lib_board_support: XMOS board support

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XMOS

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1 Introduction

This repo contains board specific hardware configuration code for various *XMOS* evaluation and development kits. By keeping the board-specific code in a dedicated repository various applications need not replicate commonly used code such as initialisation of on-board peripherals and in addition any updates or fixes can easily be rolled out to all dependent applications.

2 Supported Boards

The following boards are supported in this repo with interfaces provided in the languages shown in the table below.

Board	Supported Languages
XK_EVK_XU316	XC / C
XK_AUDIO_316_MC_AB	XC / C
XK_AUDIO_216_MC_AB	XC / C

The following section provides specific details of the features for each of the boards supported by this library.

2.1 xcore.ai Multi-Channel Audio Board

The *XMOS xcore.ai Multichannel Audio Board* (XK-AUDIO-316-MC) is a complete hardware and software reference platform targeted at up to 32-channel USB audio applications, such as DJ decks, mixers and other musical instrument interfaces. The board can also be used to prototype products with reduced feature sets or HiFi style products.

The XK-AUDIO-316-MC is based around the XU316-1024-TQ128-C24 multicore microcontroller; a dual-tile *xcore.ai* device with an integrated High Speed USB 2.0 PHY and 16 logical cores delivering up to 2400MIPS of deterministic and responsive processing power.

Exploiting the flexible programmability of the *xcore.ai* architecture, the XK-AUDIO-316-MC supports a USB audio source, streaming 8 analogue input and 8 analogue output audio channels simultaneously - at up to 192kHz. It also supports digital input/output streams

(S/PDIF and ADAT) and MIDI. Ideal for consumer and professional USB audio interfaces. The board can also be used for testing general purpose audio DSP activities - mixing, filtering, etc.

For full details regarding the hardware please refer to xcore.ai Multichannel Audio Platform Hardware Manual.

Hardware Features

The location of the various features of the *xcore.ai Multichannel Audio Board* (XK-AUDIO-316-MC) is shown in Fig. 1.

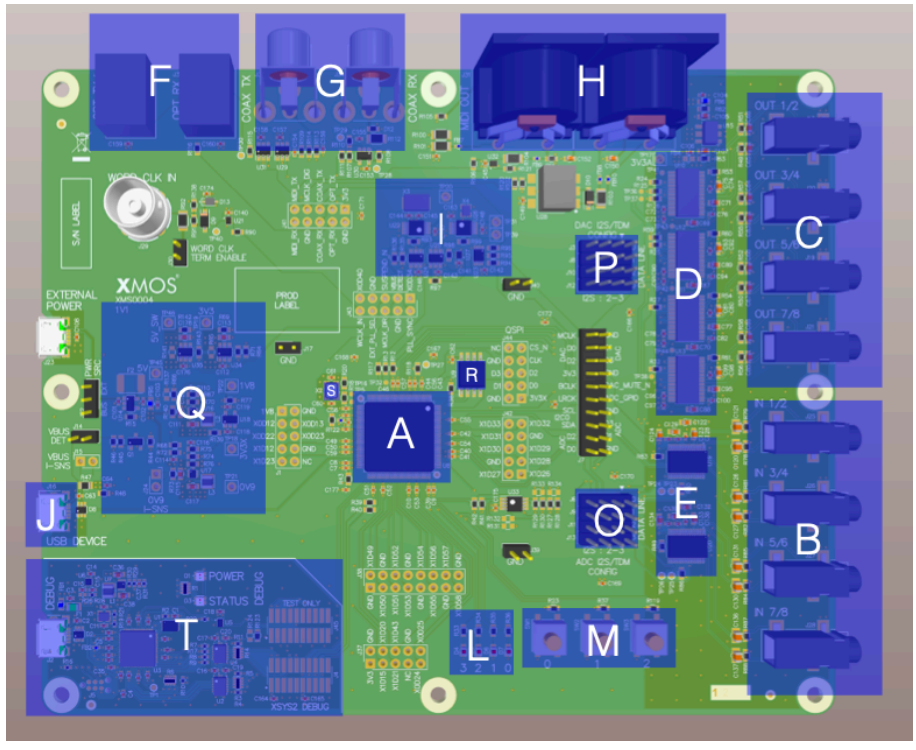

Fig. 1: xcore.ai Multichannel Audio Board hardware features

It includes the following features:

