

# 24-bit Audio CODEC

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Digital Circuit Lab

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# Outline

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- Introduction to Audio Signal
- Architecture Overview
- Device Initialization
- Device Operation



# Introduction to Audio Signal

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# Introduction

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- An audio signal is a representation of sound, typically as an **electrical voltage**.
- Audio signals have frequencies in the audio frequency range of roughly **20 to 20,000 Hz**.
- Audio signals may be **synthesized** directly, or may originate at a transducer such as a **microphone**.
- **Loudspeakers** or **headphones** convert an electrical audio signal into sound.



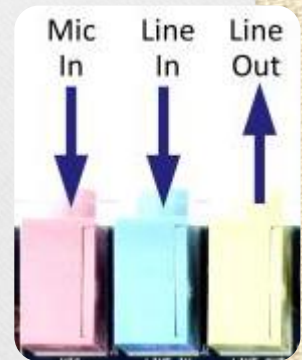
# Line Level (1/2)

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- Line level is the **specified strength** of an audio signal used to transmit analog sound between audio components.
- As opposed to line level, there are **weaker** audio signals, such as those from **microphones**, and **stronger** signals, such as those used to drive headphones and loudspeakers.

# Line Level (2/2)

- Consumer electronic devices concerned with audio (for example sound cards) often have a connector labeled "line in" and/or "line out."
- The line in/out connections on a consumer-oriented computer sound card are unbalanced, with a 3.5 mm (1/8") 3-conductor TRS minijack connector providing ground, left channel, and right channel.
- Professional equipment commonly uses balanced connections on 6.35 mm (1/4") phone jacks or XLR connectors.





# Line In V.S. Mic In

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- A line input level signal typically has a voltage ranging from 0.3 to 2 Volts.
- A microphone input level signal is more often in the range from 5 to 50 mV (millivolts).
  - Need microphone input level boost.



# Phone Connector (1/3)

- In electronics, a phone connector is a common family of connector typically used for analog signals, primarily **audio**.
  - It is also termed an **audio jack**, **phone jack**, etc.
- It is cylindrical in shape, typically with **two**, **three** or **four** contacts.
- Three-contact versions are known as **TRS connectors**, where T stands for "**tip**", R stands for "**ring**" and S stands for "**sleeve**".
- Similarly, two- and four-contact versions are called **TS** and **TRRS** connectors respectively.

