# De1-SoC Board Audio Output Using the WM8731 Audio Codec

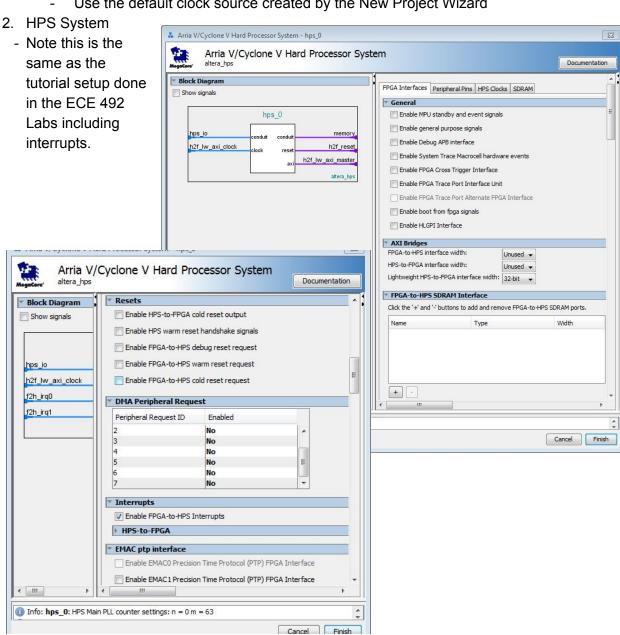
### Introduction

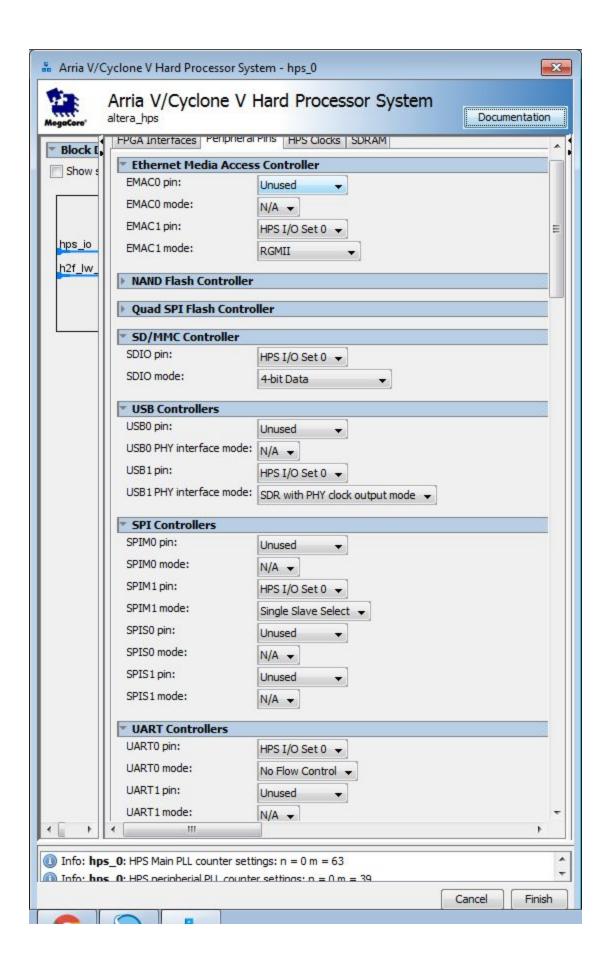
These notes will show how to get the WM8731 audio codec configured and running, as well as outputting a simple frequency sine wave.

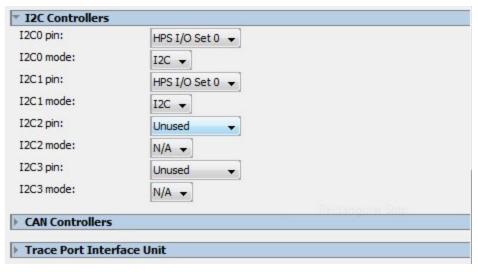
## **Hardware Procedure**

The following images show the minimum required components in the Qsys system as well as the required settings.

