kcontrol

struct snd_soc_dapm_context * **snd_soc_dapm_kcontrol_dapm**(struct snd_kcontrol * *kcontrol*)
Returns the dapm context associated to a kcontrol

Parameters

struct snd_kcontrol * kcontrol The kcontrol

Note

This function must only be used on kcontrols that are known to have been registered for a CODEC. Otherwise the behaviour is undefined.

int **snd_soc_dapm_force_bias_level**(struct snd_soc_dapm_context * *dapm*, enum snd_soc_bias_level *level*)
Sets the DAPM bias level

Parameters

struct snd_soc_dapm_context * dapm The DAPM context for which to set the level

enum snd_soc_bias_level level The level to set

Description

Forces the DAPM bias level to a specific state. It will call the bias level callback of DAPM context with the specified level. This will even happen if the context is already at the same level. Furthermore it will not go through the normal bias level sequencing, meaning any intermediate states between the current and the target state will not be entered.

Note that the change in bias level is only temporary and the next time *snd_soc_dapm_sync()* is called the state will be set to the level as determined by the DAPM core. The function is mainly intended to be used to used during probe or resume from suspend to power up the device so initialization can be done, before the DAPM core takes over.

int **snd_soc_dapm_set_bias_level**(struct snd_soc_dapm_context * *dapm*, enum snd_soc_bias_level *level*)
set the bias level for the system

Parameters

struct snd_soc_dapm_context * dapm DAPM context

enum snd_soc_bias_level level level to configure

Description

Configure the bias (power) levels for the SoC audio device.

Returns 0 for success else error.

int **snd_soc_dapm_dai_get_connected_widgets**(struct snd_soc_dai * *dai*, int *stream*, struct snd_soc_dapm_widget_list ** *list*, bool (*custom_stop_condition)(struct snd_soc_dapm_widget *, enum *snd_soc_dapm_direction*))
query audio path and it's widgets.

Parameters

struct snd_soc_dai * dai the soc DAI.

int stream stream direction.

struct snd_soc_dapm_widget_list ** list list of active widgets for this stream.

bool (*)(struct snd_soc_dapm_widget *, enum snd_soc_dapm_direction) custom_stop_condition
(optional) a function meant to stop the widget graph walk based on custom logic.

Description

Queries DAPM graph as to whether a valid audio stream path exists for the initial stream specified by name. This takes into account current mixer and mux kcontrol settings. Creates list of valid widgets.

Optionally, can be supplied with a function acting as a stopping condition. This function takes the dapm widget currently being examined and the walk direction as an arguments, it should return true if the walk should be stopped and false otherwise.

Returns the number of valid paths or negative error.

int **snd_soc_dapm_sync_unlocked**(struct snd_soc_dapm_context * *dapm*)
scan and power dapm paths

Parameters

struct snd_soc_dapm_context * dapm DAPM context

Description

Walks all dapm audio paths and powers widgets according to their stream or path usage.

Requires external locking.

Returns 0 for success.

int **snd_soc_dapm_sync**(struct snd_soc_dapm_context * *dapm*)
scan and power dapm paths

Parameters

struct snd_soc_dapm_context * dapm DAPM context

Description

Walks all dapm audio paths and powers widgets according to their stream or path usage.

Returns 0 for success.

int **snd_soc_dapm_add_routes**(struct snd_soc_dapm_context * *dapm*, const struct snd_soc_dapm_route * *route*, int *num*)
Add routes between DAPM widgets

Parameters

struct snd_soc_dapm_context * dapm DAPM context

const struct snd_soc_dapm_route * route audio routes

int num number of routes

Description

Connects 2 dapm widgets together via a named audio path. The sink is the widget receiving the audio signal, whilst the source is the sender of the audio signal.

Returns 0 for success else error. On error all resources can be freed with a call to `snd_soc_card_free()`.

int **snd_soc_dapm_del_routes**(struct snd_soc_dapm_context * *dapm*, const struct snd_soc_dapm_route * *route*, int *num*)
Remove routes between DAPM widgets

Parameters

struct snd_soc_dapm_context * dapm DAPM context

const struct snd_soc_dapm_route * route audio routes

int num number of routes

Description

Removes routes from the DAPM context.

int **snd_soc_dapm_weak_routes**(struct snd_soc_dapm_context * *dapm*, const struct snd_soc_dapm_route * *route*, int *num*)
Mark routes between DAPM widgets as weak

Parameters

struct snd_soc_dapm_context * dapm DAPM context

const struct snd_soc_dapm_route * route audio routes

int num number of routes

Description

Mark existing routes matching those specified in the passed array as being weak, meaning that they are ignored for the purpose of power decisions. The main intended use case is for sidetone paths which couple audio between other independent paths if they are both active in order to make the combination work better at the user level but which aren't intended to be "used".

Note that CODEC drivers should not use this as sidetone type paths can frequently also be used as bypass paths.

int **snd_soc_dapm_new_widgets**(struct snd_soc_card * *card*)
add new dapm widgets

Parameters

struct snd_soc_card * card card to be checked for new dapm widgets

Description

Checks the codec for any new dapm widgets and creates them if found.

Returns 0 for success.

int **snd_soc_dapm_get_volsw**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_value * *ucontrol*)
dapm mixer get callback

Parameters

struct snd_kcontrol * kcontrol mixer control

struct snd_ctl_elem_value * ucontrol control element information

Description

Callback to get the value of a dapm mixer control.

Returns 0 for success.

int **snd_soc_dapm_put_volsw**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_value * *ucontrol*)
dapm mixer set callback

Parameters

struct snd_kcontrol * kcontrol mixer control

struct snd_ctl_elem_value * ucontrol control element information

Description

Callback to set the value of a dapm mixer control.

Returns 0 for success.

int **snd_soc_dapm_get_enum_double**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_value * *ucontrol*)
dapm enumerated double mixer get callback

Parameters

struct snd_kcontrol * kcontrol mixer control

struct snd_ctl_elem_value * ucontrol control element information

Description

Callback to get the value of a dapm enumerated double mixer control.

Returns 0 for success.

int **snd_soc_dapm_put_enum_double**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_value * *ucontrol*)
dapm enumerated double mixer set callback

Parameters

struct snd_kcontrol * kcontrol mixer control

struct snd_ctl_elem_value * ucontrol control element information

Description

Callback to set the value of a dapm enumerated double mixer control.

Returns 0 for success.

int **snd_soc_dapm_info_pin_switch**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_info * *uinfo*)
Info for a pin switch

Parameters

struct snd_kcontrol * kcontrol mixer control

struct snd_ctl_elem_info * uinfo control element information

Description

Callback to provide information about a pin switch control.

int **snd_soc_dapm_get_pin_switch**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_value * *ucontrol*)
Get information for a pin switch

Parameters

struct snd_kcontrol * kcontrol mixer control

struct snd_ctl_elem_value * ucontrol Value

int **snd_soc_dapm_put_pin_switch**(struct snd_kcontrol * *kcontrol*, struct snd_ctl_elem_value * *ucontrol*)
Set information for a pin switch

Parameters

struct snd_kcontrol * kcontrol mixer control

struct snd_ctl_elem_value * ucontrol Value

int **snd_soc_dapm_new_controls**(struct snd_soc_dapm_context * *dapm*, const struct snd_soc_dapm_widget * *widget*, int *num*)
create new dapm controls

Parameters

struct snd_soc_dapm_context * dapm DAPM context

const struct snd_soc_dapm_widget * widget widget array

int num number of widgets

Description

Creates new DAPM controls based upon the templates.

Returns 0 for success else error.

void **snd_soc_dapm_stream_event**(struct snd_soc_pcm_runtime * *rtd*, int *stream*, int *event*)
send a stream event to the dapm core

Parameters

struct snd_soc_pcm_runtime * rtd PCM runtime data

int stream stream name

int event stream event

Description

Sends a stream event to the dapm core. The core then makes any necessary widget power changes.

Returns 0 for success else error.

int **snd_soc_dapm_enable_pin_unlocked**(struct snd_soc_dapm_context * *dapm*, const char * *pin*)
enable pin.

Parameters

struct snd_soc_dapm_context * dapm DAPM context

const char * pin pin name

Description

Enables input/output pin and its parents or children widgets iff there is a valid audio route and active audio stream.

Requires external locking.

NOTE

snd_soc_dapm_sync() needs to be called after this for DAPM to do any widget power switching.

int **snd_soc_dapm_enable_pin**(struct snd_soc_dapm_context * *dapm*, const char * *pin*)
enable pin.

Parameters

struct snd_soc_dapm_context * dapm DAPM context

const char * pin pin name

Description

Enables input/output pin and its parents or children widgets iff there is a valid audio route and active audio stream.

NOTE

snd_soc_dapm_sync() needs to be called after this for DAPM to do any widget power switching.

int **snd_soc_dapm_force_enable_pin_unlocked**(struct snd_soc_dapm_context * *dapm*, const char * *pin*)
force a pin to be enabled

Parameters

struct snd_soc_dapm_context * dapm DAPM context

const char * pin pin name

Description

Enables input/output pin regardless of any other state. This is intended for use with microphone bias supplies used in microphone jack detection.

Requires external locking.

NOTE

snd_soc_dapm_sync() needs to be called after this for DAPM to do any widget power switching.

int **snd_soc_dapm_force_enable_pin**(struct snd_soc_dapm_context * *dapm*, const char * *pin*)
force a pin to be enabled

Parameters

struct snd_soc_dapm_context * dapm DAPM context

const char * pin pin name

Description

Enables input/output pin regardless of any other state. This is intended for use with microphone bias supplies used in microphone jack detection.

NOTE

snd_soc_dapm_sync() needs to be called after this for DAPM to do any widget power switching.

int **snd_soc_dapm_disable_pin_unlocked**(struct snd_soc_dapm_context * *dapm*, const char * *pin*)
disable pin.

Parameters

struct snd_soc_dapm_context * dapm DAPM context

const char * pin pin name

Description

Disables input/output pin and its parents or children widgets.

Requires external locking.

NOTE

snd_soc_dapm_sync() needs to be called after this for DAPM to do any widget power switching.

int **snd_soc_dapm_disable_pin**(struct snd_soc_dapm_context * *dapm*, const char * *pin*)
disable pin.

Parameters

struct snd_soc_dapm_context * dapm DAPM context

const char * pin pin name

Description

Disables input/output pin and its parents or children widgets.

NOTE

snd_soc_dapm_sync() needs to be called after this for DAPM to do any widget power switching.

int **snd_soc_dapm_nc_pin_unlocked**(struct snd_soc_dapm_context * *dapm*, const char * *pin*)
permanently disable pin.

Parameters

struct snd_soc_dapm_context * dapm DAPM context

const char * pin pin name

Description

Marks the specified pin as being not connected, disabling it along any parent or child widgets. At present this is identical to *snd_soc_dapm_disable_pin()* but in future it will be extended to do additional things such as disabling controls which only affect paths through the pin.

Requires external locking.

NOTE

snd_soc_dapm_sync() needs to be called after this for DAPM to do any widget power switching.

int **snd_soc_dapm_nc_pin**(struct snd_soc_dapm_context * *dapm*, const char * *pin*)
permanently disable pin.

Parameters

struct snd_soc_dapm_context * dapm DAPM context

const char * pin pin name

Description

Marks the specified pin as being not connected, disabling it along any parent or child widgets. At present this is identical to *snd_soc_dapm_disable_pin()* but in future it will be extended to do additional things such as disabling controls which only affect paths through the pin.

NOTE

snd_soc_dapm_sync() needs to be called after this for DAPM to do any widget power switching.

int **snd_soc_dapm_get_pin_status**(struct snd_soc_dapm_context * *dapm*, const char * *pin*)
get audio pin status

Parameters

struct snd_soc_dapm_context * dapm DAPM context

const char * pin audio signal pin endpoint (or start point)

Description

Get audio pin status - connected or disconnected.

Returns 1 for connected otherwise 0.

int **snd_soc_dapm_ignore_suspend**(struct snd_soc_dapm_context * *dapm*, const char * *pin*)
ignore suspend status for DAPM endpoint

Parameters

struct snd_soc_dapm_context * dapm DAPM context

const char * pin audio signal pin endpoint (or start point)

Description

Mark the given endpoint or pin as ignoring suspend. When the system is disabled a path between two endpoints flagged as ignoring suspend will not be disabled. The path must already be enabled via normal means at suspend time, it will not be turned on if it was not already enabled.

void **snd_soc_dapm_free**(struct snd_soc_dapm_context * *dapm*)
free dapm resources

Parameters

struct snd_soc_dapm_context * dapm DAPM context

Description

Free all dapm widgets and resources.

ASoC DMA Engine API

int **snd_dmaengine_pcm_prepare_slave_config**(struct snd_pcm_substream * *substream*, struct snd_pcm_hw_params * *params*, struct dma_slave_config * *slave_config*)
Generic prepare_slave_config callback

Parameters

struct snd_pcm_substream * substream PCM substream

struct snd_pcm_hw_params * params hw_params

struct dma_slave_config * slave_config DMA slave config to prepare

Description

This function can be used as a generic prepare_slave_config callback for platforms which make use of the snd_dmaengine_dai_dma_data struct for their DAI DMA data. Internally the function will first call snd_hwparams_to_dma_slave_config to fill in the slave config based on the hw_params, followed by snd_dmaengine_set_config_from_dai_data to fill in the remaining fields based on the DAI DMA data.

int **snd_dmaengine_pcm_register**(struct device * *dev*, const struct *snd_dmaengine_pcm_config* * *config*, unsigned int *flags*)
Register a dmaengine based PCM device

Parameters

struct device * dev The parent device for the PCM device

const struct snd_dmaengine_pcm_config * config Platform specific PCM configuration

unsigned int flags Platform specific quirks

void **snd_dmaengine_pcm_unregister**(struct device * *dev*)
Removes a dmaengine based PCM device

Parameters

struct device * dev Parent device the PCM was register with

Description

Removes a dmaengine based PCM device previously registered with snd_dmaengine_pcm_register.

Miscellaneous Functions

Hardware-Dependent Devices API

int **snd_hwdep_new**(struct snd_card * *card*, char * *id*, int *device*, struct snd_hwdep ** *rhwdep*)
create a new hwdep instance

Parameters

struct snd_card * card the card instance

char * id the id string

int device the device index (zero-based)

struct snd_hwdep ** rhwdep the pointer to store the new hwdep instance

Description

Creates a new hwdep instance with the given index on the card. The callbacks (hwdep->ops) must be set on the returned instance after this call manually by the caller.

Return

Zero if successful, or a negative error code on failure.

Jack Abstraction Layer API

enum **snd_jack_types**
Jack types which can be reported

Constants

SND_JACK_HEADPHONE Headphone

SND_JACK_MICROPHONE Microphone

SND_JACK_HEADSET Headset

SND_JACK_LINEOUT Line out

SND_JACK_MECHANICAL Mechanical switch

SND_JACK_VIDEOOUT Video out

SND_JACK_AVOUT AV (Audio Video) out

SND_JACK_LINEIN Line in

SND_JACK_BTN_0 Button 0

SND_JACK_BTN_1 Button 1

SND_JACK_BTN_2 Button 2

SND_JACK_BTN_3 Button 3

SND_JACK_BTN_4 Button 4

SND_JACK_BTN_5 Button 5

Description

These values are used as a bitmask.

Note that this must be kept in sync with the lookup table in sound/core/jack.c.

int **snd_jack_add_new_kctl**(struct snd_jack * *jack*, const char * *name*, int *mask*)
Create a new snd_jack_kctl and add it to jack

Parameters

struct snd_jack * jack the jack instance which the kctl will attaching to

const char * name the name for the snd_kcontrol object

int mask a bitmask of enum snd_jack_type values that can be detected by this snd_jack_kctl object.

Description

Creates a new snd_kcontrol object and adds it to the jack kctl_list.

Return

Zero if successful, or a negative error code on failure.

int **snd_jack_new**(struct snd_card * *card*, const char * *id*, int *type*, struct snd_jack ** *jjack*, bool *initial_kctl*, bool *phantom_jack*)
Create a new jack

Parameters

struct snd_card * card the card instance

const char * id an identifying string for this jack

int type a bitmask of enum snd_jack_type values that can be detected by this jack

struct snd_jack ** jjack Used to provide the allocated jack object to the caller.

bool initial_kctl if true, create a kcontrol and add it to the jack list.

bool phantom_jack Don't create a input device for phantom jacks.

Description

Creates a new jack object.

Return

Zero if successful, or a negative error code on failure. On success **jjack** will be initialised.

void **snd_jack_set_parent**(struct snd_jack * *jack*, struct device * *parent*)
Set the parent device for a jack

Parameters

struct snd_jack * jack The jack to configure

struct device * parent The device to set as parent for the jack.

Description

Set the parent for the jack devices in the device tree. This function is only valid prior to registration of the jack. If no parent is configured then the parent device will be the sound card.

int **snd_jack_set_key**(struct snd_jack * *jack*, enum *snd_jack_types* *type*, int *keytype*)
Set a key mapping on a jack

Parameters

struct snd_jack * jack The jack to configure

enum snd_jack_types type Jack report type for this key

int keytype Input layer key type to be reported

Description

Map a SND_JACK_BTN_* button type to an input layer key, allowing reporting of keys on accessories via the jack abstraction. If no mapping is provided but keys are enabled in the jack type then BTN_n numeric buttons will be reported.

If jacks are not reporting via the input API this call will have no effect.

Note that this is intended to be use by simple devices with small numbers of keys that can be reported. It is also possible to access the input device directly - devices with complex input capabilities on accessories should consider doing this rather than using this abstraction.

This function may only be called prior to registration of the jack.

Return

Zero if successful, or a negative error code on failure.

void **snd_jack_report**(struct snd_jack * *jack*, int *status*)
Report the current status of a jack

Parameters

struct snd_jack * jack The jack to report status for

int status The current status of the jack

int **snd_soc_component_set_jack**(struct snd_soc_component * *component*, struct snd_soc_jack * *jack*, void * *data*)
configure component jack.

Parameters

struct snd_soc_component * component COMPONENTs

struct snd_soc_jack * jack structure to use for the jack

void * data can be used if codec driver need extra data for configuring jack

Description

Configures and enables jack detection function.

int **snd_soc_card_jack_new**(struct snd_soc_card * *card*, const char * *id*, int *type*, struct snd_soc_jack * *jack*, struct *snd_soc_jack_pin* * *pins*, unsigned int *num_pins*)
Create a new jack

Parameters

struct snd_soc_card * card ASoC card

const char * id an identifying string for this jack

int type a bitmask of enum snd_jack_type values that can be detected by this jack

struct snd_soc_jack * jack structure to use for the jack

struct snd_soc_jack_pin * pins Array of jack pins to be added to the jack or NULL

unsigned int num_pins Number of elements in the **pins** array

Description

Creates a new jack object.

Returns zero if successful, or a negative error code on failure. On success jack will be initialised.

void **snd_soc_jack_report**(struct snd_soc_jack * *jack*, int *status*, int *mask*)
Report the current status for a jack

Parameters

struct snd_soc_jack * jack the jack

int status a bitmask of enum snd_jack_type values that are currently detected.

int mask a bitmask of enum snd_jack_type values that being reported.

Description

If configured using *snd_soc_jack_add_pins()* then the associated DAPM pins will be enabled or disabled as appropriate and DAPM synchronised.

Note

This function uses mutexes and should be called from a context which can sleep (such as a workqueue).

int **snd_soc_jack_add_zones**(struct snd_soc_jack * *jack*, int *count*, struct *snd_soc_jack_zone* * *zones*)
Associate voltage zones with jack

Parameters

struct snd_soc_jack * jack ASoC jack

int count Number of zones

struct snd_soc_jack_zone * zones Array of zones

Description

After this function has been called the zones specified in the array will be associated with the jack.

int **snd_soc_jack_get_type**(struct snd_soc_jack * *jack*, int *micbias_voltage*)
Based on the mic bias value, this function returns the type of jack from the zones declared in the jack type

Parameters

struct snd_soc_jack * jack ASoC jack

int micbias_voltage mic bias voltage at adc channel when jack is plugged in

Description

Based on the mic bias value passed, this function helps identify the type of jack from the already declared jack zones

int **snd_soc_jack_add_pins**(struct snd_soc_jack * *jack*, int *count*, struct *snd_soc_jack_pin* * *pins*)
Associate DAPM pins with an ASoC jack

Parameters

struct snd_soc_jack * jack ASoC jack

int count Number of pins

struct snd_soc_jack_pin * pins Array of pins

Description

After this function has been called the DAPM pins specified in the pins array will have their status updated to reflect the current state of the jack whenever the jack status is updated.

void **snd_soc_jack_notifier_register**(struct snd_soc_jack * *jack*, struct notifier_block * *nb*)
Register a notifier for jack status

Parameters

struct snd_soc_jack * jack ASoC jack

struct notifier_block * nb Notifier block to register

Description

Register for notification of the current status of the jack. Note that it is not possible to report additional jack events in the callback from the notifier, this is intended to support applications such as enabling electrical detection only when a mechanical detection event has occurred.

void **snd_soc_jack_notifier_unregister**(struct snd_soc_jack * *jack*, struct notifier_block * *nb*)
Unregister a notifier for jack status

Parameters

struct snd_soc_jack * jack ASoC jack

struct notifier_block * nb Notifier block to unregister

Description

Stop notifying for status changes.

int **snd_soc_jack_add_gpios**(struct snd_soc_jack * *jack*, int *count*, struct *snd_soc_jack_gpio* * *gpios*)
Associate GPIO pins with an ASoC jack

Parameters

struct snd_soc_jack * jack ASoC jack

int count number of pins

struct snd_soc_jack_gpio * gpios array of gpio pins

Description

This function will request gpio, set data direction and request irq for each gpio in the array.

int **snd_soc_jack_add_gpiods**(struct device * *gpiod_dev*, struct snd_soc_jack * *jack*, int *count*, struct *snd_soc_jack_gpio* * *gpios*)
Associate GPIO descriptor pins with an ASoC jack

Parameters

struct device * gpiod_dev GPIO consumer device

struct snd_soc_jack * jack ASoC jack

int count number of pins

struct snd_soc_jack_gpio * gpios array of gpio pins

Description

This function will request gpio, set data direction and request irq for each gpio in the array.

void **snd_soc_jack_free_gpios**(struct snd_soc_jack * *jack*, int *count*, struct *snd_soc_jack_gpio* * *gpios*)
Release GPIO pins' resources of an ASoC jack

Parameters

struct snd_soc_jack * jack ASoC jack

int count number of pins

struct snd_soc_jack_gpio * gpios array of gpio pins

Description

Release gpio and irq resources for gpio pins associated with an ASoC jack.

ISA DMA Helpers

void **snd_dma_program**(unsigned long *dma*, unsigned long *addr*, unsigned int *size*, unsigned short *mode*)
program an ISA DMA transfer

Parameters

unsigned long dma the dma number

unsigned long addr the physical address of the buffer

unsigned int size the DMA transfer size

unsigned short mode the DMA transfer mode, DMA_MODE_XXX

Description

Programs an ISA DMA transfer for the given buffer.

void **snd_dma_disable**(unsigned long *dma*)
stop the ISA DMA transfer

Parameters

unsigned long dma the dma number

Description

Stops the ISA DMA transfer.

unsigned int **snd_dma_pointer**(unsigned long *dma*, unsigned int *size*)
return the current pointer to DMA transfer buffer in bytes

Parameters

unsigned long dma the dma number

unsigned int size the dma transfer size

Return

The current pointer in DMA transfer buffer in bytes.

Other Helper Macros

snd_printk(*fmt*, *...*)
printk wrapper

Parameters

fmt format string

... variable arguments

Description

Works like `printk()` but prints the file and the line of the caller when configured with CONFIG_SND_VERBOSE_PRINTK.

snd_printd(*fmt*, *...*)
debug printk

Parameters

fmt format string

... variable arguments

Description

Works like *snd_printk()* for debugging purposes. Ignored when CONFIG_SND_DEBUG is not set.

snd_BUG()
give a BUG warning message and stack trace

Parameters

Description

Calls `WARN()` if CONFIG_SND_DEBUG is set. Ignored when CONFIG_SND_DEBUG is not set.

snd_printd_ratelimit()

Parameters

snd_BUG_ON(*cond*)
debugging check macro

Parameters

cond condition to evaluate

Description

Has the same behavior as WARN_ON when CONFIG_SND_DEBUG is set, otherwise just evaluates the conditional and returns the value.

snd_printdd(*format*, *...*)
debug printk

Parameters

format format string

... variable arguments

Description

Works like *snd_printk()* for debugging purposes. Ignored when CONFIG_SND_DEBUG_VERBOSE is not set.

Writing an ALSA Driver

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Preface

This document describes how to write an ALSA (Advanced Linux Sound Architecture) driver. The document focuses mainly on PCI soundcards. In the case of other device types, the API might be different, too. However, at least the ALSA kernel API is consistent, and therefore it would be still a bit help for writing them.

This document targets people who already have enough C language skills and have basic linux kernel programming knowledge. This document doesn't explain the general topic of linux kernel coding and doesn't cover low-level driver implementation details. It only describes the standard way to write a PCI sound driver on ALSA.

If you are already familiar with the older ALSA ver.0.5.x API, you can check the drivers such as `sound/pci/es1938.c` or `sound/pci/maestro3.c` which have also almost the same code-base in the ALSA 0.5.x tree, so you can compare the differences.

This document is still a draft version. Any feedback and corrections, please!!

File Tree Structure

General

The ALSA drivers are provided in two ways.

One is the trees provided as a tarball or via cvs from the ALSA's ftp site, and another is the 2.6 (or later) Linux kernel tree. To synchronize both, the ALSA driver tree is split into two different trees: alsa-kernel and alsa-driver. The former contains purely the source code for the Linux 2.6 (or later) tree. This tree is designed only for compilation on 2.6 or later environment. The latter, alsa-driver, contains many subtle

files for compiling ALSA drivers outside of the Linux kernel tree, wrapper functions for older 2.2 and 2.4 kernels, to adapt the latest kernel API, and additional drivers which are still in development or in tests. The drivers in alsa-driver tree will be moved to alsa-kernel (and eventually to the 2.6 kernel tree) when they are finished and confirmed to work fine.

The file tree structure of ALSA driver is depicted below. Both alsa-kernel and alsa-driver have almost the same file structure, except for "core" directory. It's named as "acore" in alsa-driver tree.

```
sound
        /core
                /oss
                /seq
                        /oss
                        /instr
        /ioctl32
        /include
        /drivers
                /mpu401
                /opl3
        /i2c
                /l3
        /synth
                /emux
        /pci
                /(cards)
        /isa
                /(cards)
        /arm
        /ppc
        /sparc
        /usb
        /pcmcia /(cards)
        /oss
```

core directory

This directory contains the middle layer which is the heart of ALSA drivers. In this directory, the native ALSA modules are stored. The sub-directories contain different modules and are dependent upon the kernel config.

core/oss

The codes for PCM and mixer OSS emulation modules are stored in this directory. The rawmidi OSS emulation is included in the ALSA rawmidi code since it's quite small. The sequencer code is stored in `core/seq/oss` directory (see *below*).

core/ioctl32

This directory contains the 32bit-ioctl wrappers for 64bit architectures such like x86-64, ppc64 and sparc64. For 32bit and alpha architectures, these are not compiled.

core/seq

This directory and its sub-directories are for the ALSA sequencer. This directory contains the sequencer core and primary sequencer modules such like snd-seq-midi, snd-seq-virmidi, etc. They are compiled only when `CONFIG_SND_SEQUENCER` is set in the kernel config.

core/seq/oss

This contains the OSS sequencer emulation codes.

core/seq/instr

This directory contains the modules for the sequencer instrument layer.

include directory

This is the place for the public header files of ALSA drivers, which are to be exported to user-space, or included by several files at different directories. Basically, the private header files should not be placed in this directory, but you may still find files there, due to historical reasons :)

drivers directory

This directory contains code shared among different drivers on different architectures. They are hence supposed not to be architecture-specific. For example, the dummy pcm driver and the serial MIDI driver are found in this directory. In the sub-directories, there is code for components which are independent from bus and cpu architectures.

drivers/mpu401

The MPU401 and MPU401-UART modules are stored here.

drivers/opl3 and opl4

The OPL3 and OPL4 FM-synth stuff is found here.

i2c directory

This contains the ALSA i2c components.

Although there is a standard i2c layer on Linux, ALSA has its own i2c code for some cards, because the soundcard needs only a simple operation and the standard i2c API is too complicated for such a purpose.

i2c/l3

This is a sub-directory for ARM L3 i2c.

synth directory

This contains the synth middle-level modules.

So far, there is only Emu8000/Emu10k1 synth driver under the `synth/emux` sub-directory.

pci directory

This directory and its sub-directories hold the top-level card modules for PCI soundcards and the code specific to the PCI BUS.

The drivers compiled from a single file are stored directly in the pci directory, while the drivers with several source files are stored on their own sub-directory (e.g. emu10k1, ice1712).

isa directory

This directory and its sub-directories hold the top-level card modules for ISA soundcards.

arm, ppc, and sparc directories

They are used for top-level card modules which are specific to one of these architectures.

usb directory

This directory contains the USB-audio driver. In the latest version, the USB MIDI driver is integrated in the usb-audio driver.

pcmcia directory

The PCMCIA, especially PCCard drivers will go here. CardBus drivers will be in the pci directory, because their API is identical to that of standard PCI cards.

oss directory

The OSS/Lite source files are stored here in Linux 2.6 (or later) tree. In the ALSA driver tarball, this directory is empty, of course :)

Basic Flow for PCI Drivers

Outline

The minimum flow for PCI soundcards is as follows:

- define the PCI ID table (see the section *PCI Entries*).
- create `probe` callback.
- create `remove` callback.
- create a `struct pci_driver` structure containing the three pointers above.
- create an `init` function just calling the `pci_register_driver()` to register the pci_driver table defined above.
- create an `exit` function to call the `pci_unregister_driver()` function.

Full Code Example

The code example is shown below. Some parts are kept unimplemented at this moment but will be filled in the next sections. The numbers in the comment lines of the `snd_mychip_probe()` function refer to details explained in the following section.

```
#include <linux/init.h>
#include <linux/pci.h>
#include <linux/slab.h>
#include <sound/core.h>
#include <sound/initval.h>

/* module parameters (see "Module Parameters") */
/* SNDRV_CARDS: maximum number of cards supported by this module */
static int index[SNDRV_CARDS] = SNDRV_DEFAULT_IDX;
```

```
static char *id[SNDRV_CARDS] = SNDRV_DEFAULT_STR;
static bool enable[SNDRV_CARDS] = SNDRV_DEFAULT_ENABLE_PNP;

/* definition of the chip-specific record */
struct mychip {
        struct snd_card *card;
        /* the rest of the implementation will be in section
         * "PCI Resource Management"
         */
};

/* chip-specific destructor
 * (see "PCI Resource Management")
 */
static int snd_mychip_free(struct mychip *chip)
{
        .... /* will be implemented later... */
}

/* component-destructor
 * (see "Management of Cards and Components")
 */
static int snd_mychip_dev_free(struct snd_device *device)
{
        return snd_mychip_free(device->device_data);
}

/* chip-specific constructor
 * (see "Management of Cards and Components")
 */
static int snd_mychip_create(struct snd_card *card,
                             struct pci_dev *pci,
                             struct mychip **rchip)
{
        struct mychip *chip;
        int err;
        static struct snd_device_ops ops = {
               .dev_free = snd_mychip_dev_free,
        };

        *rchip = NULL;

        /* check PCI availability here
         * (see "PCI Resource Management")
         */
        ....

        /* allocate a chip-specific data with zero filled */
        chip = kzalloc(sizeof(*chip), GFP_KERNEL);
        if (chip == NULL)
                return -ENOMEM;

        chip->card = card;

        /* rest of initialization here; will be implemented
         * later, see "PCI Resource Management"
         */
        ....

        err = snd_device_new(card, SNDRV_DEV_LOWLEVEL, chip, &ops);
        if (err < 0) {
                snd_mychip_free(chip);
                return err;
```

```
        }

        *rchip = chip;
        return 0;
}

/* constructor -- see "Driver Constructor" sub-section */
static int snd_mychip_probe(struct pci_dev *pci,
                            const struct pci_device_id *pci_id)
{
        static int dev;
        struct snd_card *card;
        struct mychip *chip;
        int err;

        /* (1) */
        if (dev >= SNDRV_CARDS)
                return -ENODEV;
        if (!enable[dev]) {
                dev++;
                return -ENOENT;
        }

        /* (2) */
        err = snd_card_new(&pci->dev, index[dev], id[dev], THIS_MODULE,
                           0, &card);
        if (err < 0)
                return err;

        /* (3) */
        err = snd_mychip_create(card, pci, &chip);
        if (err < 0) {
                snd_card_free(card);
                return err;
        }

        /* (4) */
        strcpy(card->driver, "My Chip");
        strcpy(card->shortname, "My Own Chip 123");
        sprintf(card->longname, "%s at 0x%lx irq %i",
                card->shortname, chip->ioport, chip->irq);

        /* (5) */
        .... /* implemented later */

        /* (6) */
        err = snd_card_register(card);
        if (err < 0) {
                snd_card_free(card);
                return err;
        }

        /* (7) */
        pci_set_drvdata(pci, card);
        dev++;
        return 0;
}

/* destructor -- see the "Destructor" sub-section */
static void snd_mychip_remove(struct pci_dev *pci)
{
        snd_card_free(pci_get_drvdata(pci));
        pci_set_drvdata(pci, NULL);
```

```
}
```

Driver Constructor

The real constructor of PCI drivers is the probe callback. The probe callback and other component-constructors which are called from the probe callback cannot be used with the __init prefix because any PCI device could be a hotplug device.

In the probe callback, the following scheme is often used.

1) Check and increment the device index.

```
static int dev;
....
if (dev >= SNDRV_CARDS)
        return -ENODEV;
if (!enable[dev]) {
        dev++;
        return -ENOENT;
}
```

where enable[dev] is the module option.

Each time the probe callback is called, check the availability of the device. If not available, simply increment the device index and returns. dev will be incremented also later (*step 7*).

2) Create a card instance

```
struct snd_card *card;
int err;
....
err = snd_card_new(&pci->dev, index[dev], id[dev], THIS_MODULE,
                   0, &card);
```

The details will be explained in the section *Management of Cards and Components*.

3) Create a main component

In this part, the PCI resources are allocated.

```
struct mychip *chip;
....
err = snd_mychip_create(card, pci, &chip);
if (err < 0) {
        snd_card_free(card);
        return err;
}
```

The details will be explained in the section *PCI Resource Management*.

4) Set the driver ID and name strings.

```
strcpy(card->driver, "My Chip");
strcpy(card->shortname, "My Own Chip 123");
sprintf(card->longname, "%s at 0x%lx irq %i",
        card->shortname, chip->ioport, chip->irq);
```

The driver field holds the minimal ID string of the chip. This is used by alsa-lib's configurator, so keep it simple but unique. Even the same driver can have different driver IDs to distinguish the functionality of each chip type.

The shortname field is a string shown as more verbose name. The longname field contains the information shown in `/proc/asound/cards`.

5) Create other components, such as mixer, MIDI, etc.

Here you define the basic components such as *PCM*, mixer (e.g. *AC97*), MIDI (e.g. *MPU-401*), and other interfaces. Also, if you want a *proc file*, define it here, too.

6) Register the card instance.

```
err = snd_card_register(card);
if (err < 0) {
        snd_card_free(card);
        return err;
}
```

Will be explained in the section *Management of Cards and Components*, too.

7) Set the PCI driver data and return zero.

```
pci_set_drvdata(pci, card);
dev++;
return 0;
```

In the above, the card record is stored. This pointer is used in the remove callback and power-management callbacks, too.

Destructor

The destructor, remove callback, simply releases the card instance. Then the ALSA middle layer will release all the attached components automatically.

It would be typically like the following:

```
static void snd_mychip_remove(struct pci_dev *pci)
{
        snd_card_free(pci_get_drvdata(pci));
        pci_set_drvdata(pci, NULL);
}
```

The above code assumes that the card pointer is set to the PCI driver data.

Header Files

For the above example, at least the following include files are necessary.

```
#include <linux/init.h>
#include <linux/pci.h>
#include <linux/slab.h>
#include <sound/core.h>
#include <sound/initval.h>
```

where the last one is necessary only when module options are defined in the source file. If the code is split into several files, the files without module options don't need them.

In addition to these headers, you'll need <linux/interrupt.h> for interrupt handling, and <asm/io.h> for I/O access. If you use the mdelay() or udelay() functions, you'll need to include <linux/delay.h> too.

The ALSA interfaces like the PCM and control APIs are defined in other <sound/xxx.h> header files. They have to be included after <sound/core.h>.

Management of Cards and Components

Card Instance

For each soundcard, a "card" record must be allocated.

A card record is the headquarters of the soundcard. It manages the whole list of devices (components) on the soundcard, such as PCM, mixers, MIDI, synthesizer, and so on. Also, the card record holds the ID and the name strings of the card, manages the root of proc files, and controls the power-management states and hotplug disconnections. The component list on the card record is used to manage the correct release of resources at destruction.

As mentioned above, to create a card instance, call *snd_card_new()*.

```
struct snd_card *card;
int err;
err = snd_card_new(&pci->dev, index, id, module, extra_size, &card);
```

The function takes six arguments: the parent device pointer, the card-index number, the id string, the module pointer (usually THIS_MODULE), the size of extra-data space, and the pointer to return the card instance. The extra_size argument is used to allocate card->private_data for the chip-specific data. Note that these data are allocated by *snd_card_new()*.

The first argument, the pointer of struct struct device, specifies the parent device. For PCI devices, typically &pci-> is passed there.

Components

After the card is created, you can attach the components (devices) to the card instance. In an ALSA driver, a component is represented as a struct snd_device object. A component can be a PCM instance, a control interface, a raw MIDI interface, etc. Each such instance has one component entry.

A component can be created via *snd_device_new()* function.

```
snd_device_new(card, SNDRV_DEV_XXX, chip, &ops);
```

This takes the card pointer, the device-level (SNDRV_DEV_XXX), the data pointer, and the callback pointers (&ops). The device-level defines the type of components and the order of registration and de-registration. For most components, the device-level is already defined. For a user-defined component, you can use SNDRV_DEV_LOWLEVEL.

This function itself doesn't allocate the data space. The data must be allocated manually beforehand, and its pointer is passed as the argument. This pointer (chip in the above example) is used as the identifier for the instance.

Each pre-defined ALSA component such as ac97 and pcm calls *snd_device_new()* inside its constructor. The destructor for each component is defined in the callback pointers. Hence, you don't need to take care of calling a destructor for such a component.

If you wish to create your own component, you need to set the destructor function to the dev_free callback in the ops, so that it can be released automatically via *snd_card_free()*. The next example will show an implementation of chip-specific data.

Chip-Specific Data

Chip-specific information, e.g. the I/O port address, its resource pointer, or the irq number, is stored in the chip-specific record.

```
struct mychip {
        ....
};
```

In general, there are two ways of allocating the chip record.

1. Allocating via `snd_card_new()`.

As mentioned above, you can pass the extra-data-length to the 5th argument of *snd_card_new()*, i.e.

```
err = snd_card_new(&pci->dev, index[dev], id[dev], THIS_MODULE,
                   sizeof(struct mychip), &card);
```

`struct mychip` is the type of the chip record.

In return, the allocated record can be accessed as

```
struct mychip *chip = card->private_data;
```

With this method, you don't have to allocate twice. The record is released together with the card instance.

2. Allocating an extra device.

After allocating a card instance via *snd_card_new()* (with `0` on the 4th arg), call `kzalloc()`.

```
struct snd_card *card;
struct mychip *chip;
err = snd_card_new(&pci->dev, index[dev], id[dev], THIS_MODULE,
                   0, &card);
.....
chip = kzalloc(sizeof(*chip), GFP_KERNEL);
```

The chip record should have the field to hold the card pointer at least,

```
struct mychip {
        struct snd_card *card;
        ....
};
```

Then, set the card pointer in the returned chip instance.

```
chip->card = card;
```

Next, initialize the fields, and register this chip record as a low-level device with a specified ops,

```
static struct snd_device_ops ops = {
        .dev_free =        snd_mychip_dev_free,
};
....
snd_device_new(card, SNDRV_DEV_LOWLEVEL, chip, &ops);
```

`snd_mychip_dev_free()` is the device-destructor function, which will call the real destructor.

```
static int snd_mychip_dev_free(struct snd_device *device)
{
        return snd_mychip_free(device->device_data);
}
```

where snd_mychip_free() is the real destructor.

Registration and Release

After all components are assigned, register the card instance by calling *snd_card_register()*. Access to the device files is enabled at this point. That is, before *snd_card_register()* is called, the components are safely inaccessible from external side. If this call fails, exit the probe function after releasing the card via *snd_card_free()*.

For releasing the card instance, you can call simply *snd_card_free()*. As mentioned earlier, all components are released automatically by this call.

For a device which allows hotplugging, you can use *snd_card_free_when_closed()*. This one will postpone the destruction until all devices are closed.

PCI Resource Management

Full Code Example

In this section, we'll complete the chip-specific constructor, destructor and PCI entries. Example code is shown first, below.

```
struct mychip {
        struct snd_card *card;
        struct pci_dev *pci;

        unsigned long port;
        int irq;
};

static int snd_mychip_free(struct mychip *chip)
{
        /* disable hardware here if any */
        .... /* (not implemented in this document) */

        /* release the irq */
        if (chip->irq >= 0)
                free_irq(chip->irq, chip);
        /* release the I/O ports & memory */
        pci_release_regions(chip->pci);
        /* disable the PCI entry */
        pci_disable_device(chip->pci);
        /* release the data */
        kfree(chip);
        return 0;
}

/* chip-specific constructor */
static int snd_mychip_create(struct snd_card *card,
                             struct pci_dev *pci,
                             struct mychip **rchip)
{
        struct mychip *chip;
        int err;
        static struct snd_device_ops ops = {
               .dev_free = snd_mychip_dev_free,
        };

        *rchip = NULL;

        /* initialize the PCI entry */
```

```
        err = pci_enable_device(pci);
        if (err < 0)
                return err;
        /* check PCI availability (28bit DMA) */
        if (pci_set_dma_mask(pci, DMA_BIT_MASK(28)) < 0 ||
            pci_set_consistent_dma_mask(pci, DMA_BIT_MASK(28)) < 0) {
                printk(KERN_ERR "error to set 28bit mask DMA\n");
                pci_disable_device(pci);
                return -ENXIO;
        }

        chip = kzalloc(sizeof(*chip), GFP_KERNEL);
        if (chip == NULL) {
                pci_disable_device(pci);
                return -ENOMEM;
        }

        /* initialize the stuff */
        chip->card = card;
        chip->pci = pci;
        chip->irq = -1;

        /* (1) PCI resource allocation */
        err = pci_request_regions(pci, "My Chip");
        if (err < 0) {
                kfree(chip);
                pci_disable_device(pci);
                return err;
        }
        chip->port = pci_resource_start(pci, 0);
        if (request_irq(pci->irq, snd_mychip_interrupt,
                        IRQF_SHARED, KBUILD_MODNAME, chip)) {
                printk(KERN_ERR "cannot grab irq %d\n", pci->irq);
                snd_mychip_free(chip);
                return -EBUSY;
        }
        chip->irq = pci->irq;

        /* (2) initialization of the chip hardware */
        .... /*   (not implemented in this document) */

        err = snd_device_new(card, SNDRV_DEV_LOWLEVEL, chip, &ops);
        if (err < 0) {
                snd_mychip_free(chip);
                return err;
        }

        *rchip = chip;
        return 0;
}

/* PCI IDs */
static struct pci_device_id snd_mychip_ids[] = {
        { PCI_VENDOR_ID_FOO, PCI_DEVICE_ID_BAR,
          PCI_ANY_ID, PCI_ANY_ID, 0, 0, 0, },
        ....
        { 0, }
};
MODULE_DEVICE_TABLE(pci, snd_mychip_ids);

/* pci_driver definition */
static struct pci_driver driver = {
        .name = KBUILD_MODNAME,
```

```
        .id_table = snd_mychip_ids,
        .probe = snd_mychip_probe,
        .remove = snd_mychip_remove,
};

/* module initialization */
static int __init alsa_card_mychip_init(void)
{
        return pci_register_driver(&driver);
}

/* module clean up */
static void __exit alsa_card_mychip_exit(void)
{
        pci_unregister_driver(&driver);
}

module_init(alsa_card_mychip_init)
module_exit(alsa_card_mychip_exit)

EXPORT_NO_SYMBOLS; /* for old kernels only */
```

Some Hafta's

The allocation of PCI resources is done in the `probe` function, and usually an extra `xxx_create()` function is written for this purpose.

In the case of PCI devices, you first have to call the `pci_enable_device()` function before allocating resources. Also, you need to set the proper PCI DMA mask to limit the accessed I/O range. In some cases, you might need to call `pci_set_master()` function, too.