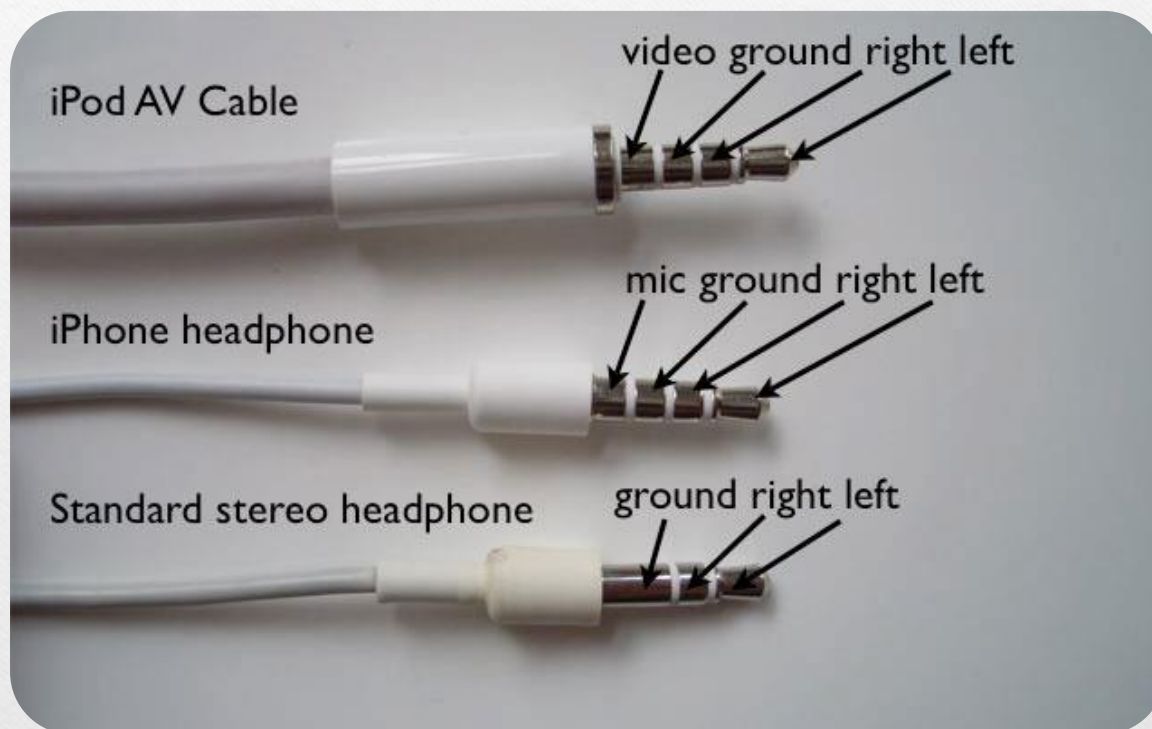


# Phone Connector (2/3)

- Modern phone connectors are available in **three** standard sizes.
  - 2.5 mm mono (TS)
  - 3.5 mm mono (TS)
  - 3.5 mm stereo (TRS)
  - 6.35 mm (1/4 in) (TRS)