- A: *xcore.ai* (XU316-1024-TQ128-C24) device
- B: 8 line level analog inputs (3.5mm stereo jacks)
- C: 8 line level analog outputs (3.5mm stereo jacks)
- D: 384kHz 24 bit audio DACs
- E: 192kHz 24 bit audio ADCs
- F: Optical connections for digital interface (e.g. S/PDIF and ADAT)
- G: Coaxial connections for digital interfaces (e.g. S/PDIF)
- H: MIDI in and out connections
- I: Flexible audio master clock generation
- J: USB 2.0 micro-B jacks
- L: 4 general purpose LEDs

- M: 3 general purpose buttons
- O: Flexible I²S/TDM input data routing
- P: Flexible I²S/TDM output data routing
- Q: Integrated power supply
- R: Quad-SPI boot ROM
- S: 24MHz Crystal
- T: Integrated XTAG4 debugger

Analogue Input & Output

A total of eight single-ended analog input channels are provided via 3.5mm stereo jacks. These inputs feed into a pair of quad-channel PCM1865 ADCs from Texas Instruments.

A total of eight single-ended analog output channels are provided. These are fed from four PCM5122 stereo DAC's from Texas instruments.

All ADC's and DAC's are configured via an I²C bus. Due to an clash of device addresses a I²C multiplexor is used.

The four digital I²S/TDM input and output channels are mapped to the xCORE input/outputs through a header array. These jumpers allow channel selection when the ADCs/-DACs are used in TDM mode.

Digital Input & Output

Optical and coaxial digital audio transmitters are used to provide digital audio input output in formats such as IEC60958 consumer mode (S/PDIF) and ADAT. The output data streams from the *xcore* are re-clocked using the external master clock to synchronise the data into the audio clock domain. This is achieved using simple external D-type flip-flops.

MIDI

MIDI input and output is provided on the board via standard 5-pin DIN connectors compliant to the MIDI specification. The signals are buffered using 5V line drivers and are then connected ports on the xCORE, via a 5V to 3.3V buffer. A 1-bit port is used for receive and a 4-bit port is used for transmit. A pull-up resistor on the MIDI output ensures there is no MIDI output when the *xcore* device is not actively driving the output.

Audio Clocking

In order to accommodate a multitude of clocking options a flexible clocking scheme is provided for the audio subsystem.

Three methods of generating an audio master clock are provided on the board:

- A Cirrus Logic CS2100-CP PLL device. The CS2100 features both a clock generator and clock multiplier/jitter reduced clock frequency synthesizer (clean up) and can generate a low jitter audio clock based on a synchronisation signal provided by the *xcore*
- A Skyworks Si5351B PLL device. The Si5351 is an I²C configurable clock generator that is suited for replacing crystals, crystal oscillators, VCXOs, phase-locked loops (PLLs), and fanout buffers.

- *xcore.ai* devices are equipped with a secondary (or *application*) PLL which can be used to generate audio clocks.

Selecting between these methods is done via writing to bits 6 and 7 of PORT 8D on tile[0]. See *Control I/O*.

Note: `lib_board_support` currently only supports the *xcore.ai* secondary PLL and CS2100 device

Control I/O

4 bits of PORT 8C are used to control external hardware on the board. This is described in *PORT 8C functionality*.

Table 1: PORT 8C functionality

Bit(s)	Functionality	0	1
[0:3]	Unused		
4	Enable 3v3 power for digital (inverted)	Enabled	Disabled
5	Enable 3v3 power for analogue	Disabled	Enabled
6	PLL Select	CS2100	Si5351B
7	Master clock direction	Output	Input

Note: To use the *xcore* application PLL bit 7 should be set to 0. To use one of the external PLL's bit 7 should be set to 1.