Suppose the 28bit mask, and the code to be added would be like:

```
err = pci_enable_device(pci);
if (err < 0)
        return err;
if (pci_set_dma_mask(pci, DMA_BIT_MASK(28)) < 0 ||
    pci_set_consistent_dma_mask(pci, DMA_BIT_MASK(28)) < 0) {
        printk(KERN_ERR "error to set 28bit mask DMA\n");
        pci_disable_device(pci);
        return -ENXIO;
}
```

Resource Allocation

The allocation of I/O ports and irqs is done via standard kernel functions. Unlike ALSA ver.0.5.x., there are no helpers for that. And these resources must be released in the destructor function (see below). Also, on ALSA 0.9.x, you don't need to allocate (pseudo-)DMA for PCI like in ALSA 0.5.x.

Now assume that the PCI device has an I/O port with 8 bytes and an interrupt. Then `struct mychip` will have the following fields:

```
struct mychip {
        struct snd_card *card;

        unsigned long port;
        int irq;
};
```

For an I/O port (and also a memory region), you need to have the resource pointer for the standard resource management. For an irq, you have to keep only the irq number (integer). But you need to initialize this

number as -1 before actual allocation, since irq 0 is valid. The port address and its resource pointer can be initialized as null by `kzalloc()` automatically, so you don't have to take care of resetting them.

The allocation of an I/O port is done like this:

```
err = pci_request_regions(pci, "My Chip");
if (err < 0) {
        kfree(chip);
        pci_disable_device(pci);
        return err;
}
chip->port = pci_resource_start(pci, 0);
```

It will reserve the I/O port region of 8 bytes of the given PCI device. The returned value, `chip->res_port`, is allocated via `kmalloc()` by `request_region()`. The pointer must be released via `kfree()`, but there is a problem with this. This issue will be explained later.

The allocation of an interrupt source is done like this:

```
if (request_irq(pci->irq, snd_mychip_interrupt,
                IRQF_SHARED, KBUILD_MODNAME, chip)) {
        printk(KERN_ERR "cannot grab irq %d\n", pci->irq);
        snd_mychip_free(chip);
        return -EBUSY;
}
chip->irq = pci->irq;
```

where `snd_mychip_interrupt()` is the interrupt handler defined *later*. Note that `chip->irq` should be defined only when `request_irq()` succeeded.

On the PCI bus, interrupts can be shared. Thus, `IRQF_SHARED` is used as the interrupt flag of `request_irq()`.

The last argument of `request_irq()` is the data pointer passed to the interrupt handler. Usually, the chip-specific record is used for that, but you can use what you like, too.

I won't give details about the interrupt handler at this point, but at least its appearance can be explained now. The interrupt handler looks usually like the following:

```
static irqreturn_t snd_mychip_interrupt(int irq, void *dev_id)
{
        struct mychip *chip = dev_id;
        ....
        return IRQ_HANDLED;
}
```

Now let's write the corresponding destructor for the resources above. The role of destructor is simple: disable the hardware (if already activated) and release the resources. So far, we have no hardware part, so the disabling code is not written here.

To release the resources, the "check-and-release" method is a safer way. For the interrupt, do like this:

```
if (chip->irq >= 0)
        free_irq(chip->irq, chip);
```

Since the irq number can start from 0, you should initialize `chip->irq` with a negative value (e.g. -1), so that you can check the validity of the irq number as above.

When you requested I/O ports or memory regions via `pci_request_region()` or `pci_request_regions()` like in this example, release the resource(s) using the corresponding function, `pci_release_region()` or `pci_release_regions()`.

```
pci_release_regions(chip->pci);
```

When you requested manually via request_region() or request_mem_region(), you can release it via release_resource(). Suppose that you keep the resource pointer returned from request_region() in chip->res_port, the release procedure looks like:

```
release_and_free_resource(chip->res_port);
```

Don't forget to call pci_disable_device() before the end.

And finally, release the chip-specific record.

```
kfree(chip);
```

We didn't implement the hardware disabling part in the above. If you need to do this, please note that the destructor may be called even before the initialization of the chip is completed. It would be better to have a flag to skip hardware disabling if the hardware was not initialized yet.

When the chip-data is assigned to the card using *snd_device_new()* with SNDRV_DEV_LOWLELVEL , its destructor is called at the last. That is, it is assured that all other components like PCMs and controls have already been released. You don't have to stop PCMs, etc. explicitly, but just call low-level hardware stopping.

The management of a memory-mapped region is almost as same as the management of an I/O port. You'll need three fields like the following:

```
struct mychip {
        ....
        unsigned long iobase_phys;
        void __iomem *iobase_virt;
};
```

and the allocation would be like below:

```
if ((err = pci_request_regions(pci, "My Chip")) < 0) {
        kfree(chip);
        return err;
}
chip->iobase_phys = pci_resource_start(pci, 0);
chip->iobase_virt = ioremap_nocache(chip->iobase_phys,
                                    pci_resource_len(pci, 0));
```

and the corresponding destructor would be:

```
static int snd_mychip_free(struct mychip *chip)
{
        ....
        if (chip->iobase_virt)
                iounmap(chip->iobase_virt);
        ....
        pci_release_regions(chip->pci);
        ....
}
```

PCI Entries

So far, so good. Let's finish the missing PCI stuff. At first, we need a struct pci_device_id table for this chipset. It's a table of PCI vendor/device ID number, and some masks.

For example,

```
static struct pci_device_id snd_mychip_ids[] = {
        { PCI_VENDOR_ID_FOO, PCI_DEVICE_ID_BAR,
          PCI_ANY_ID, PCI_ANY_ID, 0, 0, 0, },
        ....
```

```
        { 0, }
};
MODULE_DEVICE_TABLE(pci, snd_mychip_ids);
```

The first and second fields of the `struct pci_device_id` structure are the vendor and device IDs. If you have no reason to filter the matching devices, you can leave the remaining fields as above. The last field of the `struct pci_device_id` struct contains private data for this entry. You can specify any value here, for example, to define specific operations for supported device IDs. Such an example is found in the intel8x0 driver.

The last entry of this list is the terminator. You must specify this all-zero entry.

Then, prepare the `struct pci_driver` record:

```
static struct pci_driver driver = {
        .name = KBUILD_MODNAME,
        .id_table = snd_mychip_ids,
        .probe = snd_mychip_probe,
        .remove = snd_mychip_remove,
};
```

The `probe` and `remove` functions have already been defined in the previous sections. The `name` field is the name string of this device. Note that you must not use a slash "/" in this string.

And at last, the module entries:

```
static int __init alsa_card_mychip_init(void)
{
        return pci_register_driver(&driver);
}

static void __exit alsa_card_mychip_exit(void)
{
        pci_unregister_driver(&driver);
}

module_init(alsa_card_mychip_init)
module_exit(alsa_card_mychip_exit)
```

Note that these module entries are tagged with `__init` and `__exit` prefixes.

Oh, one thing was forgotten. If you have no exported symbols, you need to declare it in 2.2 or 2.4 kernels (it's not necessary in 2.6 kernels).

```
EXPORT_NO_SYMBOLS;
```

That's all!

PCM Interface

General

The PCM middle layer of ALSA is quite powerful and it is only necessary for each driver to implement the low-level functions to access its hardware.

For accessing to the PCM layer, you need to include `<sound/pcm.h>` first. In addition, `<sound/pcm_params.h>` might be needed if you access to some functions related with hw_param.

Each card device can have up to four pcm instances. A pcm instance corresponds to a pcm device file. The limitation of number of instances comes only from the available bit size of the Linux's device numbers. Once when 64bit device number is used, we'll have more pcm instances available.

A pcm instance consists of pcm playback and capture streams, and each pcm stream consists of one or more pcm substreams. Some soundcards support multiple playback functions. For example, emu10k1 has a PCM playback of 32 stereo substreams. In this case, at each open, a free substream is (usually) automatically chosen and opened. Meanwhile, when only one substream exists and it was already opened, the successful open will either block or error with EAGAIN according to the file open mode. But you don't have to care about such details in your driver. The PCM middle layer will take care of such work.

Full Code Example

The example code below does not include any hardware access routines but shows only the skeleton, how to build up the PCM interfaces.

```
#include <sound/pcm.h>
....

/* hardware definition */
static struct snd_pcm_hardware snd_mychip_playback_hw = {
        .info = (SNDRV_PCM_INFO_MMAP |
                 SNDRV_PCM_INFO_INTERLEAVED |
                 SNDRV_PCM_INFO_BLOCK_TRANSFER |
                 SNDRV_PCM_INFO_MMAP_VALID),
        .formats =          SNDRV_PCM_FMTBIT_S16_LE,
        .rates =            SNDRV_PCM_RATE_8000_48000,
        .rate_min =         8000,
        .rate_max =         48000,
        .channels_min =     2,
        .channels_max =     2,
        .buffer_bytes_max = 32768,
        .period_bytes_min = 4096,
        .period_bytes_max = 32768,
        .periods_min =      1,
        .periods_max =      1024,
};

/* hardware definition */
static struct snd_pcm_hardware snd_mychip_capture_hw = {
        .info = (SNDRV_PCM_INFO_MMAP |
                 SNDRV_PCM_INFO_INTERLEAVED |
                 SNDRV_PCM_INFO_BLOCK_TRANSFER |
                 SNDRV_PCM_INFO_MMAP_VALID),
        .formats =          SNDRV_PCM_FMTBIT_S16_LE,
        .rates =            SNDRV_PCM_RATE_8000_48000,
        .rate_min =         8000,
        .rate_max =         48000,
        .channels_min =     2,
        .channels_max =     2,
        .buffer_bytes_max = 32768,
        .period_bytes_min = 4096,
        .period_bytes_max = 32768,
        .periods_min =      1,
        .periods_max =      1024,
};

/* open callback */
static int snd_mychip_playback_open(struct snd_pcm_substream *substream)
{
        struct mychip *chip = snd_pcm_substream_chip(substream);
        struct snd_pcm_runtime *runtime = substream->runtime;

        runtime->hw = snd_mychip_playback_hw;
        /* more hardware-initialization will be done here */
        ....
```

```
        return 0;
}

/* close callback */
static int snd_mychip_playback_close(struct snd_pcm_substream *substream)
{
        struct mychip *chip = snd_pcm_substream_chip(substream);
        /* the hardware-specific codes will be here */
        ....
        return 0;

}

/* open callback */
static int snd_mychip_capture_open(struct snd_pcm_substream *substream)
{
        struct mychip *chip = snd_pcm_substream_chip(substream);
        struct snd_pcm_runtime *runtime = substream->runtime;

        runtime->hw = snd_mychip_capture_hw;
        /* more hardware-initialization will be done here */
        ....
        return 0;
}

/* close callback */
static int snd_mychip_capture_close(struct snd_pcm_substream *substream)
{
        struct mychip *chip = snd_pcm_substream_chip(substream);
        /* the hardware-specific codes will be here */
        ....
        return 0;

}

/* hw_params callback */
static int snd_mychip_pcm_hw_params(struct snd_pcm_substream *substream,
                            struct snd_pcm_hw_params *hw_params)
{
        return snd_pcm_lib_malloc_pages(substream,
                                    params_buffer_bytes(hw_params));
}

/* hw_free callback */
static int snd_mychip_pcm_hw_free(struct snd_pcm_substream *substream)
{
        return snd_pcm_lib_free_pages(substream);
}

/* prepare callback */
static int snd_mychip_pcm_prepare(struct snd_pcm_substream *substream)
{
        struct mychip *chip = snd_pcm_substream_chip(substream);
        struct snd_pcm_runtime *runtime = substream->runtime;

        /* set up the hardware with the current configuration
         * for example...
         */
        mychip_set_sample_format(chip, runtime->format);
        mychip_set_sample_rate(chip, runtime->rate);
        mychip_set_channels(chip, runtime->channels);
        mychip_set_dma_setup(chip, runtime->dma_addr,
                            chip->buffer_size,
```

```
                                chip->period_size);
        return 0;
}

/* trigger callback */
static int snd_mychip_pcm_trigger(struct snd_pcm_substream *substream,
                                  int cmd)
{
        switch (cmd) {
        case SNDRV_PCM_TRIGGER_START:
                /* do something to start the PCM engine */
                ....
                break;
        case SNDRV_PCM_TRIGGER_STOP:
                /* do something to stop the PCM engine */
                ....
                break;
        default:
                return -EINVAL;
        }
}

/* pointer callback */
static snd_pcm_uframes_t
snd_mychip_pcm_pointer(struct snd_pcm_substream *substream)
{
        struct mychip *chip = snd_pcm_substream_chip(substream);
        unsigned int current_ptr;

        /* get the current hardware pointer */
        current_ptr = mychip_get_hw_pointer(chip);
        return current_ptr;
}

/* operators */
static struct snd_pcm_ops snd_mychip_playback_ops = {
        .open =        snd_mychip_playback_open,
        .close =       snd_mychip_playback_close,
        .ioctl =       snd_pcm_lib_ioctl,
        .hw_params =   snd_mychip_pcm_hw_params,
        .hw_free =     snd_mychip_pcm_hw_free,
        .prepare =     snd_mychip_pcm_prepare,
        .trigger =     snd_mychip_pcm_trigger,
        .pointer =     snd_mychip_pcm_pointer,
};

/* operators */
static struct snd_pcm_ops snd_mychip_capture_ops = {
        .open =        snd_mychip_capture_open,
        .close =       snd_mychip_capture_close,
        .ioctl =       snd_pcm_lib_ioctl,
        .hw_params =   snd_mychip_pcm_hw_params,
        .hw_free =     snd_mychip_pcm_hw_free,
        .prepare =     snd_mychip_pcm_prepare,
        .trigger =     snd_mychip_pcm_trigger,
        .pointer =     snd_mychip_pcm_pointer,
};

/*
 *  definitions of capture are omitted here...
 */

/* create a pcm device */
```

```
static int snd_mychip_new_pcm(struct mychip *chip)
{
        struct snd_pcm *pcm;
        int err;

        err = snd_pcm_new(chip->card, "My Chip", 0, 1, 1, &pcm);
        if (err < 0)
                return err;
        pcm->private_data = chip;
        strcpy(pcm->name, "My Chip");
        chip->pcm = pcm;
        /* set operators */
        snd_pcm_set_ops(pcm, SNDRV_PCM_STREAM_PLAYBACK,
                        &snd_mychip_playback_ops);
        snd_pcm_set_ops(pcm, SNDRV_PCM_STREAM_CAPTURE,
                        &snd_mychip_capture_ops);
        /* pre-allocation of buffers */
        /* NOTE: this may fail */
        snd_pcm_lib_preallocate_pages_for_all(pcm, SNDRV_DMA_TYPE_DEV,
                                              snd_dma_pci_data(chip->pci),
                                              64*1024, 64*1024);
        return 0;
}
```

PCM Constructor

A pcm instance is allocated by the *snd_pcm_new()* function. It would be better to create a constructor for pcm, namely,

```
static int snd_mychip_new_pcm(struct mychip *chip)
{
        struct snd_pcm *pcm;
        int err;

        err = snd_pcm_new(chip->card, "My Chip", 0, 1, 1, &pcm);
        if (err < 0)
                return err;
        pcm->private_data = chip;
        strcpy(pcm->name, "My Chip");
        chip->pcm = pcm;
        ....
        return 0;
}
```

The *snd_pcm_new()* function takes four arguments. The first argument is the card pointer to which this pcm is assigned, and the second is the ID string.

The third argument (`index`, 0 in the above) is the index of this new pcm. It begins from zero. If you create more than one pcm instances, specify the different numbers in this argument. For example, `index = 1` for the second PCM device.

The fourth and fifth arguments are the number of substreams for playback and capture, respectively. Here 1 is used for both arguments. When no playback or capture substreams are available, pass 0 to the corresponding argument.

If a chip supports multiple playbacks or captures, you can specify more numbers, but they must be handled properly in open/close, etc. callbacks. When you need to know which substream you are referring to, then it can be obtained from `struct snd_pcm_substream` data passed to each callback as follows:

```
struct snd_pcm_substream *substream;
int index = substream->number;
```

After the pcm is created, you need to set operators for each pcm stream.

```
snd_pcm_set_ops(pcm, SNDRV_PCM_STREAM_PLAYBACK,
                &snd_mychip_playback_ops);
snd_pcm_set_ops(pcm, SNDRV_PCM_STREAM_CAPTURE,
                &snd_mychip_capture_ops);
```

The operators are defined typically like this:

```
static struct snd_pcm_ops snd_mychip_playback_ops = {
        .open =         snd_mychip_pcm_open,
        .close =        snd_mychip_pcm_close,
        .ioctl =        snd_pcm_lib_ioctl,
        .hw_params =    snd_mychip_pcm_hw_params,
        .hw_free =      snd_mychip_pcm_hw_free,
        .prepare =      snd_mychip_pcm_prepare,
        .trigger =      snd_mychip_pcm_trigger,
        .pointer =      snd_mychip_pcm_pointer,
};
```

All the callbacks are described in the *Operators* subsection.

After setting the operators, you probably will want to pre-allocate the buffer. For the pre-allocation, simply call the following:

```
snd_pcm_lib_preallocate_pages_for_all(pcm, SNDRV_DMA_TYPE_DEV,
                                      snd_dma_pci_data(chip->pci),
                                      64*1024, 64*1024);
```

It will allocate a buffer up to 64kB as default. Buffer management details will be described in the later section *Buffer and Memory Management*.

Additionally, you can set some extra information for this pcm in `pcm->info_flags`. The available values are defined as `SNDRV_PCM_INFO_XXX` in `<sound/asound.h>`, which is used for the hardware definition (described later). When your soundchip supports only half-duplex, specify like this:

```
pcm->info_flags = SNDRV_PCM_INFO_HALF_DUPLEX;
```

... And the Destructor?

The destructor for a pcm instance is not always necessary. Since the pcm device will be released by the middle layer code automatically, you don't have to call the destructor explicitly.

The destructor would be necessary if you created special records internally and needed to release them. In such a case, set the destructor function to `pcm->private_free`:

```
static void mychip_pcm_free(struct snd_pcm *pcm)
{
        struct mychip *chip = snd_pcm_chip(pcm);
        /* free your own data */
        kfree(chip->my_private_pcm_data);
        /* do what you like else */
        ....
}

static int snd_mychip_new_pcm(struct mychip *chip)
{
        struct snd_pcm *pcm;
        ....
        /* allocate your own data */
        chip->my_private_pcm_data = kmalloc(...);
        /* set the destructor */
        pcm->private_data = chip;
```

```
        pcm->private_free = mychip_pcm_free;
        ....
}
```

Runtime Pointer - The Chest of PCM Information

When the PCM substream is opened, a PCM runtime instance is allocated and assigned to the substream. This pointer is accessible via `substream->runtime`. This runtime pointer holds most information you need to control the PCM: the copy of hw_params and sw_params configurations, the buffer pointers, mmap records, spinlocks, etc.

The definition of runtime instance is found in <sound/pcm.h>. Here are the contents of this file:

```
struct _snd_pcm_runtime {
        /* -- Status -- */
        struct snd_pcm_substream *trigger_master;
        snd_timestamp_t trigger_tstamp;       /* trigger timestamp */
        int overrange;
        snd_pcm_uframes_t avail_max;
        snd_pcm_uframes_t hw_ptr_base;        /* Position at buffer restart */
        snd_pcm_uframes_t hw_ptr_interrupt; /* Position at interrupt time*/

        /* -- HW params -- */
        snd_pcm_access_t access;      /* access mode */
        snd_pcm_format_t format;      /* SNDRV_PCM_FORMAT_* */
        snd_pcm_subformat_t subformat;        /* subformat */
        unsigned int rate;            /* rate in Hz */
        unsigned int channels;                /* channels */
        snd_pcm_uframes_t period_size;        /* period size */
        unsigned int periods;         /* periods */
        snd_pcm_uframes_t buffer_size;        /* buffer size */
        unsigned int tick_time;               /* tick time */
        snd_pcm_uframes_t min_align;  /* Min alignment for the format */
        size_t byte_align;
        unsigned int frame_bits;
        unsigned int sample_bits;
        unsigned int info;
        unsigned int rate_num;
        unsigned int rate_den;

        /* -- SW params -- */
        struct timespec tstamp_mode;  /* mmap timestamp is updated */
        unsigned int period_step;
        unsigned int sleep_min;               /* min ticks to sleep */
        snd_pcm_uframes_t start_threshold;
        snd_pcm_uframes_t stop_threshold;
        snd_pcm_uframes_t silence_threshold; /* Silence filling happens when
                                                noise is nearest than this */
        snd_pcm_uframes_t silence_size;       /* Silence filling size */
        snd_pcm_uframes_t boundary;   /* pointers wrap point */

        snd_pcm_uframes_t silenced_start;
        snd_pcm_uframes_t silenced_size;

        snd_pcm_sync_id_t sync;               /* hardware synchronization ID */

        /* -- mmap -- */
        volatile struct snd_pcm_mmap_status *status;
        volatile struct snd_pcm_mmap_control *control;
        atomic_t mmap_count;
```

```
        /* -- locking / scheduling -- */
        spinlock_t lock;
        wait_queue_head_t sleep;
        struct timer_list tick_timer;
        struct fasync_struct *fasync;

        /* -- private section -- */
        void *private_data;
        void (*private_free)(struct snd_pcm_runtime *runtime);

        /* -- hardware description -- */
        struct snd_pcm_hardware hw;
        struct snd_pcm_hw_constraints hw_constraints;

        /* -- timer -- */
        unsigned int timer_resolution;        /* timer resolution */

        /* -- DMA -- */
        unsigned char *dma_area;      /* DMA area */
        dma_addr_t dma_addr;          /* physical bus address (not accessible from main CPU) */
        size_t dma_bytes;             /* size of DMA area */

        struct snd_dma_buffer *dma_buffer_p;  /* allocated buffer */

#if defined(CONFIG_SND_PCM_OSS) || defined(CONFIG_SND_PCM_OSS_MODULE)
        /* -- OSS things -- */
        struct snd_pcm_oss_runtime oss;
#endif
};
```

For the operators (callbacks) of each sound driver, most of these records are supposed to be read-only. Only the PCM middle-layer changes / updates them. The exceptions are the hardware description (hw) DMA buffer information and the private data. Besides, if you use the standard buffer allocation method via *snd_pcm_lib_malloc_pages()*, you don't need to set the DMA buffer information by yourself.

In the sections below, important records are explained.

Hardware Description

The hardware descriptor (`struct snd_pcm_hardware`) contains the definitions of the fundamental hardware configuration. Above all, you'll need to define this in the *PCM open callback*. Note that the runtime instance holds the copy of the descriptor, not the pointer to the existing descriptor. That is, in the open callback, you can modify the copied descriptor (`runtime->hw`) as you need. For example, if the maximum number of channels is 1 only on some chip models, you can still use the same hardware descriptor and change the channels_max later:

```
struct snd_pcm_runtime *runtime = substream->runtime;
...
runtime->hw = snd_mychip_playback_hw; /* common definition */
if (chip->model == VERY_OLD_ONE)
        runtime->hw.channels_max = 1;
```

Typically, you'll have a hardware descriptor as below:

```
static struct snd_pcm_hardware snd_mychip_playback_hw = {
        .info = (SNDRV_PCM_INFO_MMAP |
                 SNDRV_PCM_INFO_INTERLEAVED |
                 SNDRV_PCM_INFO_BLOCK_TRANSFER |
                 SNDRV_PCM_INFO_MMAP_VALID),
        .formats =          SNDRV_PCM_FMTBIT_S16_LE,
        .rates =            SNDRV_PCM_RATE_8000_48000,
```

```
        .rate_min =         8000,
        .rate_max =         48000,
        .channels_min =     2,
        .channels_max =     2,
        .buffer_bytes_max = 32768,
        .period_bytes_min = 4096,
        .period_bytes_max = 32768,
        .periods_min =      1,
        .periods_max =      1024,
};
```

- The `info` field contains the type and capabilities of this pcm. The bit flags are defined in <sound/asound.h> as `SNDRV_PCM_INFO_XXX`. Here, at least, you have to specify whether the mmap is supported and which interleaved format is supported. When the hardware supports mmap, add the `SNDRV_PCM_INFO_MMAP` flag here. When the hardware supports the interleaved or the non-interleaved formats, `SNDRV_PCM_INFO_INTERLEAVED` or `SNDRV_PCM_INFO_NONINTERLEAVED` flag must be set, respectively. If both are supported, you can set both, too.

  In the above example, `MMAP_VALID` and `BLOCK_TRANSFER` are specified for the OSS mmap mode. Usually both are set. Of course, `MMAP_VALID` is set only if the mmap is really supported.

  The other possible flags are `SNDRV_PCM_INFO_PAUSE` and `SNDRV_PCM_INFO_RESUME`. The `PAUSE` bit means that the pcm supports the "pause" operation, while the `RESUME` bit means that the pcm supports the full "suspend/resume" operation. If the `PAUSE` flag is set, the `trigger` callback below must handle the corresponding (pause push/release) commands. The suspend/resume trigger commands can be defined even without the `RESUME` flag. See *Power Management* section for details.

  When the PCM substreams can be synchronized (typically, synchronized start/stop of a playback and a capture streams), you can give `SNDRV_PCM_INFO_SYNC_START`, too. In this case, you'll need to check the linked-list of PCM substreams in the trigger callback. This will be described in the later section.

- `formats` field contains the bit-flags of supported formats (`SNDRV_PCM_FMTBIT_XXX`). If the hardware supports more than one format, give all or'ed bits. In the example above, the signed 16bit little-endian format is specified.

- `rates` field contains the bit-flags of supported rates (`SNDRV_PCM_RATE_XXX`). When the chip supports continuous rates, pass `CONTINUOUS` bit additionally. The pre-defined rate bits are provided only for typical rates. If your chip supports unconventional rates, you need to add the `KNOT` bit and set up the hardware constraint manually (explained later).

- `rate_min` and `rate_max` define the minimum and maximum sample rate. This should correspond somehow to `rates` bits.

- `channel_min` and `channel_max` define, as you might already expected, the minimum and maximum number of channels.

- `buffer_bytes_max` defines the maximum buffer size in bytes. There is no `buffer_bytes_min` field, since it can be calculated from the minimum period size and the minimum number of periods. Meanwhile, `period_bytes_min` and define the minimum and maximum size of the period in bytes. `periods_max` and `periods_min` define the maximum and minimum number of periods in the buffer.

  The "period" is a term that corresponds to a fragment in the OSS world. The period defines the size at which a PCM interrupt is generated. This size strongly depends on the hardware. Generally, the smaller period size will give you more interrupts, that is, more controls. In the case of capture, this size defines the input latency. On the other hand, the whole buffer size defines the output latency for the playback direction.

- There is also a field `fifo_size`. This specifies the size of the hardware FIFO, but currently it is neither used in the driver nor in the alsa-lib. So, you can ignore this field.

PCM Configurations

Ok, let's go back again to the PCM runtime records. The most frequently referred records in the runtime instance are the PCM configurations. The PCM configurations are stored in the runtime instance after the application sends hw_params data via alsa-lib. There are many fields copied from hw_params and sw_params structs. For example, `format` holds the format type chosen by the application. This field contains the enum value `SNDRV_PCM_FORMAT_XXX`.

One thing to be noted is that the configured buffer and period sizes are stored in "frames" in the runtime. In the ALSA world, `1 frame = channels \* samples-size`. For conversion between frames and bytes, you can use the *frames_to_bytes()* and *bytes_to_frames()* helper functions.

```
period_bytes = frames_to_bytes(runtime, runtime->period_size);
```

Also, many software parameters (sw_params) are stored in frames, too. Please check the type of the field. `snd_pcm_uframes_t` is for the frames as unsigned integer while `snd_pcm_sframes_t` is for the frames as signed integer.

DMA Buffer Information

The DMA buffer is defined by the following four fields, `dma_area`, `dma_addr`, `dma_bytes` and `dma_private`. The `dma_area` holds the buffer pointer (the logical address). You can call `memcpy()` from/to this pointer. Meanwhile, `dma_addr` holds the physical address of the buffer. This field is specified only when the buffer is a linear buffer. `dma_bytes` holds the size of buffer in bytes. `dma_private` is used for the ALSA DMA allocator.

If you use a standard ALSA function, *snd_pcm_lib_malloc_pages()*, for allocating the buffer, these fields are set by the ALSA middle layer, and you should *not* change them by yourself. You can read them but not write them. On the other hand, if you want to allocate the buffer by yourself, you'll need to manage it in hw_params callback. At least, `dma_bytes` is mandatory. `dma_area` is necessary when the buffer is mmapped. If your driver doesn't support mmap, this field is not necessary. `dma_addr` is also optional. You can use dma_private as you like, too.

Running Status

The running status can be referred via `runtime->status`. This is the pointer to the `struct snd_pcm_mmap_status` record. For example, you can get the current DMA hardware pointer via `runtime->status->hw_ptr`.

The DMA application pointer can be referred via `runtime->control`, which points to the `struct snd_pcm_mmap_control` record. However, accessing directly to this value is not recommended.

Private Data

You can allocate a record for the substream and store it in `runtime->private_data`. Usually, this is done in the *PCM open callback*. Don't mix this with `pcm->private_data`. The `pcm->private_data` usually points to the chip instance assigned statically at the creation of PCM, while the `runtime->private_data` points to a dynamic data structure created at the PCM open callback.

```
static int snd_xxx_open(struct snd_pcm_substream *substream)
{
        struct my_pcm_data *data;
        ....
        data = kmalloc(sizeof(*data), GFP_KERNEL);
        substream->runtime->private_data = data;
        ....
}
```

The allocated object must be released in the *close callback*.

Operators

OK, now let me give details about each pcm callback (ops). In general, every callback must return 0 if successful, or a negative error number such as -EINVAL. To choose an appropriate error number, it is advised to check what value other parts of the kernel return when the same kind of request fails.

The callback function takes at least the argument with struct snd_pcm_substream pointer. To retrieve the chip record from the given substream instance, you can use the following macro.

```
int xxx() {
        struct mychip *chip = snd_pcm_substream_chip(substream);
        ....
}
```

The macro reads substream->private_data, which is a copy of pcm->private_data. You can override the former if you need to assign different data records per PCM substream. For example, the cmi8330 driver assigns different private_data for playback and capture directions, because it uses two different codecs (SB- and AD-compatible) for different directions.

PCM open callback

```
static int snd_xxx_open(struct snd_pcm_substream *substream);
```

This is called when a pcm substream is opened.

At least, here you have to initialize the runtime->hw record. Typically, this is done by like this:

```
static int snd_xxx_open(struct snd_pcm_substream *substream)
{
        struct mychip *chip = snd_pcm_substream_chip(substream);
        struct snd_pcm_runtime *runtime = substream->runtime;

        runtime->hw = snd_mychip_playback_hw;
        return 0;
}
```

where snd_mychip_playback_hw is the pre-defined hardware description.

You can allocate a private data in this callback, as described in *Private Data* section.

If the hardware configuration needs more constraints, set the hardware constraints here, too. See *Constraints* for more details.

close callback

```
static int snd_xxx_close(struct snd_pcm_substream *substream);
```

Obviously, this is called when a pcm substream is closed.

Any private instance for a pcm substream allocated in the open callback will be released here.

```
static int snd_xxx_close(struct snd_pcm_substream *substream)
{
        ....
        kfree(substream->runtime->private_data);
        ....
}
```

ioctl callback

This is used for any special call to pcm ioctls. But usually you can pass a generic ioctl callback, *snd_pcm_lib_ioctl()*.

hw_params callback

```
static int snd_xxx_hw_params(struct snd_pcm_substream *substream,
                             struct snd_pcm_hw_params *hw_params);
```

This is called when the hardware parameter (hw_params) is set up by the application, that is, once when the buffer size, the period size, the format, etc. are defined for the pcm substream.

Many hardware setups should be done in this callback, including the allocation of buffers.

Parameters to be initialized are retrieved by params_xxx() macros. To allocate buffer, you can call a helper function,

```
snd_pcm_lib_malloc_pages(substream, params_buffer_bytes(hw_params));
```

snd_pcm_lib_malloc_pages() is available only when the DMA buffers have been pre-allocated. See the section *Buffer Types* for more details.

Note that this and prepare callbacks may be called multiple times per initialization. For example, the OSS emulation may call these callbacks at each change via its ioctl.

Thus, you need to be careful not to allocate the same buffers many times, which will lead to memory leaks! Calling the helper function above many times is OK. It will release the previous buffer automatically when it was already allocated.

Another note is that this callback is non-atomic (schedulable) as default, i.e. when no nonatomic flag set. This is important, because the trigger callback is atomic (non-schedulable). That is, mutexes or any schedule-related functions are not available in trigger callback. Please see the subsection *Atomicity* for details.

hw_free callback

```
static int snd_xxx_hw_free(struct snd_pcm_substream *substream);
```

This is called to release the resources allocated via hw_params. For example, releasing the buffer via *snd_pcm_lib_malloc_pages()* is done by calling the following:

```
snd_pcm_lib_free_pages(substream);
```

This function is always called before the close callback is called. Also, the callback may be called multiple times, too. Keep track whether the resource was already released.

prepare callback

```
static int snd_xxx_prepare(struct snd_pcm_substream *substream);
```

This callback is called when the pcm is “prepared”. You can set the format type, sample rate, etc. here. The difference from hw_params is that the prepare callback will be called each time *snd_pcm_prepare()* is called, i.e. when recovering after underruns, etc.

Note that this callback is now non-atomic. You can use schedule-related functions safely in this callback.

In this and the following callbacks, you can refer to the values via the runtime record, substream->runtime. For example, to get the current rate, format or channels, access to runtime->rate,

runtime->format or runtime->channels, respectively. The physical address of the allocated buffer is set to runtime->dma_area. The buffer and period sizes are in runtime->buffer_size and runtime->period_size, respectively.

Be careful that this callback will be called many times at each setup, too.

trigger callback

```
static int snd_xxx_trigger(struct snd_pcm_substream *substream, int cmd);
```

This is called when the pcm is started, stopped or paused.

Which action is specified in the second argument, SNDRV_PCM_TRIGGER_XXX in <sound/pcm.h>. At least, the START and STOP commands must be defined in this callback.

```
switch (cmd) {
case SNDRV_PCM_TRIGGER_START:
        /* do something to start the PCM engine */
        break;
case SNDRV_PCM_TRIGGER_STOP:
        /* do something to stop the PCM engine */
        break;
default:
        return -EINVAL;
}
```

When the pcm supports the pause operation (given in the info field of the hardware table), the PAUSE_PUSH and PAUSE_RELEASE commands must be handled here, too. The former is the command to pause the pcm, and the latter to restart the pcm again.

When the pcm supports the suspend/resume operation, regardless of full or partial suspend/resume support, the SUSPEND and RESUME commands must be handled, too. These commands are issued when the power-management status is changed. Obviously, the SUSPEND and RESUME commands suspend and resume the pcm substream, and usually, they are identical to the STOP and START commands, respectively. See the *Power Management* section for details.

As mentioned, this callback is atomic as default unless nonatomic flag set, and you cannot call functions which may sleep. The trigger callback should be as minimal as possible, just really triggering the DMA. The other stuff should be initialized hw_params and prepare callbacks properly beforehand.

pointer callback

```
static snd_pcm_uframes_t snd_xxx_pointer(struct snd_pcm_substream *substream)
```

This callback is called when the PCM middle layer inquires the current hardware position on the buffer. The position must be returned in frames, ranging from 0 to buffer_size - 1.

This is called usually from the buffer-update routine in the pcm middle layer, which is invoked when *snd_pcm_period_elapsed()* is called in the interrupt routine. Then the pcm middle layer updates the position and calculates the available space, and wakes up the sleeping poll threads, etc.

This callback is also atomic as default.

copy_user, copy_kernel and fill_silence ops

These callbacks are not mandatory, and can be omitted in most cases. These callbacks are used when the hardware buffer cannot be in the normal memory space. Some chips have their own buffer on the hardware which is not mappable. In such a case, you have to transfer the data manually from the memory buffer to the hardware buffer. Or, if the buffer is non-contiguous on both physical and virtual memory spaces, these callbacks must be defined, too.

If these two callbacks are defined, copy and set-silence operations are done by them. The detailed will be described in the later section *Buffer and Memory Management*.

ack callback

This callback is also not mandatory. This callback is called when the `appl_ptr` is updated in read or write operations. Some drivers like emu10k1-fx and cs46xx need to track the current `appl_ptr` for the internal buffer, and this callback is useful only for such a purpose.

This callback is atomic as default.

page callback

This callback is optional too. This callback is used mainly for non-contiguous buffers. The mmap calls this callback to get the page address. Some examples will be explained in the later section *Buffer and Memory Management*, too.

PCM Interrupt Handler

The rest of pcm stuff is the PCM interrupt handler. The role of PCM interrupt handler in the sound driver is to update the buffer position and to tell the PCM middle layer when the buffer position goes across the prescribed period size. To inform this, call the *snd_pcm_period_elapsed()* function.

There are several types of sound chips to generate the interrupts.

Interrupts at the period (fragment) boundary

This is the most frequently found type: the hardware generates an interrupt at each period boundary. In this case, you can call *snd_pcm_period_elapsed()* at each interrupt.

snd_pcm_period_elapsed() takes the substream pointer as its argument. Thus, you need to keep the substream pointer accessible from the chip instance. For example, define `substream` field in the chip record to hold the current running substream pointer, and set the pointer value at open callback (and reset at `close` callback).

If you acquire a spinlock in the interrupt handler, and the lock is used in other pcm callbacks, too, then you have to release the lock before calling *snd_pcm_period_elapsed()*, because *snd_pcm_period_elapsed()* calls other pcm callbacks inside.

Typical code would be like:

```
static irqreturn_t snd_mychip_interrupt(int irq, void *dev_id)
{
        struct mychip *chip = dev_id;
        spin_lock(&chip->lock);
        ....
        if (pcm_irq_invoked(chip)) {
                /* call updater, unlock before it */
                spin_unlock(&chip->lock);
                snd_pcm_period_elapsed(chip->substream);
                spin_lock(&chip->lock);
                /* acknowledge the interrupt if necessary */
        }
        ....
        spin_unlock(&chip->lock);
        return IRQ_HANDLED;
}
```

High frequency timer interrupts

This happens when the hardware doesn't generate interrupts at the period boundary but issues timer interrupts at a fixed timer rate (e.g. es1968 or ymfpci drivers). In this case, you need to check the current hardware position and accumulate the processed sample length at each interrupt. When the accumulated size exceeds the period size, call *snd_pcm_period_elapsed()* and reset the accumulator.

Typical code would be like the following.

```
static irqreturn_t snd_mychip_interrupt(int irq, void *dev_id)
{
        struct mychip *chip = dev_id;
        spin_lock(&chip->lock);
        ....
        if (pcm_irq_invoked(chip)) {
                unsigned int last_ptr, size;
                /* get the current hardware pointer (in frames) */
                last_ptr = get_hw_ptr(chip);
                /* calculate the processed frames since the
                 * last update
                 */
                if (last_ptr < chip->last_ptr)
                        size = runtime->buffer_size + last_ptr
                                 - chip->last_ptr;
                else
                        size = last_ptr - chip->last_ptr;
                /* remember the last updated point */
                chip->last_ptr = last_ptr;
                /* accumulate the size */
                chip->size += size;
                /* over the period boundary? */
                if (chip->size >= runtime->period_size) {
                        /* reset the accumulator */
                        chip->size %= runtime->period_size;
                        /* call updater */
                        spin_unlock(&chip->lock);
                        snd_pcm_period_elapsed(substream);
                        spin_lock(&chip->lock);
                }
                /* acknowledge the interrupt if necessary */
        }
        ....
        spin_unlock(&chip->lock);
        return IRQ_HANDLED;
}
```

On calling `snd_pcm_period_elapsed()`

In both cases, even if more than one period are elapsed, you don't have to call *snd_pcm_period_elapsed()* many times. Call only once. And the pcm layer will check the current hardware pointer and update to the latest status.

Atomicity

One of the most important (and thus difficult to debug) problems in kernel programming are race conditions. In the Linux kernel, they are usually avoided via spin-locks, mutexes or semaphores. In general, if a race condition can happen in an interrupt handler, it has to be managed atomically, and you have to use a spinlock to protect the critical session. If the critical section is not in interrupt handler code and if taking a relatively long time to execute is acceptable, you should use mutexes or semaphores instead.

As already seen, some pcm callbacks are atomic and some are not. For example, the hw_params callback is non-atomic, while trigger callback is atomic. This means, the latter is called already in a spinlock held by the PCM middle layer. Please take this atomicity into account when you choose a locking scheme in the callbacks.

In the atomic callbacks, you cannot use functions which may call schedule() or go to sleep(). Semaphores and mutexes can sleep, and hence they cannot be used inside the atomic callbacks (e.g. trigger callback). To implement some delay in such a callback, please use udelay() or mdelay().

All three atomic callbacks (trigger, pointer, and ack) are called with local interrupts disabled.

The recent changes in PCM core code, however, allow all PCM operations to be non-atomic. This assumes that the all caller sides are in non-atomic contexts. For example, the function *snd_pcm_period_elapsed()* is called typically from the interrupt handler. But, if you set up the driver to use a threaded interrupt handler, this call can be in non-atomic context, too. In such a case, you can set nonatomic filed of struct snd_pcm object after creating it. When this flag is set, mutex and rwsem are used internally in the PCM core instead of spin and rwlocks, so that you can call all PCM functions safely in a non-atomic context.

Constraints

If your chip supports unconventional sample rates, or only the limited samples, you need to set a constraint for the condition.

For example, in order to restrict the sample rates in the some supported values, use *snd_pcm_hw_constraint_list()*. You need to call this function in the open callback.

```
static unsigned int rates[] =
        {4000, 10000, 22050, 44100};
static struct snd_pcm_hw_constraint_list constraints_rates = {
        .count = ARRAY_SIZE(rates),
        .list = rates,
        .mask = 0,
};

static int snd_mychip_pcm_open(struct snd_pcm_substream *substream)
{
        int err;
        ....
        err = snd_pcm_hw_constraint_list(substream->runtime, 0,
                                         SNDRV_PCM_HW_PARAM_RATE,
                                         &constraints_rates);
        if (err < 0)
                return err;
        ....
}
```

There are many different constraints. Look at sound/pcm.h for a complete list. You can even define your own constraint rules. For example, let's suppose my_chip can manage a substream of 1 channel if and only if the format is S16_LE, otherwise it supports any format specified in the struct snd_pcm_hardware structure (or in any other constraint_list). You can build a rule like this:

```
static int hw_rule_channels_by_format(struct snd_pcm_hw_params *params,
                                      struct snd_pcm_hw_rule *rule)
{
        struct snd_interval *c = hw_param_interval(params,
                        SNDRV_PCM_HW_PARAM_CHANNELS);
        struct snd_mask *f = hw_param_mask(params, SNDRV_PCM_HW_PARAM_FORMAT);
        struct snd_interval ch;

        snd_interval_any(&ch);
        if (f->bits[0] == SNDRV_PCM_FMTBIT_S16_LE) {
                ch.min = ch.max = 1;
```

```
                ch.integer = 1;
                return snd_interval_refine(c, &ch);
        }
        return 0;
}
```

Then you need to call this function to add your rule:

```
snd_pcm_hw_rule_add(substream->runtime, 0, SNDRV_PCM_HW_PARAM_CHANNELS,
                    hw_rule_channels_by_format, NULL,
                    SNDRV_PCM_HW_PARAM_FORMAT, -1);
```

The rule function is called when an application sets the PCM format, and it refines the number of channels accordingly. But an application may set the number of channels before setting the format. Thus you also need to define the inverse rule:

```
static int hw_rule_format_by_channels(struct snd_pcm_hw_params *params,
                                      struct snd_pcm_hw_rule *rule)
{
        struct snd_interval *c = hw_param_interval(params,
              SNDRV_PCM_HW_PARAM_CHANNELS);
        struct snd_mask *f = hw_param_mask(params, SNDRV_PCM_HW_PARAM_FORMAT);
        struct snd_mask fmt;

        snd_mask_any(&fmt);    /* Init the struct */
        if (c->min < 2) {
                fmt.bits[0] &= SNDRV_PCM_FMTBIT_S16_LE;
                return snd_mask_refine(f, &fmt);
        }
        return 0;
}
```

... and in the open callback:

```
snd_pcm_hw_rule_add(substream->runtime, 0, SNDRV_PCM_HW_PARAM_FORMAT,
                    hw_rule_format_by_channels, NULL,
                    SNDRV_PCM_HW_PARAM_CHANNELS, -1);
```

I won't give more details here, rather I would like to say, "Luke, use the source."

Control Interface

General

The control interface is used widely for many switches, sliders, etc. which are accessed from user-space. Its most important use is the mixer interface. In other words, since ALSA 0.9.x, all the mixer stuff is implemented on the control kernel API.

ALSA has a well-defined AC97 control module. If your chip supports only the AC97 and nothing else, you can skip this section.