# Phone Connector (3/3)

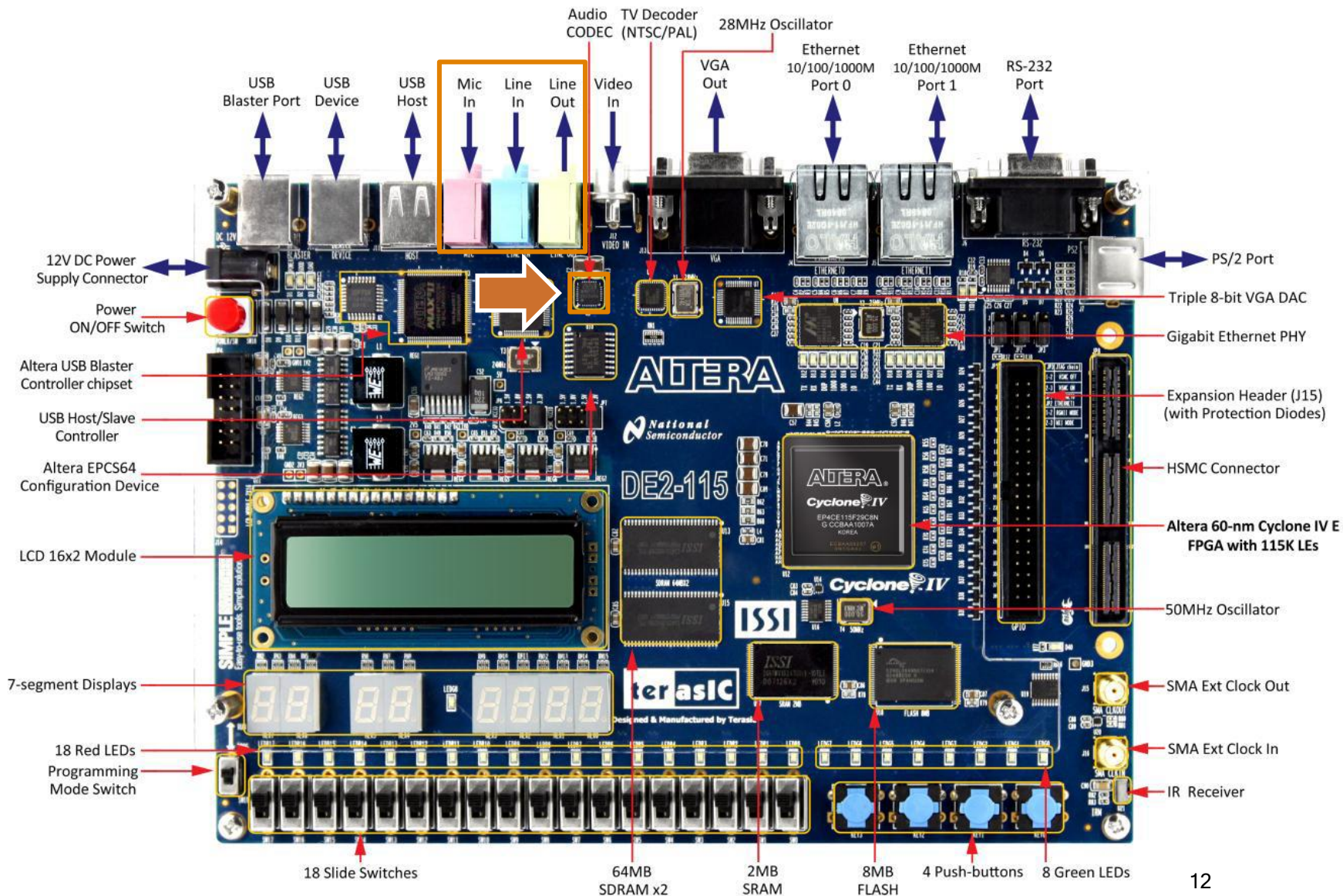




# Architecture Overview