LEDs, Buttons and Other IO

All programmable I/O on the board is configured for 3.3 volts.

Four green LED's and three push buttons are provided for general purpose user interfacing.

The LEDs are connected to PORT 4F and the buttons are connected to bits [0:2] of PORT 4E, both on tile 0. Bit 3 of this port is connected to the (currently unused) ADC interrupt line.

The board also includes support for an AES11 format Word Clock input via 75 ohm BNC. The software does not currently support any functionality related to this and it is provided for future expansion.

All spare I/O is brought out and made available on 0.1" headers for easy connection of expansion boards etc.

Power

The board is capable of acting as a USB2.0 self or bus powered device. If bus powered, the board takes power from the `USB DEVICE` connector (micro-B receptacle). If self powered, board takes power from `EXTERNAL POWER` input (micro-B receptacle).

A power source select jumper (marked `PWR SRC`) is used to select between bus and self-powered configuration.

Note: To remain USB compliant the software should be properly configured for bus vs self powered operation

Debug

For convenience the board includes an on-board xTAG4 for debugging via JTAG/xSCOPE. This is accessed via the USB (micro-B) receptacle marked DEBUG.

2.2 xcore-200 Multi-Channel Audio Board

The XMOS xcore-200 Multi-channel Audio board (XK-AUDIO-216-MC) is a complete hardware and reference software platform targeted at up to 32-channel USB and networked audio applications, such as DJ decks and mixers.

The XK-AUDIO-216-MC is based around the XE216-512-TQ128 multicore microcontroller; an dual-tile xcore-200 device with an integrated High Speed USB 2.0 PHY, RGMII (Gigabit Ethernet) interface and 16 logical cores delivering up to 2000MIPS of deterministic and responsive processing power.

Exploiting the flexible programmability of the *xcore-200* architecture, the XK-AUDIO-216-MC supports either USB or network audio source, streaming 8 analogue input and 8 analogue output audio channels simultaneously - at up to 192kHz.

For full details regarding the hardware please refer to xcore-200 Multichannel Audio Platform Hardware Manual.

Analogue Input & Output

A total of eight single-ended analog input channels are provided via 3.5mm stereo jacks. Each is fed into a CirrusLogic CS5368 ADC. Similarly a total of eight single-ended analog output channels are provided. Each is fed into a CirrusLogic CS4384 DAC.

The four digital I²S/TDM input and output channels are mapped to the *xcore* input/outputs through a header array. This jumper allows channel selection when the ADC/DAC is used in TDM mode

Digital Input & Output

Optical and coaxial digital audio transmitters are used to provide digital audio input output in formats such as IEC60958 consumer mode (S/PDIF) and ADAT. The output data streams from the *xcore-200* are re-clocked using the external master clock to synchronise the data into the audio clock domain. This is achieved using simple external D-type flip-flops.

MIDI

MIDI I/O is provided on the board via standard 5-pin DIN connectors. The signals are buffered using 5V line drivers and are then connected to 1-bit ports on the *xcore-200*, via a 5V to 3.3V buffer.

Audio Clocking

A flexible clocking scheme is provided for both audio and other system services. In order to accommodate a multitude of clocking options, the low-jitter master clock is generated locally using a frequency multiplier PLL chip. The chip used is a Phaselink PL611-01, which is pre-programmed to provide a 24MHz clock from its CLK0 output, and either 24.576 MHz or 22.5792MHz from its CLK1 output.

The 24MHz fixed output is provided to the *xcore-200* device as the main processor clock. It also provides the reference clock to a Cirrus Logic CS2100, which provides a very low jitter audio clock from a synchronisation signal provided from the *xcore-200*.

Either the locally generated clock (from the PL611) or the recovered low jitter clock (from the CS2100) may be selected to clock the audio stages; the *xcore-200*, the ADC/DAC and Digital output stages. Selection is controlled via an additional I/O, bit 5 of PORT 8C.