The control API is defined in `<sound/control.h>`. Include this file if you want to add your own controls.

Definition of Controls

To create a new control, you need to define the following three callbacks: `info`, `get` and `put`. Then, define a `struct snd_kcontrol_new` record, such as:

```
static struct snd_kcontrol_new my_control = {
        .iface = SNDRV_CTL_ELEM_IFACE_MIXER,
        .name = "PCM Playback Switch",
```

```
        .index = 0,
        .access = SNDRV_CTL_ELEM_ACCESS_READWRITE,
        .private_value = 0xffff,
        .info = my_control_info,
        .get = my_control_get,
        .put = my_control_put
};
```

The `iface` field specifies the control type, `SNDRV_CTL_ELEM_IFACE_XXX`, which is usually `MIXER`. Use `CARD` for global controls that are not logically part of the mixer. If the control is closely associated with some specific device on the sound card, use `HWDEP`, `PCM`, `RAWMIDI`, `TIMER`, or `SEQUENCER`, and specify the device number with the `device` and `subdevice` fields.

The `name` is the name identifier string. Since ALSA 0.9.x, the control name is very important, because its role is classified from its name. There are pre-defined standard control names. The details are described in the *Control Names* subsection.

The `index` field holds the index number of this control. If there are several different controls with the same name, they can be distinguished by the index number. This is the case when several codecs exist on the card. If the index is zero, you can omit the definition above.

The `access` field contains the access type of this control. Give the combination of bit masks, `SNDRV_CTL_ELEM_ACCESS_XXX`, there. The details will be explained in the *Access Flags* subsection.

The `private_value` field contains an arbitrary long integer value for this record. When using the generic `info`, `get` and `put` callbacks, you can pass a value through this field. If several small numbers are necessary, you can combine them in bitwise. Or, it's possible to give a pointer (casted to unsigned long) of some record to this field, too.

The `tlv` field can be used to provide metadata about the control; see the *Metadata* subsection.

The other three are *Control Callbacks*.

Control Names

There are some standards to define the control names. A control is usually defined from the three parts as "SOURCE DIRECTION FUNCTION".

The first, `SOURCE`, specifies the source of the control, and is a string such as "Master", "PCM", "CD" and "Line". There are many pre-defined sources.

The second, `DIRECTION`, is one of the following strings according to the direction of the control: "Playback", "Capture", "Bypass Playback" and "Bypass Capture". Or, it can be omitted, meaning both playback and capture directions.

The third, `FUNCTION`, is one of the following strings according to the function of the control: "Switch", "Volume" and "Route".

The example of control names are, thus, "Master Capture Switch" or "PCM Playback Volume".

There are some exceptions:

Global capture and playback

"Capture Source", "Capture Switch" and "Capture Volume" are used for the global capture (input) source, switch and volume. Similarly, "Playback Switch" and "Playback Volume" are used for the global output gain switch and volume.

Tone-controls

tone-control switch and volumes are specified like “Tone Control - XXX”, e.g. “Tone Control - Switch”, “Tone Control - Bass”, “Tone Control - Center”.

3D controls

3D-control switches and volumes are specified like “3D Control - XXX”, e.g. “3D Control - Switch”, “3D Control - Center”, “3D Control - Space”.

Mic boost

Mic-boost switch is set as “Mic Boost” or “Mic Boost (6dB)”.

More precise information can be found in `Documentation/sound/designs/control-names.rst`.

Access Flags

The access flag is the bitmask which specifies the access type of the given control. The default access type is `SNDRV_CTL_ELEM_ACCESS_READWRITE`, which means both read and write are allowed to this control. When the access flag is omitted (i.e. = 0), it is considered as `READWRITE` access as default.

When the control is read-only, pass `SNDRV_CTL_ELEM_ACCESS_READ` instead. In this case, you don’t have to define the `put` callback. Similarly, when the control is write-only (although it’s a rare case), you can use the `WRITE` flag instead, and you don’t need the `get` callback.

If the control value changes frequently (e.g. the VU meter), `VOLATILE` flag should be given. This means that the control may be changed without *Change notification*. Applications should poll such a control constantly.

When the control is inactive, set the `INACTIVE` flag, too. There are `LOCK` and `OWNER` flags to change the write permissions.

Control Callbacks

info callback

The `info` callback is used to get detailed information on this control. This must store the values of the given `struct snd_ctl_elem_info` object. For example, for a boolean control with a single element:

```
static int snd_myctl_mono_info(struct snd_kcontrol *kcontrol,
                               struct snd_ctl_elem_info *uinfo)
{
        uinfo->type = SNDRV_CTL_ELEM_TYPE_BOOLEAN;
        uinfo->count = 1;
        uinfo->value.integer.min = 0;
        uinfo->value.integer.max = 1;
        return 0;
}
```

The `type` field specifies the type of the control. There are `BOOLEAN`, `INTEGER`, `ENUMERATED`, `BYTES`, `IEC958` and `INTEGER64`. The `count` field specifies the number of elements in this control. For example, a stereo volume would have count = 2. The `value` field is a union, and the values stored are depending on the type. The boolean and integer types are identical.

The enumerated type is a bit different from others. You’ll need to set the string for the currently given item index.

```
static int snd_myctl_enum_info(struct snd_kcontrol *kcontrol,
                          struct snd_ctl_elem_info *uinfo)
{
        static char *texts[4] = {
                "First", "Second", "Third", "Fourth"
        };
        uinfo->type = SNDRV_CTL_ELEM_TYPE_ENUMERATED;
        uinfo->count = 1;
        uinfo->value.enumerated.items = 4;
        if (uinfo->value.enumerated.item > 3)
                uinfo->value.enumerated.item = 3;
        strcpy(uinfo->value.enumerated.name,
               texts[uinfo->value.enumerated.item]);
        return 0;
}
```

The above callback can be simplified with a helper function, *snd_ctl_enum_info()*. The final code looks like below. (You can pass ARRAY_SIZE(texts) instead of 4 in the third argument; it's a matter of taste.)

```
static int snd_myctl_enum_info(struct snd_kcontrol *kcontrol,
                          struct snd_ctl_elem_info *uinfo)
{
        static char *texts[4] = {
                "First", "Second", "Third", "Fourth"
        };
        return snd_ctl_enum_info(uinfo, 1, 4, texts);
}
```

Some common info callbacks are available for your convenience: *snd_ctl_boolean_mono_info()* and *snd_ctl_boolean_stereo_info()*. Obviously, the former is an info callback for a mono channel boolean item, just like snd_myctl_mono_info() above, and the latter is for a stereo channel boolean item.

get callback

This callback is used to read the current value of the control and to return to user-space.

For example,

```
static int snd_myctl_get(struct snd_kcontrol *kcontrol,
                         struct snd_ctl_elem_value *ucontrol)
{
        struct mychip *chip = snd_kcontrol_chip(kcontrol);
        ucontrol->value.integer.value[0] = get_some_value(chip);
        return 0;
}
```

The value field depends on the type of control as well as on the info callback. For example, the sb driver uses this field to store the register offset, the bit-shift and the bit-mask. The private_value field is set as follows:

```
.private_value = reg | (shift << 16) | (mask << 24)
```

and is retrieved in callbacks like

```
static int snd_sbmixer_get_single(struct snd_kcontrol *kcontrol,
                                  struct snd_ctl_elem_value *ucontrol)
{
        int reg = kcontrol->private_value & 0xff;
        int shift = (kcontrol->private_value >> 16) & 0xff;
        int mask = (kcontrol->private_value >> 24) & 0xff;
        ....
}
```

In the get callback, you have to fill all the elements if the control has more than one elements, i.e. `count > 1`. In the example above, we filled only one element (`value.integer.value[0]`) since it's assumed as `count = 1`.

put callback

This callback is used to write a value from user-space.

For example,

```
static int snd_myctl_put(struct snd_kcontrol *kcontrol,
                         struct snd_ctl_elem_value *ucontrol)
{
        struct mychip *chip = snd_kcontrol_chip(kcontrol);
        int changed = 0;
        if (chip->current_value !=
             ucontrol->value.integer.value[0]) {
                change_current_value(chip,
                            ucontrol->value.integer.value[0]);
                changed = 1;
        }
        return changed;
}
```

As seen above, you have to return 1 if the value is changed. If the value is not changed, return 0 instead. If any fatal error happens, return a negative error code as usual.

As in the get callback, when the control has more than one elements, all elements must be evaluated in this callback, too.

Callbacks are not atomic

All these three callbacks are basically not atomic.

Control Constructor

When everything is ready, finally we can create a new control. To create a control, there are two functions to be called, *snd_ctl_new1()* and *snd_ctl_add()*.

In the simplest way, you can do like this:

```
err = snd_ctl_add(card, snd_ctl_new1(&my_control, chip));
if (err < 0)
        return err;
```

where `my_control` is the `struct snd_kcontrol_new` object defined above, and chip is the object pointer to be passed to kcontrol->private_data which can be referred to in callbacks.

snd_ctl_new1() allocates a new `struct snd_kcontrol` instance, and *snd_ctl_add()* assigns the given control component to the card.

Change Notification

If you need to change and update a control in the interrupt routine, you can call *snd_ctl_notify()*. For example,

```
snd_ctl_notify(card, SNDRV_CTL_EVENT_MASK_VALUE, id_pointer);
```

This function takes the card pointer, the event-mask, and the control id pointer for the notification. The event-mask specifies the types of notification, for example, in the above example, the change of control values is notified. The id pointer is the pointer of `struct snd_ctl_elem_id` to be notified. You can find some examples in `es1938.c` or `es1968.c` for hardware volume interrupts.

Metadata

To provide information about the dB values of a mixer control, use on of the `DECLARE_TLV_xxx` macros from <sound/tlv.h> to define a variable containing this information, set the `tlv.p` field to point to this variable, and include the `SNDRV_CTL_ELEM_ACCESS_TLV_READ` flag in the `access` field; like this:

```
static DECLARE_TLV_DB_SCALE(db_scale_my_control, -4050, 150, 0);

static struct snd_kcontrol_new my_control = {
        ...
        .access = SNDRV_CTL_ELEM_ACCESS_READWRITE |
                  SNDRV_CTL_ELEM_ACCESS_TLV_READ,
        ...
        .tlv.p = db_scale_my_control,
};
```

The `DECLARE_TLV_DB_SCALE()` macro defines information about a mixer control where each step in the control's value changes the dB value by a constant dB amount. The first parameter is the name of the variable to be defined. The second parameter is the minimum value, in units of 0.01 dB. The third parameter is the step size, in units of 0.01 dB. Set the fourth parameter to 1 if the minimum value actually mutes the control.

The `DECLARE_TLV_DB_LINEAR()` macro defines information about a mixer control where the control's value affects the output linearly. The first parameter is the name of the variable to be defined. The second parameter is the minimum value, in units of 0.01 dB. The third parameter is the maximum value, in units of 0.01 dB. If the minimum value mutes the control, set the second parameter to `TLV_DB_GAIN_MUTE`.

API for AC97 Codec

General

The ALSA AC97 codec layer is a well-defined one, and you don't have to write much code to control it. Only low-level control routines are necessary. The AC97 codec API is defined in <sound/ac97_codec.h>.

Full Code Example

```
struct mychip {
        ....
        struct snd_ac97 *ac97;
        ....
};

static unsigned short snd_mychip_ac97_read(struct snd_ac97 *ac97,
                                           unsigned short reg)
{
        struct mychip *chip = ac97->private_data;
        ....
        /* read a register value here from the codec */
        return the_register_value;
}

static void snd_mychip_ac97_write(struct snd_ac97 *ac97,
                                  unsigned short reg, unsigned short val)
```

```
{
        struct mychip *chip = ac97->private_data;
        ....
        /* write the given register value to the codec */
}

static int snd_mychip_ac97(struct mychip *chip)
{
        struct snd_ac97_bus *bus;
        struct snd_ac97_template ac97;
        int err;
        static struct snd_ac97_bus_ops ops = {
                .write = snd_mychip_ac97_write,
                .read = snd_mychip_ac97_read,
        };

        err = snd_ac97_bus(chip->card, 0, &ops, NULL, &bus);
        if (err < 0)
                return err;
        memset(&ac97, 0, sizeof(ac97));
        ac97.private_data = chip;
        return snd_ac97_mixer(bus, &ac97, &chip->ac97);
}
```

AC97 Constructor

To create an ac97 instance, first call *snd_ac97_bus()* with an ac97_bus_ops_t record with callback functions.

```
struct snd_ac97_bus *bus;
static struct snd_ac97_bus_ops ops = {
      .write = snd_mychip_ac97_write,
      .read = snd_mychip_ac97_read,
};

snd_ac97_bus(card, 0, &ops, NULL, &pbus);
```

The bus record is shared among all belonging ac97 instances.

And then call *snd_ac97_mixer()* with an `struct snd_ac97_template` record together with the bus pointer created above.

```
struct snd_ac97_template ac97;
int err;

memset(&ac97, 0, sizeof(ac97));
ac97.private_data = chip;
snd_ac97_mixer(bus, &ac97, &chip->ac97);
```

where chip->ac97 is a pointer to a newly created ac97_t instance. In this case, the chip pointer is set as the private data, so that the read/write callback functions can refer to this chip instance. This instance is not necessarily stored in the chip record. If you need to change the register values from the driver, or need the suspend/resume of ac97 codecs, keep this pointer to pass to the corresponding functions.

AC97 Callbacks

The standard callbacks are `read` and `write`. Obviously they correspond to the functions for read and write accesses to the hardware low-level codes.

The `read` callback returns the register value specified in the argument.

```
static unsigned short snd_mychip_ac97_read(struct snd_ac97 *ac97,
                                           unsigned short reg)
{
        struct mychip *chip = ac97->private_data;
        ....
        return the_register_value;
}
```

Here, the chip can be cast from `ac97->private_data`.

Meanwhile, the `write` callback is used to set the register value

```
static void snd_mychip_ac97_write(struct snd_ac97 *ac97,
                      unsigned short reg, unsigned short val)
```

These callbacks are non-atomic like the control API callbacks.

There are also other callbacks: `reset`, `wait` and `init`.

The `reset` callback is used to reset the codec. If the chip requires a special kind of reset, you can define this callback.

The `wait` callback is used to add some waiting time in the standard initialization of the codec. If the chip requires the extra waiting time, define this callback.

The `init` callback is used for additional initialization of the codec.

Updating Registers in The Driver

If you need to access to the codec from the driver, you can call the following functions: *snd_ac97_write()*, *snd_ac97_read()*, *snd_ac97_update()* and *snd_ac97_update_bits()*.

Both *snd_ac97_write()* and *snd_ac97_update()* functions are used to set a value to the given register (AC97_XXX). The difference between them is that *snd_ac97_update()* doesn't write a value if the given value has been already set, while *snd_ac97_write()* always rewrites the value.

```
snd_ac97_write(ac97, AC97_MASTER, 0x8080);
snd_ac97_update(ac97, AC97_MASTER, 0x8080);
```

snd_ac97_read() is used to read the value of the given register. For example,

```
value = snd_ac97_read(ac97, AC97_MASTER);
```

snd_ac97_update_bits() is used to update some bits in the given register.

```
snd_ac97_update_bits(ac97, reg, mask, value);
```

Also, there is a function to change the sample rate (of a given register such as AC97_PCM_FRONT_DAC_RATE) when VRA or DRA is supported by the codec: *snd_ac97_set_rate()*.

```
snd_ac97_set_rate(ac97, AC97_PCM_FRONT_DAC_RATE, 44100);
```

The following registers are available to set the rate: AC97_PCM_MIC_ADC_RATE, AC97_PCM_FRONT_DAC_RATE, AC97_PCM_LR_ADC_RATE, AC97_SPDIF. When AC97_SPDIF is specified, the register is not really changed but the corresponding IEC958 status bits will be updated.

Clock Adjustment

In some chips, the clock of the codec isn't 48000 but using a PCI clock (to save a quartz!). In this case, change the field `bus->clock` to the corresponding value. For example, intel8x0 and es1968 drivers have their own function to read from the clock.

Proc Files

The ALSA AC97 interface will create a proc file such as `/proc/asound/card0/codec97#0/ac97#0-0` and `ac97#0-0+regs`. You can refer to these files to see the current status and registers of the codec.

Multiple Codecs

When there are several codecs on the same card, you need to call *snd_ac97_mixer()* multiple times with `ac97.num=1` or greater. The `num` field specifies the codec number.

If you set up multiple codecs, you either need to write different callbacks for each codec or check `ac97->num` in the callback routines.

MIDI (MPU401-UART) Interface

General

Many soundcards have built-in MIDI (MPU401-UART) interfaces. When the soundcard supports the standard MPU401-UART interface, most likely you can use the ALSA MPU401-UART API. The MPU401-UART API is defined in `<sound/mpu401.h>`.

Some soundchips have a similar but slightly different implementation of mpu401 stuff. For example, emu10k1 has its own mpu401 routines.

MIDI Constructor

To create a rawmidi object, call *snd_mpu401_uart_new()*.

```
struct snd_rawmidi *rmidi;
snd_mpu401_uart_new(card, 0, MPU401_HW_MPU401, port, info_flags,
                    irq, &rmidi);
```

The first argument is the card pointer, and the second is the index of this component. You can create up to 8 rawmidi devices.

The third argument is the type of the hardware, `MPU401_HW_XXX`. If it's not a special one, you can use `MPU401_HW_MPU401`.

The 4th argument is the I/O port address. Many backward-compatible MPU401 have an I/O port such as 0x330. Or, it might be a part of its own PCI I/O region. It depends on the chip design.

The 5th argument is a bitflag for additional information. When the I/O port address above is part of the PCI I/O region, the MPU401 I/O port might have been already allocated (reserved) by the driver itself. In such a case, pass a bit flag `MPU401_INFO_INTEGRATED`, and the mpu401-uart layer will allocate the I/O ports by itself.

When the controller supports only the input or output MIDI stream, pass the `MPU401_INFO_INPUT` or `MPU401_INFO_OUTPUT` bitflag, respectively. Then the rawmidi instance is created as a single stream.

`MPU401_INFO_MMIO` bitflag is used to change the access method to MMIO (via readb and writeb) instead of iob and outb. In this case, you have to pass the iomapped address to *snd_mpu401_uart_new()*.

When `MPU401_INFO_TX_IRQ` is set, the output stream isn't checked in the default interrupt handler. The driver needs to call *snd_mpu401_uart_interrupt_tx()* by itself to start processing the output stream in the irq handler.

If the MPU-401 interface shares its interrupt with the other logical devices on the card, set `MPU401_INFO_IRQ_HOOK` (see *below*).

Usually, the port address corresponds to the command port and port + 1 corresponds to the data port. If not, you may change the `cport` field of `struct snd_mpu401` manually afterward. However, `struct`

snd_mpu401 pointer is not returned explicitly by *snd_mpu401_uart_new()*. You need to cast rmidi->private_data to struct snd_mpu401 explicitly,

```
struct snd_mpu401 *mpu;
mpu = rmidi->private_data;
```

and reset the cport as you like:

```
mpu->cport = my_own_control_port;
```

The 6th argument specifies the ISA irq number that will be allocated. If no interrupt is to be allocated (because your code is already allocating a shared interrupt, or because the device does not use interrupts), pass -1 instead. For a MPU-401 device without an interrupt, a polling timer will be used instead.

MIDI Interrupt Handler

When the interrupt is allocated in *snd_mpu401_uart_new()*, an exclusive ISA interrupt handler is automatically used, hence you don't have anything else to do than creating the mpu401 stuff. Otherwise, you have to set MPU401_INFO_IRQ_HOOK, and call *snd_mpu401_uart_interrupt()* explicitly from your own interrupt handler when it has determined that a UART interrupt has occurred.

In this case, you need to pass the private_data of the returned rawmidi object from *snd_mpu401_uart_new()* as the second argument of *snd_mpu401_uart_interrupt()*.

```
snd_mpu401_uart_interrupt(irq, rmidi->private_data, regs);
```

RawMIDI Interface

Overview

The raw MIDI interface is used for hardware MIDI ports that can be accessed as a byte stream. It is not used for synthesizer chips that do not directly understand MIDI.

ALSA handles file and buffer management. All you have to do is to write some code to move data between the buffer and the hardware.

The rawmidi API is defined in <sound/rawmidi.h>.

RawMIDI Constructor

To create a rawmidi device, call the *snd_rawmidi_new()* function:

```
struct snd_rawmidi *rmidi;
err = snd_rawmidi_new(chip->card, "MyMIDI", 0, outs, ins, &rmidi);
if (err < 0)
        return err;
rmidi->private_data = chip;
strcpy(rmidi->name, "My MIDI");
rmidi->info_flags = SNDRV_RAWMIDI_INFO_OUTPUT |
                    SNDRV_RAWMIDI_INFO_INPUT |
                    SNDRV_RAWMIDI_INFO_DUPLEX;
```

The first argument is the card pointer, the second argument is the ID string.

The third argument is the index of this component. You can create up to 8 rawmidi devices.

The fourth and fifth arguments are the number of output and input substreams, respectively, of this device (a substream is the equivalent of a MIDI port).

Set the `info_flags` field to specify the capabilities of the device. Set `SNDRV_RAWMIDI_INFO_OUTPUT` if there is at least one output port, `SNDRV_RAWMIDI_INFO_INPUT` if there is at least one input port, and `SNDRV_RAWMIDI_INFO_DUPLEX` if the device can handle output and input at the same time.

After the rawmidi device is created, you need to set the operators (callbacks) for each substream. There are helper functions to set the operators for all the substreams of a device:

```
snd_rawmidi_set_ops(rmidi, SNDRV_RAWMIDI_STREAM_OUTPUT, &snd_mymidi_output_ops);
snd_rawmidi_set_ops(rmidi, SNDRV_RAWMIDI_STREAM_INPUT, &snd_mymidi_input_ops);
```

The operators are usually defined like this:

```
static struct snd_rawmidi_ops snd_mymidi_output_ops = {
        .open =    snd_mymidi_output_open,
        .close =   snd_mymidi_output_close,
        .trigger = snd_mymidi_output_trigger,
};
```

These callbacks are explained in the *RawMIDI Callbacks* section.

If there are more than one substream, you should give a unique name to each of them:

```
struct snd_rawmidi_substream *substream;
list_for_each_entry(substream,
                    &rmidi->streams[SNDRV_RAWMIDI_STREAM_OUTPUT].substreams,
                    list {
        sprintf(substream->name, "My MIDI Port %d", substream->number + 1);
}
/* same for SNDRV_RAWMIDI_STREAM_INPUT */
```

RawMIDI Callbacks

In all the callbacks, the private data that you've set for the rawmidi device can be accessed as `substream->rmidi->private_data`.

If there is more than one port, your callbacks can determine the port index from the struct snd_rawmidi_substream data passed to each callback:

```
struct snd_rawmidi_substream *substream;
int index = substream->number;
```

RawMIDI open callback

```
static int snd_xxx_open(struct snd_rawmidi_substream *substream);
```

This is called when a substream is opened. You can initialize the hardware here, but you shouldn't start transmitting/receiving data yet.

RawMIDI close callback

```
static int snd_xxx_close(struct snd_rawmidi_substream *substream);
```

Guess what.

The open and `close` callbacks of a rawmidi device are serialized with a mutex, and can sleep.

Rawmidi trigger callback for output substreams

```
static void snd_xxx_output_trigger(struct snd_rawmidi_substream *substream, int up);
```

This is called with a nonzero up parameter when there is some data in the substream buffer that must be transmitted.

To read data from the buffer, call *snd_rawmidi_transmit_peek()*. It will return the number of bytes that have been read; this will be less than the number of bytes requested when there are no more data in the buffer. After the data have been transmitted successfully, call *snd_rawmidi_transmit_ack()* to remove the data from the substream buffer:

```
unsigned char data;
while (snd_rawmidi_transmit_peek(substream, &data, 1) == 1) {
        if (snd_mychip_try_to_transmit(data))
                snd_rawmidi_transmit_ack(substream, 1);
        else
                break; /* hardware FIFO full */
}
```

If you know beforehand that the hardware will accept data, you can use the *snd_rawmidi_transmit()* function which reads some data and removes them from the buffer at once:

```
while (snd_mychip_transmit_possible()) {
        unsigned char data;
        if (snd_rawmidi_transmit(substream, &data, 1) != 1)
                break; /* no more data */
        snd_mychip_transmit(data);
}
```

If you know beforehand how many bytes you can accept, you can use a buffer size greater than one with the *snd_rawmidi_transmit*()* functions.

The `trigger` callback must not sleep. If the hardware FIFO is full before the substream buffer has been emptied, you have to continue transmitting data later, either in an interrupt handler, or with a timer if the hardware doesn't have a MIDI transmit interrupt.

The `trigger` callback is called with a zero up parameter when the transmission of data should be aborted.

RawMIDI trigger callback for input substreams

```
static void snd_xxx_input_trigger(struct snd_rawmidi_substream *substream, int up);
```

This is called with a nonzero up parameter to enable receiving data, or with a zero up parameter do disable receiving data.

The `trigger` callback must not sleep; the actual reading of data from the device is usually done in an interrupt handler.

When data reception is enabled, your interrupt handler should call *snd_rawmidi_receive()* for all received data:

```
void snd_mychip_midi_interrupt(...)
{
        while (mychip_midi_available()) {
                unsigned char data;
                data = mychip_midi_read();
                snd_rawmidi_receive(substream, &data, 1);
        }
}
```

drain callback

```
static void snd_xxx_drain(struct snd_rawmidi_substream *substream);
```

This is only used with output substreams. This function should wait until all data read from the substream buffer have been transmitted. This ensures that the device can be closed and the driver unloaded without losing data.

This callback is optional. If you do not set `drain` in the struct snd_rawmidi_ops structure, ALSA will simply wait for 50 milliseconds instead.

Miscellaneous Devices

FM OPL3

The FM OPL3 is still used in many chips (mainly for backward compatibility). ALSA has a nice OPL3 FM control layer, too. The OPL3 API is defined in <sound/opl3.h>.

FM registers can be directly accessed through the direct-FM API, defined in <sound/asound_fm.h>. In ALSA native mode, FM registers are accessed through the Hardware-Dependent Device direct-FM extension API, whereas in OSS compatible mode, FM registers can be accessed with the OSS direct-FM compatible API in /dev/dmfmX device.

To create the OPL3 component, you have two functions to call. The first one is a constructor for the `opl3_t` instance.

```
struct snd_opl3 *opl3;
snd_opl3_create(card, lport, rport, OPL3_HW_OPL3_XXX,
                integrated, &opl3);
```

The first argument is the card pointer, the second one is the left port address, and the third is the right port address. In most cases, the right port is placed at the left port + 2.

The fourth argument is the hardware type.

When the left and right ports have been already allocated by the card driver, pass non-zero to the fifth argument (`integrated`). Otherwise, the opl3 module will allocate the specified ports by itself.

When the accessing the hardware requires special method instead of the standard I/O access, you can create opl3 instance separately with `snd_opl3_new()`.

```
struct snd_opl3 *opl3;
snd_opl3_new(card, OPL3_HW_OPL3_XXX, &opl3);
```

Then set `command`, `private_data` and `private_free` for the private access function, the private data and the destructor. The `l_port` and `r_port` are not necessarily set. Only the command must be set properly. You can retrieve the data from the `opl3->private_data` field.

After creating the opl3 instance via `snd_opl3_new()`, call `snd_opl3_init()` to initialize the chip to the proper state. Note that `snd_opl3_create()` always calls it internally.

If the opl3 instance is created successfully, then create a hwdep device for this opl3.

```
struct snd_hwdep *opl3hwdep;
snd_opl3_hwdep_new(opl3, 0, 1, &opl3hwdep);
```

The first argument is the `opl3_t` instance you created, and the second is the index number, usually 0.

The third argument is the index-offset for the sequencer client assigned to the OPL3 port. When there is an MPU401-UART, give 1 for here (UART always takes 0).

Hardware-Dependent Devices

Some chips need user-space access for special controls or for loading the micro code. In such a case, you can create a hwdep (hardware-dependent) device. The hwdep API is defined in <sound/hwdep.h>. You can find examples in opl3 driver or isa/sb/sb16_csp.c.

The creation of the hwdep instance is done via *snd_hwdep_new()*.

```
struct snd_hwdep *hw;
snd_hwdep_new(card, "My HWDEP", 0, &hw);
```

where the third argument is the index number.

You can then pass any pointer value to the private_data. If you assign a private data, you should define the destructor, too. The destructor function is set in the private_free field.

```
struct mydata *p = kmalloc(sizeof(*p), GFP_KERNEL);
hw->private_data = p;
hw->private_free = mydata_free;
```

and the implementation of the destructor would be:

```
static void mydata_free(struct snd_hwdep *hw)
{
        struct mydata *p = hw->private_data;
        kfree(p);
}
```

The arbitrary file operations can be defined for this instance. The file operators are defined in the ops table. For example, assume that this chip needs an ioctl.

```
hw->ops.open = mydata_open;
hw->ops.ioctl = mydata_ioctl;
hw->ops.release = mydata_release;
```

And implement the callback functions as you like.

IEC958 (S/PDIF)

Usually the controls for IEC958 devices are implemented via the control interface. There is a macro to compose a name string for IEC958 controls, SNDRV_CTL_NAME_IEC958() defined in <include/asound.h>.

There are some standard controls for IEC958 status bits. These controls use the type SNDRV_CTL_ELEM_TYPE_IEC958, and the size of element is fixed as 4 bytes array (value.iec958.status[x]). For the info callback, you don't specify the value field for this type (the count field must be set, though).

"IEC958 Playback Con Mask" is used to return the bit-mask for the IEC958 status bits of consumer mode. Similarly, "IEC958 Playback Pro Mask" returns the bitmask for professional mode. They are read-only controls, and are defined as MIXER controls (iface = SNDRV_CTL_ELEM_IFACE_MIXER).

Meanwhile, "IEC958 Playback Default" control is defined for getting and setting the current default IEC958 bits. Note that this one is usually defined as a PCM control (iface = SNDRV_CTL_ELEM_IFACE_PCM), although in some places it's defined as a MIXER control.

In addition, you can define the control switches to enable/disable or to set the raw bit mode. The implementation will depend on the chip, but the control should be named as "IEC958 xxx", preferably using the SNDRV_CTL_NAME_IEC958() macro.

You can find several cases, for example, pci/emu10k1, pci/ice1712, or pci/cmipci.c.

Buffer and Memory Management

Buffer Types

ALSA provides several different buffer allocation functions depending on the bus and the architecture. All these have a consistent API. The allocation of physically-contiguous pages is done via `snd_malloc_xxx_pages()` function, where xxx is the bus type.

The allocation of pages with fallback is `snd_malloc_xxx_pages_fallback()`. This function tries to allocate the specified pages but if the pages are not available, it tries to reduce the page sizes until enough space is found.

The release the pages, call `snd_free_xxx_pages()` function.

Usually, ALSA drivers try to allocate and reserve a large contiguous physical space at the time the module is loaded for the later use. This is called "pre-allocation". As already written, you can call the following function at pcm instance construction time (in the case of PCI bus).

```
snd_pcm_lib_preallocate_pages_for_all(pcm, SNDRV_DMA_TYPE_DEV,
                                      snd_dma_pci_data(pci), size, max);
```

where `size` is the byte size to be pre-allocated and the `max` is the maximum size to be changed via the `prealloc` proc file. The allocator will try to get an area as large as possible within the given size.

The second argument (type) and the third argument (device pointer) are dependent on the bus. In the case of the ISA bus, pass `snd_dma_isa_data()` as the third argument with `SNDRV_DMA_TYPE_DEV` type. For the continuous buffer unrelated to the bus can be pre-allocated with `SNDRV_DMA_TYPE_CONTINUOUS` type and the `snd_dma_continuous_data(GFP_KERNEL)` device pointer, where `GFP_KERNEL` is the kernel allocation flag to use. For the PCI scatter-gather buffers, use `SNDRV_DMA_TYPE_DEV_SG` with `snd_dma_pci_data(pci)` (see the *Non-Contiguous Buffers* section).

Once the buffer is pre-allocated, you can use the allocator in the `hw_params` callback:

```
snd_pcm_lib_malloc_pages(substream, size);
```

Note that you have to pre-allocate to use this function.

External Hardware Buffers

Some chips have their own hardware buffers and the DMA transfer from the host memory is not available. In such a case, you need to either 1) copy/set the audio data directly to the external hardware buffer, or 2) make an intermediate buffer and copy/set the data from it to the external hardware buffer in interrupts (or in tasklets, preferably).

The first case works fine if the external hardware buffer is large enough. This method doesn't need any extra buffers and thus is more effective. You need to define the `copy_user` and `copy_kernel` callbacks for the data transfer, in addition to `fill_silence` callback for playback. However, there is a drawback: it cannot be mmapped. The examples are GUS's GF1 PCM or emu8000's wavetable PCM.

The second case allows for mmap on the buffer, although you have to handle an interrupt or a tasklet to transfer the data from the intermediate buffer to the hardware buffer. You can find an example in the vxpocket driver.

Another case is when the chip uses a PCI memory-map region for the buffer instead of the host memory. In this case, mmap is available only on certain architectures like the Intel one. In non-mmap mode, the data cannot be transferred as in the normal way. Thus you need to define the `copy_user`, `copy_kernel` and `fill_silence` callbacks as well, as in the cases above. The examples are found in `rme32.c` and `rme96.c`.

The implementation of the `copy_user`, `copy_kernel` and `silence` callbacks depends upon whether the hardware supports interleaved or non-interleaved samples. The `copy_user` callback is defined like below, a bit differently depending whether the direction is playback or capture:

```
static int playback_copy_user(struct snd_pcm_substream *substream,
            int channel, unsigned long pos,
            void __user *src, unsigned long count);
```

```
static int capture_copy_user(struct snd_pcm_substream *substream,
             int channel, unsigned long pos,
             void __user *dst, unsigned long count);
```

In the case of interleaved samples, the second argument (`channel`) is not used. The third argument (`pos`) points the current position offset in bytes.

The meaning of the fourth argument is different between playback and capture. For playback, it holds the source data pointer, and for capture, it's the destination data pointer.

The last argument is the number of bytes to be copied.

What you have to do in this callback is again different between playback and capture directions. In the playback case, you copy the given amount of data (`count`) at the specified pointer (`src`) to the specified offset (`pos`) on the hardware buffer. When coded like memcpy-like way, the copy would be like:

```
my_memcpy_from_user(my_buffer + pos, src, count);
```

For the capture direction, you copy the given amount of data (`count`) at the specified offset (`pos`) on the hardware buffer to the specified pointer (`dst`).

```
my_memcpy_to_user(dst, my_buffer + pos, count);
```

Here the functions are named as `from_user` and `to_user` because it's the user-space buffer that is passed to these callbacks. That is, the callback is supposed to copy from/to the user-space data directly to/from the hardware buffer.

Careful readers might notice that these callbacks receive the arguments in bytes, not in frames like other callbacks. It's because it would make coding easier like the examples above, and also it makes easier to unify both the interleaved and non-interleaved cases, as explained in the following.

In the case of non-interleaved samples, the implementation will be a bit more complicated. The callback is called for each channel, passed by the second argument, so totally it's called for N-channels times per transfer.

The meaning of other arguments are almost same as the interleaved case. The callback is supposed to copy the data from/to the given user-space buffer, but only for the given channel. For the detailed implementations, please check `isa/gus/gus_pcm.c` or "pci/rme9652/rme9652.c" as examples.

The above callbacks are the copy from/to the user-space buffer. There are some cases where we want copy from/to the kernel-space buffer instead. In such a case, `copy_kernel` callback is called. It'd look like:

```
static int playback_copy_kernel(struct snd_pcm_substream *substream,
             int channel, unsigned long pos,
             void *src, unsigned long count);
static int capture_copy_kernel(struct snd_pcm_substream *substream,
             int channel, unsigned long pos,
             void *dst, unsigned long count);
```

As found easily, the only difference is that the buffer pointer is without `__user` prefix; that is, a kernel-buffer pointer is passed in the fourth argument. Correspondingly, the implementation would be a version without the user-copy, such as:

```
my_memcpy(my_buffer + pos, src, count);
```

Usually for the playback, another callback `fill_silence` is defined. It's implemented in a similar way as the copy callbacks above:

```
static int silence(struct snd_pcm_substream *substream, int channel,
                   unsigned long pos, unsigned long count);
```

The meanings of arguments are the same as in the `copy_user` and `copy_kernel` callbacks, although there is no buffer pointer argument. In the case of interleaved samples, the channel argument has no meaning, as well as on `copy_*` callbacks.

The role of `fill_silence` callback is to set the given amount (`count`) of silence data at the specified offset (`pos`) on the hardware buffer. Suppose that the data format is signed (that is, the silent-data is 0), and the implementation using a memset-like function would be like:

```
my_memset(my_buffer + pos, 0, count);
```

In the case of non-interleaved samples, again, the implementation becomes a bit more complicated, as it's called N-times per transfer for each channel. See, for example, `isa/gus/gus_pcm.c`.

Non-Contiguous Buffers

If your hardware supports the page table as in emu10k1 or the buffer descriptors as in via82xx, you can use the scatter-gather (SG) DMA. ALSA provides an interface for handling SG-buffers. The API is provided in `<sound/pcm.h>`.

For creating the SG-buffer handler, call *snd_pcm_lib_preallocate_pages()* or *snd_pcm_lib_preallocate_pages_for_all()* with `SNDRV_DMA_TYPE_DEV_SG` in the PCM constructor like other PCI pre-allocator. You need to pass `snd_dma_pci_data(pci)`, where pci is the `struct pci_dev` pointer of the chip as well. The `struct snd_sg_buf` instance is created as `substream->dma_private`. You can cast the pointer like:

```
struct snd_sg_buf *sgbuf = (struct snd_sg_buf *)substream->dma_private;
```

Then call *snd_pcm_lib_malloc_pages()* in the `hw_params` callback as well as in the case of normal PCI buffer. The SG-buffer handler will allocate the non-contiguous kernel pages of the given size and map them onto the virtually contiguous memory. The virtual pointer is addressed in runtime->dma_area. The physical address (`runtime->dma_addr`) is set to zero, because the buffer is physically non-contiguous. The physical address table is set up in `sgbuf->table`. You can get the physical address at a certain offset via *snd_pcm_sgbuf_get_addr()*.

When a SG-handler is used, you need to set *snd_pcm_sgbuf_ops_page()* as the page callback. (See *page callback* section.)

To release the data, call *snd_pcm_lib_free_pages()* in the `hw_free` callback as usual.

Vmalloc'ed Buffers

It's possible to use a buffer allocated via `vmalloc()`, for example, for an intermediate buffer. Since the allocated pages are not contiguous, you need to set the page callback to obtain the physical address at every offset.

The implementation of page callback would be like this:

```
#include <linux/vmalloc.h>

/* get the physical page pointer on the given offset */
static struct page *mychip_page(struct snd_pcm_substream *substream,
                                unsigned long offset)
{
        void *pageptr = substream->runtime->dma_area + offset;
        return vmalloc_to_page(pageptr);
}
```

Proc Interface

ALSA provides an easy interface for procfs. The proc files are very useful for debugging. I recommend you set up proc files if you write a driver and want to get a running status or register dumps. The API is found in `<sound/info.h>`.

To create a proc file, call `snd_card_proc_new()`.

```
struct snd_info_entry *entry;
int err = snd_card_proc_new(card, "my-file", &entry);
```

where the second argument specifies the name of the proc file to be created. The above example will create a file `my-file` under the card directory, e.g. `/proc/asound/card0/my-file`.

Like other components, the proc entry created via `snd_card_proc_new()` will be registered and released automatically in the card registration and release functions.

When the creation is successful, the function stores a new instance in the pointer given in the third argument. It is initialized as a text proc file for read only. To use this proc file as a read-only text file as it is, set the read callback with a private data via `snd_info_set_text_ops()`.

```
snd_info_set_text_ops(entry, chip, my_proc_read);
```

where the second argument (`chip`) is the private data to be used in the callbacks. The third parameter specifies the read buffer size and the fourth (`my_proc_read`) is the callback function, which is defined like

```
static void my_proc_read(struct snd_info_entry *entry,
                         struct snd_info_buffer *buffer);
```

In the read callback, use `snd_iprintf()` for output strings, which works just like normal `printf()`. For example,

```
static void my_proc_read(struct snd_info_entry *entry,
                         struct snd_info_buffer *buffer)
{
        struct my_chip *chip = entry->private_data;

        snd_iprintf(buffer, "This is my chip!\n");
        snd_iprintf(buffer, "Port = %ld\n", chip->port);
}
```

The file permissions can be changed afterwards. As default, it's set as read only for all users. If you want to add write permission for the user (root as default), do as follows:

```
entry->mode = S_IFREG | S_IRUGO | S_IWUSR;
```

and set the write buffer size and the callback

```
entry->c.text.write = my_proc_write;
```

For the write callback, you can use *snd_info_get_line()* to get a text line, and *snd_info_get_str()* to retrieve a string from the line. Some examples are found in `core/oss/mixer_oss.c`, core/oss/and `pcm_oss.c`.

For a raw-data proc-file, set the attributes as follows:

```
static struct snd_info_entry_ops my_file_io_ops = {
        .read = my_file_io_read,
};

entry->content = SNDRV_INFO_CONTENT_DATA;
entry->private_data = chip;
entry->c.ops = &my_file_io_ops;
entry->size = 4096;
entry->mode = S_IFREG | S_IRUGO;
```

For the raw data, `size` field must be set properly. This specifies the maximum size of the proc file access.

The read/write callbacks of raw mode are more direct than the text mode. You need to use a low-level I/O functions such as `copy_from/to_user()` to transfer the data.

```
static ssize_t my_file_io_read(struct snd_info_entry *entry,
                               void *file_private_data,
                               struct file *file,
                               char *buf,
                               size_t count,
                               loff_t pos)
{
        if (copy_to_user(buf, local_data + pos, count))
                return -EFAULT;
        return count;
}
```

If the size of the info entry has been set up properly, `count` and `pos` are guaranteed to fit within 0 and the given size. You don't have to check the range in the callbacks unless any other condition is required.

Power Management

If the chip is supposed to work with suspend/resume functions, you need to add power-management code to the driver. The additional code for power-management should be ifdef-ed with `CONFIG_PM`.

If the driver *fully* supports suspend/resume that is, the device can be properly resumed to its state when suspend was called, you can set the `SNDRV_PCM_INFO_RESUME` flag in the pcm info field. Usually, this is possible when the registers of the chip can be safely saved and restored to RAM. If this is set, the trigger callback is called with `SNDRV_PCM_TRIGGER_RESUME` after the resume callback completes.

Even if the driver doesn't support PM fully but partial suspend/resume is still possible, it's still worthy to implement suspend/resume callbacks. In such a case, applications would reset the status by calling *snd_pcm_prepare()* and restart the stream appropriately. Hence, you can define suspend/resume callbacks below but don't set `SNDRV_PCM_INFO_RESUME` info flag to the PCM.

Note that the trigger with SUSPEND can always be called when *snd_pcm_suspend_all()* is called, regardless of the `SNDRV_PCM_INFO_RESUME` flag. The RESUME flag affects only the behavior of `snd_pcm_resume()`. (Thus, in theory, `SNDRV_PCM_TRIGGER_RESUME` isn't needed to be handled in the trigger callback when no `SNDRV_PCM_INFO_RESUME` flag is set. But, it's better to keep it for compatibility reasons.)

In the earlier version of ALSA drivers, a common power-management layer was provided, but it has been removed. The driver needs to define the suspend/resume hooks according to the bus the device is connected to. In the case of PCI drivers, the callbacks look like below:

```
#ifdef CONFIG_PM
static int snd_my_suspend(struct pci_dev *pci, pm_message_t state)
{
        .... /* do things for suspend */
        return 0;
}
static int snd_my_resume(struct pci_dev *pci)
{
        .... /* do things for suspend */
        return 0;
}
#endif
```

The scheme of the real suspend job is as follows.

1. Retrieve the card and the chip data.
2. Call `snd_power_change_state()` with `SNDRV_CTL_POWER_D3hot` to change the power status.
3. Call *snd_pcm_suspend_all()* to suspend the running PCM streams.
4. If AC97 codecs are used, call *snd_ac97_suspend()* for each codec.
5. Save the register values if necessary.

6. Stop the hardware if necessary.
7. Disable the PCI device by calling `pci_disable_device()`. Then, call `pci_save_state()` at last.

A typical code would be like:

```
static int mychip_suspend(struct pci_dev *pci, pm_message_t state)
{
        /* (1) */
        struct snd_card *card = pci_get_drvdata(pci);
        struct mychip *chip = card->private_data;
        /* (2) */
        snd_power_change_state(card, SNDRV_CTL_POWER_D3hot);
        /* (3) */
        snd_pcm_suspend_all(chip->pcm);
        /* (4) */
        snd_ac97_suspend(chip->ac97);
        /* (5) */
        snd_mychip_save_registers(chip);
        /* (6) */
        snd_mychip_stop_hardware(chip);
        /* (7) */
        pci_disable_device(pci);
        pci_save_state(pci);
        return 0;
}
```

The scheme of the real resume job is as follows.