- 1. Clock
  - Use the default clock source created by the New Project Wizard

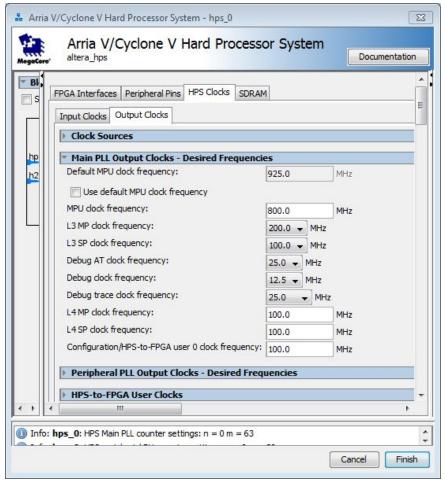


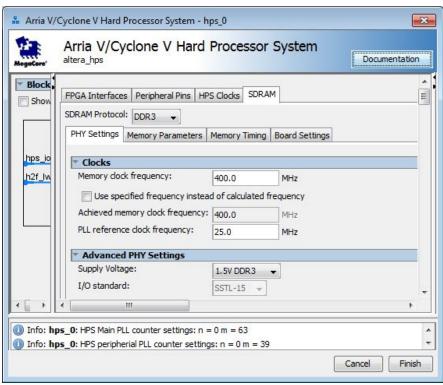


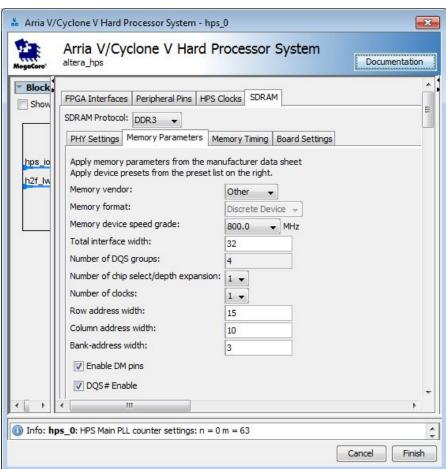


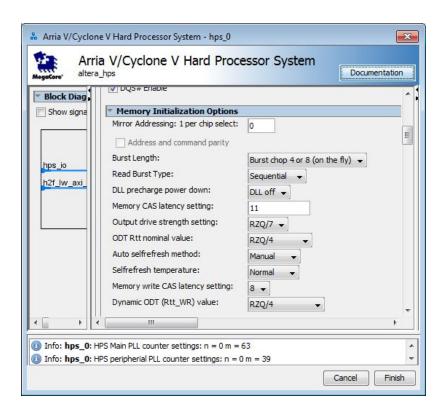
In the Peripheral Mux Table click on the GPIO09, GPIO35, GPIO40, GPIO48, GPIO53, GPIO54 and GPIO61 pins to appropriately map them manually.

Under the HPS Clocks tab leave the Input Clocks tab as the default. The output clocks are as follows:



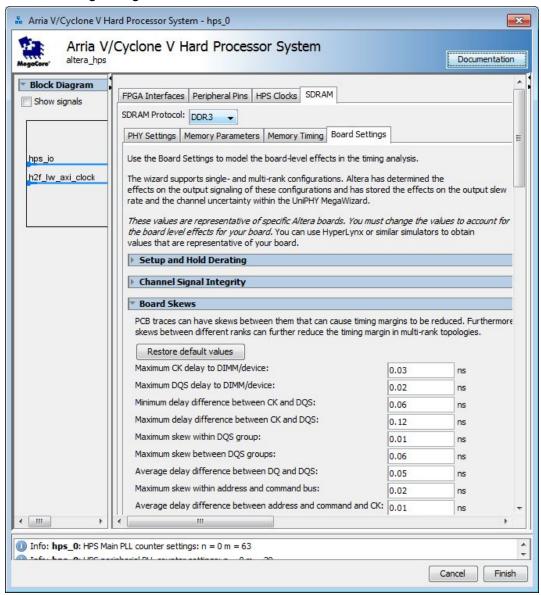




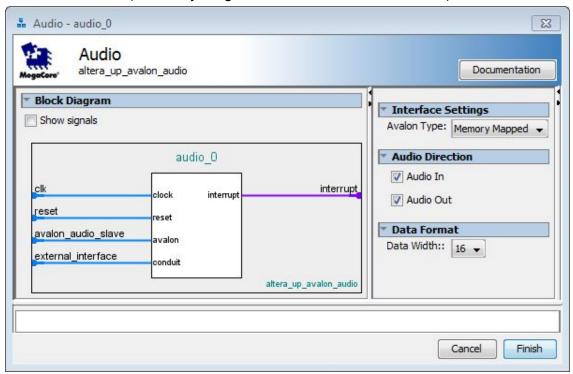


Under Board Settings -> Setup and Hold Derating and Channel Signal Integrity, leave Altera's default settings checked.