LEDs, Buttons and Other IO

An array of 4*4 green LEDs, 3 buttons and a switch are provided for general purpose user interfacing. The LED array is driven by eight signals each controlling one of 4 rows and 4 columns.

A standard *XMOS* xSYS interface is provided to allow host debug of the board via JTAG.

2.3 xcore.ai Evaluation Kit

The *XMOS xcore.ai Evaluation Kit* (XK-EVK-XU316) is an evaluation board for the *xcore.ai* multi-core microcontroller from *XMOS*.

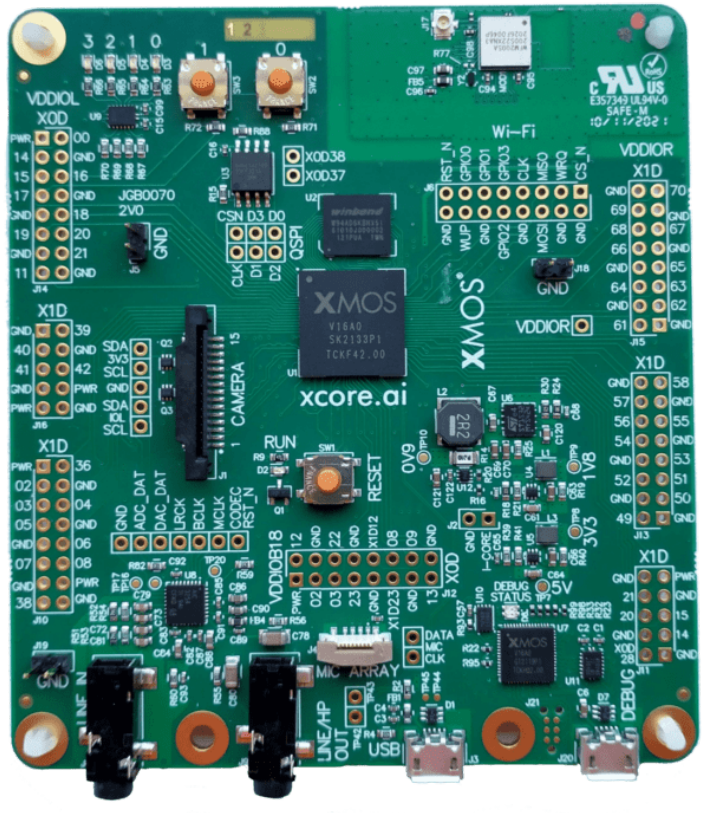

Fig. 2: *xcore.ai* Evaluation Kit

The XK-EVK-XU316 allows testing in multiple application scenarios and provides a good general software development board for simple tests and demos. The XK-EVK-XU316 comprises an *xcore.ai* processor with a set of I/O devices and connectors arranged around it, as shown in Fig. 3.

External hardware features board include, four general purpose LEDs, two general purpose push-button switches, a PDM microphone connector, audio codec with line-in and line-out jack, QSPI flash memory, LPDDR1 external memory 58 GPIO connections from tile 0 and 1, micro USB for power and host connection, MIPI connector for a MIPI camera, integrated *xTAG* debug adapter and a reset switch with LED to indicate running.

For full details regarding the hardware please refer to XK-EVK-XU316 xcore.ai Evaluation Kit Manual.