1. Retrieve the card and the chip data.
2. Set up PCI. First, call `pci_restore_state()`. Then enable the pci device again by calling `pci_enable_device()`. Call `pci_set_master()` if necessary, too.
3. Re-initialize the chip.
4. Restore the saved registers if necessary.
5. Resume the mixer, e.g. calling *snd_ac97_resume()*.
6. Restart the hardware (if any).
7. Call `snd_power_change_state()` with `SNDRV_CTL_POWER_D0` to notify the processes.

A typical code would be like:

```
static int mychip_resume(struct pci_dev *pci)
{
        /* (1) */
        struct snd_card *card = pci_get_drvdata(pci);
        struct mychip *chip = card->private_data;
        /* (2) */
        pci_restore_state(pci);
        pci_enable_device(pci);
        pci_set_master(pci);
        /* (3) */
        snd_mychip_reinit_chip(chip);
        /* (4) */
        snd_mychip_restore_registers(chip);
        /* (5) */
        snd_ac97_resume(chip->ac97);
        /* (6) */
        snd_mychip_restart_chip(chip);
        /* (7) */
        snd_power_change_state(card, SNDRV_CTL_POWER_D0);
        return 0;
}
```

As shown in the above, it's better to save registers after suspending the PCM operations via *snd_pcm_suspend_all()* or *snd_pcm_suspend()*. It means that the PCM streams are already stopped when the register snapshot is taken. But, remember that you don't have to restart the PCM stream in the resume callback. It'll be restarted via trigger call with SNDRV_PCM_TRIGGER_RESUME when necessary.

OK, we have all callbacks now. Let's set them up. In the initialization of the card, make sure that you can get the chip data from the card instance, typically via private_data field, in case you created the chip data individually.

```
static int snd_mychip_probe(struct pci_dev *pci,
                            const struct pci_device_id *pci_id)
{
        ....
        struct snd_card *card;
        struct mychip *chip;
        int err;
        ....
        err = snd_card_new(&pci->dev, index[dev], id[dev], THIS_MODULE,
                           0, &card);
        ....
        chip = kzalloc(sizeof(*chip), GFP_KERNEL);
        ....
        card->private_data = chip;
        ....
}
```

When you created the chip data with *snd_card_new()*, it's anyway accessible via private_data field.

```
static int snd_mychip_probe(struct pci_dev *pci,
                            const struct pci_device_id *pci_id)
{
        ....
        struct snd_card *card;
        struct mychip *chip;
        int err;
        ....
        err = snd_card_new(&pci->dev, index[dev], id[dev], THIS_MODULE,
                           sizeof(struct mychip), &card);
        ....
        chip = card->private_data;
        ....
}
```

If you need a space to save the registers, allocate the buffer for it here, too, since it would be fatal if you cannot allocate a memory in the suspend phase. The allocated buffer should be released in the corresponding destructor.

And next, set suspend/resume callbacks to the pci_driver.

```
static struct pci_driver driver = {
        .name = KBUILD_MODNAME,
        .id_table = snd_my_ids,
        .probe = snd_my_probe,
        .remove = snd_my_remove,
#ifdef CONFIG_PM
        .suspend = snd_my_suspend,
        .resume = snd_my_resume,
#endif
};
```

Module Parameters

There are standard module options for ALSA. At least, each module should have the `index`, `id` and `enable` options.

If the module supports multiple cards (usually up to 8 = `SNDRV_CARDS` cards), they should be arrays. The default initial values are defined already as constants for easier programming:

```
static int index[SNDRV_CARDS] = SNDRV_DEFAULT_IDX;
static char *id[SNDRV_CARDS] = SNDRV_DEFAULT_STR;
static int enable[SNDRV_CARDS] = SNDRV_DEFAULT_ENABLE_PNP;
```

If the module supports only a single card, they could be single variables, instead. `enable` option is not always necessary in this case, but it would be better to have a dummy option for compatibility.

The module parameters must be declared with the standard `module_param()()`, `module_param_array()()` and `MODULE_PARM_DESC()` macros.

The typical coding would be like below:

```
#define CARD_NAME "My Chip"

module_param_array(index, int, NULL, 0444);
MODULE_PARM_DESC(index, "Index value for " CARD_NAME " soundcard.");
module_param_array(id, charp, NULL, 0444);
MODULE_PARM_DESC(id, "ID string for " CARD_NAME " soundcard.");
module_param_array(enable, bool, NULL, 0444);
MODULE_PARM_DESC(enable, "Enable " CARD_NAME " soundcard.");
```

Also, don't forget to define the module description, classes, license and devices. Especially, the recent modprobe requires to define the module license as GPL, etc., otherwise the system is shown as "tainted".

```
MODULE_DESCRIPTION("My Chip");
MODULE_LICENSE("GPL");
MODULE_SUPPORTED_DEVICE("{{Vendor,My Chip Name}}");
```

How To Put Your Driver Into ALSA Tree

General

So far, you've learned how to write the driver codes. And you might have a question now: how to put my own driver into the ALSA driver tree? Here (finally :) the standard procedure is described briefly.

Suppose that you create a new PCI driver for the card "xyz". The card module name would be snd-xyz. The new driver is usually put into the alsa-driver tree, `alsa-driver/pci` directory in the case of PCI cards. Then the driver is evaluated, audited and tested by developers and users. After a certain time, the driver will go to the alsa-kernel tree (to the corresponding directory, such as `alsa-kernel/pci`) and eventually will be integrated into the Linux 2.6 tree (the directory would be `linux/sound/pci`).

In the following sections, the driver code is supposed to be put into alsa-driver tree. The two cases are covered: a driver consisting of a single source file and one consisting of several source files.

Driver with A Single Source File

1. Modify alsa-driver/pci/Makefile

   Suppose you have a file xyz.c. Add the following two lines

```
snd-xyz-objs := xyz.o
obj-$(CONFIG_SND_XYZ) += snd-xyz.o
```

2. Create the Kconfig entry

   Add the new entry of Kconfig for your xyz driver. config SND_XYZ tristate “Foobar XYZ” depends on SND select SND_PCM help Say Y here to include support for Foobar XYZ soundcard. To compile this driver as a module, choose M here: the module will be called snd-xyz. the line, select SND_PCM, specifies that the driver xyz supports PCM. In addition to SND_PCM, the following components are supported for select command: SND_RAWMIDI, SND_TIMER, SND_HWDEP, SND_MPU401_UART, SND_OPL3_LIB, SND_OPL4_LIB, SND_VX_LIB, SND_AC97_CODEC. Add the select command for each supported component.

   Note that some selections imply the lowlevel selections. For example, PCM includes TIMER, MPU401_UART includes RAWMIDI, AC97_CODEC includes PCM, and OPL3_LIB includes HWDEP. You don’t need to give the lowlevel selections again.

   For the details of Kconfig script, refer to the kbuild documentation.

3. Run cvscompile script to re-generate the configure script and build the whole stuff again.

Drivers with Several Source Files

Suppose that the driver snd-xyz have several source files. They are located in the new subdirectory, pci/xyz.

1. Add a new directory (xyz) in `alsa-driver/pci/Makefile` as below

```
obj-$(CONFIG_SND) += xyz/
```

2. Under the directory xyz, create a Makefile

```
ifndef SND_TOPDIR
SND_TOPDIR=../..
endif

include $(SND_TOPDIR)/toplevel.config
include $(SND_TOPDIR)/Makefile.conf

snd-xyz-objs := xyz.o abc.o def.o

obj-$(CONFIG_SND_XYZ) += snd-xyz.o

include $(SND_TOPDIR)/Rules.make
```

3. Create the Kconfig entry

   This procedure is as same as in the last section.

4. Run cvscompile script to re-generate the configure script and build the whole stuff again.

Useful Functions

`snd_printk()` and friends

ALSA provides a verbose version of the `printk()` function. If a kernel config `CONFIG_SND_VERBOSE_PRINTK` is set, this function prints the given message together with the file name and the line of the caller. The KERN_XXX prefix is processed as well as the original `printk()` does, so it’s recommended to add this prefix, e.g. snd_printk(KERN_ERR “Oh my, sorry, it’s extremely bad!\n”);

There are also `printk()`‘s for debugging. *snd_printd()* can be used for general debugging purposes. If `CONFIG_SND_DEBUG` is set, this function is compiled, and works just like *snd_printk()*. If the ALSA is compiled without the debugging flag, it’s ignored.

snd_printdd() is compiled in only when CONFIG_SND_DEBUG_VERBOSE is set. Please note that CONFIG_SND_DEBUG_VERBOSE is not set as default even if you configure the alsa-driver with `--with-debug=full` option. You need to give explicitly `--with-debug=detect` option instead.

snd_BUG()

It shows the BUG? message and stack trace as well as *snd_BUG_ON()* at the point. It's useful to show that a fatal error happens there.

When no debug flag is set, this macro is ignored.

snd_BUG_ON()

snd_BUG_ON() macro is similar with WARN_ON() macro. For example, snd_BUG_ON(!pointer); or it can be used as the condition, if (snd_BUG_ON(non_zero_is_bug)) return -EINVAL;

The macro takes an conditional expression to evaluate. When CONFIG_SND_DEBUG, is set, if the expression is non-zero, it shows the warning message such as BUG? (xxx) normally followed by stack trace. In both cases it returns the evaluated value.

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CHAPTER

TWO

DESIGNS AND IMPLEMENTATIONS

Standard ALSA Control Names

This document describes standard names of mixer controls.

Standard Syntax

Syntax: [LOCATION] SOURCE [CHANNEL] [DIRECTION] FUNCTION

DIRECTION

<nothing>	both directions
Playback	one direction
Capture	one direction
Bypass Playback	one direction
Bypass Capture	one direction

FUNCTION

Switch	on/off switch
Volume	amplifier
Route	route control, hardware specific

CHANNEL

<nothing>	channel independent, or applies to all channels
Front	front left/right channels
Surround	rear left/right in 4.0/5.1 surround
CLFE	C/LFE channels
Center	center cannel
LFE	LFE channel
Side	side left/right for 7.1 surround

LOCATION (Physical location of source)

Front	front position
Rear	rear position
Dock	on docking station
Internal	internal

SOURCE

Master	
Master Mono	
Hardware Master	
Speaker	internal speaker
Bass Speaker	internal LFE speaker
Headphone	
Line Out	
Beep	beep generator
Phone	
Phone Input	
Phone Output	
Synth	
FM	
Mic	
Headset Mic	mic part of combined headset jack - 4-pin headphone + mic
Headphone Mic	mic part of either/or - 3-pin headphone or mic
Line	input only, use “Line Out” for output
CD	
Video	
Zoom Video	
Aux	
PCM	
PCM Pan	
Loopback	
Analog Loopback	D/A -> A/D loopback
Digital Loopback	playback -> capture loopback - without analog path
Mono	
Mono Output	
Multi	
ADC	
Wave	
Music	
I2S	
IEC958	
HDMI	
SPDIF	output only
SPDIF In	
Digital In	
HDMI/DP	either HDMI or DisplayPort

Exceptions (deprecated)

[Analogue|Digital] Capture Source	
[Analogue|Digital] Capture Switch	aka input gain switch
[Analogue|Digital] Capture Volume	aka input gain volume
[Analogue|Digital] Playback Switch	aka output gain switch
[Analogue|Digital] Playback Volume	aka output gain volume
Tone Control - Switch	
Tone Control - Bass	
Tone Control - Treble	
3D Control - Switch	
3D Control - Center	
3D Control - Depth	
3D Control - Wide	
3D Control - Space	
3D Control - Level	
Mic Boost [(?dB)]	

PCM interface

Sample Clock Source	{ “Word”, “Internal”, “AutoSync” }
Clock Sync Status	{ “Lock”, “Sync”, “No Lock” }
External Rate	external capture rate
Capture Rate	capture rate taken from external source

IEC958 (S/PDIF) interface

IEC958 [...] [Playback|Capture] Switch	turn on/off the IEC958 interface
IEC958 [...] [Playback|Capture] Volume	digital volume control
IEC958 [...] [Playback|Capture] Default	default or global value - read/write
IEC958 [...] [Playback|Capture] Mask	consumer and professional mask
IEC958 [...] [Playback|Capture] Con Mask	consumer mask
IEC958 [...] [Playback|Capture] Pro Mask	professional mask
IEC958 [...] [Playback|Capture] PCM Stream	the settings assigned to a PCM stream
IEC958 Q-subcode [Playback|Capture] Default	Q-subcode bits
IEC958 Preamble [Playback|Capture] Default	burst preamble words (4*16bits)

ALSA PCM channel-mapping API

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General

The channel mapping API allows user to query the possible channel maps and the current channel map, also optionally to modify the channel map of the current stream.

A channel map is an array of position for each PCM channel. Typically, a stereo PCM stream has a channel map of `{ front_left, front_right }` while a 4.0 surround PCM stream has a channel map of `{ front left, front right, rear left, rear right }`.

The problem, so far, was that we had no standard channel map explicitly, and applications had no way to know which channel corresponds to which (speaker) position. Thus, applications applied wrong channels

for 5.1 outputs, and you hear suddenly strange sound from rear. Or, some devices secretly assume that center/LFE is the third/fourth channels while others that C/LFE as 5th/6th channels.

Also, some devices such as HDMI are configurable for different speaker positions even with the same number of total channels. However, there was no way to specify this because of lack of channel map specification. These are the main motivations for the new channel mapping API.

Design

Actually, "the channel mapping API" doesn't introduce anything new in the kernel/user-space ABI perspective. It uses only the existing control element features.

As a ground design, each PCM substream may contain a control element providing the channel mapping information and configuration. This element is specified by:

- iface = SNDRV_CTL_ELEM_IFACE_PCM
- name = "Playback Channel Map" or "Capture Channel Map"
- device = the same device number for the assigned PCM substream
- index = the same index number for the assigned PCM substream

Note the name is different depending on the PCM substream direction.

Each control element provides at least the TLV read operation and the read operation. Optionally, the write operation can be provided to allow user to change the channel map dynamically.

TLV

The TLV operation gives the list of available channel maps. A list item of a channel map is usually a TLV of `type data-bytes ch0 ch1 ch2...` where type is the TLV type value, the second argument is the total bytes (not the numbers) of channel values, and the rest are the position value for each channel.

As a TLV type, either SNDRV_CTL_TLVT_CHMAP_FIXED, SNDRV_CTL_TLV_CHMAP_VAR or SNDRV_CTL_TLVT_CHMAP_PAIRED can be used. The _FIXED type is for a channel map with the fixed channel position while the latter two are for flexible channel positions. _VAR type is for a channel map where all channels are freely swappable and _PAIRED type is where pair-wise channels are swappable. For example, when you have {FL/FR/RL/RR} channel map, _PAIRED type would allow you to swap only {RL/RR/FL/FR} while _VAR type would allow even swapping FL and RR.

These new TLV types are defined in `sound/tlv.h`.

The available channel position values are defined in `sound/asound.h`, here is a cut:

```
/* channel positions */
enum {
      SNDRV_CHMAP_UNKNOWN = 0,
      SNDRV_CHMAP_NA,         /* N/A, silent */
      SNDRV_CHMAP_MONO,       /* mono stream */
      /* this follows the alsa-lib mixer channel value + 3 */
      SNDRV_CHMAP_FL,         /* front left */
      SNDRV_CHMAP_FR,         /* front right */
      SNDRV_CHMAP_RL,         /* rear left */
      SNDRV_CHMAP_RR,         /* rear right */
      SNDRV_CHMAP_FC,         /* front center */
      SNDRV_CHMAP_LFE,        /* LFE */
      SNDRV_CHMAP_SL,         /* side left */
      SNDRV_CHMAP_SR,         /* side right */
      SNDRV_CHMAP_RC,         /* rear center */
      /* new definitions */
      SNDRV_CHMAP_FLC,        /* front left center */
      SNDRV_CHMAP_FRC,        /* front right center */
      SNDRV_CHMAP_RLC,        /* rear left center */
```

```
      SNDRV_CHMAP_RRC,        /* rear right center */
      SNDRV_CHMAP_FLW,        /* front left wide */
      SNDRV_CHMAP_FRW,        /* front right wide */
      SNDRV_CHMAP_FLH,        /* front left high */
      SNDRV_CHMAP_FCH,        /* front center high */
      SNDRV_CHMAP_FRH,        /* front right high */
      SNDRV_CHMAP_TC,         /* top center */
      SNDRV_CHMAP_TFL,        /* top front left */
      SNDRV_CHMAP_TFR,        /* top front right */
      SNDRV_CHMAP_TFC,        /* top front center */
      SNDRV_CHMAP_TRL,        /* top rear left */
      SNDRV_CHMAP_TRR,        /* top rear right */
      SNDRV_CHMAP_TRC,        /* top rear center */
      SNDRV_CHMAP_LAST = SNDRV_CHMAP_TRC,
};
```

When a PCM stream can provide more than one channel map, you can provide multiple channel maps in a TLV container type. The TLV data to be returned will contain such as:

```
SNDRV_CTL_TLVT_CONTAINER 96
    SNDRV_CTL_TLVT_CHMAP_FIXED 4 SNDRV_CHMAP_FC
    SNDRV_CTL_TLVT_CHMAP_FIXED 8 SNDRV_CHMAP_FL SNDRV_CHMAP_FR
    SNDRV_CTL_TLVT_CHMAP_FIXED 16 NDRV_CHMAP_FL SNDRV_CHMAP_FR \
        SNDRV_CHMAP_RL SNDRV_CHMAP_RR
```

The channel position is provided in LSB 16bits. The upper bits are used for bit flags.

```
#define SNDRV_CHMAP_POSITION_MASK       0xffff
#define SNDRV_CHMAP_PHASE_INVERSE       (0x01 << 16)
#define SNDRV_CHMAP_DRIVER_SPEC         (0x02 << 16)
```

SNDRV_CHMAP_PHASE_INVERSE indicates the channel is phase inverted, (thus summing left and right channels would result in almost silence). Some digital mic devices have this.

When SNDRV_CHMAP_DRIVER_SPEC is set, all the channel position values don't follow the standard definition above but driver-specific.

Read Operation

The control read operation is for providing the current channel map of the given stream. The control element returns an integer array containing the position of each channel.

When this is performed before the number of the channel is specified (i.e. hw_params is set), it should return all channels set to UNKNOWN.

Write Operation

The control write operation is optional, and only for devices that can change the channel configuration on the fly, such as HDMI. User needs to pass an integer value containing the valid channel positions for all channels of the assigned PCM substream.

This operation is allowed only at PCM PREPARED state. When called in other states, it shall return an error.

ALSA Compress-Offload API

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Overview

Since its early days, the ALSA API was defined with PCM support or constant bitrates payloads such as IEC61937 in mind. Arguments and returned values in frames are the norm, making it a challenge to extend the existing API to compressed data streams.

In recent years, audio digital signal processors (DSP) were integrated in system-on-chip designs, and DSPs are also integrated in audio codecs. Processing compressed data on such DSPs results in a dramatic reduction of power consumption compared to host-based processing. Support for such hardware has not been very good in Linux, mostly because of a lack of a generic API available in the mainline kernel.

Rather than requiring a compatibility break with an API change of the ALSA PCM interface, a new 'Compressed Data' API is introduced to provide a control and data-streaming interface for audio DSPs.

The design of this API was inspired by the 2-year experience with the Intel Moorestown SOC, with many corrections required to upstream the API in the mainline kernel instead of the staging tree and make it usable by others.

Requirements

The main requirements are:

- separation between byte counts and time. Compressed formats may have a header per file, per frame, or no header at all. The payload size may vary from frame-to-frame. As a result, it is not possible to estimate reliably the duration of audio buffers when handling compressed data. Dedicated mechanisms are required to allow for reliable audio-video synchronization, which requires precise reporting of the number of samples rendered at any given time.
- Handling of multiple formats. PCM data only requires a specification of the sampling rate, number of channels and bits per sample. In contrast, compressed data comes in a variety of formats. Audio DSPs may also provide support for a limited number of audio encoders and decoders embedded in firmware, or may support more choices through dynamic download of libraries.
- Focus on main formats. This API provides support for the most popular formats used for audio and video capture and playback. It is likely that as audio compression technology advances, new formats will be added.
- Handling of multiple configurations. Even for a given format like AAC, some implementations may support AAC multichannel but HE-AAC stereo. Likewise WMA10 level M3 may require too much memory and cpu cycles. The new API needs to provide a generic way of listing these formats.
- Rendering/Grabbing only. This API does not provide any means of hardware acceleration, where PCM samples are provided back to user-space for additional processing. This API focuses instead on streaming compressed data to a DSP, with the assumption that the decoded samples are routed to a physical output or logical back-end.
- Complexity hiding. Existing user-space multimedia frameworks all have existing enums/structures for each compressed format. This new API assumes the existence of a platform-specific compatibility layer to expose, translate and make use of the capabilities of the audio DSP, eg. Android HAL or PulseAudio sinks. By construction, regular applications are not supposed to make use of this API.

Design

The new API shares a number of concepts with the PCM API for flow control. Start, pause, resume, drain and stop commands have the same semantics no matter what the content is.

The concept of memory ring buffer divided in a set of fragments is borrowed from the ALSA PCM API. However, only sizes in bytes can be specified.

Seeks/trick modes are assumed to be handled by the host.

The notion of rewinds/forwards is not supported. Data committed to the ring buffer cannot be invalidated, except when dropping all buffers.

The Compressed Data API does not make any assumptions on how the data is transmitted to the audio DSP. DMA transfers from main memory to an embedded audio cluster or to a SPI interface for external DSPs are possible. As in the ALSA PCM case, a core set of routines is exposed; each driver implementer will have to write support for a set of mandatory routines and possibly make use of optional ones.

The main additions are

get_caps This routine returns the list of audio formats supported. Querying the codecs on a capture stream will return encoders, decoders will be listed for playback streams.

get_codec_caps For each codec, this routine returns a list of capabilities. The intent is to make sure all the capabilities correspond to valid settings, and to minimize the risks of configuration failures. For example, for a complex codec such as AAC, the number of channels supported may depend on a specific profile. If the capabilities were exposed with a single descriptor, it may happen that a specific combination of profiles/channels/formats may not be supported. Likewise, embedded DSPs have limited memory and cpu cycles, it is likely that some implementations make the list of capabilities dynamic and dependent on existing workloads. In addition to codec settings, this routine returns the minimum buffer size handled by the implementation. This information can be a function of the DMA buffer sizes, the number of bytes required to synchronize, etc, and can be used by userspace to define how much needs to be written in the ring buffer before playback can start.

set_params This routine sets the configuration chosen for a specific codec. The most important field in the parameters is the codec type; in most cases decoders will ignore other fields, while encoders will strictly comply to the settings

get_params This routines returns the actual settings used by the DSP. Changes to the settings should remain the exception.

get_timestamp The timestamp becomes a multiple field structure. It lists the number of bytes transferred, the number of samples processed and the number of samples rendered/grabbed. All these values can be used to determine the average bitrate, figure out if the ring buffer needs to be refilled or the delay due to decoding/encoding/io on the DSP.

Note that the list of codecs/profiles/modes was derived from the OpenMAX AL specification instead of reinventing the wheel. Modifications include: - Addition of FLAC and IEC formats - Merge of encoder/decoder capabilities - Profiles/modes listed as bitmasks to make descriptors more compact - Addition of set_params for decoders (missing in OpenMAX AL) - Addition of AMR/AMR-WB encoding modes (missing in OpenMAX AL) - Addition of format information for WMA - Addition of encoding options when required (derived from OpenMAX IL) - Addition of rateControlSupported (missing in OpenMAX AL)

Gapless Playback

When playing thru an album, the decoders have the ability to skip the encoder delay and padding and directly move from one track content to another. The end user can perceive this as gapless playback as we don't have silence while switching from one track to another

Also, there might be low-intensity noises due to encoding. Perfect gapless is difficult to reach with all types of compressed data, but works fine with most music content. The decoder needs to know the encoder delay and encoder padding. So we need to pass this to DSP. This metadata is extracted from ID3/MP4 headers and are not present by default in the bitstream, hence the need for a new interface to pass this information to the DSP. Also DSP and userspace needs to switch from one track to another and start using data for second track.

The main additions are:

set_metadata This routine sets the encoder delay and encoder padding. This can be used by decoder to strip the silence. This needs to be set before the data in the track is written.

set_next_track This routine tells DSP that metadata and write operation sent after this would correspond to subsequent track

partial drain This is called when end of file is reached. The userspace can inform DSP that EOF is reached and now DSP can start skipping padding delay. Also next write data would belong to next track

Sequence flow for gapless would be: - Open - Get caps / codec caps - Set params - Set metadata of the first track - Fill data of the first track - Trigger start - User-space finished sending all, - Indicate next track data by sending set_next_track - Set metadata of the next track - then call partial_drain to flush most of buffer in DSP - Fill data of the next track - DSP switches to second track

(note: order for partial_drain and write for next track can be reversed as well)

Not supported

- Support for VoIP/circuit-switched calls is not the target of this API. Support for dynamic bit-rate changes would require a tight coupling between the DSP and the host stack, limiting power savings.
- Packet-loss concealment is not supported. This would require an additional interface to let the decoder synthesize data when frames are lost during transmission. This may be added in the future.
- Volume control/routing is not handled by this API. Devices exposing a compressed data interface will be considered as regular ALSA devices; volume changes and routing information will be provided with regular ALSA kcontrols.
- Embedded audio effects. Such effects should be enabled in the same manner, no matter if the input was PCM or compressed.
- multichannel IEC encoding. Unclear if this is required.
- Encoding/decoding acceleration is not supported as mentioned above. It is possible to route the output of a decoder to a capture stream, or even implement transcoding capabilities. This routing would be enabled with ALSA kcontrols.
- Audio policy/resource management. This API does not provide any hooks to query the utilization of the audio DSP, nor any preemption mechanisms.
- No notion of underrun/overrun. Since the bytes written are compressed in nature and data written/read doesn't translate directly to rendered output in time, this does not deal with underrun/overrun and maybe dealt in user-library

Credits

- Mark Brown and Liam Girdwood for discussions on the need for this API
- Harsha Priya for her work on intel_sst compressed API
- Rakesh Ughreja for valuable feedback
- Sing Nallasellan, Sikkandar Madar and Prasanna Samaga for demonstrating and quantifying the benefits of audio offload on a real platform.

ALSA PCM Timestamping

The ALSA API can provide two different system timestamps:

- Trigger_tstamp is the system time snapshot taken when the .trigger callback is invoked. This snapshot is taken by the ALSA core in the general case, but specific hardware may have synchronization capabilities or conversely may only be able to provide a correct estimate with a delay. In the latter two cases, the low-level driver is responsible for updating the trigger_tstamp at the most appropriate and precise moment. Applications should not rely solely on the first trigger_tstamp but update their internal calculations if the driver provides a refined estimate with a delay.

- tstamp is the current system timestamp updated during the last event or application query. The difference (tstamp - trigger_tstamp) defines the elapsed time.

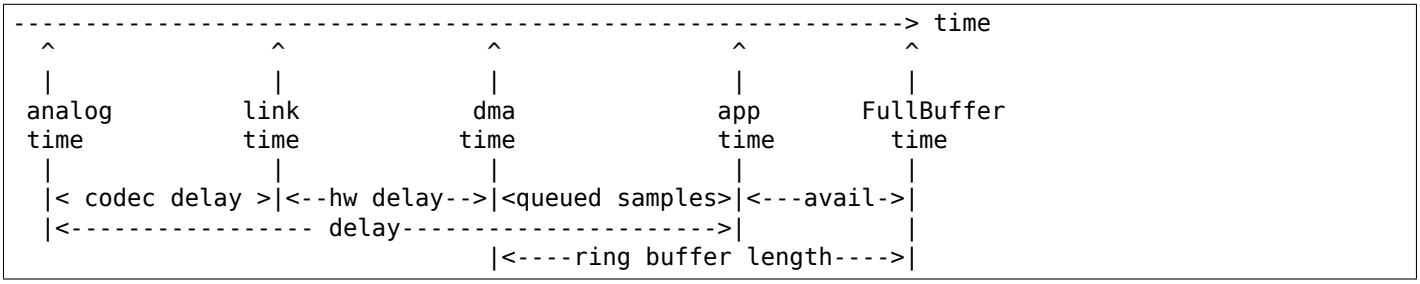
The ALSA API provides two basic pieces of information, avail and delay, which combined with the trigger and current system timestamps allow for applications to keep track of the 'fullness' of the ring buffer and the amount of queued samples.

The use of these different pointers and time information depends on the application needs:

- `avail` reports how much can be written in the ring buffer
- `delay` reports the time it will take to hear a new sample after all queued samples have been played out.

When timestamps are enabled, the avail/delay information is reported along with a snapshot of system time. Applications can select from `CLOCK_REALTIME` (NTP corrections including going backwards), `CLOCK_MONOTONIC` (NTP corrections but never going backwards), `CLOCK_MONOTIC_RAW` (without NTP corrections) and change the mode dynamically with sw_params

The ALSA API also provide an audio_tstamp which reflects the passage of time as measured by different components of audio hardware. In ascii-art, this could be represented as follows (for the playback case):



The analog time is taken at the last stage of the playback, as close as possible to the actual transducer

The link time is taken at the output of the SoC/chipset as the samples are pushed on a link. The link time can be directly measured if supported in hardware by sample counters or wallclocks (e.g. with HDAudio 24MHz or PTP clock for networked solutions) or indirectly estimated (e.g. with the frame counter in USB).

The DMA time is measured using counters - typically the least reliable of all measurements due to the bursty nature of DMA transfers.

The app time corresponds to the time tracked by an application after writing in the ring buffer.

The application can query the hardware capabilities, define which audio time it wants reported by selecting the relevant settings in audio_tstamp_config fields, thus get an estimate of the timestamp accuracy. It can also request the delay-to-analog be included in the measurement. Direct access to the link time is very interesting on platforms that provide an embedded DSP; measuring directly the link time with dedicated hardware, possibly synchronized with system time, removes the need to keep track of internal DSP processing times and latency.

In case the application requests an audio tstamp that is not supported in hardware/low-level driver, the type is overridden as DEFAULT and the timestamp will report the DMA time based on the hw_pointer value.

For backwards compatibility with previous implementations that did not provide timestamp selection, with a zero-valued COMPAT timestamp type the results will default to the HDAudio wall clock for playback streams and to the DMA time (hw_ptr) in all other cases.

The audio timestamp accuracy can be returned to user-space, so that appropriate decisions are made:

- for dma time (default), the granularity of the transfers can be inferred from the steps between updates and in turn provide information on how much the application pointer can be rewound safely.
- the link time can be used to track long-term drifts between audio and system time using the (tstamp-trigger_tstamp)/audio_tstamp ratio, the precision helps define how much smoothing/low-pass filtering is required. The link time can be either reset on startup or reported as is (the latter being useful to compare progress of different streams - but may require the wallclock to be always running and

not wrap-around during idle periods). If supported in hardware, the absolute link time could also be used to define a precise start time (patches WIP)

- including the delay in the audio timestamp may counter-intuitively not increase the precision of timestamps, e.g. if a codec includes variable-latency DSP processing or a chain of hardware components the delay is typically not known with precision.

The accuracy is reported in nanosecond units (using an unsigned 32-bit word), which gives a max precision of 4.29s, more than enough for audio applications...

Due to the varied nature of timestamping needs, even for a single application, the audio_tstamp_config can be changed dynamically. In the STATUS ioctl, the parameters are read-only and do not allow for any application selection. To work around this limitation without impacting legacy applications, a new STATUS_EXT ioctl is introduced with read/write parameters. ALSA-lib will be modified to make use of STATUS_EXT and effectively deprecate STATUS.

The ALSA API only allows for a single audio timestamp to be reported at a time. This is a conscious design decision, reading the audio timestamps from hardware registers or from IPC takes time, the more timestamps are read the more imprecise the combined measurements are. To avoid any interpretation issues, a single (system, audio) timestamp is reported. Applications that need different timestamps will be required to issue multiple queries and perform an interpolation of the results

In some hardware-specific configuration, the system timestamp is latched by a low-level audio subsystem, and the information provided back to the driver. Due to potential delays in the communication with the hardware, there is a risk of misalignment with the avail and delay information. To make sure applications are not confused, a driver_timestamp field is added in the snd_pcm_status structure; this timestamp shows when the information is put together by the driver before returning from the STATUS and STATUS_EXT ioctl. in most cases this driver_timestamp will be identical to the regular system tstamp.

Examples of typestamping with HDaudio:

1. DMA timestamp, no compensation for DMA+analog delay

```
$ ./audio_time  -p --ts_type=1
playback: systime: 341121338 nsec, audio time 342000000 nsec,         systime delta -878662
playback: systime: 426236663 nsec, audio time 427187500 nsec,         systime delta -950837
playback: systime: 597080580 nsec, audio time 598000000 nsec,         systime delta -919420
playback: systime: 682059782 nsec, audio time 683020833 nsec,         systime delta -961051
playback: systime: 852896415 nsec, audio time 853854166 nsec,         systime delta -957751
playback: systime: 937903344 nsec, audio time 938854166 nsec,         systime delta -950822
```

2. DMA timestamp, compensation for DMA+analog delay

```
$ ./audio_time  -p --ts_type=1 -d
playback: systime: 341053347 nsec, audio time 341062500 nsec,         systime delta -9153
playback: systime: 426072447 nsec, audio time 426062500 nsec,         systime delta 9947
playback: systime: 596899518 nsec, audio time 596895833 nsec,         systime delta 3685
playback: systime: 681915317 nsec, audio time 681916666 nsec,         systime delta -1349
playback: systime: 852741306 nsec, audio time 852750000 nsec,         systime delta -8694
```

3. link timestamp, compensation for DMA+analog delay

```
$ ./audio_time  -p --ts_type=2 -d
playback: systime: 341060004 nsec, audio time 341062791 nsec,         systime delta -2787
playback: systime: 426242074 nsec, audio time 426244875 nsec,         systime delta -2801
playback: systime: 597080992 nsec, audio time 597084583 nsec,         systime delta -3591
playback: systime: 682084512 nsec, audio time 682088291 nsec,         systime delta -3779
playback: systime: 852936229 nsec, audio time 852940916 nsec,         systime delta -4687
playback: systime: 938107562 nsec, audio time 938112708 nsec,         systime delta -5146
```

Example 1 shows that the timestamp at the DMA level is close to 1ms ahead of the actual playback time (as a side time this sort of measurement can help define rewind safeguards). Compensating for the DMA-link delay in example 2 helps remove the hardware buffering but the information is still very jittery, with up to one sample of error. In example 3 where the timestamps are measured with the link wallclock, the timestamps show a monotonic behavior and a lower dispersion.

Example 3 and 4 are with USB audio class. Example 3 shows a high offset between audio time and system time due to buffering. Example 4 shows how compensating for the delay exposes a 1ms accuracy (due to the use of the frame counter by the driver)

Example 3: DMA timestamp, no compensation for delay, delta of ~5ms

```
$ ./audio_time -p -Dhw:1 -t1
playback: systime: 120174019 nsec, audio time 125000000 nsec,         systime delta -4825981
playback: systime: 245041136 nsec, audio time 250000000 nsec,         systime delta -4958864
playback: systime: 370106088 nsec, audio time 375000000 nsec,         systime delta -4893912
playback: systime: 495040065 nsec, audio time 500000000 nsec,         systime delta -4959935
playback: systime: 620038179 nsec, audio time 625000000 nsec,         systime delta -4961821
playback: systime: 745087741 nsec, audio time 750000000 nsec,         systime delta -4912259
playback: systime: 870037336 nsec, audio time 875000000 nsec,         systime delta -4962664
```

Example 4: DMA timestamp, compensation for delay, delay of ~1ms

```
$ ./audio_time -p -Dhw:1 -t1 -d
playback: systime: 120190520 nsec, audio time 120000000 nsec,         systime delta 190520
playback: systime: 245036740 nsec, audio time 244000000 nsec,         systime delta 1036740
playback: systime: 370034081 nsec, audio time 369000000 nsec,         systime delta 1034081
playback: systime: 495159907 nsec, audio time 494000000 nsec,         systime delta 1159907
playback: systime: 620098824 nsec, audio time 619000000 nsec,         systime delta 1098824
playback: systime: 745031847 nsec, audio time 744000000 nsec,         systime delta 1031847
```

ALSA Jack Controls

Why we need Jack kcontrols

ALSA uses kcontrols to export audio controls(switch, volume, Mux, ...) to user space. This means userspace applications like pulseaudio can switch off headphones and switch on speakers when no headphones are pluged in.

The old ALSA jack code only created input devices for each registered jack. These jack input devices are not readable by userspace devices that run as non root.

The new jack code creates embedded jack kcontrols for each jack that can be read by any process.

This can be combined with UCM to allow userspace to route audio more intelligently based on jack insertion or removal events.

Jack Kcontrol Internals

Each jack will have a kcontrol list, so that we can create a kcontrol and attach it to the jack, at jack creation stage. We can also add a kcontrol to an existing jack, at anytime when required.

Those kcontrols will be freed automatically when the Jack is freed.

How to use jack kcontrols

In order to keep compatibility, snd_jack_new() has been modified by adding two params:

initial_kctl if true, create a kcontrol and add it to the jack list.

phantom_jack Don't create a input device for phantom jacks.

HDA jacks can set phantom_jack to true in order to create a phantom jack and set initial_kctl to true to create an initial kcontrol with the correct id.

ASoC jacks should set initial_kctl as false. The pin name will be assigned as the jack kcontrol name.

Tracepoints in ALSA

2017/07/02 Takasahi Sakamoto

Tracepoints in ALSA PCM core

ALSA PCM core registers `snd_pcm` subsystem to kernel tracepoint system. This subsystem includes two categories of tracepoints; for state of PCM buffer and for processing of PCM hardware parameters. These tracepoints are available when corresponding kernel configurations are enabled. When `CONFIG_SND_DEBUG` is enabled, the latter tracepoints are available. When additional `SND_PCM_XRUN_DEBUG` is enabled too, the former trace points are enabled.

Tracepoints for state of PCM buffer

This category includes four tracepoints; `hwptr`, `applptr`, `xrun` and `hw_ptr_error`.

Tracepoints for processing of PCM hardware parameters

This category includes two tracepoints; `hw_mask_param` and `hw_interval_param`.

In a design of ALSA PCM core, data transmission is abstracted as PCM substream. Applications manage PCM substream to maintain data transmission for PCM frames. Before starting the data transmission, applications need to configure PCM substream. In this procedure, PCM hardware parameters are decided by interaction between applications and ALSA PCM core. Once decided, runtime of the PCM substream keeps the parameters.

The parameters are described in `struct snd_pcm_hw_params`. This structure includes several types of parameters. Applications set preferable value to these parameters, then execute ioctl(2) with SNDRV_PCM_IOCTL_HW_REFINE or SNDRV_PCM_IOCTL_HW_PARAMS. The former is used just for refining available set of parameters. The latter is used for an actual decision of the parameters.

The `struct snd_pcm_hw_params` structure has below members:

`flags` Configurable. ALSA PCM core and some drivers handle this flag to select convenient parameters or change their behaviour.

`masks` Configurable. This type of parameter is described in `struct snd_mask` and represent mask values. As of PCM protocol v2.0.13, three types are defined.

- SNDRV_PCM_HW_PARAM_ACCESS
- SNDRV_PCM_HW_PARAM_FORMAT
- SNDRV_PCM_HW_PARAM_SUBFORMAT

`intervals` Configurable. This type of parameter is described in `struct snd_interval` and represent values with a range. As of PCM protocol v2.0.13, twelve types are defined.

- SNDRV_PCM_HW_PARAM_SAMPLE_BITS
- SNDRV_PCM_HW_PARAM_FRAME_BITS
- SNDRV_PCM_HW_PARAM_CHANNELS
- SNDRV_PCM_HW_PARAM_RATE
- SNDRV_PCM_HW_PARAM_PERIOD_TIME
- SNDRV_PCM_HW_PARAM_PERIOD_SIZE
- SNDRV_PCM_HW_PARAM_PERIOD_BYTES
- SNDRV_PCM_HW_PARAM_PERIODS

- SNDRV_PCM_HW_PARAM_BUFFER_TIME
- SNDRV_PCM_HW_PARAM_BUFFER_SIZE
- SNDRV_PCM_HW_PARAM_BUFFER_BYTES
- SNDRV_PCM_HW_PARAM_TICK_TIME

rmask Configurable. This is evaluated at ioctl(2) with SNDRV_PCM_IOCTL_HW_REFINE only. Applications can select which mask/interval parameter can be changed by ALSA PCM core. For SNDRV_PCM_IOCTL_HW_PARAMS, this mask is ignored and all of parameters are going to be changed.

cmask Read-only. After returning from ioctl(2), buffer in user space for `struct snd_pcm_hw_params` includes result of each operation. This mask represents which mask/interval parameter is actually changed.

info Read-only. This represents hardware/driver capabilities as bit flags with SNDRV_PCM_INFO_XXX. Typically, applications execute ioctl(2) with SNDRV_PCM_IOCTL_HW_REFINE to retrieve this flag, then decide candidates of parameters and execute ioctl(2) with SNDRV_PCM_IOCTL_HW_PARAMS to configure PCM substream.

msbits Read-only. This value represents available bit width in MSB side of a PCM sample. When a parameter of SNDRV_PCM_HW_PARAM_SAMPLE_BITS was decided as a fixed number, this value is also calculated according to it. Else, zero. But this behaviour depends on implementations in driver side.

rate_num Read-only. This value represents numerator of sampling rate in fraction notation. Basically, when a parameter of SNDRV_PCM_HW_PARAM_RATE was decided as a single value, this value is also calculated according to it. Else, zero. But this behaviour depends on implementations in driver side.

rate_den Read-only. This value represents denominator of sampling rate in fraction notation. Basically, when a parameter of SNDRV_PCM_HW_PARAM_RATE was decided as a single value, this value is also calculated according to it. Else, zero. But this behaviour depends on implementations in driver side.

fifo_size Read-only. This value represents the size of FIFO in serial sound interface of hardware. Basically, each driver can assigns a proper value to this parameter but some drivers intentionally set zero with a care of hardware design or data transmission protocol.

ALSA PCM core handles buffer of `struct snd_pcm_hw_params` when applications execute ioctl(2) with SNDRV_PCM_HW_REFINE or SNDRV_PCM_HW_PARAMS. Parameters in the buffer are changed according to `struct snd_pcm_hardware` and rules of constraints in the runtime. The structure describes capabilities of handled hardware. The rules describes dependencies on which a parameter is decided according to several parameters. A rule has a callback function, and drivers can register arbitrary functions to compute the target parameter. ALSA PCM core registers some rules to the runtime as a default.

Each driver can join in the interaction as long as it prepared for two stuffs in a callback of `struct snd_pcm_ops.open`.

1. In the callback, drivers are expected to change a member of `struct snd_pcm_hardware` type in the runtime, according to capacities of corresponding hardware.
2. In the same callback, drivers are also expected to register additional rules of constraints into the runtime when several parameters have dependencies due to hardware design.

The driver can refers to result of the interaction in a callback of `struct snd_pcm_ops.hw_params`, however it should not change the content.

Tracepoints in this category are designed to trace changes of the mask/interval parameters. When ALSA PCM core changes them, `hw_mask_param` or `hw_interval_param` event is probed according to type of the changed parameter.

ALSA PCM core also has a pretty print format for each of the tracepoints. Below is an example for `hw_mask_param`.

```
hw_mask_param: pcmC0D0p 001/023 FORMAT 00000000000000000000001000000044 00000000000000000000001000000044
```

Below is an example for `hw_interval_param`.

```
hw_interval_param: pcmC0D0p 000/023 BUFFER_SIZE 0 0 [0 4294967295] 0 1 [0 4294967295]
```

The first three fields are common. They represent name of ALSA PCM character device, rules of constraint and name of the changed parameter, in order. The field for rules of constraint consists of two sub-fields; index of applied rule and total number of rules added to the runtime. As an exception, the index 000 means that the parameter is changed by ALSA PCM core, regardless of the rules.

The rest of field represent state of the parameter before/after changing. These fields are different according to type of the parameter. For parameters of mask type, the fields represent hexadecimal dump of content of the parameter. For parameters of interval type, the fields represent values of each member of `empty`, `integer`, `openmin`, `min`, `max`, `openmax` in `struct snd_interval` in this order.

Tracepoints in drivers

Some drivers have tracepoints for developers' convenience. For them, please refer to each documentation or implementation.

Proc Files of ALSA Drivers

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General

ALSA has its own proc tree, /proc/asound. Many useful information are found in this tree. When you encounter a problem and need debugging, check the files listed in the following sections.

Each card has its subtree cardX, where X is from 0 to 7. The card-specific files are stored in the `card*` subdirectories.