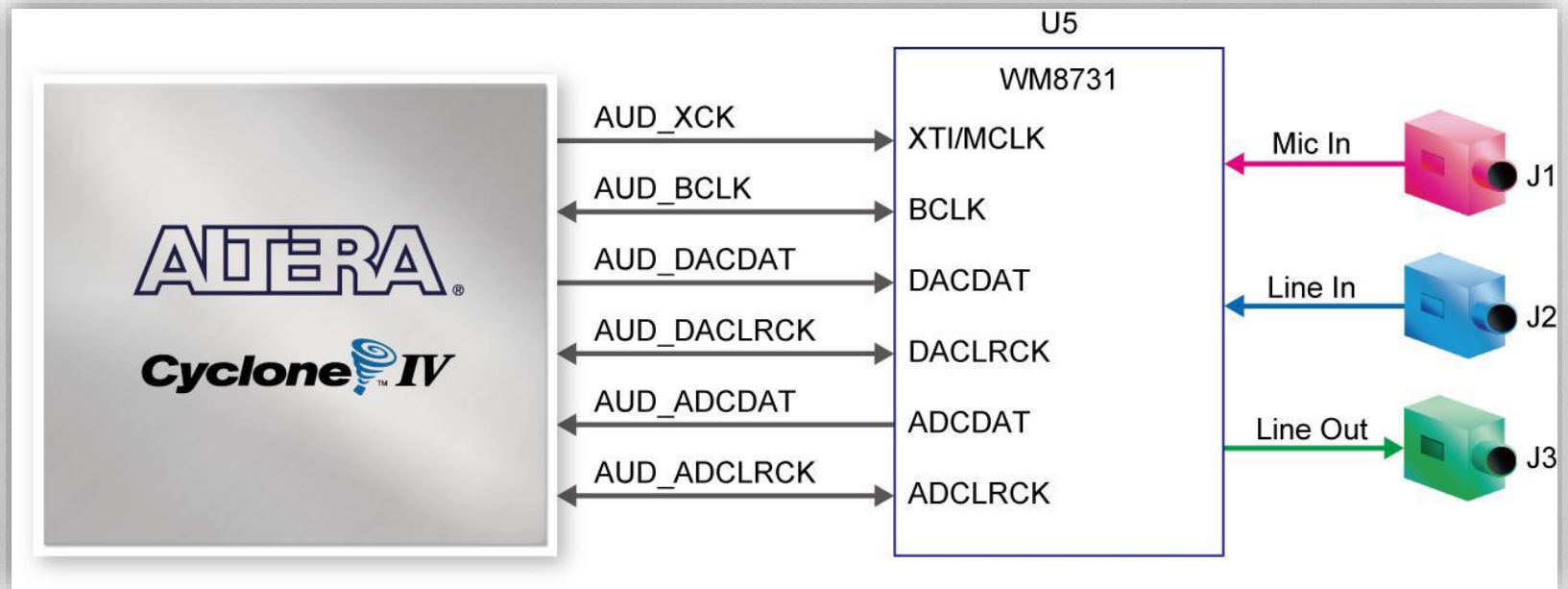
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# Schematic Diagram