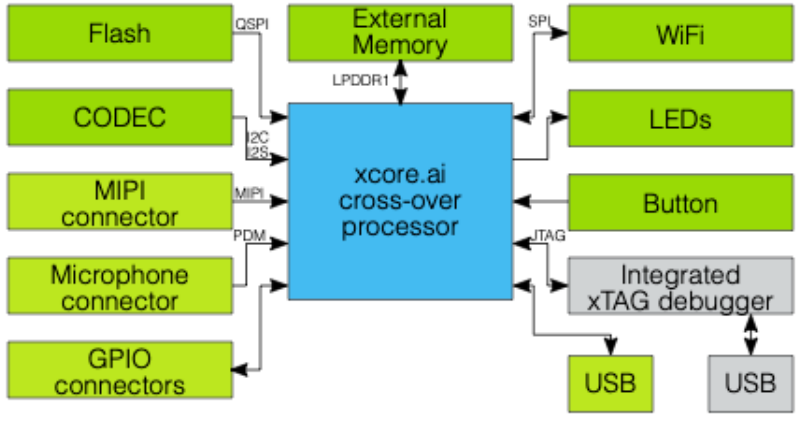

Fig. 3: *xcore.ai* Evaluation Kit block diagram

Warning: The *xcore.ai Evaluation Kit* is a general purpose evaluation platform and should be considered an "example" rather than a fully fledged reference design.

Analogue Audio Input & Output

A stereo CODEC (TLV320AIC3204), connected to the xcore.ai device via an I²S interface, provides analogue input/output functionality at line level.

The audio CODEC is are configured by the *xcore.ai* device via an I²C bus.

Audio Clocking

xcore.ai devices are equipped with a secondary (or *application*) PLL which is used to generate the audio clocks for the CODEC.

LEDs, Buttons and Other IO

Four green LED's and two push buttons are provided for general purpose user interfacing.

The LEDs are connected to PORT 4C and the buttons are connected to bits [0:1] of PORT 4D.

All spare I/O is brought out and made available on 0.1" headers for easy connection of expansion boards etc.

Power

The XK-EVK-XU316 requires a 5V power source that is normally provided through the micro-USB cable J3. The voltage is converted by on-board regulators to the 0V9, 1V8 and 3V3 supplies used by the components.

The board should therefore be configured to present itself as a bus powered device when connected to an active USB host.

Debug

For convenience the board includes an on-board xTAG4 for debugging via JTAG/xSCOPE. This is accessed via the USB (micro-B) receptacle marked **DEBUG**.

3 Usage

This repo supports the *XMOS* build system; *XCommon CMake*. To use the library add `lib_board_support` to an applications *CMakeLists.txt* file using the *APP_DEPENDENT_MODULES* entry. The application must provide a relevant `xn` file, although example `xn` files are provided in alongside this libray (see *xn_files* directory).

The application must use the APIs for the specific board that it is using. To ensure that only the correct sources for the board in use get compiled in, it is necessary to set the preprocessor value `BOARD_SUPPORT_BOARD` in the project to one of the available boards listed in *api/boards/boards_utils.h*. This can be done in the app with the following snippet of cmake:

```
set(APP_COMPILER_FLAGS
    -DBOARD_SUPPORT_BOARD=XK_AUDIO_316_MC_AB  # Change value to select board, see api/boards/boards_utils.h for
↪available boards
)
```

From the application where board initialisation of configuration is done it is necessary to include the relevant header file. For example:

```
#include "xk_audio_316_mc_ab/board.h"
```

From then onwards the code may call the relevant API functions to setup and configure the board hardware. Examples are provided in the *examples* directory of this repo.

Note that in some cases, the *xcore* tile that calls the configuration function (usually from I²S initialisation) is different from the tile where I²C master is placed. Since I²C master is required by most audio CODECs for configuration and *xcore* tiles can only communicate with each other via channels, a remote server is needed to provide the I²C setup. This usually takes the form of a task which is run on a thread placed on the I²C tile and is controlled via a channel from the other tile where I²S resides. The cross-tile channel must be declared at the top-level XC main function. The included examples provide a reference for this using both XC and C.

4 Application Programmer Interface

This section contains the details of the API support by *lib_board_support*. The API is broken down into 2 sections:

1. *Boards*: This includes subdirectories for each supported board which need to be included in your application.
2. *Drivers*: This includes sources for configuring peripheral devices which may be on one or more of the supported boards.

4.1 Common API

This section contains the list of supported boards, one of which needs to be globally defined as `BOARD_SUPPORT_BOARD` in the project.