Global Information

cards Shows the list of currently configured ALSA drivers, index, the id string, short and long descriptions.

version Shows the version string and compile date.

modules Lists the module of each card

devices Lists the ALSA native device mappings.

meminfo Shows the status of allocated pages via ALSA drivers. Appears only when `CONFIG_SND_DEBUG=y`.

hwdep Lists the currently available hwdep devices in format of `<card>-<device>: <name>`

pcm Lists the currently available PCM devices in format of `<card>-<device>: <id>: <name> : <sub-streams>`

timer Lists the currently available timer devices

oss/devices Lists the OSS device mappings.

oss/sndstat Provides the output compatible with /dev/sndstat. You can symlink this to /dev/sndstat.

Card Specific Files

The card-specific files are found in `/proc/asound/card*` directories. Some drivers (e.g. cmipci) have their own proc entries for the register dump, etc (e.g. `/proc/asound/card*/cmipci` shows the register dump). These files would be really helpful for debugging.

When PCM devices are available on this card, you can see directories like pcm0p or pcm1c. They hold the PCM information for each PCM stream. The number after `pcm` is the PCM device number from 0, and the last p or c means playback or capture direction. The files in this subtree is described later.

The status of MIDI I/O is found in `midi*` files. It shows the device name and the received/transmitted bytes through the MIDI device.

When the card is equipped with AC97 codecs, there are codec97#* subdirectories (described later).

When the OSS mixer emulation is enabled (and the module is loaded), oss_mixer file appears here, too. This shows the current mapping of OSS mixer elements to the ALSA control elements. You can change the mapping by writing to this device. Read OSS-Emulation.txt for details.

PCM Proc Files

card*/pcm*/info The general information of this PCM device: card #, device #, substreams, etc.

card*/pcm*/xrun_debug This file appears when `CONFIG_SND_DEBUG=y` and `CONFIG_PCM_XRUN_DEBUG=y`. This shows the status of xrun (= buffer overrun/xrun) and invalid PCM position debug/check of ALSA PCM middle layer. It takes an integer value, can be changed by writing to this file, such as:

```
# echo 5 > /proc/asound/card0/pcm0p/xrun_debug
```

The value consists of the following bit flags:

- bit 0 = Enable XRUN/jiffies debug messages
- bit 1 = Show stack trace at XRUN / jiffies check
- bit 2 = Enable additional jiffies check

When the bit 0 is set, the driver will show the messages to kernel log when an xrun is detected. The debug message is shown also when the invalid H/W pointer is detected at the update of periods (usually called from the interrupt handler).

When the bit 1 is set, the driver will show the stack trace additionally. This may help the debugging.

Since 2.6.30, this option can enable the hwptr check using jiffies. This detects spontaneous invalid pointer callback values, but can be lead to too much corrections for a (mostly buggy) hardware that doesn't give smooth pointer updates. This feature is enabled via the bit 2.

card*/pcm*/sub*/info The general information of this PCM sub-stream.

card*/pcm*/sub*/status The current status of this PCM sub-stream, elapsed time, H/W position, etc.

card*/pcm*/sub*/hw_params The hardware parameters set for this sub-stream.

card*/pcm*/sub*/sw_params The soft parameters set for this sub-stream.

card*/pcm*/sub*/prealloc The buffer pre-allocation information.

card*/pcm*/sub*/xrun_injection Triggers an XRUN to the running stream when any value is written to this proc file. Used for fault injection. This entry is write-only.

AC97 Codec Information

card*/codec97#*/ac97#?-? Shows the general information of this AC97 codec chip, such as name, capabilities, set up.

card*/codec97#0/ac97#?-?+regs Shows the AC97 register dump. Useful for debugging.

When CONFIG_SND_DEBUG is enabled, you can write to this file for changing an AC97 register directly. Pass two hex numbers. For example,

```
# echo 02 9f1f > /proc/asound/card0/codec97#0/ac97#0-0+regs
```

USB Audio Streams

card*/stream* Shows the assignment and the current status of each audio stream of the given card. This information is very useful for debugging.

HD-Audio Codecs

card*/codec#* Shows the general codec information and the attribute of each widget node.

card*/eld#* Available for HDMI or DisplayPort interfaces. Shows ELD(EDID Like Data) info retrieved from the attached HDMI sink, and describes its audio capabilities and configurations.

Some ELD fields may be modified by doing `echo name hex_value > eld#*`. Only do this if you are sure the HDMI sink provided value is wrong. And if that makes your HDMI audio work, please report to us so that we can fix it in future kernel releases.

Sequencer Information

seq/drivers Lists the currently available ALSA sequencer drivers.

seq/clients Shows the list of currently available sequencer clients and ports. The connection status and the running status are shown in this file, too.

seq/queues Lists the currently allocated/running sequencer queues.

seq/timer Lists the currently allocated/running sequencer timers.

seq/oss Lists the OSS-compatible sequencer stuffs.

Help For Debugging?

When the problem is related with PCM, first try to turn on xrun_debug mode. This will give you the kernel messages when and where xrun happened.

If it's really a bug, report it with the following information:

- the name of the driver/card, show in `/proc/asound/cards`
- the register dump, if available (e.g. `card*/cmipci`)

when it's a PCM problem,

- set-up of PCM, shown in hw_parms, sw_params, and status in the PCM sub-stream directory

when it's a mixer problem,

- AC97 proc files, `codec97#*/*` files

for USB audio/midi,

- output of `lsusb -v`
- `stream*` files in card directory

The ALSA bug-tracking system is found at: https://bugtrack.alsa-project.org/alsa-bug/

Notes on Power-Saving Mode

AC97 and HD-audio drivers have the automatic power-saving mode. This feature is enabled via Kconfig CONFIG_SND_AC97_POWER_SAVE and CONFIG_SND_HDA_POWER_SAVE options, respectively.

With the automatic power-saving, the driver turns off the codec power appropriately when no operation is required. When no applications use the device and/or no analog loopback is set, the power disablement is done fully or partially. It'll save a certain power consumption, thus good for laptops (even for desktops).

The time-out for automatic power-off can be specified via power_save module option of snd-ac97-codec and snd-hda-intel modules. Specify the time-out value in seconds. 0 means to disable the automatic power-saving. The default value of timeout is given via CONFIG_SND_AC97_POWER_SAVE_DEFAULT and CONFIG_SND_HDA_POWER_SAVE_DEFAULT Kconfig options. Setting this to 1 (the minimum value) isn't recommended because many applications try to reopen the device frequently. 10 would be a good choice for normal operations.

The power_save option is exported as writable. This means you can adjust the value via sysfs on the fly. For example, to turn on the automatic power-save mode with 10 seconds, write to /sys/modules/snd_ac97_codec/parameters/power_save (usually as root):

```
# echo 10 > /sys/modules/snd_ac97_codec/parameters/power_save
```

Note that you might hear click noise/pop when changing the power state. Also, it often takes certain time to wake up from the power-down to the active state. These are often hardly to fix, so don't report extra bug reports unless you have a fix patch ;-)

For HD-audio interface, there is another module option, power_save_controller. This enables/disables the power-save mode of the controller side. Setting this on may reduce a bit more power consumption, but might result in longer wake-up time and click noise. Try to turn it off when you experience such a thing too often.

Notes on Kernel OSS-Emulation

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Modules

ALSA provides a powerful OSS emulation on the kernel. The OSS emulation for PCM, mixer and sequencer devices is implemented as add-on kernel modules, snd-pcm-oss, snd-mixer-oss and snd-seq-oss. When you need to access the OSS PCM, mixer or sequencer devices, the corresponding module has to be loaded.

These modules are loaded automatically when the corresponding service is called. The alias is defined sound-service-x-y, where x and y are the card number and the minor unit number. Usually you don't have to define these aliases by yourself.

Only necessary step for auto-loading of OSS modules is to define the card alias in /etc/modprobe.d/alsa.conf, such as:

```
alias sound-slot-0 snd-emu10k1
```

As the second card, define sound-slot-1 as well. Note that you can't use the aliased name as the target name (i.e. alias sound-slot-0 snd-card-0 doesn't work any more like the old modutils).

The currently available OSS configuration is shown in /proc/asound/oss/sndstat. This shows in the same syntax of /dev/sndstat, which is available on the commercial OSS driver. On ALSA, you can symlink /dev/sndstat to this proc file.

Please note that the devices listed in this proc file appear only after the corresponding OSS-emulation module is loaded. Don't worry even if "NOT ENABLED IN CONFIG" is shown in it.

Device Mapping

ALSA supports the following OSS device files:

```
PCM:
        /dev/dspX
        /dev/adspX

Mixer:
        /dev/mixerX

MIDI:
        /dev/midi0X
        /dev/amidi0X

Sequencer:
        /dev/sequencer
        /dev/sequencer2 (aka /dev/music)
```

where X is the card number from 0 to 7.

(NOTE: Some distributions have the device files like /dev/midi0 and /dev/midi1. They are NOT for OSS but for tclmidi, which is a totally different thing.)

Unlike the real OSS, ALSA cannot use the device files more than the assigned ones. For example, the first card cannot use /dev/dsp1 or /dev/dsp2, but only /dev/dsp0 and /dev/adsp0.

As seen above, PCM and MIDI may have two devices. Usually, the first PCM device (`hw:0,0` in ALSA) is mapped to /dev/dsp and the secondary device (`hw:0,1`) to /dev/adsp (if available). For MIDI, /dev/midi and /dev/amidi, respectively.

You can change this device mapping via the module options of snd-pcm-oss and snd-rawmidi. In the case of PCM, the following options are available for snd-pcm-oss:

dsp_map PCM device number assigned to /dev/dspX (default = 0)

adsp_map PCM device number assigned to /dev/adspX (default = 1)

For example, to map the third PCM device (`hw:0,2`) to /dev/adsp0, define like this:

```
options snd-pcm-oss adsp_map=2
```

The options take arrays. For configuring the second card, specify two entries separated by comma. For example, to map the third PCM device on the second card to /dev/adsp1, define like below:

```
options snd-pcm-oss adsp_map=0,2
```

To change the mapping of MIDI devices, the following options are available for snd-rawmidi:

midi_map MIDI device number assigned to /dev/midi0X (default = 0)

amidi_map MIDI device number assigned to /dev/amidi0X (default = 1)

For example, to assign the third MIDI device on the first card to /dev/midi00, define as follows:

```
options snd-rawmidi midi_map=2
```

PCM Mode

As default, ALSA emulates the OSS PCM with so-called plugin layer, i.e. tries to convert the sample format, rate or channels automatically when the card doesn't support it natively. This will lead to some problems for some applications like quake or wine, especially if they use the card only in the MMAP mode.

In such a case, you can change the behavior of PCM per application by writing a command to the proc file. There is a proc file for each PCM stream, `/proc/asound/cardX/pcmY[cp]/oss`, where X is the card

number (zero-based), Y the PCM device number (zero-based), and p is for playback and c for capture, respectively. Note that this proc file exists only after snd-pcm-oss module is loaded.

The command sequence has the following syntax:

```
app_name fragments fragment_size [options]
```

app_name is the name of application with (higher priority) or without path. `fragments` specifies the number of fragments or zero if no specific number is given. `fragment_size` is the size of fragment in bytes or zero if not given. `options` is the optional parameters. The following options are available:

disable the application tries to open a pcm device for this channel but does not want to use it.

direct don't use plugins

block force block open mode

non-block force non-block open mode

partial-frag write also partial fragments (affects playback only)

no-silence do not fill silence ahead to avoid clicks

The `disable` option is useful when one stream direction (playback or capture) is not handled correctly by the application although the hardware itself does support both directions. The `direct` option is used, as mentioned above, to bypass the automatic conversion and useful for MMAP-applications. For example, to playback the first PCM device without plugins for quake, send a command via echo like the following:

```
% echo "quake 0 0 direct" > /proc/asound/card0/pcm0p/oss
```

While quake wants only playback, you may append the second command to notify driver that only this direction is about to be allocated:

```
% echo "quake 0 0 disable" > /proc/asound/card0/pcm0c/oss
```

The permission of proc files depend on the module options of snd. As default it's set as root, so you'll likely need to be superuser for sending the command above.

The block and non-block options are used to change the behavior of opening the device file.

As default, ALSA behaves as original OSS drivers, i.e. does not block the file when it's busy. The -EBUSY error is returned in this case.

This blocking behavior can be changed globally via nonblock_open module option of snd-pcm-oss. For using the blocking mode as default for OSS devices, define like the following:

```
options snd-pcm-oss nonblock_open=0
```

The `partial-frag` and `no-silence` commands have been added recently. Both commands are for optimization use only. The former command specifies to invoke the write transfer only when the whole fragment is filled. The latter stops writing the silence data ahead automatically. Both are disabled as default.

You can check the currently defined configuration by reading the proc file. The read image can be sent to the proc file again, hence you can save the current configuration

```
% cat /proc/asound/card0/pcm0p/oss > /somewhere/oss-cfg
```

and restore it like

```
% cat /somewhere/oss-cfg > /proc/asound/card0/pcm0p/oss
```

Also, for clearing all the current configuration, send `erase` command as below:

```
% echo "erase" > /proc/asound/card0/pcm0p/oss
```

Mixer Elements

Since ALSA has completely different mixer interface, the emulation of OSS mixer is relatively complicated. ALSA builds up a mixer element from several different ALSA (mixer) controls based on the name string. For example, the volume element SOUND_MIXER_PCM is composed from “PCM Playback Volume” and “PCM Playback Switch” controls for the playback direction and from “PCM Capture Volume” and “PCM Capture Switch” for the capture directory (if exists). When the PCM volume of OSS is changed, all the volume and switch controls above are adjusted automatically.

As default, ALSA uses the following control for OSS volumes:

OSS volume	ALSA control	Index
SOUND_MIXER_VOLUME	Master	0
SOUND_MIXER_BASS	Tone Control - Bass	0
SOUND_MIXER_TREBLE	Tone Control - Treble	0
SOUND_MIXER_SYNTH	Synth	0
SOUND_MIXER_PCM	PCM	0
SOUND_MIXER_SPEAKER	PC Speaker	0
SOUND_MIXER_LINE	Line	0
SOUND_MIXER_MIC	Mic	0
SOUND_MIXER_CD	CD	0
SOUND_MIXER_IMIX	Monitor Mix	0
SOUND_MIXER_ALTPCM	PCM	1
SOUND_MIXER_RECLEV	(not assigned)	
SOUND_MIXER_IGAIN	Capture	0
SOUND_MIXER_OGAIN	Playback	0
SOUND_MIXER_LINE1	Aux	0
SOUND_MIXER_LINE2	Aux	1
SOUND_MIXER_LINE3	Aux	2
SOUND_MIXER_DIGITAL1	Digital	0
SOUND_MIXER_DIGITAL2	Digital	1
SOUND_MIXER_DIGITAL3	Digital	2
SOUND_MIXER_PHONEIN	Phone	0
SOUND_MIXER_PHONEOUT	Phone	1
SOUND_MIXER_VIDEO	Video	0
SOUND_MIXER_RADIO	Radio	0
SOUND_MIXER_MONITOR	Monitor	0

The second column is the base-string of the corresponding ALSA control. In fact, the controls with XXX `[Playback|Capture] [Volume|Switch]` will be checked in addition.

The current assignment of these mixer elements is listed in the proc file, /proc/asound/cardX/oss_mixer, which will be like the following

```
VOLUME "Master" 0
BASS "" 0
TREBLE "" 0
SYNTH "" 0
PCM "PCM" 0
...
```

where the first column is the OSS volume element, the second column the base-string of the corresponding ALSA control, and the third the control index. When the string is empty, it means that the corresponding OSS control is not available.

For changing the assignment, you can write the configuration to this proc file. For example, to map “Wave Playback” to the PCM volume, send the command like the following:

```
% echo 'VOLUME "Wave Playback" 0' > /proc/asound/card0/oss_mixer
```

The command is exactly as same as listed in the proc file. You can change one or more elements, one

volume per line. In the last example, both “Wave Playback Volume” and “Wave Playback Switch” will be affected when PCM volume is changed.

Like the case of PCM proc file, the permission of proc files depend on the module options of snd. you’ll likely need to be superuser for sending the command above.

As well as in the case of PCM proc file, you can save and restore the current mixer configuration by reading and writing the whole file image.

Duplex Streams

Note that when attempting to use a single device file for playback and capture, the OSS API provides no way to set the format, sample rate or number of channels different in each direction. Thus

```
io_handle = open("device", O_RDWR)
```

will only function correctly if the values are the same in each direction.

To use different values in the two directions, use both

```
input_handle = open("device", O_RDONLY)
output_handle = open("device", O_WRONLY)
```

and set the values for the corresponding handle.

Unsupported Features

MMAP on ICE1712 driver

ICE1712 supports only the unconventional format, interleaved 10-channels 24bit (packed in 32bit) format. Therefore you cannot mmap the buffer as the conventional (mono or 2-channels, 8 or 16bit) format on OSS.

OSS Sequencer Emulation on ALSA

Copyright (c) 1998,1999 by Takashi Iwai

ver.0.1.8; Nov. 16, 1999

Description

This directory contains the OSS sequencer emulation driver on ALSA. Note that this program is still in the development state.

What this does - it provides the emulation of the OSS sequencer, access via `/dev/sequencer` and `/dev/music` devices. The most of applications using OSS can run if the appropriate ALSA sequencer is prepared.

The following features are emulated by this driver:

- Normal sequencer and MIDI events:

  They are converted to the ALSA sequencer events, and sent to the corresponding port.

- Timer events:

  The timer is not selectable by ioctl. The control rate is fixed to 100 regardless of HZ. That is, even on Alpha system, a tick is always 1/100 second. The base rate and tempo can be changed in `/dev/music`.

- Patch loading:

  It purely depends on the synth drivers whether it's supported since the patch loading is realized by callback to the synth driver.

- I/O controls:

  Most of controls are accepted. Some controls are dependent on the synth driver, as well as even on original OSS.

Furthermore, you can find the following advanced features:

- Better queue mechanism:

  The events are queued before processing them.

- Multiple applications:

  You can run two or more applications simultaneously (even for OSS sequencer)! However, each MIDI device is exclusive - that is, if a MIDI device is opened once by some application, other applications can't use it. No such a restriction in synth devices.

- Real-time event processing:

  The events can be processed in real time without using out of bound ioctl. To switch to real-time mode, send ABSTIME 0 event. The followed events will be processed in real-time without queued. To switch off the real-time mode, send RELTIME 0 event.

- `/proc` interface:

  The status of applications and devices can be shown via `/proc/asound/seq/oss` at any time. In the later version, configuration will be changed via `/proc` interface, too.

Installation

Run configure script with both sequencer support (`--with-sequencer=yes`) and OSS emulation (`--with-oss=yes`) options. A module `snd-seq-oss.o` will be created. If the synth module of your sound card supports for OSS emulation (so far, only Emu8000 driver), this module will be loaded automatically. Otherwise, you need to load this module manually.

At beginning, this module probes all the MIDI ports which have been already connected to the sequencer. Once after that, the creation and deletion of ports are watched by announcement mechanism of ALSA sequencer.

The available synth and MIDI devices can be found in proc interface. Run `cat /proc/asound/seq/oss`, and check the devices. For example, if you use an AWE64 card, you'll see like the following:

```
OSS sequencer emulation version 0.1.8
ALSA client number 63
ALSA receiver port 0

Number of applications: 0

Number of synth devices: 1
synth 0: [EMU8000]
  type 0x1 : subtype 0x20 : voices 32
  capabilties : ioctl enabled / load_patch enabled

Number of MIDI devices: 3
midi 0: [Emu8000 Port-0] ALSA port 65:0
  capability write / opened none

midi 1: [Emu8000 Port-1] ALSA port 65:1
  capability write / opened none
```

```
midi 2: [0: MPU-401 (UART)] ALSA port 64:0
  capability read/write / opened none
```

Note that the device number may be different from the information of /proc/asound/oss-devices or ones of the original OSS driver. Use the device number listed in /proc/asound/seq/oss to play via OSS sequencer emulation.

Using Synthesizer Devices

Run your favorite program. I've tested playmidi-2.4, awemidi-0.4.3, gmod-3.1 and xmp-1.1.5. You can load samples via /dev/sequencer like sfxload, too.

If the lowlevel driver supports multiple access to synth devices (like Emu8000 driver), two or more applications are allowed to run at the same time.

Using MIDI Devices

So far, only MIDI output was tested. MIDI input was not checked at all, but hopefully it will work. Use the device number listed in /proc/asound/seq/oss. Be aware that these numbers are mostly different from the list in /proc/asound/oss-devices.

Module Options

The following module options are available:

maxqlen specifies the maximum read/write queue length. This queue is private for OSS sequencer, so that it is independent from the queue length of ALSA sequencer. Default value is 1024.

seq_oss_debug specifies the debug level and accepts zero (= no debug message) or positive integer. Default value is 0.

Queue Mechanism

OSS sequencer emulation uses an ALSA priority queue. The events from /dev/sequencer are processed and put onto the queue specified by module option.

All the events from /dev/sequencer are parsed at beginning. The timing events are also parsed at this moment, so that the events may be processed in real-time. Sending an event ABSTIME 0 switches the operation mode to real-time mode, and sending an event RELTIME 0 switches it off. In the real-time mode, all events are dispatched immediately.

The queued events are dispatched to the corresponding ALSA sequencer ports after scheduled time by ALSA sequencer dispatcher.

If the write-queue is full, the application sleeps until a certain amount (as default one half) becomes empty in blocking mode. The synchronization to write timing was implemented, too.

The input from MIDI devices or echo-back events are stored on read FIFO queue. If application reads /dev/sequencer in blocking mode, the process will be awaked.

Interface to Synthesizer Device

Registration

To register an OSS synthesizer device, use snd_seq_oss_synth_register() function:

```
int snd_seq_oss_synth_register(char *name, int type, int subtype, int nvoices,
        snd_seq_oss_callback_t *oper, void *private_data)
```

The arguments name, type, subtype and nvoices are used for making the appropriate synth_info structure for ioctl. The return value is an index number of this device. This index must be remembered for unregister. If registration is failed, -errno will be returned.

To release this device, call snd_seq_oss_synth_unregister() function:

```
int snd_seq_oss_synth_unregister(int index)
```

where the index is the index number returned by register function.

Callbacks

OSS synthesizer devices have capability for sample downloading and ioctls like sample reset. In OSS emulation, these special features are realized by using callbacks. The registration argument oper is used to specify these callbacks. The following callback functions must be defined:

```
snd_seq_oss_callback_t:
 int (*open)(snd_seq_oss_arg_t *p, void *closure);
 int (*close)(snd_seq_oss_arg_t *p);
 int (*ioctl)(snd_seq_oss_arg_t *p, unsigned int cmd, unsigned long arg);
 int (*load_patch)(snd_seq_oss_arg_t *p, int format, const char *buf, int offs, int count);
 int (*reset)(snd_seq_oss_arg_t *p);
```

Except for open and close callbacks, they are allowed to be NULL.

Each callback function takes the argument type snd_seq_oss_arg_t as the first argument.

```
struct snd_seq_oss_arg_t {
    int app_index;
    int file_mode;
    int seq_mode;
    snd_seq_addr_t addr;
    void *private_data;
    int event_passing;
};
```

The first three fields, app_index, file_mode and seq_mode are initialized by OSS sequencer. The app_index is the application index which is unique to each application opening OSS sequencer. The file_mode is bit-flags indicating the file operation mode. See seq_oss.h for its meaning. The seq_mode is sequencer operation mode. In the current version, only SND_OSSSEQ_MODE_SYNTH is used.

The next two fields, addr and private_data, must be filled by the synth driver at open callback. The addr contains the address of ALSA sequencer port which is assigned to this device. If the driver allocates memory for private_data, it must be released in close callback by itself.

The last field, event_passing, indicates how to translate note-on / off events. In PROCESS_EVENTS mode, the note 255 is regarded as velocity change, and key pressure event is passed to the port. In PASS_EVENTS mode, all note on/off events are passed to the port without modified. PROCESS_KEYPRESS mode checks the note above 128 and regards it as key pressure event (mainly for Emu8000 driver).

Open Callback

The open is called at each time this device is opened by an application using OSS sequencer. This must not be NULL. Typically, the open callback does the following procedure:

1. Allocate private data record.
2. Create an ALSA sequencer port.

3. Set the new port address on `arg->addr`.
4. Set the private data record pointer on `arg->private_data`.

Note that the type bit-flags in port_info of this synth port must NOT contain `TYPE_MIDI_GENERIC` bit. Instead, `TYPE_SPECIFIC` should be used. Also, `CAP_SUBSCRIPTION` bit should NOT be included, too. This is necessary to tell it from other normal MIDI devices. If the open procedure succeeded, return zero. Otherwise, return -errno.

Ioctl Callback

The `ioctl` callback is called when the sequencer receives device-specific ioctls. The following two ioctls should be processed by this callback:

IOCTL_SEQ_RESET_SAMPLES reset all samples on memory – return 0

IOCTL_SYNTH_MEMAVL return the available memory size

FM_4OP_ENABLE can be ignored usually

The other ioctls are processed inside the sequencer without passing to the lowlevel driver.

Load_Patch Callback

The `load_patch` callback is used for sample-downloading. This callback must read the data on user-space and transfer to each device. Return 0 if succeeded, and -errno if failed. The format argument is the patch key in patch_info record. The buf is user-space pointer where patch_info record is stored. The offs can be ignored. The count is total data size of this sample data.

Close Callback

The `close` callback is called when this device is closed by the application. If any private data was allocated in open callback, it must be released in the close callback. The deletion of ALSA port should be done here, too. This callback must not be NULL.

Reset Callback

The `reset` callback is called when sequencer device is reset or closed by applications. The callback should turn off the sounds on the relevant port immediately, and initialize the status of the port. If this callback is undefined, OSS seq sends a `HEARTBEAT` event to the port.

Events

Most of the events are processed by sequencer and translated to the adequate ALSA sequencer events, so that each synth device can receive by input_event callback of ALSA sequencer port. The following ALSA events should be implemented by the driver:

ALSA event	Original OSS events
NOTEON	SEQ_NOTEON, MIDI_NOTEON
NOTE	SEQ_NOTEOFF, MIDI_NOTEOFF
KEYPRESS	MIDI_KEY_PRESSURE
CHANPRESS	SEQ_AFTERTOUCH, MIDI_CHN_PRESSURE
PGMCHANGE	SEQ_PGMCHANGE, MIDI_PGM_CHANGE
PITCHBEND	SEQ_CONTROLLER(CTRL_PITCH_BENDER), MIDI_PITCH_BEND
CONTROLLER	MIDI_CTL_CHANGE, SEQ_BALANCE (with CTL_PAN)
CONTROL14	SEQ_CONTROLLER
REGPARAM	SEQ_CONTROLLER(CTRL_PITCH_BENDER_RANGE)
SYSEX	SEQ_SYSEX

The most of these behavior can be realized by MIDI emulation driver included in the Emu8000 lowlevel driver. In the future release, this module will be independent.

Some OSS events (SEQ_PRIVATE and SEQ_VOLUME events) are passed as event type SND_SEQ_OSS_PRIVATE. The OSS sequencer passes these event 8 byte packets without any modification. The lowlevel driver should process these events appropriately.

Interface to MIDI Device

Since the OSS emulation probes the creation and deletion of ALSA MIDI sequencer ports automatically by receiving announcement from ALSA sequencer, the MIDI devices don't need to be registered explicitly like synth devices. However, the MIDI port_info registered to ALSA sequencer must include a group name SND_SEQ_GROUP_DEVICE and a capability-bit CAP_READ or CAP_WRITE. Also, subscription capabilities, CAP_SUBS_READ or CAP_SUBS_WRITE, must be defined, too. If these conditions are not satisfied, the port is not registered as OSS sequencer MIDI device.

The events via MIDI devices are parsed in OSS sequencer and converted to the corresponding ALSA sequencer events. The input from MIDI sequencer is also converted to MIDI byte events by OSS sequencer. This works just a reverse way of seq_midi module.

Known Problems / TODO's

- Patch loading via ALSA instrument layer is not implemented yet.

CHAPTER

THREE

ALSA SOC LAYER

The documentation is spilt into the following sections:-

ALSA SoC Layer Overview

The overall project goal of the ALSA System on Chip (ASoC) layer is to provide better ALSA support for embedded system-on-chip processors (e.g. pxa2xx, au1x00, iMX, etc) and portable audio codecs. Prior to the ASoC subsystem there was some support in the kernel for SoC audio, however it had some limitations:-

- Codec drivers were often tightly coupled to the underlying SoC CPU. This is not ideal and leads to code duplication - for example, Linux had different wm8731 drivers for 4 different SoC platforms.
- There was no standard method to signal user initiated audio events (e.g. Headphone/Mic insertion, Headphone/Mic detection after an insertion event). These are quite common events on portable devices and often require machine specific code to re-route audio, enable amps, etc., after such an event.
- Drivers tended 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 was also no support for saving power via changing codec oversampling rates, bias currents, etc.

ASoC 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 its audio interface capabilities with the core and are subsequently matched and configured when the application hardware parameters are known.
- Dynamic Audio Power Management (DAPM). DAPM automatically sets the codec to its 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.
- Machine specific controls: Allow machines to add controls to the sound card (e.g. volume control for speaker amplifier).

To achieve all this, ASoC basically splits an embedded audio system into multiple re-usable component drivers :-

- Codec class drivers: The codec class driver is platform independent and contains audio controls, audio interface capabilities, codec DAPM definition and codec IO functions. This class extends to BT, FM and MODEM ICs if required. Codec class drivers should be generic code that can run on any architecture and machine.

- Platform class drivers: The platform class driver includes the audio DMA engine driver, digital audio interface (DAI) drivers (e.g. I2S, AC97, PCM) and any audio DSP drivers for that platform.
- Machine class driver: The machine driver class acts as the glue that describes and binds the other component drivers together to form an ALSA "sound card device". It handles any machine specific controls and machine level audio events (e.g. turning on an amp at start of playback).

ASoC Codec Class Driver

The codec class driver is generic and hardware independent code that configures the codec, FM, MODEM, BT or external DSP to provide audio capture and playback. It should contain no code that is specific to the target platform or machine. All platform and machine specific code should be added to the platform and machine drivers respectively.

Each codec class driver *must* provide the following features:-

1. Codec DAI and PCM configuration
2. Codec control IO - using RegMap API
3. Mixers and audio controls
4. Codec audio operations
5. DAPM description.
6. DAPM event handler.

Optionally, codec drivers can also provide:-

7. DAC Digital mute control.

Its probably best to use this guide in conjunction with the existing codec driver code in sound/soc/codecs/

ASoC Codec driver breakdown

Codec DAI and PCM configuration

Each codec driver must have a struct snd_soc_dai_driver to define its DAI and PCM capabilities and operations. This struct is exported so that it can be registered with the core by your machine driver.

e.g.

```
static struct snd_soc_dai_ops wm8731_dai_ops = {
      .prepare        = wm8731_pcm_prepare,
      .hw_params      = wm8731_hw_params,
      .shutdown       = wm8731_shutdown,
      .digital_mute   = wm8731_mute,
      .set_sysclk     = wm8731_set_dai_sysclk,
      .set_fmt        = wm8731_set_dai_fmt,
};

struct snd_soc_dai_driver wm8731_dai = {
      .name = "wm8731-hifi",
      .playback = {
               .stream_name = "Playback",
               .channels_min = 1,
               .channels_max = 2,
               .rates = WM8731_RATES,
               .formats = WM8731_FORMATS,},
      .capture = {
               .stream_name = "Capture",
               .channels_min = 1,
```

```
                .channels_max = 2,
                .rates = WM8731_RATES,
                .formats = WM8731_FORMATS,},
        .ops = &wm8731_dai_ops,
        .symmetric_rates = 1,
};
```

Codec control IO

The codec can usually be controlled via an I2C or SPI style interface (AC97 combines control with data in the DAI). The codec driver should use the Regmap API for all codec IO. Please see include/linux/regmap.h and existing codec drivers for example regmap usage.

Mixers and audio controls

All the codec mixers and audio controls can be defined using the convenience macros defined in soc.h.

```
#define SOC_SINGLE(xname, reg, shift, mask, invert)
```

Defines a single control as follows:-

```
xname = Control name e.g. "Playback Volume"
reg = codec register
shift = control bit(s) offset in register
mask = control bit size(s) e.g. mask of 7 = 3 bits
invert = the control is inverted
```

Other macros include:-

```
#define SOC_DOUBLE(xname, reg, shift_left, shift_right, mask, invert)
```

A stereo control

```
#define SOC_DOUBLE_R(xname, reg_left, reg_right, shift, mask, invert)
```

A stereo control spanning 2 registers

```
#define SOC_ENUM_SINGLE(xreg, xshift, xmask, xtexts)
```

Defines an single enumerated control as follows:-

```
xreg = register
xshift = control bit(s) offset in register
xmask = control bit(s) size
xtexts = pointer to array of strings that describe each setting

#define SOC_ENUM_DOUBLE(xreg, xshift_l, xshift_r, xmask, xtexts)
```

Defines a stereo enumerated control

Codec Audio Operations

The codec driver also supports the following ALSA PCM operations:-

```
/* SoC audio ops */
struct snd_soc_ops {
      int (*startup)(struct snd_pcm_substream *);
      void (*shutdown)(struct snd_pcm_substream *);
      int (*hw_params)(struct snd_pcm_substream *, struct snd_pcm_hw_params *);
      int (*hw_free)(struct snd_pcm_substream *);
```

```
      int (*prepare)(struct snd_pcm_substream *);
};
```

Please refer to the ALSA driver PCM documentation for details. http://www.alsa-project.org/~iwai/writing-an-alsa-driver/

DAPM description

The Dynamic Audio Power Management description describes the codec power components and their relationships and registers to the ASoC core. Please read dapm.rst for details of building the description.

Please also see the examples in other codec drivers.

DAPM event handler

This function is a callback that handles codec domain PM calls and system domain PM calls (e.g. suspend and resume). It is used to put the codec to sleep when not in use.

Power states:-

```
SNDRV_CTL_POWER_D0: /* full On */
/* vref/mid, clk and osc on, active */

SNDRV_CTL_POWER_D1: /* partial On */
SNDRV_CTL_POWER_D2: /* partial On */

SNDRV_CTL_POWER_D3hot: /* Off, with power */
/* everything off except vref/vmid, inactive */

SNDRV_CTL_POWER_D3cold: /* Everything Off, without power */
```

Codec DAC digital mute control

Most codecs have a digital mute before the DACs that can be used to minimise any system noise. The mute stops any digital data from entering the DAC.

A callback can be created that is called by the core for each codec DAI when the mute is applied or freed.

i.e.

```
static int wm8974_mute(struct snd_soc_dai *dai, int mute)
{
      struct snd_soc_component *component = dai->component;
      u16 mute_reg = snd_soc_component_read32(component, WM8974_DAC) & 0xffbf;

      if (mute)
              snd_soc_component_write(component, WM8974_DAC, mute_reg | 0x40);
      else
              snd_soc_component_write(component, WM8974_DAC, mute_reg);
      return 0;
}
```

ASoC Digital Audio Interface (DAI)

ASoC currently supports the three main Digital Audio Interfaces (DAI) found on SoC controllers and portable audio CODECs today, namely AC97, I2S and PCM.

AC97

AC97 is a five wire interface commonly found on many PC sound cards. It is now also popular in many portable devices. This DAI has a reset line and time multiplexes its data on its SDATA_OUT (playback) and SDATA_IN (capture) lines. The bit clock (BCLK) is always driven by the CODEC (usually 12.288MHz) and the frame (FRAME) (usually 48kHz) is always driven by the controller. Each AC97 frame is 21uS long and is divided into 13 time slots.

The AC97 specification can be found at : http://www.intel.com/p/en_US/business/design

I2S

I2S is a common 4 wire DAI used in HiFi, STB and portable devices. The Tx and Rx lines are used for audio transmission, whilst the bit clock (BCLK) and left/right clock (LRC) synchronise the link. I2S is flexible in that either the controller or CODEC can drive (master) the BCLK and LRC clock lines. Bit clock usually varies depending on the sample rate and the master system clock (SYSCLK). LRCLK is the same as the sample rate. A few devices support separate ADC and DAC LRCLKs, this allows for simultaneous capture and playback at different sample rates.

I2S has several different operating modes:-

I2S MSB is transmitted on the falling edge of the first BCLK after LRC transition.

Left Justified MSB is transmitted on transition of LRC.

Right Justified MSB is transmitted sample size BCLKs before LRC transition.

PCM

PCM is another 4 wire interface, very similar to I2S, which can support a more flexible protocol. It has bit clock (BCLK) and sync (SYNC) lines that are used to synchronise the link whilst the Tx and Rx lines are used to transmit and receive the audio data. Bit clock usually varies depending on sample rate whilst sync runs at the sample rate. PCM also supports Time Division Multiplexing (TDM) in that several devices can use the bus simultaneously (this is sometimes referred to as network mode).

Common PCM operating modes:-

Mode A MSB is transmitted on falling edge of first BCLK after FRAME/SYNC.

Mode B MSB is transmitted on rising edge of FRAME/SYNC.

Dynamic Audio Power Management for Portable Devices

Description

Dynamic Audio Power Management (DAPM) is designed to allow portable Linux devices to use the minimum amount of power within the audio subsystem at all times. It is independent of other kernel PM and as such, can easily co-exist with the other PM systems.

DAPM is also completely transparent to all user space applications as all power switching is done within the ASoC core. No code changes or recompiling are required for user space applications. DAPM makes power switching decisions based upon any audio stream (capture/playback) activity and audio mixer settings within the device.

DAPM spans the whole machine. It covers power control within the entire audio subsystem, this includes internal codec power blocks and machine level power systems.

There are 4 power domains within DAPM

Codec bias domain VREF, VMID (core codec and audio power)

Usually controlled at codec probe/remove and suspend/resume, although can be set at stream time if power is not needed for sidetone, etc.

Platform/Machine domain physically connected inputs and outputs

Is platform/machine and user action specific, is configured by the machine driver and responds to asynchronous events e.g when HP are inserted

Path domain audio subsystem signal paths

Automatically set when mixer and mux settings are changed by the user. e.g. alsamixer, amixer.

Stream domain DACs and ADCs.

Enabled and disabled when stream playback/capture is started and stopped respectively. e.g. aplay, arecord.

All DAPM power switching decisions are made automatically by consulting an audio routing map of the whole machine. This map is specific to each machine and consists of the interconnections between every audio component (including internal codec components). All audio components that effect power are called widgets hereafter.

DAPM Widgets

Audio DAPM widgets fall into a number of types:-

Mixer Mixes several analog signals into a single analog signal.

Mux An analog switch that outputs only one of many inputs.

PGA A programmable gain amplifier or attenuation widget.

ADC Analog to Digital Converter

DAC Digital to Analog Converter

Switch An analog switch

Input A codec input pin

Output A codec output pin

Headphone Headphone (and optional Jack)

Mic Mic (and optional Jack)

Line Line Input/Output (and optional Jack)

Speaker Speaker

Supply Power or clock supply widget used by other widgets.

Regulator External regulator that supplies power to audio components.

Clock External clock that supplies clock to audio components.

AIF IN Audio Interface Input (with TDM slot mask).

AIF OUT Audio Interface Output (with TDM slot mask).

Siggen Signal Generator.

DAI IN Digital Audio Interface Input.

DAI OUT Digital Audio Interface Output.

DAI Link DAI Link between two DAI structures

Pre Special PRE widget (exec before all others)

Post Special POST widget (exec after all others)

Buffer Inter widget audio data buffer within a DSP.

Scheduler DSP internal scheduler that schedules component/pipeline processing work.

Effect Widget that performs an audio processing effect.

SRC Sample Rate Converter within DSP or CODEC

ASRC Asynchronous Sample Rate Converter within DSP or CODEC

Encoder Widget that encodes audio data from one format (usually PCM) to another usually more compressed format.

Decoder Widget that decodes audio data from a compressed format to an uncompressed format like PCM.

(Widgets are defined in include/sound/soc-dapm.h)

Widgets can be added to the sound card by any of the component driver types. There are convenience macros defined in soc-dapm.h that can be used to quickly build a list of widgets of the codecs and machines DAPM widgets.

Most widgets have a name, register, shift and invert. Some widgets have extra parameters for stream name and kcontrols.

Stream Domain Widgets

Stream Widgets relate to the stream power domain and only consist of ADCs (analog to digital converters), DACs (digital to analog converters), AIF IN and AIF OUT.

Stream widgets have the following format:-

```
SND_SOC_DAPM_DAC(name, stream name, reg, shift, invert),
SND_SOC_DAPM_AIF_IN(name, stream, slot, reg, shift, invert)
```

NOTE: the stream name must match the corresponding stream name in your codec snd_soc_codec_dai.

e.g. stream widgets for HiFi playback and capture

```
SND_SOC_DAPM_DAC("HiFi DAC", "HiFi Playback", REG, 3, 1),
SND_SOC_DAPM_ADC("HiFi ADC", "HiFi Capture", REG, 2, 1),
```

e.g. stream widgets for AIF

```
SND_SOC_DAPM_AIF_IN("AIF1RX", "AIF1 Playback", 0, SND_SOC_NOPM, 0, 0),
SND_SOC_DAPM_AIF_OUT("AIF1TX", "AIF1 Capture", 0, SND_SOC_NOPM, 0, 0),
```

Path Domain Widgets

Path domain widgets have a ability to control or affect the audio signal or audio paths within the audio subsystem. They have the following form:-

```
SND_SOC_DAPM_PGA(name, reg, shift, invert, controls, num_controls)
```

Any widget kcontrols can be set using the controls and num_controls members.

e.g. Mixer widget (the kcontrols are declared first)

```
/* Output Mixer */
static const snd_kcontrol_new_t wm8731_output_mixer_controls[] = {
SOC_DAPM_SINGLE("Line Bypass Switch", WM8731_APANA, 3, 1, 0),
SOC_DAPM_SINGLE("Mic Sidetone Switch", WM8731_APANA, 5, 1, 0),
SOC_DAPM_SINGLE("HiFi Playback Switch", WM8731_APANA, 4, 1, 0),
};
```

```
SND_SOC_DAPM_MIXER("Output Mixer", WM8731_PWR, 4, 1, wm8731_output_mixer_controls,
      ARRAY_SIZE(wm8731_output_mixer_controls)),
```

If you don't want the mixer elements prefixed with the name of the mixer widget, you can use SND_SOC_DAPM_MIXER_NAMED_CTL instead. the parameters are the same as for SND_SOC_DAPM_MIXER.

Machine domain Widgets

Machine widgets are different from codec widgets in that they don't have a codec register bit associated with them. A machine widget is assigned to each machine audio component (non codec or DSP) that can be independently powered. e.g.

- Speaker Amp
- Microphone Bias
- Jack connectors

A machine widget can have an optional call back.

e.g. Jack connector widget for an external Mic that enables Mic Bias when the Mic is inserted:-:

```
static int spitz_mic_bias(struct snd_soc_dapm_widget* w, int event)
{
      gpio_set_value(SPITZ_GPIO_MIC_BIAS, SND_SOC_DAPM_EVENT_ON(event));
      return 0;
}

SND_SOC_DAPM_MIC("Mic Jack", spitz_mic_bias),
```

Codec (BIAS) Domain

The codec bias power domain has no widgets and is handled by the codecs DAPM event handler. This handler is called when the codec powerstate is changed wrt to any stream event or by kernel PM events.

Virtual Widgets

Sometimes widgets exist in the codec or machine audio map that don't have any corresponding soft power control. In this case it is necessary to create a virtual widget - a widget with no control bits e.g.

```
SND_SOC_DAPM_MIXER("AC97 Mixer", SND_SOC_DAPM_NOPM, 0, 0, NULL, 0),
```

This can be used to merge to signal paths together in software.

After all the widgets have been defined, they can then be added to the DAPM subsystem individually with a call to snd_soc_dapm_new_control().

Codec/DSP Widget Interconnections

Widgets are connected to each other within the codec, platform and machine by audio paths (called interconnections). Each interconnection must be defined in order to create a map of all audio paths between widgets.

This is easiest with a diagram of the codec or DSP (and schematic of the machine audio system), as it requires joining widgets together via their audio signal paths.

e.g., from the WM8731 output mixer (wm8731.c)

The WM8731 output mixer has 3 inputs (sources)

1. Line Bypass Input
2. DAC (HiFi playback)
3. Mic Sidetone Input

Each input in this example has a kcontrol associated with it (defined in example above) and is connected to the output mixer via its kcontrol name. We can now connect the destination widget (wrt audio signal) with its source widgets.

```
/* output mixer */
{"Output Mixer", "Line Bypass Switch", "Line Input"},
{"Output Mixer", "HiFi Playback Switch", "DAC"},
{"Output Mixer", "Mic Sidetone Switch", "Mic Bias"},
```

So we have :-

- Destination Widget <=== Path Name <=== Source Widget, or
- Sink, Path, Source, or
- `Output Mixer` is connected to the DAC via the `HiFi Playback Switch`.