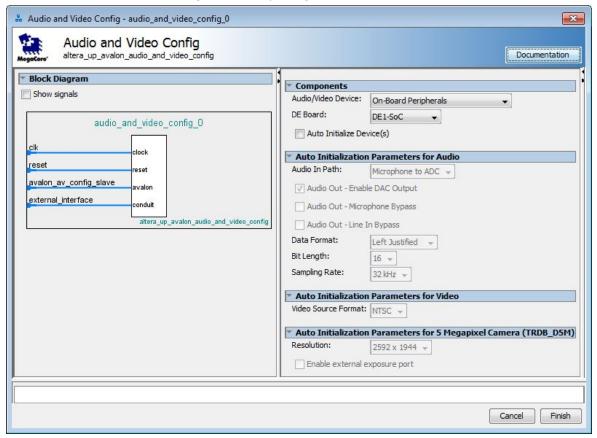
The remaining configuration is as follows:



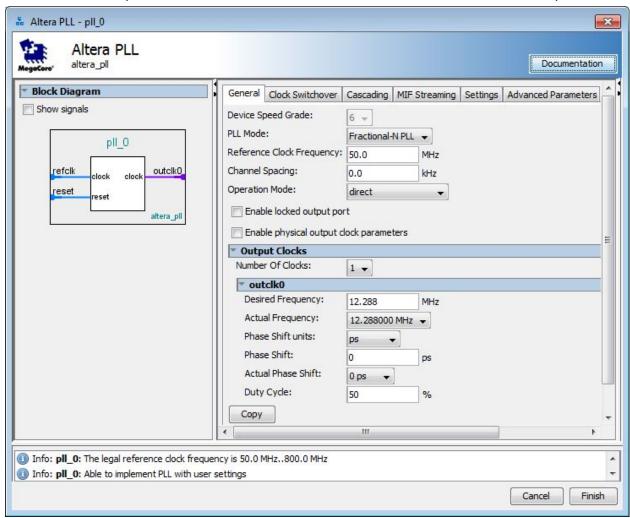
3. Audio Core (University Program -> Audio and Video -> Audio)



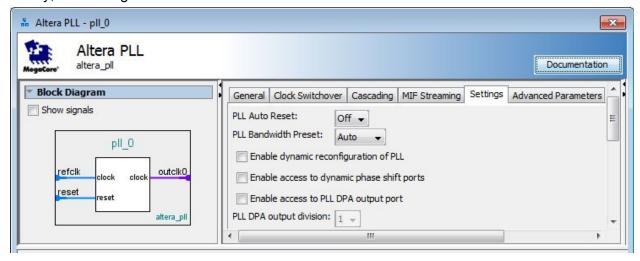
4. Audio and Video config (University Program -> Audio & Video -> Audio and Video Config)



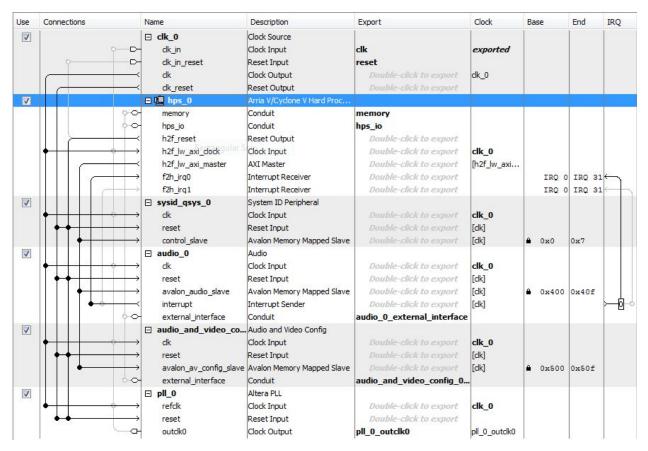
5. Altera PLL (Basic Functions -> Clocks; PLLs and Resets -> PLL -> Altera PLL)



Under the Clock Switchover, Cascading, and MIF Streaming tabs, nothing needs to be enabled Finally, the Settings tab is as follows:



Once these components are added, connect them according to the following image and generate the VHDL in Qsys. Be sure to also include the sysid\_qsys\_0 component with the default configurations of 0x0.



Note you will have four warnings pertaining to interrupts upon successful generation. This is fine and will still provide functional audio.

Also note, the base addresses for the audio components are set at 0x400 and 0x500. You may change these to suit your design as long as they are updated in the audio.h and audio\_cfg.h files (discussed later).

Next, ensure your toplevel VHDL file in Quartus is connected as follows. A template is provided in audio.vhd for reference.