NULL_BOARD

Define representing Null board i.e. no board in use

XK_AUDIO_216_MC_AB

Define representing XK-AUDIO-216-MC Board

XK_AUDIO_316_MC_AB

Define representing XK-AUDIO-316-MC Board

XK_EVK_XU316

Define representing XK-EVK-XU316 board

BOARD_SUPPORT_N_BOARDS

Total number of boards supported by the library

BOARD_SUPPORT_BOARD

Define that should be set to the current board type in use

Default value: NULL_BOARD

4.2 XK_AUDIO_316_MC_AB API

struct **xk_audio_316_mc_ab_config_t**

Configuration struct type for setting the hardware profile.

Public Members

xk_audio_316_mc_ab_mclk_modes_t **clk_mode**

See xk_audio_316_mc_ab_mclk_modes_t for available clock mode options.

char **dac_is_clock_master**

Boolean setting for whether the DAC or the xcore.ai is I2S clock master. Set to 0 to make the xcore.ai master.

unsigned **default_mclk**

Nominal clock frequency in MHz. Standard rates are supported between 11.2896 MHz and 49.152 MHz.

unsigned **pll_sync_freq**

When the CLK_CS2100 is used, this defines the nominal reference clock frequency for multiplication by the PLL. This value is ignored when the CS2100 is not used.

xk_audio_316_mc_ab_pcm_format_t **pcm_format**

See xk_audio_316_mc_ab_pcm_format_t for available data frame options.

unsigned **i2s_n_bits**

Number of bits per data frame in I2S.

unsigned **i2s_chans_per_frame**

This defines the number of audio channels per frame (a frame is a complete cycle of FSYNC or LRCLK).

enum **xk_audio_316_mc_ab_mclk_modes_t**

Type of clock to be instantiated. This may be a fixed clock using the application PLL, an adjustable clock using the CS2100 external PLL or an adjustable or fixed clock using the on-chip application PLL.

Values:

enumerator **CLK_FIXED**

enumerator **CLK_CS2100**

enumerator **CLK_PLL**

enum **xk_audio_316_mc_ab_pcm_format_t**

Formats supported by the DAC and ADC. Either I2S using multiple data lines or TDM supporting multi-channel using a single data line.

Values:

enumerator **AUD_316_PCM_FORMAT_I2S**

enumerator **AUD_316_PCM_FORMAT_TDM**

port **p_scl**

I2C interface ports

port **p_sda**

void **xk_audio_316_mc_ab_i2c_master**(SERVER_INTERFACE(i2c_master_if, i2c[1]))

Starts an I2C master task. Must be started from tile[0] *after* *xk_audio_316_mc_ab_board_setup()* and *before* and tile[1] HW calls.

Parameters

- **i2c** – client side of I2C master interface connection.

void **xk_audio_316_mc_ab_board_setup**(const REFERENCE_PARAM(*xk_audio_316_mc_ab_config_t*, config))

Performs the required port operations to enable and the audio hardware on the platform. Must be called from tile[0] and *before* *xk_audio_316_mc_ab_AudioHwInit()* is called.

Parameters

- **config** – Reference to the *xk_audio_316_mc_ab_config_t* configuration struct.

void **xk_audio_316_mc_ab_AudioHwInit**(CLIENT_INTERFACE(i2c_master_if, i2c), const REFERENCE_PARAM(*xk_audio_316_mc_ab_config_t*, config))

Initialises the audio hardware ready for a configuration. Must be called once *after* *xk_audio_316_mc_ab_board_setup()*.

Parameters

- **i2c** – Client side of I2C master interface connection.
- **config** – Reference to the *xk_audio_316_mc_ab_config_t* hardware configuration struct.

void **xk_audio_316_mc_ab_AudioHwConfig**(CLIENT_INTERFACE(i2c_master_if, i2c), const REFERENCE_PARAM(*xk_audio_316_mc_ab_config_t*, config), unsigned samFreq, unsigned mClk, unsigned dsdMode, unsigned sampRes_DAC, unsigned sampRes_ADC)

Configures the audio hardware following initialisation. This is typically called each time a sample rate or stream format change occurs.