When there is no path name connecting widgets (e.g. a direct connection) we pass NULL for the path name.

Interconnections are created with a call to:-

```
snd_soc_dapm_connect_input(codec, sink, path, source);
```

Finally, snd_soc_dapm_new_widgets(codec) must be called after all widgets and interconnections have been registered with the core. This causes the core to scan the codec and machine so that the internal DAPM state matches the physical state of the machine.

Machine Widget Interconnections

Machine widget interconnections are created in the same way as codec ones and directly connect the codec pins to machine level widgets.

e.g. connects the speaker out codec pins to the internal speaker.

```
/* ext speaker connected to codec pins LOUT2, ROUT2  */
{"Ext Spk", NULL , "ROUT2"},
{"Ext Spk", NULL , "LOUT2"},
```

This allows the DAPM to power on and off pins that are connected (and in use) and pins that are NC respectively.

Endpoint Widgets

An endpoint is a start or end point (widget) of an audio signal within the machine and includes the codec. e.g.

- Headphone Jack
- Internal Speaker
- Internal Mic
- Mic Jack
- Codec Pins

Endpoints are added to the DAPM graph so that their usage can be determined in order to save power. e.g. NC codecs pins will be switched OFF, unconnected jacks can also be switched OFF.

DAPM Widget Events

Some widgets can register their interest with the DAPM core in PM events. e.g. A Speaker with an amplifier registers a widget so the amplifier can be powered only when the spk is in use.

```
/* turn speaker amplifier on/off depending on use */
static int corgi_amp_event(struct snd_soc_dapm_widget *w, int event)
{
      gpio_set_value(CORGI_GPIO_APM_ON, SND_SOC_DAPM_EVENT_ON(event));
      return 0;
}

/* corgi machine dapm widgets */
static const struct snd_soc_dapm_widget wm8731_dapm_widgets =
      SND_SOC_DAPM_SPK("Ext Spk", corgi_amp_event);
```

Please see soc-dapm.h for all other widgets that support events.

Event types

The following event types are supported by event widgets.

```
/* dapm event types */
#define SND_SOC_DAPM_PRE_PMU  0x1     /* before widget power up */
#define SND_SOC_DAPM_POST_PMU 0x2             /* after widget power up */
#define SND_SOC_DAPM_PRE_PMD  0x4     /* before widget power down */
#define SND_SOC_DAPM_POST_PMD 0x8             /* after widget power down */
#define SND_SOC_DAPM_PRE_REG  0x10    /* before audio path setup */
#define SND_SOC_DAPM_POST_REG 0x20    /* after audio path setup */
```

ASoC Platform Driver

An ASoC platform driver class can be divided into audio DMA drivers, SoC DAI drivers and DSP drivers. The platform drivers only target the SoC CPU and must have no board specific code.

Audio DMA

The platform DMA driver optionally supports the following ALSA operations:-

```
/* SoC audio ops */
struct snd_soc_ops {
      int (*startup)(struct snd_pcm_substream *);
      void (*shutdown)(struct snd_pcm_substream *);
      int (*hw_params)(struct snd_pcm_substream *, struct snd_pcm_hw_params *);
      int (*hw_free)(struct snd_pcm_substream *);
      int (*prepare)(struct snd_pcm_substream *);
      int (*trigger)(struct snd_pcm_substream *, int);
};
```

The platform driver exports its DMA functionality via struct snd_soc_component_driver:-

```
struct snd_soc_component_driver {
      const char *name;

      ...
      int (*probe)(struct snd_soc_component *);
      void (*remove)(struct snd_soc_component *);
      int (*suspend)(struct snd_soc_component *);
```

```
      int (*resume)(struct snd_soc_component *);

      /* pcm creation and destruction */
      int (*pcm_new)(struct snd_soc_pcm_runtime *);
      void (*pcm_free)(struct snd_pcm *);

      ...
      const struct snd_pcm_ops *ops;
      const struct snd_compr_ops *compr_ops;
      ...
};
```

Please refer to the ALSA driver documentation for details of audio DMA. http://www.alsa-project.org/~iwai/writing-an-alsa-driver/

An example DMA driver is soc/pxa/pxa2xx-pcm.c

SoC DAI Drivers

Each SoC DAI driver must provide the following features:-

1. Digital audio interface (DAI) description
2. Digital audio interface configuration
3. PCM's description
4. SYSCLK configuration
5. Suspend and resume (optional)

Please see codec.rst for a description of items 1 - 4.

SoC DSP Drivers

Each SoC DSP driver usually supplies the following features :-

1. DAPM graph
2. Mixer controls
3. DMA IO to/from DSP buffers (if applicable)
4. Definition of DSP front end (FE) PCM devices.

Please see DPCM.txt for a description of item 4.

ASoC Machine Driver

The ASoC machine (or board) driver is the code that glues together all the component drivers (e.g. codecs, platforms and DAIs). It also describes the relationships between each component which include audio paths, GPIOs, interrupts, clocking, jacks and voltage regulators.

The machine driver can contain codec and platform specific code. It registers the audio subsystem with the kernel as a platform device and is represented by the following struct:-

```
/* SoC machine */
struct snd_soc_card {
      char *name;

      ...
```

```
      int (*probe)(struct platform_device *pdev);
      int (*remove)(struct platform_device *pdev);

      /* the pre and post PM functions are used to do any PM work before and
       * after the codec and DAIs do any PM work. */
      int (*suspend_pre)(struct platform_device *pdev, pm_message_t state);
      int (*suspend_post)(struct platform_device *pdev, pm_message_t state);
      int (*resume_pre)(struct platform_device *pdev);
      int (*resume_post)(struct platform_device *pdev);

      ...

      /* CPU <--> Codec DAI links  */
      struct snd_soc_dai_link *dai_link;
      int num_links;

      ...
};
```

probe()/remove()

probe/remove are optional. Do any machine specific probe here.

suspend()/resume()

The machine driver has pre and post versions of suspend and resume to take care of any machine audio tasks that have to be done before or after the codec, DAIs and DMA is suspended and resumed. Optional.

Machine DAI Configuration

The machine DAI configuration glues all the codec and CPU DAIs together. It can also be used to set up the DAI system clock and for any machine related DAI initialisation e.g. the machine audio map can be connected to the codec audio map, unconnected codec pins can be set as such.

struct snd_soc_dai_link is used to set up each DAI in your machine. e.g.

```
/* corgi digital audio interface glue - connects codec <--> CPU */
static struct snd_soc_dai_link corgi_dai = {
      .name = "WM8731",
      .stream_name = "WM8731",
      .cpu_dai_name = "pxa-is2-dai",
      .codec_dai_name = "wm8731-hifi",
      .platform_name = "pxa-pcm-audio",
      .codec_name = "wm8713-codec.0-001a",
      .init = corgi_wm8731_init,
      .ops = &corgi_ops,
};
```

struct snd_soc_card then sets up the machine with its DAIs. e.g.

```
/* corgi audio machine driver */
static struct snd_soc_card snd_soc_corgi = {
      .name = "Corgi",
      .dai_link = &corgi_dai,
      .num_links = 1,
};
```

Machine Power Map

The machine driver can optionally extend the codec power map and to become an audio power map of the audio subsystem. This allows for automatic power up/down of speaker/HP amplifiers, etc. Codec pins can be connected to the machines jack sockets in the machine init function.

Machine Controls

Machine specific audio mixer controls can be added in the DAI init function.

Audio Pops and Clicks

Pops and clicks are unwanted audio artifacts caused by the powering up and down of components within the audio subsystem. This is noticeable on PCs when an audio module is either loaded or unloaded (at module load time the sound card is powered up and causes a popping noise on the speakers).

Pops and clicks can be more frequent on portable systems with DAPM. This is because the components within the subsystem are being dynamically powered depending on the audio usage and this can subsequently cause a small pop or click every time a component power state is changed.

Minimising Playback Pops and Clicks

Playback pops in portable audio subsystems cannot be completely eliminated currently, however future audio codec hardware will have better pop and click suppression. Pops can be reduced within playback by powering the audio components in a specific order. This order is different for startup and shutdown and follows some basic rules:-

```
Startup Order :- DAC --> Mixers --> Output PGA --> Digital Unmute

Shutdown Order :- Digital Mute --> Output PGA --> Mixers --> DAC
```

This assumes that the codec PCM output path from the DAC is via a mixer and then a PGA (programmable gain amplifier) before being output to the speakers.

Minimising Capture Pops and Clicks

Capture artifacts are somewhat easier to get rid as we can delay activating the ADC until all the pops have occurred. This follows similar power rules to playback in that components are powered in a sequence depending upon stream startup or shutdown.

```
Startup Order - Input PGA --> Mixers --> ADC

Shutdown Order - ADC --> Mixers --> Input PGA
```

Zipper Noise

An unwanted zipper noise can occur within the audio playback or capture stream when a volume control is changed near its maximum gain value. The zipper noise is heard when the gain increase or decrease changes the mean audio signal amplitude too quickly. It can be minimised by enabling the zero cross setting for each volume control. The ZC forces the gain change to occur when the signal crosses the zero amplitude line.

Audio Clocking

This text describes the audio clocking terms in ASoC and digital audio in general. Note: Audio clocking can be complex!

Master Clock

Every audio subsystem is driven by a master clock (sometimes referred to as MCLK or SYSCLK). This audio master clock can be derived from a number of sources (e.g. crystal, PLL, CPU clock) and is responsible for producing the correct audio playback and capture sample rates.

Some master clocks (e.g. PLLs and CPU based clocks) are configurable in that their speed can be altered by software (depending on the system use and to save power). Other master clocks are fixed at a set frequency (i.e. crystals).

DAI Clocks

The Digital Audio Interface is usually driven by a Bit Clock (often referred to as BCLK). This clock is used to drive the digital audio data across the link between the codec and CPU.

The DAI also has a frame clock to signal the start of each audio frame. This clock is sometimes referred to as LRC (left right clock) or FRAME. This clock runs at exactly the sample rate (LRC = Rate).

Bit Clock can be generated as follows:-

- BCLK = MCLK / x, or
- BCLK = LRC * x, or
- BCLK = LRC * Channels * Word Size

This relationship depends on the codec or SoC CPU in particular. In general it is best to configure BCLK to the lowest possible speed (depending on your rate, number of channels and word size) to save on power.

It is also desirable to use the codec (if possible) to drive (or master) the audio clocks as it usually gives more accurate sample rates than the CPU.

ASoC jack detection

ALSA has a standard API for representing physical jacks to user space, the kernel side of which can be seen in include/sound/jack.h. ASoC provides a version of this API adding two additional features:

- It allows more than one jack detection method to work together on one user visible jack. In embedded systems it is common for multiple to be present on a single jack but handled by separate bits of hardware.
- Integration with DAPM, allowing DAPM endpoints to be updated automatically based on the detected jack status (eg, turning off the headphone outputs if no headphones are present).

This is done by splitting the jacks up into three things working together: the jack itself represented by a struct snd_soc_jack, sets of snd_soc_jack_pins representing DAPM endpoints to update and blocks of code providing jack reporting mechanisms.

For example, a system may have a stereo headset jack with two reporting mechanisms, one for the headphone and one for the microphone. Some systems won't be able to use their speaker output while a headphone is connected and so will want to make sure to update both speaker and headphone when the headphone jack status changes.

The jack - struct snd_soc_jack

This represents a physical jack on the system and is what is visible to user space. The jack itself is completely passive, it is set up by the machine driver and updated by jack detection methods.

Jacks are created by the machine driver calling snd_soc_jack_new().

snd_soc_jack_pin

These represent a DAPM pin to update depending on some of the status bits supported by the jack. Each snd_soc_jack has zero or more of these which are updated automatically. They are created by the machine driver and associated with the jack using snd_soc_jack_add_pins(). The status of the endpoint may configured to be the opposite of the jack status if required (eg, enabling a built in microphone if a microphone is not connected via a jack).

Jack detection methods

Actual jack detection is done by code which is able to monitor some input to the system and update a jack by calling snd_soc_jack_report(), specifying a subset of bits to update. The jack detection code should be set up by the machine driver, taking configuration for the jack to update and the set of things to report when the jack is connected.

Often this is done based on the status of a GPIO - a handler for this is provided by the snd_soc_jack_add_gpio() function. Other methods are also available, for example integrated into CODECs. One example of CODEC integrated jack detection can be see in the WM8350 driver.

Each jack may have multiple reporting mechanisms, though it will need at least one to be useful.

Machine drivers

These are all hooked together by the machine driver depending on the system hardware. The machine driver will set up the snd_soc_jack and the list of pins to update then set up one or more jack detection mechanisms to update that jack based on their current status.

Dynamic PCM

Description

Dynamic PCM allows an ALSA PCM device to digitally route its PCM audio to various digital endpoints during the PCM stream runtime. e.g. PCM0 can route digital audio to I2S DAI0, I2S DAI1 or PDM DAI2. This is useful for on SoC DSP drivers that expose several ALSA PCMs and can route to multiple DAIs.

The DPCM runtime routing is determined by the ALSA mixer settings in the same way as the analog signal is routed in an ASoC codec driver. DPCM uses a DAPM graph representing the DSP internal audio paths and uses the mixer settings to determine the patch used by each ALSA PCM.

DPCM re-uses all the existing component codec, platform and DAI drivers without any modifications.

Phone Audio System with SoC based DSP

Consider the following phone audio subsystem. This will be used in this document for all examples :-

```
| Front End PCMs    |  SoC DSP  | Back End DAIs | Audio devices |

                     *************
PCM0 <------------> *           * <----DAI0-----> Codec Headset
                     *           *
PCM1 <------------> *           * <----DAI1-----> Codec Speakers
                     *   DSP     *
PCM2 <------------> *           * <----DAI2-----> MODEM
                     *           *
PCM3 <------------> *           * <----DAI3-----> BT
                     *           *
                     *           * <----DAI4-----> DMIC
                     *           *
                     *           * <----DAI5-----> FM
                     *************
```

This diagram shows a simple smart phone audio subsystem. It supports Bluetooth, FM digital radio, Speakers, Headset Jack, digital microphones and cellular modem. This sound card exposes 4 DSP front end (FE) ALSA PCM devices and supports 6 back end (BE) DAIs. Each FE PCM can digitally route audio data to any of the BE DAIs. The FE PCM devices can also route audio to more than 1 BE DAI.

Example - DPCM Switching playback from DAI0 to DAI1

Audio is being played to the Headset. After a while the user removes the headset and audio continues playing on the speakers.

Playback on PCM0 to Headset would look like :-

```
                     *************
PCM0 <============> *           * <====DAI0=====> Codec Headset
                     *           *
PCM1 <------------> *           * <----DAI1-----> Codec Speakers
                     *   DSP     *
PCM2 <------------> *           * <----DAI2-----> MODEM
                     *           *
PCM3 <------------> *           * <----DAI3-----> BT
                     *           *
                     *           * <----DAI4-----> DMIC
                     *           *
                     *           * <----DAI5-----> FM
                     *************
```

The headset is removed from the jack by user so the speakers must now be used :-

```
                     *************
PCM0 <============> *           * <----DAI0-----> Codec Headset
                     *           *
PCM1 <------------> *           * <====DAI1=====> Codec Speakers
                     *   DSP     *
PCM2 <------------> *           * <----DAI2-----> MODEM
                     *           *
PCM3 <------------> *           * <----DAI3-----> BT
                     *           *
                     *           * <----DAI4-----> DMIC
                     *           *
                     *           * <----DAI5-----> FM
                     *************
```

The audio driver processes this as follows :-

1. Machine driver receives Jack removal event.
2. Machine driver OR audio HAL disables the Headset path.

3. DPCM runs the PCM trigger(stop), hw_free(), shutdown() operations on DAI0 for headset since the path is now disabled.
4. Machine driver or audio HAL enables the speaker path.
5. DPCM runs the PCM ops for startup(), hw_params(), prepapre() and trigger(start) for DAI1 Speakers since the path is enabled.

In this example, the machine driver or userspace audio HAL can alter the routing and then DPCM will take care of managing the DAI PCM operations to either bring the link up or down. Audio playback does not stop during this transition.

DPCM machine driver

The DPCM enabled ASoC machine driver is similar to normal machine drivers except that we also have to :-

1. Define the FE and BE DAI links.
2. Define any FE/BE PCM operations.
3. Define widget graph connections.

FE and BE DAI links

```
| Front End PCMs    |  SoC DSP  | Back End DAIs | Audio devices |

                     *************
PCM0 <------------> *           * <----DAI0-----> Codec Headset
                     *           *
PCM1 <------------> *           * <----DAI1-----> Codec Speakers
                     *   DSP     *
PCM2 <------------> *           * <----DAI2-----> MODEM
                     *           *
PCM3 <------------> *           * <----DAI3-----> BT
                     *           *
                     *           * <----DAI4-----> DMIC
                     *           *
                     *           * <----DAI5-----> FM
                     *************
```

For the example above we have to define 4 FE DAI links and 6 BE DAI links. The FE DAI links are defined as follows :-

```
static struct snd_soc_dai_link machine_dais[] = {
      {
               .name = "PCM0 System",
               .stream_name = "System Playback",
               .cpu_dai_name = "System Pin",
               .platform_name = "dsp-audio",
               .codec_name = "snd-soc-dummy",
               .codec_dai_name = "snd-soc-dummy-dai",
               .dynamic = 1,
               .trigger = {SND_SOC_DPCM_TRIGGER_POST, SND_SOC_DPCM_TRIGGER_POST},
               .dpcm_playback = 1,
      },
      .....< other FE and BE DAI links here >
};
```

This FE DAI link is pretty similar to a regular DAI link except that we also set the DAI link to a DPCM FE with the `dynamic = 1`. The supported FE stream directions should also be set with the `dpcm_playback` and `dpcm_capture` flags. There is also an option to specify the ordering of the trigger call for each FE.

This allows the ASoC core to trigger the DSP before or after the other components (as some DSPs have strong requirements for the ordering DAI/DSP start and stop sequences).

The FE DAI above sets the codec and code DAIs to dummy devices since the BE is dynamic and will change depending on runtime config.

The BE DAIs are configured as follows :-

```
static struct snd_soc_dai_link machine_dais[] = {
      .....< FE DAI links here >
      {
              .name = "Codec Headset",
              .cpu_dai_name = "ssp-dai.0",
              .platform_name = "snd-soc-dummy",
              .no_pcm = 1,
              .codec_name = "rt5640.0-001c",
              .codec_dai_name = "rt5640-aif1",
              .ignore_suspend = 1,
              .ignore_pmdown_time = 1,
              .be_hw_params_fixup = hswult_ssp0_fixup,
              .ops = &haswell_ops,
              .dpcm_playback = 1,
              .dpcm_capture = 1,
      },
      .....< other BE DAI links here >
};
```

This BE DAI link connects DAI0 to the codec (in this case RT5460 AIF1). It sets the `no_pcm` flag to mark it has a BE and sets flags for supported stream directions using `dpcm_playback` and `dpcm_capture` above.

The BE has also flags set for ignoring suspend and PM down time. This allows the BE to work in a hostless mode where the host CPU is not transferring data like a BT phone call :-

```
                    *************
PCM0 <------------> *           * <----DAI0-----> Codec Headset
                    *           *
PCM1 <------------> *           * <----DAI1-----> Codec Speakers
                    *   DSP     *
PCM2 <------------> *           * <====DAI2=====> MODEM
                    *           *
PCM3 <------------> *           * <====DAI3=====> BT
                    *           *
                    *           * <----DAI4-----> DMIC
                    *           *
                    *           * <----DAI5-----> FM
                    *************
```

This allows the host CPU to sleep whilst the DSP, MODEM DAI and the BT DAI are still in operation.

A BE DAI link can also set the codec to a dummy device if the code is a device that is managed externally.

Likewise a BE DAI can also set a dummy cpu DAI if the CPU DAI is managed by the DSP firmware.

FE/BE PCM operations

The BE above also exports some PCM operations and a `fixup` callback. The fixup callback is used by the machine driver to (re)configure the DAI based upon the FE hw params. i.e. the DSP may perform SRC or ASRC from the FE to BE.

e.g. DSP converts all FE hw params to run at fixed rate of 48k, 16bit, stereo for DAI0. This means all FE hw_params have to be fixed in the machine driver for DAI0 so that the DAI is running at desired configuration regardless of the FE configuration.

```
static int dai0_fixup(struct snd_soc_pcm_runtime *rtd,
                      struct snd_pcm_hw_params *params)
{
      struct snd_interval *rate = hw_param_interval(params,
                      SNDRV_PCM_HW_PARAM_RATE);
      struct snd_interval *channels = hw_param_interval(params,
                                              SNDRV_PCM_HW_PARAM_CHANNELS);

      /* The DSP will covert the FE rate to 48k, stereo */
      rate->min = rate->max = 48000;
      channels->min = channels->max = 2;

      /* set DAI0 to 16 bit */
      snd_mask_set(&params->masks[SNDRV_PCM_HW_PARAM_FORMAT -
                                  SNDRV_PCM_HW_PARAM_FIRST_MASK],
                                  SNDRV_PCM_FORMAT_S16_LE);
      return 0;
}
```

The other PCM operation are the same as for regular DAI links. Use as necessary.

Widget graph connections

The BE DAI links will normally be connected to the graph at initialisation time by the ASoC DAPM core. However, if the BE codec or BE DAI is a dummy then this has to be set explicitly in the driver :-

```
/* BE for codec Headset -  DAI0 is dummy and managed by DSP FW */
{"DAI0 CODEC IN", NULL, "AIF1 Capture"},
{"AIF1 Playback", NULL, "DAI0 CODEC OUT"},
```

Writing a DPCM DSP driver

The DPCM DSP driver looks much like a standard platform class ASoC driver combined with elements from a codec class driver. A DSP platform driver must implement :-

1. Front End PCM DAIs - i.e. struct snd_soc_dai_driver.
2. DAPM graph showing DSP audio routing from FE DAIs to BEs.
3. DAPM widgets from DSP graph.
4. Mixers for gains, routing, etc.
5. DMA configuration.
6. BE AIF widgets.

Items 6 is important for routing the audio outside of the DSP. AIF need to be defined for each BE and each stream direction. e.g for BE DAI0 above we would have :-

```
SND_SOC_DAPM_AIF_IN("DAI0 RX", NULL, 0, SND_SOC_NOPM, 0, 0),
SND_SOC_DAPM_AIF_OUT("DAI0 TX", NULL, 0, SND_SOC_NOPM, 0, 0),
```

The BE AIF are used to connect the DSP graph to the graphs for the other component drivers (e.g. codec graph).

Hostless PCM streams

A hostless PCM stream is a stream that is not routed through the host CPU. An example of this would be a phone call from handset to modem.

```
                    *************
PCM0 <------------> *           * <----DAI0-----> Codec Headset
                    *           *
PCM1 <------------> *           * <====DAI1=====> Codec Speakers/Mic
                    *   DSP     *
PCM2 <------------> *           * <====DAI2=====> MODEM
                    *           *
PCM3 <------------> *           * <----DAI3-----> BT
                    *           *
                    *           * <----DAI4-----> DMIC
                    *           *
                    *           * <----DAI5-----> FM
                    *************
```

In this case the PCM data is routed via the DSP. The host CPU in this use case is only used for control and can sleep during the runtime of the stream.

The host can control the hostless link either by :-

1. Configuring the link as a CODEC <-> CODEC style link. In this case the link is enabled or disabled by the state of the DAPM graph. This usually means there is a mixer control that can be used to connect or disconnect the path between both DAIs.
2. Hostless FE. This FE has a virtual connection to the BE DAI links on the DAPM graph. Control is then carried out by the FE as regular PCM operations. This method gives more control over the DAI links, but requires much more userspace code to control the link. Its recommended to use CODEC<->CODEC unless your HW needs more fine grained sequencing of the PCM ops.

CODEC <-> CODEC link

This DAI link is enabled when DAPM detects a valid path within the DAPM graph. The machine driver sets some additional parameters to the DAI link i.e.

```
static const struct snd_soc_pcm_stream dai_params = {
      .formats = SNDRV_PCM_FMTBIT_S32_LE,
      .rate_min = 8000,
      .rate_max = 8000,
      .channels_min = 2,
      .channels_max = 2,
};

static struct snd_soc_dai_link dais[] = {
      < ... more DAI links above ... >
      {
              .name = "MODEM",
              .stream_name = "MODEM",
              .cpu_dai_name = "dai2",
              .codec_dai_name = "modem-aif1",
              .codec_name = "modem",
              .dai_fmt = SND_SOC_DAIFMT_I2S | SND_SOC_DAIFMT_NB_NF
                              | SND_SOC_DAIFMT_CBM_CFM,
              .params = &dai_params,
      }
      < ... more DAI links here ... >
```

These parameters are used to configure the DAI hw_params() when DAPM detects a valid path and then calls the PCM operations to start the link. DAPM will also call the appropriate PCM operations to disable the DAI when the path is no longer valid.

Hostless FE

The DAI link(s) are enabled by a FE that does not read or write any PCM data. This means creating a new FE that is connected with a virtual path to both DAI links. The DAI links will be started when the FE PCM is started and stopped when the FE PCM is stopped. Note that the FE PCM cannot read or write data in this configuration.

Creating codec to codec dai link for ALSA dapm

Mostly the flow of audio is always from CPU to codec so your system will look as below:

```
 ---------           ---------
|         |  dai    |         |
    CPU     ------->    codec
|         |         |         |
 ---------           ---------
```

In case your system looks as below:

```
                     ---------
                    |         |
                      codec-2
                    |         |
                    ---------
                         |
                        dai-2
                         |
 ----------          ---------
|          |  dai-1 |         |
    CPU      ------->  codec-1
|          |         |         |
 ----------          ---------
                         |
                        dai-3
                         |
                     ---------
                    |         |
                      codec-3
                    |         |
                     ---------
```

Suppose codec-2 is a bluetooth chip and codec-3 is connected to a speaker and you have a below scenario: codec-2 will receive the audio data and the user wants to play that audio through codec-3 without involving the CPU.This aforementioned case is the ideal case when codec to codec connection should be used.

Your dai_link should appear as below in your machine file:

```
/*
 * this pcm stream only supports 24 bit, 2 channel and
 * 48k sampling rate.
 */
static const struct snd_soc_pcm_stream dsp_codec_params = {
       .formats = SNDRV_PCM_FMTBIT_S24_LE,
       .rate_min = 48000,
       .rate_max = 48000,
       .channels_min = 2,
       .channels_max = 2,
};

{
   .name = "CPU-DSP",
   .stream_name = "CPU-DSP",
```

```
   .cpu_dai_name = "samsung-i2s.0",
   .codec_name = "codec-2,
   .codec_dai_name = "codec-2-dai_name",
   .platform_name = "samsung-i2s.0",
   .dai_fmt = SND_SOC_DAIFMT_I2S | SND_SOC_DAIFMT_NB_NF
           | SND_SOC_DAIFMT_CBM_CFM,
   .ignore_suspend = 1,
   .params = &dsp_codec_params,
},
{
   .name = "DSP-CODEC",
   .stream_name = "DSP-CODEC",
   .cpu_dai_name = "wm0010-sdi2",
   .codec_name = "codec-3,
   .codec_dai_name = "codec-3-dai_name",
   .dai_fmt = SND_SOC_DAIFMT_I2S | SND_SOC_DAIFMT_NB_NF
           | SND_SOC_DAIFMT_CBM_CFM,
   .ignore_suspend = 1,
   .params = &dsp_codec_params,
},
```

Above code snippet is motivated from sound/soc/samsung/speyside.c.

Note the “params” callback which lets the dapm know that this dai_link is a codec to codec connection.

In dapm core a route is created between cpu_dai playback widget and codec_dai capture widget for playback path and vice-versa is true for capture path. In order for this aforementioned route to get triggered, DAPM needs to find a valid endpoint which could be either a sink or source widget corresponding to playback and capture path respectively.

In order to trigger this dai_link widget, a thin codec driver for the speaker amp can be created as demonstrated in wm8727.c file, it sets appropriate constraints for the device even if it needs no control.

Make sure to name your corresponding cpu and codec playback and capture dai names ending with “Playback” and “Capture” respectively as dapm core will link and power those dais based on the name.

Note that in current device tree there is no way to mark a dai_link as codec to codec. However, it may change in future.

CHAPTER

FOUR

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

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 http://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, 0 = disable (default)

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

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.

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 MIXART.txt 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/oss/MultiSound 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 http://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.

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.

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.

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.

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 http://www.alsa-project.org/alsa-info.sh

CHAPTER

FIVE

HD-AUDIO

More Notes on HD-Audio Driver

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General

HD-audio is the new standard on-board audio component on modern PCs after AC97. Although Linux has been supporting HD-audio since long time ago, there are often problems with new machines. A part of the problem is broken BIOS, and the rest is the driver implementation. This document explains the brief trouble-shooting and debugging methods for the HD-audio hardware.

The HD-audio component consists of two parts: the controller chip and the codec chips on the HD-audio bus. Linux provides a single driver for all controllers, snd-hda-intel. Although the driver name contains a word of a well-known hardware vendor, it's not specific to it but for all controller chips by other companies. Since the HD-audio controllers are supposed to be compatible, the single snd-hda-driver should work in most cases. But, not surprisingly, there are known bugs and issues specific to each controller type. The snd-hda-intel driver has a bunch of workarounds for these as described below.

A controller may have multiple codecs. Usually you have one audio codec and optionally one modem codec. In theory, there might be multiple audio codecs, e.g. for analog and digital outputs, and the driver might not work properly because of conflict of mixer elements. This should be fixed in future if such hardware really exists.

The snd-hda-intel driver has several different codec parsers depending on the codec. It has a generic parser as a fallback, but this functionality is fairly limited until now. Instead of the generic parser, usually the codec-specific parser (coded in patch_*.c) is used for the codec-specific implementations. The details about the codec-specific problems are explained in the later sections.

If you are interested in the deep debugging of HD-audio, read the HD-audio specification at first. The specification is found on Intel's web page, for example:

- http://www.intel.com/standards/hdaudio/

HD-Audio Controller

DMA-Position Problem

The most common problem of the controller is the inaccurate DMA pointer reporting. The DMA pointer for playback and capture can be read in two ways, either via a LPIB register or via a position-buffer map. As default the driver tries to read from the io-mapped position-buffer, and falls back to LPIB if the position-buffer appears dead. However, this detection isn't perfect on some devices. In such a case, you can change the default method via `position_fix` option.

`position_fix=1` means to use LPIB method explicitly. `position_fix=2` means to use the position-buffer. `position_fix=3` means to use a combination of both methods, needed for some VIA controllers. The

capture stream position is corrected by comparing both LPIB and position-buffer values. `position_fix=4` is another combination available for all controllers, and uses LPIB for the playback and the position-buffer for the capture streams. 0 is the default value for all other controllers, the automatic check and fallback to LPIB as described in the above. If you get a problem of repeated sounds, this option might help.

In addition to that, every controller is known to be broken regarding the wake-up timing. It wakes up a few samples before actually processing the data on the buffer. This caused a lot of problems, for example, with ALSA dmix or JACK. Since 2.6.27 kernel, the driver puts an artificial delay to the wake up timing. This delay is controlled via `bdl_pos_adj` option.

When `bdl_pos_adj` is a negative value (as default), it's assigned to an appropriate value depending on the controller chip. For Intel chips, it'd be 1 while it'd be 32 for others. Usually this works. Only in case it doesn't work and you get warning messages, you should change this parameter to other values.

Codec-Probing Problem

A less often but a more severe problem is the codec probing. When BIOS reports the available codec slots wrongly, the driver gets confused and tries to access the non-existing codec slot. This often results in the total screw-up, and destructs the further communication with the codec chips. The symptom appears usually as error messages like:

```
hda_intel: azx_get_response timeout, switching to polling mode:
      last cmd=0x12345678
hda_intel: azx_get_response timeout, switching to single_cmd mode:
      last cmd=0x12345678
```

The first line is a warning, and this is usually relatively harmless. It means that the codec response isn't notified via an IRQ. The driver uses explicit polling method to read the response. It gives very slight CPU overhead, but you'd unlikely notice it.

The second line is, however, a fatal error. If this happens, usually it means that something is really wrong. Most likely you are accessing a non-existing codec slot.

Thus, if the second error message appears, try to narrow the probed codec slots via `probe_mask` option. It's a bitmask, and each bit corresponds to the codec slot. For example, to probe only the first slot, pass `probe_mask=1`. For the first and the third slots, pass `probe_mask=5` (where 5 = 1 | 4), and so on.

Since 2.6.29 kernel, the driver has a more robust probing method, so this error might happen rarely, though.

On a machine with a broken BIOS, sometimes you need to force the driver to probe the codec slots the hardware doesn't report for use. In such a case, turn the bit 8 (0x100) of `probe_mask` option on. Then the rest 8 bits are passed as the codec slots to probe unconditionally. For example, `probe_mask=0x103` will force to probe the codec slots 0 and 1 no matter what the hardware reports.

Interrupt Handling

HD-audio driver uses MSI as default (if available) since 2.6.33 kernel as MSI works better on some machines, and in general, it's better for performance. However, Nvidia controllers showed bad regressions with MSI (especially in a combination with AMD chipset), thus we disabled MSI for them.

There seem also still other devices that don't work with MSI. If you see a regression wrt the sound quality (stuttering, etc) or a lock-up in the recent kernel, try to pass `enable_msi=0` option to disable MSI. If it works, you can add the known bad device to the blacklist defined in hda_intel.c. In such a case, please report and give the patch back to the upstream developer.

HD-Audio Codec

Model Option

The most common problem regarding the HD-audio driver is the unsupported codec features or the mismatched device configuration. Most of codec-specific code has several preset models, either to override the BIOS setup or to provide more comprehensive features.

The driver checks PCI SSID and looks through the static configuration table until any matching entry is found. If you have a new machine, you may see a message like below:

```
hda_codec: ALC880: BIOS auto-probing.
```

Meanwhile, in the earlier versions, you would see a message like:

```
hda_codec: Unknown model for ALC880, trying auto-probe from BIOS...
```

Even if you see such a message, DON'T PANIC. Take a deep breath and keep your towel. First of all, it's an informational message, no warning, no error. This means that the PCI SSID of your device isn't listed in the known preset model (white-)list. But, this doesn't mean that the driver is broken. Many codec-drivers provide the automatic configuration mechanism based on the BIOS setup.

The HD-audio codec has usually "pin" widgets, and BIOS sets the default configuration of each pin, which indicates the location, the connection type, the jack color, etc. The HD-audio driver can guess the right connection judging from these default configuration values. However – some codec-support codes, such as patch_analog.c, don't support the automatic probing (yet as of 2.6.28). And, BIOS is often, yes, pretty often broken. It sets up wrong values and screws up the driver.

The preset model (or recently called as "fix-up") is provided basically to overcome such a situation. When the matching preset model is found in the white-list, the driver assumes the static configuration of that preset with the correct pin setup, etc. Thus, if you have a newer machine with a slightly different PCI SSID (or codec SSID) from the existing one, you may have a good chance to re-use the same model. You can pass the `model` option to specify the preset model instead of PCI (and codec-) SSID look-up.

What `model` option values are available depends on the codec chip. Check your codec chip from the codec proc file (see "Codec Proc-File" section below). It will show the vendor/product name of your codec chip. Then, see Documentation/sound/hd-audio/models.rst file, the section of HD-audio driver. You can find a list of codecs and `model` options belonging to each codec. For example, for Realtek ALC262 codec chip, pass `model=ultra` for devices that are compatible with Samsung Q1 Ultra.

Thus, the first thing you can do for any brand-new, unsupported and non-working HD-audio hardware is to check HD-audio codec and several different `model` option values. If you have any luck, some of them might suit with your device well.

There are a few special model option values:

- when 'nofixup' is passed, the device-specific fixups in the codec parser are skipped.
- when `generic` is passed, the codec-specific parser is skipped and only the generic parser is used.

Speaker and Headphone Output

One of the most frequent (and obvious) bugs with HD-audio is the silent output from either or both of a built-in speaker and a headphone jack. In general, you should try a headphone output at first. A speaker output often requires more additional controls like the external amplifier bits. Thus a headphone output has a slightly better chance.

Before making a bug report, double-check whether the mixer is set up correctly. The recent version of snd-hda-intel driver provides mostly "Master" volume control as well as "Front" volume (where Front indicates the front-channels). In addition, there can be individual "Headphone" and "Speaker" controls.

Ditto for the speaker output. There can be "External Amplifier" switch on some codecs. Turn on this if present.

Another related problem is the automatic mute of speaker output by headphone plugging. This feature is implemented in most cases, but not on every preset model or codec-support code.

In anyway, try a different model option if you have such a problem. Some other models may match better and give you more matching functionality. If none of the available models works, send a bug report. See the bug report section for details.

If you are masochistic enough to debug the driver problem, note the following:

- The speaker (and the headphone, too) output often requires the external amplifier. This can be set usually via EAPD verb or a certain GPIO. If the codec pin supports EAPD, you have a better chance via SET_EAPD_BTL verb (0x70c). On others, GPIO pin (mostly it's either GPIO0 or GPIO1) may turn on/off EAPD.
- Some Realtek codecs require special vendor-specific coefficients to turn on the amplifier. See patch_realtek.c.
- IDT codecs may have extra power-enable/disable controls on each analog pin. See patch_sigmatel.c.
- Very rare but some devices don't accept the pin-detection verb until triggered. Issuing GET_PIN_SENSE verb (0xf09) may result in the codec-communication stall. Some examples are found in patch_realtek.c.

Capture Problems

The capture problems are often because of missing setups of mixers. Thus, before submitting a bug report, make sure that you set up the mixer correctly. For example, both "Capture Volume" and "Capture Switch" have to be set properly in addition to the right "Capture Source" or "Input Source" selection. Some devices have "Mic Boost" volume or switch.

When the PCM device is opened via "default" PCM (without pulse-audio plugin), you'll likely have "Digital Capture Volume" control as well. This is provided for the extra gain/attenuation of the signal in software, especially for the inputs without the hardware volume control such as digital microphones. Unless really needed, this should be set to exactly 50%, corresponding to 0dB – neither extra gain nor attenuation. When you use "hw" PCM, i.e., a raw access PCM, this control will have no influence, though.

It's known that some codecs / devices have fairly bad analog circuits, and the recorded sound contains a certain DC-offset. This is no bug of the driver.

Most of modern laptops have no analog CD-input connection. Thus, the recording from CD input won't work in many cases although the driver provides it as the capture source. Use CDDA instead.

The automatic switching of the built-in and external mic per plugging is implemented on some codec models but not on every model. Partly because of my laziness but mostly lack of testers. Feel free to submit the improvement patch to the author.

Direct Debugging

If no model option gives you a better result, and you are a tough guy to fight against evil, try debugging via hitting the raw HD-audio codec verbs to the device. Some tools are available: hda-emu and hda-analyzer. The detailed description is found in the sections below. You'd need to enable hwdep for using these tools. See "Kernel Configuration" section.

Other Issues

Kernel Configuration

In general, I recommend you to enable the sound debug option, `CONFIG_SND_DEBUG=y`, no matter whether you are debugging or not. This enables snd_printd() macro and others, and you'll get additional kernel messages at probing.

In addition, you can enable `CONFIG_SND_DEBUG_VERBOSE=y`. But this will give you far more messages. Thus turn this on only when you are sure to want it.

Don't forget to turn on the appropriate CONFIG_SND_HDA_CODEC_* options. Note that each of them corresponds to the codec chip, not the controller chip. Thus, even if lspci shows the Nvidia controller, you may need to choose the option for other vendors. If you are unsure, just select all yes.

CONFIG_SND_HDA_HWDEP is a useful option for debugging the driver. When this is enabled, the driver creates hardware-dependent devices (one per each codec), and you have a raw access to the device via these device files. For example, hwC0D2 will be created for the codec slot #2 of the first card (#0). For debug-tools such as hda-verb and hda-analyzer, the hwdep device has to be enabled. Thus, it'd be better to turn this on always.

CONFIG_SND_HDA_RECONFIG is a new option, and this depends on the hwdep option above. When enabled, you'll have some sysfs files under the corresponding hwdep directory. See "HD-audio reconfiguration" section below.

CONFIG_SND_HDA_POWER_SAVE option enables the power-saving feature. See "Power-saving" section below.

Codec Proc-File

The codec proc-file is a treasure-chest for debugging HD-audio. It shows most of useful information of each codec widget.

The proc file is located in /proc/asound/card*/codec#*, one file per each codec slot. You can know the codec vendor, product id and names, the type of each widget, capabilities and so on. This file, however, doesn't show the jack sensing state, so far. This is because the jack-sensing might be depending on the trigger state.

This file will be picked up by the debug tools, and also it can be fed to the emulator as the primary codec information. See the debug tools section below.

This proc file can be also used to check whether the generic parser is used. When the generic parser is used, the vendor/product ID name will appear as "Realtek ID 0262", instead of "Realtek ALC262".

HD-Audio Reconfiguration

This is an experimental feature to allow you re-configure the HD-audio codec dynamically without reloading the driver. The following sysfs files are available under each codec-hwdep device directory (e.g. /sys/class/sound/hwC0D0):

vendor_id Shows the 32bit codec vendor-id hex number. You can change the vendor-id value by writing to this file.

subsystem_id Shows the 32bit codec subsystem-id hex number. You can change the subsystem-id value by writing to this file.

revision_id Shows the 32bit codec revision-id hex number. You can change the revision-id value by writing to this file.

afg Shows the AFG ID. This is read-only.

mfg Shows the MFG ID. This is read-only.

name Shows the codec name string. Can be changed by writing to this file.

modelname Shows the currently set `model` option. Can be changed by writing to this file.

init_verbs The extra verbs to execute at initialization. You can add a verb by writing to this file. Pass three numbers: nid, verb and parameter (separated with a space).

hints Shows / stores hint strings for codec parsers for any use. Its format is `key = value`. For example, passing `jack_detect = no` will disable the jack detection of the machine completely.

init_pin_configs Shows the initial pin default config values set by BIOS.

driver_pin_configs Shows the pin default values set by the codec parser explicitly. This doesn't show all pin values but only the changed values by the parser. That is, if the parser doesn't change the pin default config values by itself, this will contain nothing.

user_pin_configs Shows the pin default config values to override the BIOS setup. Writing this (with two numbers, NID and value) appends the new value. The given will be used instead of the initial BIOS value at the next reconfiguration time. Note that this config will override even the driver pin configs, too.

reconfig Triggers the codec re-configuration. When any value is written to this file, the driver re-initialize and parses the codec tree again. All the changes done by the sysfs entries above are taken into account.

clear Resets the codec, removes the mixer elements and PCM stuff of the specified codec, and clear all init verbs and hints.

For example, when you want to change the pin default configuration value of the pin widget 0x14 to 0x9993013f, and let the driver re-configure based on that state, run like below:

```
# echo 0x14 0x9993013f > /sys/class/sound/hwC0D0/user_pin_configs
# echo 1 > /sys/class/sound/hwC0D0/reconfig
```

Hint Strings

The codec parser have several switches and adjustment knobs for matching better with the actual codec or device behavior. Many of them can be adjusted dynamically via "hints" strings as mentioned in the section above. For example, by passing `jack_detect = no` string via sysfs or a patch file, you can disable the jack detection, thus the codec parser will skip the features like auto-mute or mic auto-switch. As a boolean value, either `yes`, `no`, `true`, `false`, `1` or `0` can be passed.

The generic parser supports the following hints:

jack_detect (bool) specify whether the jack detection is available at all on this machine; default true

inv_jack_detect (bool) indicates that the jack detection logic is inverted

trigger_sense (bool) indicates that the jack detection needs the explicit call of AC_VERB_SET_PIN_SENSE verb

inv_eapd (bool) indicates that the EAPD is implemented in the inverted logic

pcm_format_first (bool) sets the PCM format before the stream tag and channel ID

sticky_stream (bool) keep the PCM format, stream tag and ID as long as possible; default true

spdif_status_reset (bool) reset the SPDIF status bits at each time the SPDIF stream is set up

pin_amp_workaround (bool) the output pin may have multiple amp values

single_adc_amp (bool) ADCs can have only single input amps

auto_mute (bool) enable/disable the headphone auto-mute feature; default true

auto_mic (bool) enable/disable the mic auto-switch feature; default true

line_in_auto_switch (bool) enable/disable the line-in auto-switch feature; default false

need_dac_fix (bool) limits the DACs depending on the channel count

primary_hp (bool) probe headphone jacks as the primary outputs; default true

multi_io (bool) try probing multi-I/O config (e.g. shared line-in/surround, mic/clfe jacks)

multi_cap_vol (bool) provide multiple capture volumes

inv_dmic_split (bool) provide split internal mic volume/switch for phase-inverted digital mics

indep_hp (bool) provide the independent headphone PCM stream and the corresponding mixer control, if available

add_stereo_mix_input (bool) add the stereo mix (analog-loopback mix) to the input mux if available

add_jack_modes (bool) add “xxx Jack Mode” enum controls to each I/O jack for allowing to change the headphone amp and mic bias VREF capabilities

power_save_node (bool) advanced power management for each widget, controlling the power sate (D0/D3) of each widget node depending on the actual pin and stream states

power_down_unused (bool) power down the unused widgets, a subset of power_save_node, and will be dropped in future

add_hp_mic (bool) add the headphone to capture source if possible

hp_mic_detect (bool) enable/disable the hp/mic shared input for a single built-in mic case; default true

vmaster (bool) enable/disable the virtual Master control; default true

mixer_nid (int) specifies the widget NID of the analog-loopback mixer

Early Patching

When `CONFIG_SND_HDA_PATCH_LOADER=y` is set, you can pass a “patch” as a firmware file for modifying the HD-audio setup before initializing the codec. This can work basically like the reconfiguration via sysfs in the above, but it does it before the first codec configuration.