A. Declare these pins in the entity (in addition to all of the hps signals):

```
-- I2C Interface
FPGA_I2C_SCLK : out std_logic;
FPGA_I2C_SDAT : inout std_logic := 'X';
-- Audio

AUD_ADCDAT : in std_logic := 'X';
AUD_ADCLRCK : in std_logic := 'X';
AUD_BCLK : in std_logic := 'X';
AUD_DACDAT : out std_logic;
```

```
AUD_DACLRCK : in std_logic := 'X';
AUD_XCK : out std_logic
```

B. Add these ports in the architecture's component declaration

```
audio 0 external interface ADCDAT
                                       : in std logic
                                                                 := 'X':
                                                                              -- ADCDAT
audio_0_external_interface_ADCLRCK
                                        : in std_logic
                                                                 := 'X':
                                                                            -- ADCLRCK
audio 0 external interface BCLK
                                        : in std logic
                                                                 := 'X':
                                                                              -- BCLK
audio 0 external interface DACDAT
                                        : out std logic;
                                                                              -- DACDAT
audio 0 external interface DACLRCK
                                       : in std logic
                                                                 := 'X';
                                                                             -- DACLRCK
audio_and_video_config_0_external_interface_SDAT : inout std_logic := 'X';
                                                                                -- SDAT
audio and video config 0 external interface SCLK : out std logic;
                                                                                -- SCLK
pll 0 outclk0 clk
                                  : out std logic;
                                                                        -- clk
```

C. Port map the pins to the ports in the instantiation:

```
audio_and_video_config_0_external_interface_SDAT => FPGA_I2C_SDAT,
audio_and_video_config_0_external_interface_SCLK => FPGA_I2C_SCLK,
audio_0_external_interface_ADCDAT => AUD_ADCDAT,
audio_0_external_interface_ADCLRCK => AUD_ADCLRCK,
audio_0_external_interface_BCLK => AUD_BCLK,
audio_0_external_interface_DACDAT => AUD_DACDAT,
audio_0_external_interface_DACLRCK => AUD_DACLRCK,
pll_0_outclk0_clk => AUD_XCK
```

Ensure that the .qip and .tcl scripts have also been added to the project before compiling. Also be sure to run the appropriate .tcl scripts beforehand. Then you may compile the design.

A zipped archive of a template project is included with this report for your reference.

#### **Software Procedure**

While only using the LINEOUT to the speakers, the audio core has four writable registers starting at the base address. For a full description of what each register does, see the first reference. In order to send audio out, you must write to both the left and right audio channels, if only one channel is written to and one FIFO buffer remains empty, the codec will not output audio. Although the audio and video config component can be setup in Qsys, this configuration can be overridden by writing to the components registers. These can be written to by the function write\_audio\_cfg\_register(REGISTER ADDRESS, REGISTER VALUE). Below is the full register configuration used in the example code. However, something to note in the 0x8 register, is that bit 6 must be set to 0 as to enable the Slave Mode.

```
write_audio_cfg_register(0x0, 0x17);
write_audio_cfg_register(0x1, 0x17);
write_audio_cfg_register(0x2, 0x7F);
write_audio_cfg_register(0x3, 0x7F);
write_audio_cfg_register(0x4, 0x15);
```

```
write_audio_cfg_register(0x5, 0x06);
write_audio_cfg_register(0x6, 0x00);
write_audio_cfg_register(0x7, 0x4D);
write_audio_cfg_register(0x8, 0x18);
write_audio_cfg_register(0x9, 0x01);
```

This should enable all of the correct settings for the audio core and it should be ready to be written to according to the 12.288 MHz clock specified in the Altera PLL.

Finally, to ensure the software can properly communicate with the hardware, specify the base addresses in the audio.h and audio\_cfg.h files. Import all of the audio files into your project and use them as follows.

- 1. Generate a buffer of audio data of type INT32S\*
- 2. Pass the buffer and frequency to the write function using write\_audio\_data(buffer, freq); An example is shown in the attached app.c file. Please take note that the SAMPLING\_RATE specified in app.c of 32000 Hz is dependent on the 12.288 MHz clock in the PLL. Changing the sampling rate to be say 44100 Hz requires further changes in the hardware to ensure the codec is reading out bytes at the same rate that the software is producing them.

#### References

[1] University of Toronto. *Digital Embedded Systems Lab: Audio Core.* [Online]. Available:http://www-ug.eecg.utoronto.ca/desl/nios devices SoC/dev audio.html

**[2]** WOLFSON MICROELECTRONICS plc. (2005). *Portable Internet Audio CODEC with Headphone Driver and Programmable Sample Rates.* [Online].

Available: <a href="https://www.rockbox.org/wiki/pub/Main/DataSheets/WM8731\_8731L.pdf">https://www.rockbox.org/wiki/pub/Main/DataSheets/WM8731\_8731L.pdf</a>

#### Attached Files

audio.c: Provides communication driver for writing to the audio buffers.

audio.h: Header file for audio core driver.

audio\_cfg.c: Provides configurations for audio and video cores.

audio cfg.h: Header file for audio and video core drivers.

app.c: Provides an example main.c and AppTask to produce audio output.

audio.qar: Archived Quartus/Qsys project example to provide functional audio.