Parameters

- **i2c** – Client side of I2C master interface connection.
- **config** – Reference to the *xk_audio_316_mc_ab_config_t* hardware configuration struct.
- **samFreq** – The sample rate in Hertz.
- **mClk** – The master clock rate in Hertz.
- **dsdMode** – Controls whether the DAC is to be set into DSD mode (1) or PCM mode (0).
- **sampRes_DAC** – The sample resolution of the DAC output in bits. Typically 16, 24 or 32.
- **sampRes_ADC** – The sample resolution of the ADC input in bits. Typically 16, 24 or 32.

void **xk_audio_316_mc_ab_i2c_master_exit**(CLIENT_INTERFACE(i2c_master_if, i2c))

Causes the tile[0] to exit, freeing up a thread. Must be called from tile[1]. Once called, HW config calls from tile[1] will block forever. It is possible to re-start *xk_audio_316_mc_ab_i2c_master()* on tile[0] if needed to re-enable this service.

Parameters

- **i2c** – Client side of I2C master interface connection.

4.3 XK_AUDIO_216_MC_AB API

struct **xk_audio_216_mc_ab_config_t**

Configuration struct type for setting the hardware profile.

Public Members

xk_audio_216_mc_ab_clk_mode_t **clk_mode**

See xk_audio_216_mc_ab_clk_mode_t for clock mode available options.

char **codec_is_clk_master**

Boolean setting for whether the DAC or the xcore-200 is I2S clock master. Set to 0 to make the xcore-200 master.

xk_audio_216_mc_ab_usb_sel_t **usb_sel**

USB port slection - see xk_audio_216_mc_ab_usb_sel_t for options.

xk_audio_216_mc_ab_pcm_format_t **pcm_format**

See xk_audio_216_mc_ab_pcm_format_t for available pmc_format options.

unsigned **pll_sync_freq**

When the external PLL is used, this defines the nominal reference clock frequency for multiplication by the PLL.

enum **xk_audio_216_mc_ab_clk_mode_t**

Type of clock to be instantiated. This may be a fixed clock using an external generator or an adjustable clock using an external PLL (CS2100) in either digital Rx clock recovery or USB clock recovery using synchronous mode.

Values:

enumerator **AUD_216_CLK_FIXED**

enumerator **AUD_216_CLK_EXTERNAL_PLL**

enumerator **AUD_216_CLK_EXTERNAL_PLL_USB**

enum **xk_audio_216_mc_ab_pcm_format_t**

Formats supported by the DAC and ADC. Either I2S using multiple data lines or TDM supporting multi-channel using a single data line.

Values:

enumerator **AUD_216_PCM_FORMAT_I2S**

enumerator **AUD_216_PCM_FORMAT_TDM**

enum **xk_audio_216_mc_ab_usb_sel_t**

Selects which USB port to use - either type A or type B.

Values:

enumerator **AUD_216_USB_A**

enumerator **AUD_216_USB_B**

void **xk_audio_216_mc_ab_AudioHwInit**(const REFERENCE_PARAM(*xk_audio_216_mc_ab_config_t*, config))

Initialises the audio hardware ready for a configuration. Must be called once *after* *xk_audio_316_mc_ab_board_setup()*.

Parameters

- **config** – Reference to the *xk_audio_216_mc_ab_config_t* hardware configuration struct.

void **xk_audio_216_mc_ab_AudioHwConfig**(const REFERENCE_PARAM(*xk_audio_216_mc_ab_config_t*, config), unsigned samFreq, unsigned mClk, unsigned dsdMode, unsigned sampRes_DAC, unsigned sampRes_ADC)

Configures the audio hardware following initialisation. This is typically called each time a sample rate or stream format change occurs.