A patch file is a plain text file which looks like below:

```
[codec]
0x12345678 0xabcd1234 2

[model]
auto

[pincfg]
0x12 0x411111f0

[verb]
0x20 0x500 0x03
0x20 0x400 0xff

[hint]
jack_detect = no
```

The file needs to have a line `[codec]`. The next line should contain three numbers indicating the codec vendor-id (0x12345678 in the example), the codec subsystem-id (0xabcd1234) and the address (2) of the codec. The rest patch entries are applied to this specified codec until another codec entry is given. Passing 0 or a negative number to the first or the second value will make the check of the corresponding field be skipped. It’ll be useful for really broken devices that don’t initialize SSID properly.

The `[model]` line allows to change the model name of the each codec. In the example above, it will be changed to model=auto. Note that this overrides the module option.

After the `[pincfg]` line, the contents are parsed as the initial default pin-configurations just like `user_pin_configs` sysfs above. The values can be shown in user_pin_configs sysfs file, too.

Similarly, the lines after `[verb]` are parsed as `init_verbs` sysfs entries, and the lines after `[hint]` are parsed as `hints` sysfs entries, respectively.

Another example to override the codec vendor id from 0x12345678 to 0xdeadbeef is like below:

```
[codec]
0x12345678 0xabcd1234 2

[vendor_id]
0xdeadbeef
```

In the similar way, you can override the codec subsystem_id via [subsystem_id], the revision id via [revision_id] line. Also, the codec chip name can be rewritten via [chip_name] line.

```
[codec]
0x12345678 0xabcd1234 2

[subsystem_id]
0xffff1111

[revision_id]
0x10

[chip_name]
My-own NEWS-0002
```

The hd-audio driver reads the file via request_firmware(). Thus, a patch file has to be located on the appropriate firmware path, typically, /lib/firmware. For example, when you pass the option patch=hda-init.fw, the file /lib/firmware/hda-init.fw must be present.

The patch module option is specific to each card instance, and you need to give one file name for each instance, separated by commas. For example, if you have two cards, one for an on-board analog and one for an HDMI video board, you may pass patch option like below:

```
options snd-hda-intel patch=on-board-patch,hdmi-patch
```

Power-Saving

The power-saving is a kind of auto-suspend of the device. When the device is inactive for a certain time, the device is automatically turned off to save the power. The time to go down is specified via power_save module option, and this option can be changed dynamically via sysfs.

The power-saving won't work when the analog loopback is enabled on some codecs. Make sure that you mute all unneeded signal routes when you want the power-saving.

The power-saving feature might cause audible click noises at each power-down/up depending on the device. Some of them might be solvable, but some are hard, I'm afraid. Some distros such as openSUSE enables the power-saving feature automatically when the power cable is unplugged. Thus, if you hear noises, suspect first the power-saving. See /sys/module/snd_hda_intel/parameters/power_save to check the current value. If it's non-zero, the feature is turned on.

The recent kernel supports the runtime PM for the HD-audio controller chip, too. It means that the HD-audio controller is also powered up / down dynamically. The feature is enabled only for certain controller chips like Intel LynxPoint. You can enable/disable this feature forcibly by setting power_save_controller option, which is also available at /sys/module/snd_hda_intel/parameters directory.

Tracepoints

The hd-audio driver gives a few basic tracepoints. hda:hda_send_cmd traces each CORB write while hda:hda_get_response traces the response from RIRB (only when read from the codec driver). hda:hda_bus_reset traces the bus-reset due to fatal error, etc, hda:hda_unsol_event traces the unsolicited events, and hda:hda_power_down and hda:hda_power_up trace the power down/up via power-saving behavior.

Enabling all tracepoints can be done like

```
# echo 1 > /sys/kernel/debug/tracing/events/hda/enable
```

then after some commands, you can traces from /sys/kernel/debug/tracing/trace file. For example, when you want to trace what codec command is sent, enable the tracepoint like:

```
# cat /sys/kernel/debug/tracing/trace
# tracer: nop
#
#       TASK-PID    CPU#    TIMESTAMP  FUNCTION
#          | |       |          |         |
       <...>-7807  [002] 105147.774889: hda_send_cmd: [0:0] val=e3a019
       <...>-7807  [002] 105147.774893: hda_send_cmd: [0:0] val=e39019
       <...>-7807  [002] 105147.999542: hda_send_cmd: [0:0] val=e3a01a
       <...>-7807  [002] 105147.999543: hda_send_cmd: [0:0] val=e3901a
       <...>-26764 [001] 349222.837143: hda_send_cmd: [0:0] val=e3a019
       <...>-26764 [001] 349222.837148: hda_send_cmd: [0:0] val=e39019
       <...>-26764 [001] 349223.058539: hda_send_cmd: [0:0] val=e3a01a
       <...>-26764 [001] 349223.058541: hda_send_cmd: [0:0] val=e3901a
```

Here `[0:0]` indicates the card number and the codec address, and `val` shows the value sent to the codec, respectively. The value is a packed value, and you can decode it via hda-decode-verb program included in hda-emu package below. For example, the value e3a019 is to set the left output-amp value to 25.

```
% hda-decode-verb 0xe3a019
raw value = 0x00e3a019
cid = 0, nid = 0x0e, verb = 0x3a0, parm = 0x19
raw value: verb = 0x3a0, parm = 0x19
verbname = set_amp_gain_mute
amp raw val = 0xa019
output, left, idx=0, mute=0, val=25
```

Development Tree

The latest development codes for HD-audio are found on sound git tree:

- git://git.kernel.org/pub/scm/linux/kernel/git/tiwai/sound.git

The master branch or for-next branches can be used as the main development branches in general while the development for the current and next kernels are found in for-linus and for-next branches, respectively.

Sending a Bug Report

If any model or module options don't work for your device, it's time to send a bug report to the developers. Give the following in your bug report:

- Hardware vendor, product and model names
- Kernel version (and ALSA-driver version if you built externally)
- `alsa-info.sh` output; run with `--no-upload` option. See the section below about alsa-info

If it's a regression, at best, send alsa-info outputs of both working and non-working kernels. This is really helpful because we can compare the codec registers directly.

Send a bug report either the following:

kernel-bugzilla https://bugzilla.kernel.org/

alsa-devel ML alsa-devel@alsa-project.org

Debug Tools

This section describes some tools available for debugging HD-audio problems.

alsa-info

The script `alsa-info.sh` is a very useful tool to gather the audio device information. It's included in alsa-utils package. The latest version can be found on git repository:

- git://git.alsa-project.org/alsa-utils.git

The script can be fetched directly from the following URL, too:

- http://www.alsa-project.org/alsa-info.sh

Run this script as root, and it will gather the important information such as the module lists, module parameters, proc file contents including the codec proc files, mixer outputs and the control elements. As default, it will store the information onto a web server on alsa-project.org. But, if you send a bug report, it'd be better to run with `--no-upload` option, and attach the generated file.

There are some other useful options. See `--help` option output for details.

When a probe error occurs or when the driver obviously assigns a mismatched model, it'd be helpful to load the driver with `probe_only=1` option (at best after the cold reboot) and run alsa-info at this state. With this option, the driver won't configure the mixer and PCM but just tries to probe the codec slot. After probing, the proc file is available, so you can get the raw codec information before modified by the driver. Of course, the driver isn't usable with `probe_only=1`. But you can continue the configuration via hwdep sysfs file if hda-reconfig option is enabled. Using `probe_only` mask 2 skips the reset of HDA codecs (use `probe_only=3` as module option). The hwdep interface can be used to determine the BIOS codec initialization.

hda-verb

hda-verb is a tiny program that allows you to access the HD-audio codec directly. You can execute a raw HD-audio codec verb with this. This program accesses the hwdep device, thus you need to enable the kernel config `CONFIG_SND_HDA_HWDEP=y` beforehand.

The hda-verb program takes four arguments: the hwdep device file, the widget NID, the verb and the parameter. When you access to the codec on the slot 2 of the card 0, pass /dev/snd/hwC0D2 to the first argument, typically. (However, the real path name depends on the system.)

The second parameter is the widget number-id to access. The third parameter can be either a hex/digit number or a string corresponding to a verb. Similarly, the last parameter is the value to write, or can be a string for the parameter type.

```
% hda-verb /dev/snd/hwC0D0 0x12 0x701 2
nid = 0x12, verb = 0x701, param = 0x2
value = 0x0

% hda-verb /dev/snd/hwC0D0 0x0 PARAMETERS VENDOR_ID
nid = 0x0, verb = 0xf00, param = 0x0
value = 0x10ec0262

% hda-verb /dev/snd/hwC0D0 2 set_a 0xb080
nid = 0x2, verb = 0x300, param = 0xb080
value = 0x0
```

Although you can issue any verbs with this program, the driver state won't be always updated. For example, the volume values are usually cached in the driver, and thus changing the widget amp value directly via hda-verb won't change the mixer value.

The hda-verb program is included now in alsa-tools:

- git://git.alsa-project.org/alsa-tools.git

Also, the old stand-alone package is found in the ftp directory:

- ftp://ftp.suse.com/pub/people/tiwai/misc/

Also a git repository is available:

- git://git.kernel.org/pub/scm/linux/kernel/git/tiwai/hda-verb.git

See README file in the tarball for more details about hda-verb program.

hda-analyzer

hda-analyzer provides a graphical interface to access the raw HD-audio control, based on pyGTK2 binding. It's a more powerful version of hda-verb. The program gives you an easy-to-use GUI stuff for showing the widget information and adjusting the amp values, as well as the proc-compatible output.

The hda-analyzer:

- http://git.alsa-project.org/?p=alsa.git;a=tree;f=hda-analyzer

is a part of alsa.git repository in alsa-project.org:

- git://git.alsa-project.org/alsa.git

Codecgraph

Codecgraph is a utility program to generate a graph and visualizes the codec-node connection of a codec chip. It's especially useful when you analyze or debug a codec without a proper datasheet. The program parses the given codec proc file and converts to SVG via graphiz program.

The tarball and GIT trees are found in the web page at:

- http://helllabs.org/codecgraph/

hda-emu

hda-emu is an HD-audio emulator. The main purpose of this program is to debug an HD-audio codec without the real hardware. Thus, it doesn't emulate the behavior with the real audio I/O, but it just dumps the codec register changes and the ALSA-driver internal changes at probing and operating the HD-audio driver.

The program requires a codec proc-file to simulate. Get a proc file for the target codec beforehand, or pick up an example codec from the codec proc collections in the tarball. Then, run the program with the proc file, and the hda-emu program will start parsing the codec file and simulates the HD-audio driver:

```
% hda-emu codecs/stac9200-dell-d820-laptop
# Parsing..
hda_codec: Unknown model for STAC9200, using BIOS defaults
hda_codec: pin nid 08 bios pin config 40c003fa
....
```

The program gives you only a very dumb command-line interface. You can get a proc-file dump at the current state, get a list of control (mixer) elements, set/get the control element value, simulate the PCM operation, the jack plugging simulation, etc.

The program is found in the git repository below:

- git://git.kernel.org/pub/scm/linux/kernel/git/tiwai/hda-emu.git

See README file in the repository for more details about hda-emu program.

hda-jack-retask

hda-jack-retask is a user-friendly GUI program to manipulate the HD-audio pin control for jack retasking. If you have a problem about the jack assignment, try this program and check whether you can get useful

results. Once when you figure out the proper pin assignment, it can be fixed either in the driver code statically or via passing a firmware patch file (see “Early Patching” section).

The program is included in alsa-tools now:

- git://git.alsa-project.org/alsa-tools.git

HD-Audio Codec-Specific Models

ALC880

3stack 3-jack in back and a headphone out

3stack-digout 3-jack in back, a HP out and a SPDIF out

5stack 5-jack in back, 2-jack in front

5stack-digout 5-jack in back, 2-jack in front, a SPDIF out

6stack 6-jack in back, 2-jack in front

6stack-digout 6-jack with a SPDIF out

6stack-automute 6-jack with headphone jack detection

ALC260

gpio1 Enable GPIO1

coef Enable EAPD via COEF table

fujitsu Quirk for FSC S7020

fujitsu-jwse Quirk for FSC S7020 with jack modes and HP mic support

ALC262

inv-dmic Inverted internal mic workaround

ALC267/268

inv-dmic Inverted internal mic workaround

hp-eapd Disable HP EAPD on NID 0x15

ALC22x/23x/25x/269/27x/28x/29x (and vendor-specific ALC3xxx models)

laptop-amic Laptops with analog-mic input

laptop-dmic Laptops with digital-mic input

alc269-dmic Enable ALC269(VA) digital mic workaround

alc271-dmic Enable ALC271X digital mic workaround

inv-dmic Inverted internal mic workaround

headset-mic Indicates a combined headset (headphone+mic) jack

headset-mode More comprehensive headset support for ALC269 & co

headset-mode-no-hp-mic Headset mode support without headphone mic

lenovo-dock Enables docking station I/O for some Lenovos
hp-gpio-led GPIO LED support on HP laptops
hp-dock-gpio-mic1-led HP dock with mic LED support
dell-headset-multi Headset jack, which can also be used as mic-in
dell-headset-dock Headset jack (without mic-in), and also dock I/O
alc283-dac-wcaps Fixups for Chromebook with ALC283
alc283-sense-combo Combo jack sensing on ALC283
tpt440-dock Pin configs for Lenovo Thinkpad Dock support
tpt440 Lenovo Thinkpad T440s setup
tpt460 Lenovo Thinkpad T460/560 setup
dual-codecs Lenovo laptops with dual codecs
alc700-ref Intel reference board with ALC700 codec

ALC66x/67x/892

mario Chromebook mario model fixup
asus-mode1 ASUS
asus-mode2 ASUS
asus-mode3 ASUS
asus-mode4 ASUS
asus-mode5 ASUS
asus-mode6 ASUS
asus-mode7 ASUS
asus-mode8 ASUS
inv-dmic Inverted internal mic workaround
dell-headset-multi Headset jack, which can also be used as mic-in
dual-codecs Lenovo laptops with dual codecs

ALC680

N/A

ALC88x/898/1150

acer-aspire-4930g Acer Aspire 4930G/5930G/6530G/6930G/7730G
acer-aspire-8930g Acer Aspire 8330G/6935G
acer-aspire Acer Aspire others
inv-dmic Inverted internal mic workaround
no-primary-hp VAIO Z/VGC-LN51JGB workaround (for fixed speaker DAC)
dual-codecs ALC1220 dual codecs for Gaming mobos

ALC861/660

N/A

ALC861VD/660VD

N/A

CMI9880

minimal 3-jack in back

min_fp 3-jack in back, 2-jack in front

full 6-jack in back, 2-jack in front

full_dig 6-jack in back, 2-jack in front, SPDIF I/O

allout 5-jack in back, 2-jack in front, SPDIF out

auto auto-config reading BIOS (default)

AD1882 / AD1882A

3stack 3-stack mode

3stack-automute 3-stack with automute front HP (default)

6stack 6-stack mode

AD1884A / AD1883 / AD1984A / AD1984B

desktop 3-stack desktop (default) laptop laptop with HP jack sensing mobile mobile devices with HP jack sensing thinkpad Lenovo Thinkpad X300 touchsmart HP Touchsmart

AD1884

N/A

AD1981

basic 3-jack (default) hp HP nx6320 thinkpad Lenovo Thinkpad T60/X60/Z60 toshiba Toshiba U205

AD1983

N/A

AD1984

basic default configuration thinkpad Lenovo Thinkpad T61/X61 dell_desktop Dell T3400

AD1986A

3stack 3-stack, shared surrounds
laptop 2-channel only (FSC V2060, Samsung M50)
laptop-imic 2-channel with built-in mic
eapd Turn on EAPD constantly

AD1988/AD1988B/AD1989A/AD1989B

6stack 6-jack
6stack-dig ditto with SPDIF
3stack 3-jack
3stack-dig ditto with SPDIF
laptop 3-jack with hp-jack automute
laptop-dig ditto with SPDIF
auto auto-config reading BIOS (default)

Conexant 5045

cap-mix-amp Fix max input level on mixer widget
toshiba-p105 Toshiba P105 quirk
hp-530 HP 530 quirk

Conexant 5047

cap-mix-amp Fix max input level on mixer widget

Conexant 5051

lenovo-x200 Lenovo X200 quirk

Conexant 5066

stereo-dmic Workaround for inverted stereo digital mic
gpio1 Enable GPIO1 pin
headphone-mic-pin Enable headphone mic NID 0x18 without detection
tp410 Thinkpad T400 & co quirks
thinkpad Thinkpad mute/mic LED quirk
lemote-a1004 Lemote A1004 quirk
lemote-a1205 Lemote A1205 quirk
olpc-xo OLPC XO quirk
mute-led-eapd Mute LED control via EAPD
hp-dock HP dock support

mute-led-gpio Mute LED control via GPIO
hp-mic-fix Fix for headset mic pin on HP boxes

STAC9200

ref Reference board
oqo OQO Model 2
dell-d21 Dell (unknown)
dell-d22 Dell (unknown)
dell-d23 Dell (unknown)
dell-m21 Dell Inspiron 630m, Dell Inspiron 640m
dell-m22 Dell Latitude D620, Dell Latitude D820
dell-m23 Dell XPS M1710, Dell Precision M90
dell-m24 Dell Latitude 120L
dell-m25 Dell Inspiron E1505n
dell-m26 Dell Inspiron 1501
dell-m27 Dell Inspiron E1705/9400
gateway-m4 Gateway laptops with EAPD control
gateway-m4-2 Gateway laptops with EAPD control
panasonic Panasonic CF-74
auto BIOS setup (default)

STAC9205/9254

ref Reference board
dell-m42 Dell (unknown)
dell-m43 Dell Precision
dell-m44 Dell Inspiron
eapd Keep EAPD on (e.g. Gateway T1616)
auto BIOS setup (default)

STAC9220/9221

ref Reference board
3stack D945 3stack
5stack D945 5stack + SPDIF
intel-mac-v1 Intel Mac Type 1
intel-mac-v2 Intel Mac Type 2
intel-mac-v3 Intel Mac Type 3
intel-mac-v4 Intel Mac Type 4
intel-mac-v5 Intel Mac Type 5

intel-mac-auto Intel Mac (detect type according to subsystem id)
macmini Intel Mac Mini (equivalent with type 3)
macbook Intel Mac Book (eq. type 5)
macbook-pro-v1 Intel Mac Book Pro 1st generation (eq. type 3)
macbook-pro Intel Mac Book Pro 2nd generation (eq. type 3)
imac-intel Intel iMac (eq. type 2)
imac-intel-20 Intel iMac (newer version) (eq. type 3)
ecs202 ECS/PC chips
dell-d81 Dell (unknown)
dell-d82 Dell (unknown)
dell-m81 Dell (unknown)
dell-m82 Dell XPS M1210
auto BIOS setup (default)

STAC9202/9250/9251

ref Reference board, base config
m1 Some Gateway MX series laptops (NX560XL)
m1-2 Some Gateway MX series laptops (MX6453)
m2 Some Gateway MX series laptops (M255)
m2-2 Some Gateway MX series laptops
m3 Some Gateway MX series laptops
m5 Some Gateway MX series laptops (MP6954)
m6 Some Gateway NX series laptops
auto BIOS setup (default)

STAC9227/9228/9229/927x

ref Reference board
ref-no-jd Reference board without HP/Mic jack detection
3stack D965 3stack
5stack D965 5stack + SPDIF
5stack-no-fp D965 5stack without front panel
dell-3stack Dell Dimension E520
dell-bios Fixes with Dell BIOS setup
dell-bios-amic Fixes with Dell BIOS setup including analog mic
volknob Fixes with volume-knob widget 0x24
auto BIOS setup (default)

STAC92HD71B*

ref Reference board
dell-m4-1 Dell desktops
dell-m4-2 Dell desktops
dell-m4-3 Dell desktops
hp-m4 HP mini 1000
hp-dv5 HP dv series
hp-hdx HP HDX series
hp-dv4-1222nr HP dv4-1222nr (with LED support)
auto BIOS setup (default)

STAC92HD73*

ref Reference board
no-jd BIOS setup but without jack-detection
intel Intel DG45* mobos
dell-m6-amic Dell desktops/laptops with analog mics
dell-m6-dmic Dell desktops/laptops with digital mics
dell-m6 Dell desktops/laptops with both type of mics
dell-eq Dell desktops/laptops
alienware Alienware M17x
asus-mobo Pin configs for ASUS mobo with 5.1/SPDIF out
auto BIOS setup (default)

STAC92HD83*

ref Reference board
mic-ref Reference board with power management for ports
dell-s14 Dell laptop
dell-vostro-3500 Dell Vostro 3500 laptop
hp-dv7-4000 HP dv-7 4000
hp_cNB11_intquad HP CNB models with 4 speakers
hp-zephyr HP Zephyr
hp-led HP with broken BIOS for mute LED
hp-inv-led HP with broken BIOS for inverted mute LED
hp-mic-led HP with mic-mute LED
headset-jack Dell Latitude with a 4-pin headset jack
hp-envy-bass Pin fixup for HP Envy bass speaker (NID 0x0f)
hp-envy-ts-bass Pin fixup for HP Envy TS bass speaker (NID 0x10)
hp-bnb13-eq Hardware equalizer setup for HP laptops

hp-envy-ts-bass HP Envy TS bass support
auto BIOS setup (default)

STAC92HD95

hp-led LED support for HP laptops
hp-bass Bass HPF setup for HP Spectre 13

STAC9872

vaio VAIO laptop without SPDIF
auto BIOS setup (default)

Cirrus Logic CS4206/4207

mbp53 MacBook Pro 5,3
mbp55 MacBook Pro 5,5
imac27 IMac 27 Inch
imac27_122 iMac 12,2
apple Generic Apple quirk
mbp101 MacBookPro 10,1
mbp81 MacBookPro 8,1
mba42 MacBookAir 4,2
auto BIOS setup (default)

Cirrus Logic CS4208

mba6 MacBook Air 6,1 and 6,2
gpio0 Enable GPIO 0 amp
mbp11 MacBookPro 11,2
macmini MacMini 7,1
auto BIOS setup (default)

VIA VT17xx/VT18xx/VT20xx

auto BIOS setup (default)

HD-Audio Codec-Specific Mixer Controls

This file explains the codec-specific mixer controls.

Realtek codecs

Channel Mode This is an enum control to change the surround-channel setup, appears only when the surround channels are available. It gives the number of channels to be used, “2ch”, “4ch”, “6ch”, and “8ch”. According to the configuration, this also controls the jack-retasking of multi-I/O jacks.

Auto-Mute Mode This is an enum control to change the auto-mute behavior of the headphone and line-out jacks. If built-in speakers and headphone and/or line-out jacks are available on a machine, this controls appears. When there are only either headphones or line-out jacks, it gives “Disabled” and “Enabled” state. When enabled, the speaker is muted automatically when a jack is plugged.

When both headphone and line-out jacks are present, it gives “Disabled”, “Speaker Only” and “Line-Out+Speaker”. When speaker-only is chosen, plugging into a headphone or a line-out jack mutes the speakers, but not line-outs. When line-out+speaker is selected, plugging to a headphone jack mutes both speakers and line-outs.

IDT/Sigmatel codecs

Analog Loopback This control enables/disables the analog-loopback circuit. This appears only when “loopback” is set to true in a codec hint (see HD-Audio.txt). Note that on some codecs the analog-loopback and the normal PCM playback are exclusive, i.e. when this is on, you won’t hear any PCM stream.

Swap Center/LFE Swaps the center and LFE channel order. Normally, the left corresponds to the center and the right to the LFE. When this is ON, the left to the LFE and the right to the center.

Headphone as Line Out When this control is ON, treat the headphone jacks as line-out jacks. That is, the headphone won’t auto-mute the other line-outs, and no HP-amp is set to the pins.

Mic Jack Mode, Line Jack Mode, etc These enum controls the direction and the bias of the input jack pins. Depending on the jack type, it can set as “Mic In” and “Line In”, for determining the input bias, or it can be set to “Line Out” when the pin is a multi-I/O jack for surround channels.

VIA codecs

Smart 5.1 An enum control to re-task the multi-I/O jacks for surround outputs. When it’s ON, the corresponding input jacks (usually a line-in and a mic-in) are switched as the surround and the CLFE output jacks.

Independent HP When this enum control is enabled, the headphone output is routed from an individual stream (the third PCM such as hw:0,2) instead of the primary stream. In the case the headphone DAC is shared with a side or a CLFE-channel DAC, the DAC is switched to the headphone automatically.

Loopback Mixing An enum control to determine whether the analog-loopback route is enabled or not. When it’s enabled, the analog-loopback is mixed to the front-channel. Also, the same route is used for the headphone and speaker outputs. As a side-effect, when this mode is set, the individual volume controls will be no longer available for headphones and speakers because there is only one DAC connected to a mixer widget.

Dynamic Power-Control This control determines whether the dynamic power-control per jack detection is enabled or not. When enabled, the widgets power state (D0/D3) are changed dynamically depending on the jack plugging state for saving power consumptions. However, if your system doesn’t provide a proper jack-detection, this won’t work; in such a case, turn this control OFF.

Jack Detect This control is provided only for VT1708 codec which gives no proper unsolicited event per jack plug. When this is on, the driver polls the jack detection so that the headphone auto-mute can work, while turning this off would reduce the power consumption.

Conexant codecs

Auto-Mute Mode See Reatek codecs.

Analog codecs

Channel Mode This is an enum control to change the surround-channel setup, appears only when the surround channels are available. It gives the number of channels to be used, “2ch”, “4ch” and “6ch”. According to the configuration, this also controls the jack-retasking of multi-I/O jacks.

Independent HP When this enum control is enabled, the headphone output is routed from an individual stream (the third PCM such as hw:0,2) instead of the primary stream.

HD-Audio DP-MST Support

To support DP MST audio, HD Audio hdmi codec driver introduces virtual pin and dynamic pcm assignment.

Virtual pin is an extension of per_pin. The most difference of DP MST from legacy is that DP MST introduces device entry. Each pin can contain several device entries. Each device entry behaves as a pin.

As each pin may contain several device entries and each codec may contain several pins, if we use one pcm per per_pin, there will be many PCMs. The new solution is to create a few PCMs and to dynamically bind pcm to per_pin. Driver uses spec->dyn_pcm_assign flag to indicate whether to use the new solution.

PCM

To be added

Pin Initialization

Each pin may have several device entries (virtual pins). On Intel platform, the device entries number is dynamically changed. If DP MST hub is connected, it is in DP MST mode, and the device entries number is 3. Otherwise, the device entries number is 1.

To simplify the implementation, all the device entries will be initialized when bootup no matter whether it is in DP MST mode or not.

Connection list

DP MST reuses connection list code. The code can be reused because device entries on the same pin have the same connection list.

This means DP MST gets the device entry connection list without the device entry setting.

Jack

Presume:

- MST must be dyn_pcm_assign, and it is acomp (for Intel scenario);
- NON-MST may or may not be dyn_pcm_assign, it can be acomp or !acomp;

So there are the following scenarios:

1. MST (&& dyn_pcm_assign && acomp)
2. NON-MST && dyn_pcm_assign && acomp

3. NON-MST && !dyn_pcm_assign && !acomp

Below discussion will ignore MST and NON-MST difference as it doesn't impact on jack handling too much.

Driver uses struct hdmi_pcm pcm[] array in hdmi_spec and snd_jack is a member of hdmi_pcm. Each pin has one struct hdmi_pcm * pcm pointer.

For !dyn_pcm_assign, per_pin->pcm will assigned to spec->pcm[n] statically.

For dyn_pcm_assign, per_pin->pcm will assigned to spec->pcm[n] when monitor is hotplugged.

Build Jack

- dyn_pcm_assign

  Will not use hda_jack but use snd_jack in spec->pcm_rec[pcm_idx].jack directly.

- !dyn_pcm_assign

  Use hda_jack and assign spec->pcm_rec[pcm_idx].jack = jack->jack statically.

Unsolicited Event Enabling

Enable unsolicited event if !acomp.

Monitor Hotplug Event Handling

- acomp

  pin_eld_notify() -> check_presence_and_report() -> hdmi_present_sense() -> sync_eld_via_acomp().

  Use directly snd_jack_report() on spec->pcm_rec[pcm_idx].jack for both dyn_pcm_assign and !dyn_pcm_assign

- !acomp

  hdmi_unsol_event() -> hdmi_intrinsic_event() -> check_presence_and_report() -> hdmi_present_sense() -> hdmi_prepsent_sense_via_verbs()

  Use directly snd_jack_report() on spec->pcm_rec[pcm_idx].jack for dyn_pcm_assign. Use hda_jack mechanism to handle jack events.

Others to be added later

CHAPTER

SIX

CARD-SPECIFIC INFORMATION

Analog Joystick Support on ALSA Drivers

Oct. 14, 2003

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General

First of all, you need to enable GAMEPORT support on Linux kernel for using a joystick with the ALSA driver. For the details of gameport support, refer to Documentation/input/joydev/joystick.rst.

The joystick support of ALSA drivers is different between ISA and PCI cards. In the case of ISA (PnP) cards, it's usually handled by the independent module (ns558). Meanwhile, the ALSA PCI drivers have the built-in gameport support. Hence, when the ALSA PCI driver is built in the kernel, CONFIG_GAMEPORT must be 'y', too. Otherwise, the gameport support on that card will be (silently) disabled.

Some adapter modules probe the physical connection of the device at the load time. It'd be safer to plug in the joystick device before loading the module.

PCI Cards

For PCI cards, the joystick is enabled when the appropriate module option is specified. Some drivers don't need options, and the joystick support is always enabled. In the former ALSA version, there was a dynamic control API for the joystick activation. It was changed, however, to the static module options because of the system stability and the resource management.

The following PCI drivers support the joystick natively.

Driver	Module Option	Available Values
als4000	joystick_port	0 = disable (default), 1 = auto-detect, manual: any address (e.g. 0x200)
au88x0	N/A	N/A
azf3328	joystick	0 = disable, 1 = enable, -1 = auto (default)
ens1370	joystick	0 = disable (default), 1 = enable
ens1371	joystick_port	0 = disable (default), 1 = auto-detect, manual: 0x200, 0x208, 0x210, 0x218
cmipci	joystick_port	0 = disable (default), 1 = auto-detect, manual: any address (e.g. 0x200)
cs4281	N/A	N/A
cs46xx	N/A	N/A
es1938	N/A	N/A
es1968	joystick	0 = disable (default), 1 = enable
son-icvibes	N/A	N/A
trident	N/A	N/A
via82xx [1]	joystick	0 = disable (default), 1 = enable
ymfpci	joystick_port	0 = disable (default), 1 = auto-detect, manual: 0x201, 0x202, 0x204, 0x205 [2]

The following drivers don't support gameport natively, but there are additional modules. Load the corresponding module to add the gameport support.

Driver	Additional Module
emu10k1	emu10k1-gp
fm801	fm801-gp

Note: the "pcigame" and "cs461x" modules are for the OSS drivers only. These ALSA drivers (cs46xx, trident and au88x0) have the built-in gameport support.

As mentioned above, ALSA PCI drivers have the built-in gameport support, so you don't have to load ns558 module. Just load "joydev" and the appropriate adapter module (e.g. "analog").

ISA Cards

ALSA ISA drivers don't have the built-in gameport support. Instead, you need to load "ns558" module in addition to "joydev" and the adapter module (e.g. "analog").

Brief Notes on C-Media 8338/8738/8768/8770 Driver

Takashi Iwai <tiwai@suse.de>

Front/Rear Multi-channel Playback

CM8x38 chip can use ADC as the second DAC so that two different stereo channels can be used for front/rear playbacks. Since there are two DACs, both streams are handled independently unlike the 4/6ch multi- channel playbacks in the section below.

As default, ALSA driver assigns the first PCM device (i.e. hw:0,0 for card#0) for front and 4/6ch playbacks, while the second PCM device (hw:0,1) is assigned to the second DAC for rear playback.

There are slight differences between the two DACs:

- The first DAC supports U8 and S16LE formats, while the second DAC supports only S16LE.
- The second DAC supports only two channel stereo.

[1] VIA686A/B only
[2] With YMF744/754 chips, the port address can be chosen arbitrarily

Please note that the CM8x38 DAC doesn't support continuous playback rate but only fixed rates: 5512, 8000, 11025, 16000, 22050, 32000, 44100 and 48000 Hz.

The rear output can be heard only when "Four Channel Mode" switch is disabled. Otherwise no signal will be routed to the rear speakers. As default it's turned on.

Warning:

When "Four Channel Mode" switch is off, the output from rear speakers will be FULL VOLUME re less of Master and PCM volumes [a]. This might damage your audio equipment. Please discor speakers before your turn off this switch.

[a] Well.. I once got the output with correct volume (i.e. same with the front one) and was so excited. It was even with Channel" bit on and "double DAC" mode. Actually I could hear separate 4 channels from front and rear speakers! But. reboot, all was gone. It's a very pity that I didn't save the register dump at that time.. Maybe there is an unknown re to achieve this...

If your card has an extra output jack for the rear output, the rear playback should be routed there as default. If not, there is a control switch in the driver "Line-In As Rear", which you can change via alsamixer or somewhat else. When this switch is on, line-in jack is used as rear output.

There are two more controls regarding to the rear output. The "Exchange DAC" switch is used to exchange front and rear playback routes, i.e. the 2nd DAC is output from front output.

4/6 Multi-Channel Playback

The recent CM8738 chips support for the 4/6 multi-channel playback function. This is useful especially for AC3 decoding.

When the multi-channel is supported, the driver name has a suffix "-MC" such like "CMI8738-MC6". You can check this name from /proc/asound/cards.

When the 4/6-ch output is enabled, the second DAC accepts up to 6 (or 4) channels. While the dual DAC supports two different rates or formats, the 4/6-ch playback supports only the same condition for all channels. Since the multi-channel playback mode uses both DACs, you cannot operate with full-duplex.

The 4.0 and 5.1 modes are defined as the pcm "surround40" and "surround51" in alsa-lib. For example, you can play a WAV file with 6 channels like

```
% aplay -Dsurround51 sixchannels.wav
```

For programming the 4/6 channel playback, you need to specify the PCM channels as you like and set the format S16LE. For example, for playback with 4 channels,

```
snd_pcm_hw_params_set_access(pcm, hw, SND_PCM_ACCESS_RW_INTERLEAVED);
    // or mmap if you like
snd_pcm_hw_params_set_format(pcm, hw, SND_PCM_FORMAT_S16_LE);
snd_pcm_hw_params_set_channels(pcm, hw, 4);
```

and use the interleaved 4 channel data.

There are some control switches affecting to the speaker connections:

Line-In Mode an enum control to change the behavior of line-in jack. Either "Line-In", "Rear Output" or "Bass Output" can be selected. The last item is available only with model 039 or newer. When "Rear Output" is chosen, the surround channels 3 and 4 are output to line-in jack.

Mic-In Mode an enum control to change the behavior of mic-in jack. Either "Mic-In" or "Center/LFE Output" can be selected. When "Center/LFE Output" is chosen, the center and bass channels (channels 5 and 6) are output to mic-in jack.

Digital I/O

The CM8x38 provides the excellent SPDIF capability with very cheap price (yes, that's the reason I bought the card :)

The SPDIF playback and capture are done via the third PCM device (hw:0,2). Usually this is assigned to the PCM device "spdif". The available rates are 44100 and 48000 Hz. For playback with aplay, you can run like below:

```
% aplay -Dhw:0,2 foo.wav
```

or

```
% aplay -Dspdif foo.wav
```

24bit format is also supported experimentally.

The playback and capture over SPDIF use normal DAC and ADC, respectively, so you cannot playback both analog and digital streams simultaneously.

To enable SPDIF output, you need to turn on "IEC958 Output Switch" control via mixer or alsactl ("IEC958" is the official name of so-called S/PDIF). Then you'll see the red light on from the card so you know that's working obviously :) The SPDIF input is always enabled, so you can hear SPDIF input data from line-out with "IEC958 In Monitor" switch at any time (see below).

You can play via SPDIF even with the first device (hw:0,0), but SPDIF is enabled only when the proper format (S16LE), sample rate (441100 or 48000) and channels (2) are used. Otherwise it's turned off. (Also don't forget to turn on "IEC958 Output Switch", too.)

Additionally there are relevant control switches:

IEC958 Mix Analog Mix analog PCM playback and FM-OPL/3 streams and output through SPDIF. This switch appears only on old chip models (CM8738 033 and 037).

Note: without this control you can output PCM to SPDIF. This is "mixing" of streams, so e.g. it's not for AC3 output (see the next section).

IEC958 In Select Select SPDIF input, the internal CD-in (false) and the external input (true).

IEC958 Loop SPDIF input data is loop back into SPDIF output (aka bypass)

IEC958 Copyright Set the copyright bit.

IEC958 5V Select 0.5V (coax) or 5V (optical) interface. On some cards this doesn't work and you need to change the configuration with hardware dip-switch.

IEC958 In Monitor SPDIF input is routed to DAC.

IEC958 In Phase Inverse Set SPDIF input format as inverse. [FIXME: this doesn't work on all chips..]

IEC958 In Valid Set input validity flag detection.

Note: When "PCM Playback Switch" is on, you'll hear the digital output stream through analog line-out.

The AC3 (RAW DIGITAL) OUTPUT

The driver supports raw digital (typically AC3) i/o over SPDIF. This can be toggled via IEC958 playback control, but usually you need to access it via alsa-lib. See alsa-lib documents for more details.

On the raw digital mode, the "PCM Playback Switch" is automatically turned off so that non-audio data is heard from the analog line-out. Similarly the following switches are off: "IEC958 Mix Analog" and "IEC958 Loop". The switches are resumed after closing the SPDIF PCM device automatically to the previous state.

On the model 033, AC3 is implemented by the software conversion in the alsa-lib. If you need to bypass the software conversion of IEC958 subframes, pass the "soft_ac3=0" module option. This doesn't matter on the newer models.

ANALOG MIXER INTERFACE

The mixer interface on CM8x38 is similar to SB16. There are Master, PCM, Synth, CD, Line, Mic and PC Speaker playback volumes. Synth, CD, Line and Mic have playback and capture switches, too, as well as SB16.

In addition to the standard SB mixer, CM8x38 provides more functions. - PCM playback switch - PCM capture switch (to capture the data sent to DAC) - Mic Boost switch - Mic capture volume - Aux playback volume/switch and capture switch - 3D control switch

MIDI CONTROLLER

With CMI8338 chips, the MPU401-UART interface is disabled as default. You need to set the module option "mpu_port" to a valid I/O port address to enable MIDI support. Valid I/O ports are 0x300, 0x310, 0x320 and 0x330. Choose a value that doesn't conflict with other cards.

With CMI8738 and newer chips, the MIDI interface is enabled by default and the driver automatically chooses a port address.

There is *no* hardware wavetable function on this chip (except for OPL3 synth below). What's said as MIDI synth on Windows is a software synthesizer emulation. On Linux use TiMidity or other softsynth program for playing MIDI music.

FM OPL/3 Synth

The FM OPL/3 is also enabled as default only for the first card. Set "fm_port" module option for more cards.

The output quality of FM OPL/3 is, however, very weird. I don't know why..

CMI8768 and newer chips do not have the FM synth.

Joystick and Modem

The legacy joystick is supported. To enable the joystick support, pass joystick_port=1 module option. The value 1 means the auto-detection. If the auto-detection fails, try to pass the exact I/O address.

The modem is enabled dynamically via a card control switch "Modem".

Debugging Information

The registers are shown in /proc/asound/cardX/cmipci. If you have any problem (especially unexpected behavior of mixer), please attach the output of this proc file together with the bug report.

Sound Blaster Live mixer / default DSP code

The EMU10K1 chips have a DSP part which can be programmed to support various ways of sample processing, which is described here. (This article does not deal with the overall functionality of the EMU10K1 chips. See the manuals section for further details.)

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

IEC958 (S/PDIF) raw PCM

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

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

Digital mixer controls

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

Explanation of used abbreviations:

DAC digital to analog converter

ADC analog to digital converter

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

LFE low frequency effects (subwoofer signal)

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

IEC958 S/PDIF

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

`name='Wave Playback Volume',index=0`

This control is used to attenuate samples for left and right PCM FX-bus accumulators. ALSA uses accumulators 0 and 1 for left and right PCM samples. The result samples are forwarded to the front DAC PCM slots of the AC97 codec.

`name='Wave Surround Playback Volume',index=0`

This control is used to attenuate samples for left and right PCM FX-bus accumulators. ALSA uses accumulators 0 and 1 for left and right PCM samples. The result samples are forwarded to the rear I2S DACs. These DACs operates separately (they are not inside the AC97 codec).

`name='Wave Center Playback Volume',index=0`

This control is used to attenuate samples for left and right PCM FX-bus accumulators. ALSA uses accumulators 0 and 1 for left and right PCM samples. The result is mixed to mono signal (single channel) and forwarded to the ??rear?? right DAC PCM slot of the AC97 codec.

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

This control is used to attenuate samples for left and right PCM FX-bus accumulators. ALSA uses accumulators 0 and 1 for left and right PCM. The result is mixed to mono signal (single channel) and forwarded to the ??rear?? left DAC PCM slot of the AC97 codec.

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

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

name='Synth Playback Volume',index=0

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

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

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

name='Surround Playback Volume',index=0

This control is used to attenuate samples for left and right rear PCM FX-bus accumulators. ALSA uses accumulators 2 and 3 for left and right rear PCM samples. The result samples are forwarded to the rear I2S DACs. These DACs operate separately (they are not inside the AC97 codec).

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

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

name='Center Playback Volume',index=0

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

name='LFE Playback Volume',index=0

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

name='AC97 Playback Volume',index=0

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

Note:

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

name='AC97 Capture Volume',index=0

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

Note:

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

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

This control is used to attenuate samples from left and right IEC958 TTL digital inputs (usually used by a CDROM drive). The result samples are forwarded to the front DAC PCM slots of the AC97 codec.

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

name='Headphone Playback Volume',index=1

This control attenuates the samples for the headphone output.

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

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

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

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

PCM stream related controls

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

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

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

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

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

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

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

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

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

- 0 - mono, A destination attn, default 255 (no attenuation)
- 1 - mono, B destination attn, default 255 (no attenuation)
- 2 - mono, C destination attn, default 0 (mute)
- 3 - mono, D destination attn, default 0 (mute)
- 4 - left, A destination attn, default 255 (no attenuation)
- 5 - left, B destination attn, default 0 (mute)
- 6 - left, C destination attn, default 0 (mute)

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

MANUALS/PATENTS

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

LM4545.pdf AC97 Codec

m2049.pdf The EMU10K1 Digital Audio Processor

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

WIPO Patents

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

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

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

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

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

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

US 5930158 Processor with Instruction Set for Audio Effects (Jul. 27, 1999)

US 6032235 Memory initialization circuit (Tram) (Feb. 29, 2000)

US 6138207 Interpolation looping of audio samples in cache connected to system bus with prioritization and modification of bus transfers in accordance with loop ends and minimum block sizes (Oct. 24, 2000)

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

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

Sound Blaster Audigy mixer / default DSP code

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

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

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

Digital mixer controls

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

Explanation of used abbreviations:

DAC digital to analog converter

ADC analog to digital converter

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

LFE low frequency effects (subwoofer signal)

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

IEC958 S/PDIF

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

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

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

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

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

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

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

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

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

name='PCM Playback Volume',index=0

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

name='PCM Capture Volume',index=0

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

name='Music Playback Volume',index=0

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

name='Music Capture Volume',index=0

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

name='Mic Playback Volume',index=0

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

name='Mic Capture Volume',index=0

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

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

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

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

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

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

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

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

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

name='Line2 Playback Volume',index=0

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

name='Line2 Capture Volume',index=1

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

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

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

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

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

name='Aux2 Playback Volume',index=0

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

name='Aux2 Capture Volume',index=1

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

name='Front Playback Volume',index=0

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

name='Surround Playback Volume',index=0

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

name='Center Playback Volume',index=0

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

name='LFE Playback Volume',index=0

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

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

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

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

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

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

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

name='Master Playback Volume',index=0

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

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

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

PCM stream related controls

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

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

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

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

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

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

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

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

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

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

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

- 19 - right, D destination attn, default 0 (mute)
- 20 - right, E destination attn, default 0 (mute)
- 21 - right, F destination attn, default 0 (mute)
- 22 - right, G destination attn, default 0 (mute)
- 23 - right, H destination attn, default 0 (mute)

MANUALS/PATENTS

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

LM4545.pdf AC97 Codec

m2049.pdf The EMU10K1 Digital Audio Processor

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

WIPO Patents

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

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

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

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

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

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

US 5930158 Processor with Instruction Set for Audio Effects (Jul. 27, 1999)

US 6032235 Memory initialization circuit (Tram) (Feb. 29, 2000)

US 6138207 Interpolation looping of audio samples in cache connected to system bus with prioritization and modification of bus transfers in accordance with loop ends and minimum block sizes (Oct. 24, 2000)

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

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

Low latency, multichannel audio with JACK and the emu10k1/emu10k2

This document is a guide to using the emu10k1 based devices with JACK for low latency, multichannel recording functionality. All of my recent work to allow Linux users to use the full capabilities of their hardware has been inspired by the kX Project. Without their work I never would have discovered the true power of this hardware.

http://www.kxproject.com

- Lee Revell, 2005.03.30

Until recently, emu10k1 users on Linux did not have access to the same low latency, multichannel features offered by the "kX ASIO" feature of their Windows driver. As of ALSA 1.0.9 this is no more!

For those unfamiliar with kX ASIO, this consists of 16 capture and 16 playback channels. With a post 2.6.9 Linux kernel, latencies down to 64 (1.33 ms) or even 32 (0.66ms) frames should work well.

The configuration is slightly more involved than on Windows, as you have to select the correct device for JACK to use. Actually, for qjackctl users it's fairly self explanatory - select Duplex, then for capture and playback select the multichannel devices, set the in and out channels to 16, and the sample rate to 48000Hz. The command line looks like this:

```
/usr/local/bin/jackd -R -dalsa -r48000 -p64 -n2 -D -Chw:0,2 -Phw:0,3 -S
```

This will give you 16 input ports and 16 output ports.

The 16 output ports map onto the 16 FX buses (or the first 16 of 64, for the Audigy). The mapping from FX bus to physical output is described in sb-live-mixer.rst (or audigy-mixer.rst).

The 16 input ports are connected to the 16 physical inputs. Contrary to popular belief, all emu10k1 cards are multichannel cards. Which of these input channels have physical inputs connected to them depends on the card model. Trial and error is highly recommended; the pinout diagrams for the card have been reverse engineered by some enterprising kX users and are available on the internet. Meterbridge is helpful here, and the kX forums are packed with useful information.

Each input port will either correspond to a digital (SPDIF) input, an analog input, or nothing. The one exception is the SBLive! 5.1. On these devices, the second and third input ports are wired to the center/LFE output. You will still see 16 capture channels, but only 14 are available for recording inputs.

This chart, borrowed from kxfxlib/da_asio51.cpp, describes the mapping of JACK ports to FXBUS2 (multi-track recording input) and EXTOUT (physical output) channels.

JACK (& ASIO) mappings on 10k1 5.1 SBLive cards:

JACK	Epilog	FXBUS2(nr)
capture_1	asio14	FXBUS2(0xe)
capture_2	asio15	FXBUS2(0xf)
capture_3	asio0	FXBUS2(0x0)
~capture_4	Center	EXTOUT(0x11) // mapped to by Center
~capture_5	LFE	EXTOUT(0x12) // mapped to by LFE
capture_6	asio3	FXBUS2(0x3)
capture_7	asio4	FXBUS2(0x4)
capture_8	asio5	FXBUS2(0x5)
capture_9	asio6	FXBUS2(0x6)
capture_10	asio7	FXBUS2(0x7)
capture_11	asio8	FXBUS2(0x8)
capture_12	asio9	FXBUS2(0x9)
capture_13	asio10	FXBUS2(0xa)
capture_14	asio11	FXBUS2(0xb)
capture_15	asio12	FXBUS2(0xc)
capture_16	asio13	FXBUS2(0xd)

TODO: describe use of ld10k1/qlo10k1 in conjunction with JACK

VIA82xx mixer

On many VIA82xx boards, the `Input Source Select` mixer control does not work. Setting it to `Input2` on such boards will cause recording to hang, or fail with EIO (input/output error) via OSS emulation. This control should be left at `Input1` for such cards.

Guide to using M-Audio Audiophile USB with ALSA and Jack

v1.5

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This document is a guide to using the M-Audio Audiophile USB (tm) device with ALSA and JACK.

History

- v1.4 - Thibault Le Meur (2007-07-11)
  - Added Low Endianness nature of 16bits-modes found by Hakan Lennestal <Hakan.Lennestal@brfsodrahamn.se>
  - Modifying document structure
- v1.5 - Thibault Le Meur (2007-07-12) - Added AC3/DTS passthru info

Audiophile USB Specs and correct usage

This part is a reminder of important facts about the functions and limitations of the device.

The device has 4 audio interfaces, and 2 MIDI ports:

- Analog Stereo Input (Ai)
  - This port supports 2 pairs of line-level audio inputs (1/4” TS and RCA)
  - When the 1/4” TS (jack) connectors are connected, the RCA connectors are disabled
- Analog Stereo Output (Ao)
- Digital Stereo Input (Di)
- Digital Stereo Output (Do)
- Midi In (Mi)
- Midi Out (Mo)

The internal DAC/ADC has the following characteristics:

- sample depth of 16 or 24 bits
- sample rate from 8kHz to 96kHz
- Two interfaces can’t use different sample depths at the same time.

Moreover, the Audiophile USB documentation gives the following Warning: Please exit any audio application running before switching between bit depths

Due to the USB 1.1 bandwidth limitation, a limited number of interfaces can be activated at the same time depending on the audio mode selected:

- 16-bit/48kHz ==> 4 channels in + 4 channels out
  - Ai+Ao+Di+Do
- 24-bit/48kHz ==> 4 channels in + 2 channels out, or 2 channels in + 4 channels out
  - Ai+Ao+Do or Ai+Di+Ao or Ai+Di+Do or Di+Ao+Do
- 24-bit/96kHz ==> 2 channels in _or_ 2 channels out (half duplex only)
  - Ai or Ao or Di or Do

Important facts about the Digital interface:

- The Do port additionally supports surround-encoded AC-3 and DTS passthrough, though I haven't tested it under Linux
  - Note that in this setup only the Do interface can be enabled
- Apart from recording an audio digital stream, enabling the Di port is a way to synchronize the device to an external sample clock
  - As a consequence, the Di port must be enable only if an active Digital source is connected
  - Enabling Di when no digital source is connected can result in a synchronization error (for instance sound played at an odd sample rate)

Audiophile USB MIDI support in ALSA

The Audiophile USB MIDI ports will be automatically supported once the following modules have been loaded:

- snd-usb-audio
- snd-seq-midi

No additional setting is required.

Audiophile USB Audio support in ALSA

Audio functions of the Audiophile USB device are handled by the snd-usb-audio module. This module can work in a default mode (without any device-specific parameter), or in an "advanced" mode with the device-specific parameter called `device_setup`.

Default Alsa driver mode

The default behavior of the snd-usb-audio driver is to list the device capabilities at startup and activate the required mode when required by the applications: for instance if the user is recording in a 24bit-depth-mode and immediately after wants to switch to a 16bit-depth mode, the snd-usb-audio module will reconfigure the device on the fly.

This approach has the advantage to let the driver automatically switch from sample rates/depths automatically according to the user's needs. However, those who are using the device under windows know that this is not how the device is meant to work: under windows applications must be closed before using the m-audio control panel to switch the device working mode. Thus as we'll see in next section, this Default Alsa driver mode can lead to device misconfigurations.

Let's get back to the Default Alsa driver mode for now. In this case the Audiophile interfaces are mapped to alsa pcm devices in the following way (I suppose the device's index is 1):

- hw:1,0 is Ao in playback and Di in capture
- hw:1,1 is Do in playback and Ai in capture
- hw:1,2 is Do in AC3/DTS passthrough mode

In this mode, the device uses Big Endian byte-encoding so that supported audio format are S16_BE for 16-bit depth modes and S24_3BE for 24-bits depth mode.

One exception is the hw:1,2 port which was reported to be Little Endian compliant (supposedly supporting S16_LE) but processes in fact only S16_BE streams. This has been fixed in kernel 2.6.23 and above and now the hw:1,2 interface is reported to be big endian in this default driver mode.

Examples:

- playing a S24_3BE encoded raw file to the Ao port:

```
% aplay -D hw:1,0 -c2 -t raw -r48000 -fS24_3BE test.raw
```

- recording a S24_3BE encoded raw file from the Ai port:

```
% arecord -D hw:1,1 -c2  -t raw -r48000 -fS24_3BE test.raw
```

- playing a S16_BE encoded raw file to the Do port:

```
% aplay -D hw:1,1 -c2 -t raw -r48000 -fS16_BE test.raw
```

- playing an ac3 sample file to the Do port:

```
% aplay -D hw:1,2 --channels=6 ac3_S16_BE_encoded_file.raw
```

If you're happy with the default Alsa driver mode and don't experience any issue with this mode, then you can skip the following chapter.

Advanced module setup

Due to the hardware constraints described above, the device initialization made by the Alsa driver in default mode may result in a corrupted state of the device. For instance, a particularly annoying issue is that the sound captured from the Ai interface sounds distorted (as if boosted with an excessive high volume gain).

For people having this problem, the snd-usb-audio module has a new module parameter called `device_setup` (this parameter was introduced in kernel release 2.6.17)

Initializing the working mode of the Audiophile USB

As far as the Audiophile USB device is concerned, this value let the user specify:

- the sample depth
- the sample rate
- whether the Di port is used or not

When initialized with `device_setup=0x00`, the snd-usb-audio module has the same behaviour as when the parameter is omitted (see paragraph "Default Alsa driver mode" above)

Others modes are described in the following subsections.

16-bit modes

The two supported modes are:

- `device_setup=0x01`
  - 16bits 48kHz mode with Di disabled
  - Ai,Ao,Do can be used at the same time
  - hw:1,0 is not available in capture mode
  - hw:1,2 is not available
- `device_setup=0x11`
  - 16bits 48kHz mode with Di enabled
  - Ai,Ao,Di,Do can be used at the same time
  - hw:1,0 is available in capture mode

- hw:1,2 is not available

In this modes the device operates only at 16bits-modes. Before kernel 2.6.23, the devices where reported to be Big-Endian when in fact they were Little-Endian so that playing a file was a matter of using:

```
% aplay -D hw:1,1 -c2 -t raw -r48000 -fS16_BE test_S16_LE.raw
```

where “test_S16_LE.raw” was in fact a little-endian sample file.

Thanks to Hakan Lennestal (who discovered the Little-Endiannes of the device in these modes) a fix has been committed (expected in kernel 2.6.23) and Alsa now reports Little-Endian interfaces. Thus playing a file now is as simple as using:

```
% aplay -D hw:1,1 -c2 -t raw -r48000 -fS16_LE test_S16_LE.raw
```

24-bit modes

The three supported modes are:

- `device_setup=0x09`
  - 24bits 48kHz mode with Di disabled
  - Ai,Ao,Do can be used at the same time
  - hw:1,0 is not available in capture mode
  - hw:1,2 is not available
- `device_setup=0x19`
  - 24bits 48kHz mode with Di enabled
  - 3 ports from {Ai,Ao,Di,Do} can be used at the same time
  - hw:1,0 is available in capture mode and an active digital source must be connected to Di
  - hw:1,2 is not available
- `device_setup=0x0D` or `0x10`
  - 24bits 96kHz mode
  - Di is enabled by default for this mode but does not need to be connected to an active source
  - Only 1 port from {Ai,Ao,Di,Do} can be used at the same time
  - hw:1,0 is available in captured mode
  - hw:1,2 is not available

In these modes the device is only Big-Endian compliant (see “Default Alsa driver mode” above for an aplay command example)

AC3 w/ DTS passthru mode

Thanks to Hakan Lennestal, I now have a report saying that this mode works.

- `device_setup=0x03`
  - 16bits 48kHz mode with only the Do port enabled
  - AC3 with DTS passthru
  - Caution with this setup the Do port is mapped to the pcm device hw:1,0

The command line used to playback the AC3/DTS encoded .wav-files in this mode:

```
% aplay -D hw:1,0 --channels=6 ac3_S16_LE_encoded_file.raw
```

How to use the device_setup parameter

The parameter can be given:

- By manually probing the device (as root)::

```
    # modprobe -r snd-usb-audio
    # modprobe snd-usb-audio index=1 device_setup=0x09
```

- Or while configuring the modules options in your modules configuration file (typically a .conf file in /etc/modprobe.d/ directory::

```
    alias snd-card-1 snd-usb-audio
    options snd-usb-audio index=1 device_setup=0x09
```

CAUTION when initializing the device

- Correct initialization on the device requires that device_setup is given to the module BEFORE the device is turned on. So, if you use the "manual probing" method described above, take care to power-on the device AFTER this initialization.
- Failing to respect this will lead to a misconfiguration of the device. In this case turn off the device, unprobe the snd-usb-audio module, then probe it again with correct device_setup parameter and then (and only then) turn on the device again.
- If you've correctly initialized the device in a valid mode and then want to switch to another mode (possibly with another sample-depth), please use also the following procedure:
  - first turn off the device
  - de-register the snd-usb-audio module (modprobe -r)
  - change the device_setup parameter by changing the device_setup option in `/etc/modprobe.d/*.conf`
  - turn on the device
- A workaround for this last issue has been applied to kernel 2.6.23, but it may not be enough to ensure the 'stability' of the device initialization.

Technical details for hackers

This section is for hackers, wanting to understand details about the device internals and how Alsa supports it.

Audiophile USB's device_setup structure

If you want to understand the device_setup magic numbers for the Audiophile USB, you need some very basic understanding of binary computation. However, this is not required to use the parameter and you may skip this section.

The device_setup is one byte long and its structure is the following:

```
+---+---+---+---+---+---+---+---+
| b7| b6| b5| b4| b3| b2| b1| b0|
+---+---+---+---+---+---+---+---+
| 0 | 0 | 0 | Di|24B|96K|DTS|SET|
+---+---+---+---+---+---+---+---+
```

Where:

- b0 is the SET bit
  - it MUST be set if device_setup is initialized
- b1 is the DTS bit
  - it is set only for Digital output with DTS/AC3
  - this setup is not tested
- b2 is the Rate selection flag
  - When set to 1 the rate range is 48.1-96kHz
  - Otherwise the sample rate range is 8-48kHz
- b3 is the bit depth selection flag
  - When set to 1 samples are 24bits long
  - Otherwise they are 16bits long
  - Note that b2 implies b3 as the 96kHz mode is only supported for 24 bits samples
- b4 is the Digital input flag
  - When set to 1 the device assumes that an active digital source is connected
  - You shouldn't enable Di if no source is seen on the port (this leads to synchronization issues)
  - b4 is implied by b2 (since only one port is enabled at a time no synch error can occur)
- b5 to b7 are reserved for future uses, and must be set to `0`
  - might become Ao, Do, Ai, for b7, b6, b4 respectively

Caution:

- there is no check on the value you will give to device_setup
  - for instance choosing 0x05 (16bits 96kHz) will fail back to 0x09 since b2 implies b3. But _there_will_be_no_warning_ in /var/log/messages
- Hardware constraints due to the USB bus limitation aren't checked
  - choosing b2 will prepare all interfaces for 24bits/96kHz but you'll only be able to use one at the same time

USB implementation details for this device

You may safely skip this section if you're not interested in driver hacking.

This section describes some internal aspects of the device and summarizes the data I got by usb-snooping the windows and Linux drivers.

The M-Audio Audiophile USB has 7 USB Interfaces: a "USB interface":

- USB Interface nb.0
- USB Interface nb.1
  - Audio Control function
- USB Interface nb.2
  - Analog Output
- USB Interface nb.3
  - Digital Output
- USB Interface nb.4

  - Analog Input
- USB Interface nb.5
  - Digital Input
- USB Interface nb.6
  - MIDI interface compliant with the MIDIMAN quirk

Each interface has 5 altsettings (AltSet 1,2,3,4,5) except:

- Interface 3 (Digital Out) has an extra Alset nb.6
- Interface 5 (Digital In) does not have Alset nb.3 and 5

Here is a short description of the AltSettings capabilities:

- AltSettings 1 corresponds to
  - 24-bit depth, 48.1-96kHz sample mode
  - Adaptive playback (Ao and Do), Synch capture (Ai), or Asynch capture (Di)
- AltSettings 2 corresponds to
  - 24-bit depth, 8-48kHz sample mode
  - Asynch capture and playback (Ao,Ai,Do,Di)
- AltSettings 3 corresponds to
  - 24-bit depth, 8-48kHz sample mode
  - Synch capture (Ai) and Adaptive playback (Ao,Do)
- AltSettings 4 corresponds to
  - 16-bit depth, 8-48kHz sample mode
  - Asynch capture and playback (Ao,Ai,Do,Di)
- AltSettings 5 corresponds to
  - 16-bit depth, 8-48kHz sample mode
  - Synch capture (Ai) and Adaptive playback (Ao,Do)
- AltSettings 6 corresponds to
  - 16-bit depth, 8-48kHz sample mode
  - Synch playback (Do), audio format type III IEC1937_AC-3

In order to ensure a correct initialization of the device, the driver *must know* how the device will be used:

- if DTS is chosen, only Interface 2 with AltSet nb.6 must be registered
- if 96KHz only AltSets nb.1 of each interface must be selected
- if samples are using 24bits/48KHz then AltSet 2 must me used if Digital input is connected, and only AltSet nb.3 if Digital input is not connected
- if samples are using 16bits/48KHz then AltSet 4 must me used if Digital input is connected, and only AltSet nb.5 if Digital input is not connected

When device_setup is given as a parameter to the snd-usb-audio module, the parse_audio_endpoints function uses a quirk called `audiophile_skip_setting_quirk` in order to prevent AltSettings not corresponding to device_setup from being registered in the driver.

Audiophile USB and Jack support

This section deals with support of the Audiophile USB device in Jack.

There are 2 main potential issues when using Jackd with the device:

- support for Big-Endian devices in 24-bit modes
- support for 4-in / 4-out channels

Direct support in Jackd

Jack supports big endian devices only in recent versions (thanks to Andreas Steinmetz for his first big-endian patch). I can't remember exactly when this support was released into jackd, let's just say that with jackd version 0.103.0 it's almost ok (just a small bug is affecting 16bits Big-Endian devices, but since you've read carefully the above paragraphs, you're now using kernel >= 2.6.23 and your 16bits devices are now Little Endians ;-)).

You can run jackd with the following command for playback with Ao and record with Ai:

```
% jackd -R -dalsa -Phw:1,0 -r48000 -p128 -n2 -D -Chw:1,1
```

Using Alsa plughw

If you don't have a recent Jackd installed, you can downgrade to using the Alsa `plug` converter.

For instance here is one way to run Jack with 2 playback channels on Ao and 2 capture channels from Ai:

```
% jackd -R -dalsa -dplughw:1 -r48000 -p256 -n2 -D -Cplughw:1,1
```

However you may see the following warning message: You appear to be using the ALSA software "plug" layer, probably a result of using the "default" ALSA device. This is less efficient than it could be. Consider using a hardware device instead rather than using the plug layer.

Getting 2 input and/or output interfaces in Jack

As you can see, starting the Jack server this way will only enable 1 stereo input (Di or Ai) and 1 stereo output (Ao or Do).

This is due to the following restrictions:

- Jack can only open one capture device and one playback device at a time
- The Audiophile USB is seen as 2 (or three) Alsa devices: hw:1,0, hw:1,1 (and optionally hw:1,2)

If you want to get Ai+Di and/or Ao+Do support with Jack, you would need to combine the Alsa devices into one logical "complex" device.

If you want to give it a try, I recommend reading the information from this page: http://www.sound-man.co.uk/linuxaudio/ice1712multi.html It is related to another device (ice1712) but can be adapted to suit the Audiophile USB.

Enabling multiple Audiophile USB interfaces for Jackd will certainly require:

- Making sure your Jackd version has the MMAP_COMPLEX patch (see the ice1712 page)
- (maybe) patching the alsa-lib/src/pcm/pcm_multi.c file (see the ice1712 page)
- define a multi device (combination of hw:1,0 and hw:1,1) in your .asoundrc file
- start jackd with this device

I had no success in testing this for now, if you have any success with this kind of setup, please drop me an email.

Alsa driver for Digigram miXart8 and miXart8AES/EBU soundcards

Digigram <alsa@digigram.com>

GENERAL

The miXart8 is a multichannel audio processing and mixing soundcard that has 4 stereo audio inputs and 4 stereo audio outputs. The miXart8AES/EBU is the same with a add-on card that offers further 4 digital stereo audio inputs and outputs. Furthermore the add-on card offers external clock synchronisation (AES/EBU, Word Clock, Time Code and Video Synchro)

The mainboard has a PowerPC that offers onboard mpeg encoding and decoding, samplerate conversions and various effects.

The driver don't work properly at all until the certain firmwares are loaded, i.e. no PCM nor mixer devices will appear. Use the mixartloader that can be found in the alsa-tools package.

VERSION 0.1.0

One miXart8 board will be represented as 4 alsa cards, each with 1 stereo analog capture 'pcm0c' and 1 stereo analog playback 'pcm0p' device. With a miXart8AES/EBU there is in addition 1 stereo digital input 'pcm1c' and 1 stereo digital output 'pcm1p' per card.

Formats

U8, S16_LE, S16_BE, S24_3LE, S24_3BE, FLOAT_LE, FLOAT_BE Sample rates : 8000 - 48000 Hz continuously

Playback

For instance the playback devices are configured to have max. 4 substreams performing hardware mixing. This could be changed to a maximum of 24 substreams if wished. Mono files will be played on the left and right channel. Each channel can be muted for each stream to use 8 analog/digital outputs separately.

Capture

There is one substream per capture device. For instance only stereo formats are supported.

Mixer

<Master> and <Master Capture> analog volume control of playback and capture PCM.

<PCM 0-3> and <PCM Capture> digital volume control of each analog substream.

<AES 0-3> and <AES Capture> digital volume control of each AES/EBU substream.

<Monitoring> Loopback from 'pcm0c' to 'pcm0p' with digital volume and mute control.

Rem : for best audio quality try to keep a 0 attenuation on the PCM and AES volume controls which is set by 219 in the range from 0 to 255 (about 86% with alsamixer)

NOT YET IMPLEMENTED

- external clock support (AES/EBU, Word Clock, Time Code, Video Sync)
- MPEG audio formats
- mono record
- on-board effects and samplerate conversions
- linked streams

FIRMWARE

[As of 2.6.11, the firmware can be loaded automatically with hotplug when CONFIG_FW_LOADER is set. The mixartloader is necessary only for older versions or when you build the driver into kernel.]

For loading the firmware automatically after the module is loaded, use a install command. For example, add the following entry to /etc/modprobe.d/mixart.conf for miXart driver:

```
install snd-mixart /sbin/modprobe --first-time -i snd-mixart && \
                   /usr/bin/mixartloader
```

(for 2.2/2.4 kernels, add "post-install snd-mixart /usr/bin/vxloader" to /etc/modules.conf, instead.)

The firmware binaries are installed on /usr/share/alsa/firmware (or /usr/local/share/alsa/firmware, depending to the prefix option of configure). There will be a miXart.conf file, which define the dsp image files.

The firmware files are copyright by Digigram SA

COPYRIGHT

ALSA BT87x Driver

Intro

You might have noticed that the bt878 grabber cards have actually *two* PCI functions:

```
$ lspci
[ ... ]
00:0a.0 Multimedia video controller: Brooktree Corporation Bt878 (rev 02)
00:0a.1 Multimedia controller: Brooktree Corporation Bt878 (rev 02)
[ ... ]
```

The first does video, it is backward compatible to the bt848. The second does audio. snd-bt87x is a driver for the second function. It's a sound driver which can be used for recording sound (and *only* recording, no playback). As most TV cards come with a short cable which can be plugged into your sound card's line-in you probably don't need this driver if all you want to do is just watching TV...

Some cards do not bother to connect anything to the audio input pins of the chip, and some other cards use the audio function to transport MPEG video data, so it's quite possible that audio recording may not work with your card.

Driver Status

The driver is now stable. However, it doesn't know about many TV cards, and it refuses to load for cards it doesn't know.

If the driver complains ("Unknown TV card found, the audio driver will not load"), you can specify the `load_all=1` option to force the driver to try to use the audio capture function of your card. If the frequency of recorded data is not right, try to specify the `digital_rate` option with other values than the default 32000 (often it's 44100 or 64000).

If you have an unknown card, please mail the ID and board name to <alsa-devel@alsa-project.org>, regardless of whether audio capture works or not, so that future versions of this driver know about your card.

Audio modes

The chip knows two different modes (digital/analog). snd-bt87x registers two PCM devices, one for each mode. They cannot be used at the same time.

Digital audio mode

The first device (hw:X,0) gives you 16 bit stereo sound. The sample rate depends on the external source which feeds the Bt87x with digital sound via I2S interface.

Analog audio mode (A/D)

The second device (hw:X,1) gives you 8 or 16 bit mono sound. Supported sample rates are between 119466 and 448000 Hz (yes, these numbers are that high). If you've set the CONFIG_SND_BT87X_OVERCLOCK option, the maximum sample rate is 1792000 Hz, but audio data becomes unusable beyond 896000 Hz on my card.

The chip has three analog inputs. Consequently you'll get a mixer device to control these.

Have fun,

Clemens

Written by Clemens Ladisch <clemens@ladisch.de> big parts copied from btaudio.txt by Gerd Knorr <kraxel@bytesex.org>

Notes on Maya44 USB Audio Support

Note:

The following is the original document of Rainer's patch that the current maya44 code based on. Some contents might be obsoleted, but I keep here as reference – tiwai

Feb 14, 2008

Rainer Zimmermann <mail@lightshed.de>

STATE OF DEVELOPMENT

This driver is being developed on the initiative of Piotr Makowski (oponek@gmail.com) and financed by Lars Bergmann. Development is carried out by Rainer Zimmermann (mail@lightshed.de).

ESI provided a sample Maya44 card for the development work.

However, unfortunately it has turned out difficult to get detailed programming information, so I (Rainer Zimmermann) had to find out some card-specific information by experiment and conjecture. Some information (in particular, several GPIO bits) is still missing.

This is the first testing version of the Maya44 driver released to the alsa-devel mailing list (Feb 5, 2008).

The following functions work, as tested by Rainer Zimmermann and Piotr Makowski:

- playback and capture at all sampling rates
- input/output level
- crossmixing
- line/mic switch
- phantom power switch
- analogue monitor a.k.a bypass

The following functions *should* work, but are not fully tested:

- Channel 3+4 analogue - S/PDIF input switching
- S/PDIF output
- all inputs/outputs on the M/IO/DIO extension card
- internal/external clock selection

In particular, we would appreciate testing of these functions by anyone who has access to an M/IO/DIO extension card.

Things that do not seem to work:

- The level meters (“multi track”) in ‘alsamixer’ do not seem to react to signals in (if this is a bug, it would probably be in the existing ICE1724 code).
- Ardour 2.1 seems to work only via JACK, not using ALSA directly or via OSS. This still needs to be tracked down.

DRIVER DETAILS

the following files were added:

- pci/ice1724/maya44.c - Maya44 specific code
- pci/ice1724/maya44.h
- pci/ice1724/ice1724.patch
- pci/ice1724/ice1724.h.patch - PROPOSED patch to ice1724.h (see SAMPLING RATES)
- i2c/other/wm8776.c - low-level access routines for Wolfson WM8776 codecs
- include/wm8776.h

Note that the wm8776.c code is meant to be card-independent and does not actually register the codec with the ALSA infrastructure. This is done in maya44.c, mainly because some of the WM8776 controls are used in Maya44-specific ways, and should be named appropriately.

the following files were created in pci/ice1724, simply #including the corresponding file from the alsa-kernel tree:

- wtm.h
- vt1720_mobo.h
- revo.h
- prodigy192.h
- pontis.h
- phase.h
- maya44.h
- juli.h
- aureon.h
- amp.h
- envy24ht.h
- se.h
- prodigy_hifi.h

I hope this is the correct way to do things.

SAMPLING RATES

The Maya44 card (or more exactly, the Wolfson WM8776 codecs) allow a maximum sampling rate of 192 kHz for playback and 92 kHz for capture.

As the ICE1724 chip only allows one global sampling rate, this is handled as follows:

- setting the sampling rate on any open PCM device on the maya44 card will always set the *global* sampling rate for all playback and capture channels.
- In the current state of the driver, setting rates of up to 192 kHz is permitted even for capture devices.

AVOID CAPTURING AT RATES ABOVE 96kHz, even though it may appear to work. The codec cannot actually capture at such rates, meaning poor quality.

I propose some additional code for limiting the sampling rate when setting on a capture pcm device. However because of the global sampling rate, this logic would be somewhat problematic.

The proposed code (currently deactivated) is in ice1712.h.patch, ice1724.c and maya44.c (in pci/ice1712).

SOUND DEVICES

PCM devices correspond to inputs/outputs as follows (assuming Maya44 is card #0):

- hw:0,0 input - stereo, analog input 1+2
- hw:0,0 output - stereo, analog output 1+2
- hw:0,1 input - stereo, analog input 3+4 OR S/PDIF input
- hw:0,1 output - stereo, analog output 3+4 (and SPDIF out)

NAMING OF MIXER CONTROLS

(for more information about the signal flow, please refer to the block diagram on p.24 of the ESI Maya44 manual, or in the ESI windows software).

PCM (digital) output level for channel 1+2

PCM 1 same for channel 3+4

Mic Phantom+48V switch for +48V phantom power for electrostatic microphones on input 1/2.

Make sure this is not turned on while any other source is connected to input 1/2. It might damage the source and/or the maya44 card.

Mic/Line input if switch is on, input jack 1/2 is microphone input (mono), otherwise line input (stereo).

Bypass analogue bypass from ADC input to output for channel 1+2. Same as "Monitor" in the windows driver.

Bypass 1 same for channel 3+4.

Crossmix cross-mixer from channels 1+2 to channels 3+4

Crossmix 1 cross-mixer from channels 3+4 to channels 1+2

IEC958 Output switch for S/PDIF output.

This is not supported by the ESI windows driver. S/PDIF should output the same signal as channel 3+4. [untested!]

Digitial output selectors These switches allow a direct digital routing from the ADCs to the DACs. Each switch determines where the digital input data to one of the DACs comes from. They are not supported by the ESI windows driver. For normal operation, they should all be set to "PCM out".

H/W Output source channel 1

H/W 1 Output source channel 2

H/W 2 Output source channel 3

H/W 3 Output source channel 4

H/W 4 ... H/W 9 unknown function, left in to enable testing.

Possibly some of these control S/PDIF output(s). If these turn out to be unused, they will go away in later driver versions.

Selectable values for each of the digital output selectors are:

PCM out DAC output of the corresponding channel (default setting)

Input 1 ... Input 4 direct routing from ADC output of the selected input channel

Software Interface ALSA-DSP MADI Driver

(translated from German, so no good English ;-),

2004 - winfried ritsch

Full functionality has been added to the driver. Since some of the Controls and startup-options are ALSA-Standard and only the special Controls are described and discussed below.

Hardware functionality

Audio transmission

- number of channels – depends on transmission mode

  The number of channels chosen is from 1..Nmax. The reason to use for a lower number of channels is only resource allocation, since unused DMA channels are disabled and less memory is allocated. So also the throughput of the PCI system can be scaled. (Only important for low performance boards).

- Single Speed – 1..64 channels

Note:

(Note: Choosing the 56channel mode for transmission or as receiver, only 56 are transmitted/received over the MADI, but all 64 channels are available for the mixer, so channel count for the driver)

- Double Speed – 1..32 channels

Note:

Note: Choosing the 56-channel mode for transmission/receive-mode , only 28 are transmitted/received over the MADI, but all 32 channels are available for the mixer, so channel count for the driver

- Quad Speed – 1..16 channels

Note:

Choosing the 56-channel mode for transmission/receive-mode , only 14 are transmitted/received over the MADI, but all 16 channels are available for the mixer, so channel count for the driver

- Format – signed 32 Bit Little Endian (SNDRV_PCM_FMTBIT_S32_LE)
- Sample Rates –

  Single Speed – 32000, 44100, 48000

  Double Speed – 64000, 88200, 96000 (untested)

  Quad Speed – 128000, 176400, 192000 (untested)
- access-mode – MMAP (memory mapped), Not interleaved (PCM_NON-INTERLEAVED)
- buffer-sizes – 64,128,256,512,1024,2048,8192 Samples
- fragments – 2
- Hardware-pointer – 2 Modi

  The Card supports the readout of the actual Buffer-pointer, where DMA reads/writes. Since of the bulk mode of PCI it is only 64 Byte accurate. SO it is not really usable for the ALSA-mid-level functions (here the buffer-ID gives a better result), but if MMAP is used by the application. Therefore it can be configured at load-time with the parameter precise-pointer.

Hint:

(Hint: Experimenting I found that the pointer is maximum 64 to large never to small. So if you subtract 64 you always have a safe pointer for writing, which is used on this mode inside ALSA. In theory now you can get now a latency as low as 16 Samples, which is a quarter of the interrupt possibilities.)

- **Precise Pointer – off** *interrupt used for pointer-calculation*
- **Precise Pointer – on** *hardware pointer used.*

Controller

Since DSP-MADI-Mixer has 8152 Fader, it does not make sense to use the standard mixer-controls, since this would break most of (especially graphic) ALSA-Mixer GUIs. So Mixer control has be provided by a 2-dimensional controller using the hwdep-interface.

Also all 128+256 Peak and RMS-Meter can be accessed via the hwdep-interface. Since it could be a performance problem always copying and converting Peak and RMS-Levels even if you just need one, I decided to export the hardware structure, so that of needed some driver-guru can implement a memory-mapping of mixer or peak-meters over ioctl, or also to do only copying and no conversion. A test-application shows the usage of the controller.

- Latency Controls — not implemented !!!

Note:

Note: Within the windows-driver the latency is accessible of a control-panel, but buffer-sizes are controlled with ALSA from hwparams-calls and should not be changed in run-state, I did not implement it here.

- System Clock – suspended !!!!
  - Name – "System Clock Mode"
  - Access – Read Write
  - Values – "Master" "Slave"

Note:

!!!! This is a hardware-function but is in conflict with the Clock-source controller, which is a kind of ALSA-standard. I makes sense to set the card to a special mode (master at some frequency or slave), since even not using an Audio-application a studio should have working synchronisations setup. So use Clock-source-controller instead !!!!

- Clock Source
  - Name – "Sample Clock Source"
  - Access – Read Write
  - Values – "AutoSync", "Internal 32.0 kHz", "Internal 44.1 kHz", "Internal 48.0 kHz", "Internal 64.0 kHz", "Internal 88.2 kHz", "Internal 96.0 kHz"

    Choose between Master at a specific Frequency and so also the Speed-mode or Slave (Autosync). Also see "Preferred Sync Ref"

Warning: *!!!! This is no pure hardware function but was implemented by ALSA by some ALSA-drivers be so I use it also. !!!*

- Preferred Sync Ref
  - Name – "Preferred Sync Reference"
  - Access – Read Write
  - Values – "Word" "MADI"

    Within the Auto-sync-Mode the preferred Sync Source can be chosen. If it is not available another is used if possible.

Note:

Note: Since MADI has a much higher bit-rate than word-clock, the card should synchronise better in MADI Mode. But since the RME-PLL is very good, there are almost no problems with word-clock too. I never found a difference.

- TX 64 channel
  - Name – “TX 64 channels mode”
  - Access – Read Write
  - Values – 0 1

    Using 64-channel-modus (1) or 56-channel-modus for MADI-transmission (0).

Note:

Note: This control is for output only. Input-mode is detected automatically from hardware sending MADI.

- Clear TMS
  - Name – “Clear Track Marker”
  - Access – Read Write
  - Values – 0 1

    Don’t use to lower 5 Audio-bits on AES as additional Bits.
- Safe Mode oder Auto Input
  - Name – “Safe Mode”
  - Access – Read Write
  - Values – 0 1 (default on)

    If on (1), then if either the optical or coaxial connection has a failure, there is a takeover to the working one, with no sample failure. Its only useful if you use the second as a backup connection.
- Input
  - Name – “Input Select”
  - Access – Read Write
  - Values – optical coaxial

    Choosing the Input, optical or coaxial. If Safe-mode is active, this is the preferred Input.

Mixer

- Mixer
  - Name – “Mixer”
  - Access – Read Write
  - Values - <channel-number 0-127> <Value 0-65535>

    Here as a first value the channel-index is taken to get/set the corresponding mixer channel, where 0-63 are the input to output fader and 64-127 the playback to outputs fader. Value 0 is channel muted 0 and 32768 an amplification of 1.
- Chn 1-64

  fast mixer for the ALSA-mixer utils. The diagonal of the mixer-matrix is implemented from playback to output.
- Line Out
  - Name – “Line Out”

- Access – Read Write
- Values – 0 1

  Switching on and off the analog out, which has nothing to do with mixing or routing. the analog outs reflects channel 63,64.

Information (only read access)

- Sample Rate
  - Name – “System Sample Rate”
  - Access – Read-only

    getting the sample rate.
- External Rate measured
  - Name – “External Rate”
  - Access – Read only

    Should be “Autosync Rate”, but Name used is ALSA-Scheme. External Sample frequency liked used on Autosync is reported.
- MADI Sync Status
  - Name – “MADI Sync Lock Status”
  - Access – Read
  - Values – 0,1,2

    MADI-Input is 0=Unlocked, 1=Locked, or 2=Synced.
- Word Clock Sync Status
  - Name – “Word Clock Lock Status”
  - Access – Read
  - Values – 0,1,2

    Word Clock Input is 0=Unlocked, 1=Locked, or 2=Synced.
- AutoSync
  - Name – “AutoSync Reference”
  - Access – Read
  - Values – “WordClock”, “MADI”, “None”

    Sync-Reference is either “WordClock”, “MADI” or none.
- RX 64ch — noch nicht implementiert

  MADI-Receiver is in 64 channel mode oder 56 channel mode.
- AB_inp — not tested

  Used input for Auto-Input.
- actual Buffer Position — not implemented

  !!! this is a ALSA internal function, so no control is used !!!

Calling Parameter

- index int array (min = 1, max = 8)

  Index value for RME HDSPM interface. card-index within ALSA

  note: ALSA-standard

- id string array (min = 1, max = 8)

  ID string for RME HDSPM interface.

  note: ALSA-standard

- enable int array (min = 1, max = 8)

  Enable/disable specific HDSPM sound-cards.

  note: ALSA-standard

- precise_ptr int array (min = 1, max = 8)

  Enable precise pointer, or disable.

Note:

note: Use only when the application supports this (which is a special case).

- line_outs_monitor int array (min = 1, max = 8)

  Send playback streams to analog outs by default.

Note:

note: each playback channel is mixed to the same numbered output channel (routed). This is against the ALSA-convention, where all channels have to be muted on after loading the driver, but was used before on other cards, so i historically use it again)

- enable_monitor int array (min = 1, max = 8)

  Enable Analog Out on Channel 63/64 by default.

Note:

note: here the analog output is enabled (but not routed).

Serial UART 16450/16550 MIDI driver

The adaptor module parameter allows you to select either:

- 0 - Roland Soundcanvas support (default)
- 1 - Midiator MS-124T support (1)
- 2 - Midiator MS-124W S/A mode (2)
- 3 - MS-124W M/B mode support (3)
- 4 - Generic device with multiple input support (4)

For the Midiator MS-124W, you must set the physical M-S and A-B switches on the Midiator to match the driver mode you select.

In Roland Soundcanvas mode, multiple ALSA raw MIDI substreams are supported (midiCnD0-midiCnD15). Whenever you write to a different substream, the driver sends the nonstandard MIDI command sequence F5 NN, where NN is the substream number plus 1. Roland modules use this command to switch between different "parts", so this feature lets you treat each part as a distinct raw MIDI substream. The driver provides no way to send F5 00 (no selection) or to not send the F5 NN command sequence at all; perhaps it ought to.

Usage example for simple serial converter:

```
/sbin/setserial /dev/ttyS0 uart none
/sbin/modprobe snd-serial-u16550 port=0x3f8 irq=4 speed=115200
```

Usage example for Roland SoundCanvas with 4 MIDI ports:

```
/sbin/setserial /dev/ttyS0 uart none
/sbin/modprobe snd-serial-u16550 port=0x3f8 irq=4 outs=4
```

In MS-124T mode, one raw MIDI substream is supported (midiCnD0); the outs module parameter is automatically set to 1. The driver sends the same data to all four MIDI Out connectors. Set the A-B switch and the speed module parameter to match (A=19200, B=9600).

Usage example for MS-124T, with A-B switch in A position:

```
/sbin/setserial /dev/ttyS0 uart none
/sbin/modprobe snd-serial-u16550 port=0x3f8 irq=4 adaptor=1 \
                speed=19200
```

In MS-124W S/A mode, one raw MIDI substream is supported (midiCnD0); the outs module parameter is automatically set to 1. The driver sends the same data to all four MIDI Out connectors at full MIDI speed.

Usage example for S/A mode:

```
/sbin/setserial /dev/ttyS0 uart none
/sbin/modprobe snd-serial-u16550 port=0x3f8 irq=4 adaptor=2
```

In MS-124W M/B mode, the driver supports 16 ALSA raw MIDI substreams; the outs module parameter is automatically set to 16. The substream number gives a bitmask of which MIDI Out connectors the data should be sent to, with midiCnD1 sending to Out 1, midiCnD2 to Out 2, midiCnD4 to Out 3, and midiCnD8 to Out 4. Thus midiCnD15 sends the data to all 4 ports. As a special case, midiCnD0 also sends to all ports, since it is not useful to send the data to no ports. M/B mode has extra overhead to select the MIDI Out for each byte, so the aggregate data rate across all four MIDI Outs is at most one byte every 520 us, as compared with the full MIDI data rate of one byte every 320 us per port.

Usage example for M/B mode:

```
/sbin/setserial /dev/ttyS0 uart none
/sbin/modprobe snd-serial-u16550 port=0x3f8 irq=4 adaptor=3
```

The MS-124W hardware's M/A mode is currently not supported. This mode allows the MIDI Outs to act independently at double the aggregate throughput of M/B, but does not allow sending the same byte simultaneously to multiple MIDI Outs. The M/A protocol requires the driver to twiddle the modem control lines under timing constraints, so it would be a bit more complicated to implement than the other modes.

Midiator models other than MS-124W and MS-124T are currently not supported. Note that the suffix letter is significant; the MS-124 and MS-124B are not compatible, nor are the other known models MS-101, MS-101B, MS-103, and MS-114. I do have documentation (tim.mann@compaq.com) that partially covers these models, but no units to experiment with. The MS-124W support is tested with a real unit. The MS-124T support is untested, but should work.

The Generic driver supports multiple input and output substreams over a single serial port. Similar to Roland Soundcanvas mode, F5 NN is used to select the appropriate input or output stream (depending on

the data direction). Additionally, the CTS signal is used to regulate the data flow. The number of inputs is specified by the ins parameter.

Imagination Technologies SPDIF Input Controllers

The Imagination Technologies SPDIF Input controller contains the following controls:

- name='IEC958 Capture Mask',index=0

This control returns a mask that shows which of the IEC958 status bits can be read using the 'IEC958 Capture Default' control.

- name='IEC958 Capture Default',index=0

This control returns the status bits contained within the SPDIF stream that is being received. The 'IEC958 Capture Mask' shows which bits can be read from this control.

- name='SPDIF In Multi Frequency Acquire',index=0
- name='SPDIF In Multi Frequency Acquire',index=1
- name='SPDIF In Multi Frequency Acquire',index=2
- name='SPDIF In Multi Frequency Acquire',index=3

This control is used to attempt acquisition of up to four different sample rates. The active rate can be obtained by reading the 'SPDIF In Lock Frequency' control.

When the value of this control is set to {0,0,0,0}, the rate given to hw_params will determine the single rate the block will capture. Else, the rate given to hw_params will be ignored, and the block will attempt capture for each of the four sample rates set here.

If less than four rates are required, the same rate can be specified more than once

- name='SPDIF In Lock Frequency',index=0

This control returns the active capture rate, or 0 if a lock has not been acquired

- name='SPDIF In Lock TRK',index=0

This control is used to modify the locking/jitter rejection characteristics of the block. Larger values increase the locking range, but reduce jitter rejection.

- name='SPDIF In Lock Acquire Threshold',index=0

This control is used to change the threshold at which a lock is acquired.

- name='SPDIF In Lock Release Threshold',index=0

This control is used to change the threshold at which a lock is released.

Symbols

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