Parameters

- **config** – Reference to the *xk_audio_216_mc_ab_config_t* hardware configuration struct.
- **samFreq** – The sample rate in Hertz.
- **mClk** – The master clock rate in Hertz.
- **dsdMode** – Controls whether the DAC is to be set into DSD mode (1) or PCM mode (0).
- **sampRes_DAC** – The sample resolution of the DAC output in bits. Typically 16, 24 or 32.
- **sampRes_ADC** – The sample resolution of the ADC input in bits. Typically 16, 24 or 32.

4.4 XK_EVK_XU316 API

struct **xk_evk_xu316_config_t**

Public Members

unsigned **default_mclk**

xk_audio_316_mc_ab_config_t::clk_mode See xk_audio_316_mc_ab_mclk_modes_t for available clock mode options.

enum **audioHwCmd_t**

Command enumeration for channel based commands to I2C master server on other tile.

Values:

enumerator **AUDIOHW_CMD_REGWR**

enumerator **AUDIOHW_CMD_REGRD**

enumerator **AUDIOHW_CMD_EXIT**

void **xk_evk_xu316_AudioHwRemote**(chanend c)

Starts an I2C master server task. Must be started *before* the tile[1] xk_evk_xu316_AudioHwInit calls. In the background this also starts a combinable channel to interface translation task so the API may be used over a channel end however it still only occupies one thread. May be exited after config by sending AUDIOHW_CMD_EXIT if dynamic configuration is not required.

Parameters

- **c** – Server side of channel connecting I2C master server and HW config functions.

void **xk_evk_xu316_AudioHwChanInit**(chanend c)

Initialises the client side channel for remote communications with I2C. Must be called on tile[1] *before* *xk_evk_xu316_AudioHwInit()*.

Parameters

- **c** – Client side of channel connecting I2C master server and HW config functions.

void **xk_evk_xu316_AudioHwInit**(const REFERENCE_PARAM(*xk_evk_xu316_config_t*, config))

Initialises the audio hardware ready for a configuration. Must be called once *after* *xk_evk_xu316_AudioHwRemote()* and *xk_evk_xu316_AudioHwChanInit()*.

Parameters

- **config** – Reference to the *xk_audio_316_mc_ab_config_t* hardware configuration struct.

void **xk_evk_xu316_AudioHwConfig**(unsigned samFreq, unsigned mClk, unsigned dsdMode, unsigned sampRes_DAC, unsigned sampRes_ADC)

Configures the audio hardware following initialisation. This is typically called each time a sample rate or stream format change occurs.

Parameters

- **samFreq** – The sample rate in Hertz.
- **mClk** – The master clock rate in Hertz.
- **dsdMode** – Controls whether the DAC is to be set into DSD mode (1) or PCM mode (0).
- **sampRes_DAC** – The sample resolution of the DAC output in bits. Typically 16, 24 or 32.
- **sampRes_ADC** – The sample resolution of the ADC input in bits. Typically 16, 24 or 32.

5 Example Applications

Some simple example applications are provided in order to show how to use *lib_board_support*.

5.1 Simple C Usage

The applications *app_evk_316_simple_c* and *app_xk_audio_316_mc_simple_c* provide a bare-bones application where the hardware setup is called from C.

These applications run on the *XK-EVK-XU316* and *XK-AUDIO-316-MC* boards respectively.

They show how to use the cross-tile communications in conjunction with the I²C master server. The applications only setup the hardware and then exit the I²C server.

5.2 XC Usage Example

The application *app_xk_audio_316_mc_simple_xc* demonstrates calling the hardware setup API from C. It runs on the *XK-AUDIO-316-MC* board.

5.3 Building and running

To build and run an example, run the following from an XTC tools terminal to configure the build:

```
cd examples/<app_name>
cmake -G "Unix Makefiles" -B build
```

Any missing dependencies will be downloaded by the build system at this point.

The application binaries can be built using `xmake`:

```
xmake -C build
```

To run the application use the following command:

```
xrun --io bin/<app_name>.xe
```

For example:

```
cd examples/app_xk_audio_316_mc_simple_xc
cmake -G "Unix Makefiles" -B build
xmake -C build
xrun --io bin/app_xk_audio_316_mc_simple_xc.xe
```