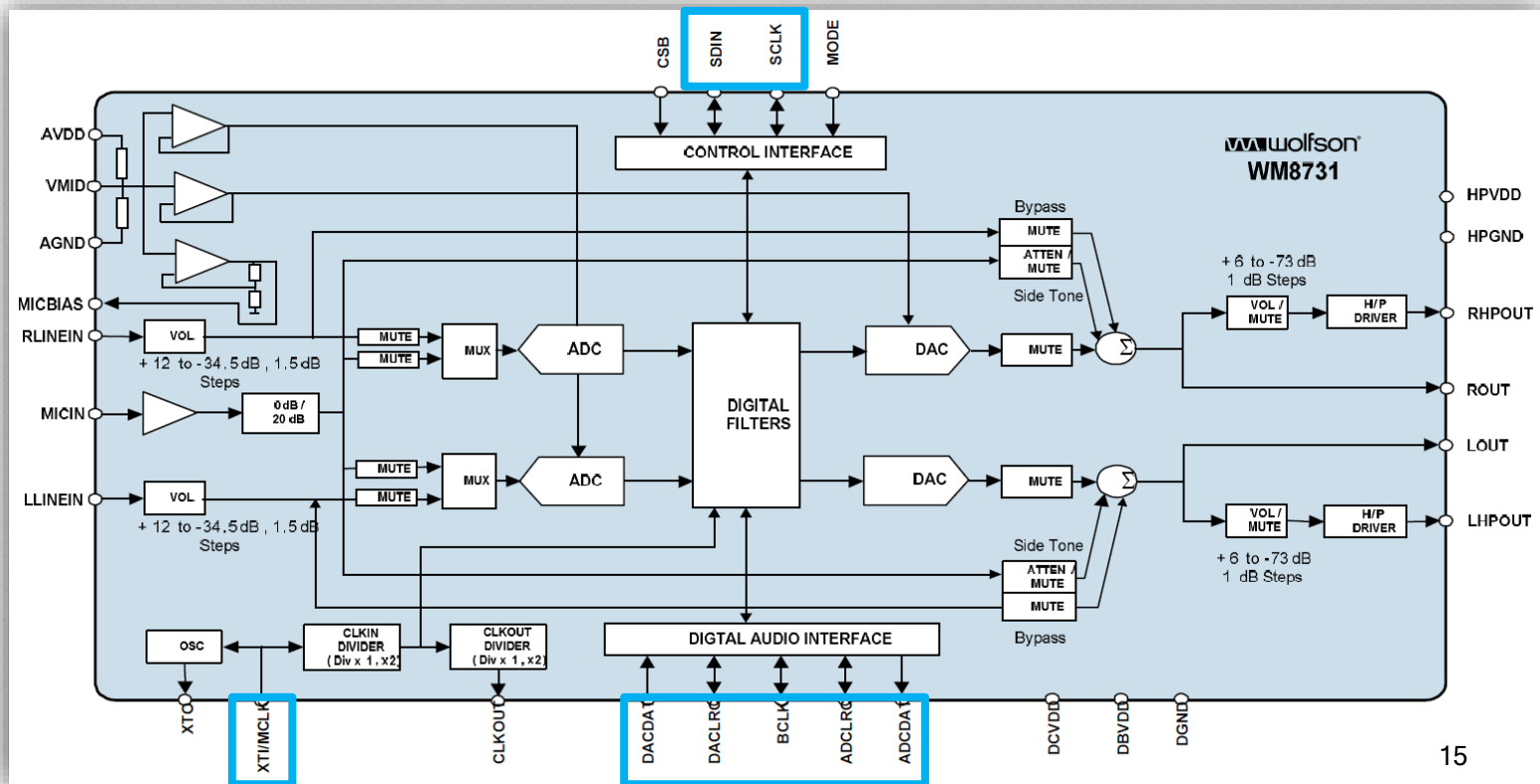
# Audio CODEC Pin Assignments

Signal Name	FPGA Pin No.	Description
I2C_SCLK	PIN_B7	I2C Clock
I2C_SDAT	PIN_A8	I2C Data
AUD_XCK	PIN_E1	Audio CODEC Chip Clock
AUD_BCLK	PIN_F2	Audio CODEC Bit-Stream Clock
AUD_ADCLRC K	PIN_C2	Audio CODEC ADC LR Clock
AUD_ADCDAT	PIN_D2	Audio CODEC ADC Data
AUD_DACLRC K	PIN_E3	Audio CODEC DAC LR Clock
AUD_DACDAT	PIN_D1	Audio CODEC DAC Data

Audio chip and TV decoder  
chip share one I2C bus

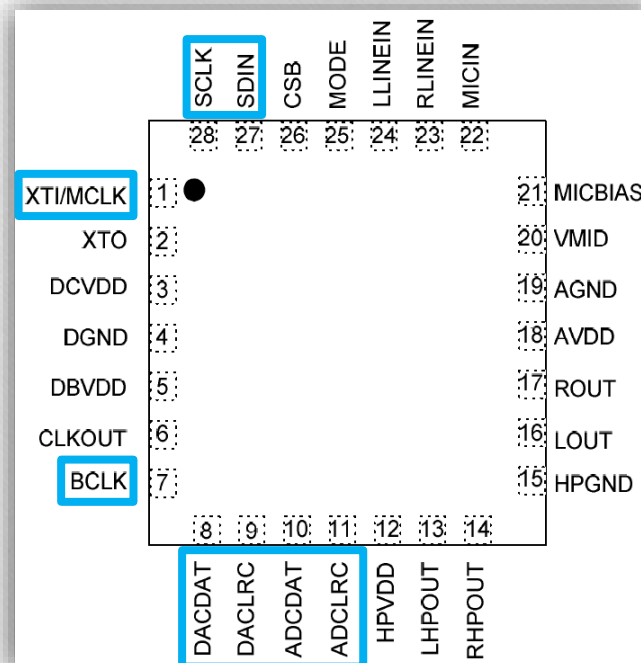


# WM8731 Block Diagram



# WM8731 Pin Configuration

- Top view of 28 pin QFN







# How to Use WM8731?

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1. Initialize the device by setting the registers via I2C bus interface.
2. After correct initialization, we can receive or transmit audio data via digital audio interface.



# Device Initialization

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# Software Control Interface (1/2)

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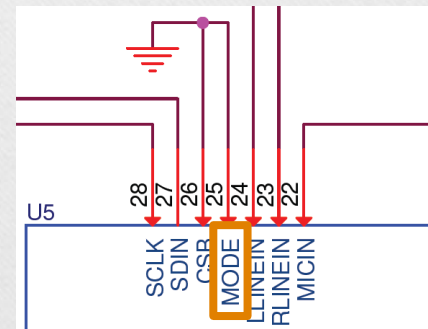
- Different modes can be configured under software control.
  - Input to ADC: Microphone
  - Sampling rate: 32kHz
  - Input audio data bit length: 16 bits
  - etc.



# Software Control Interface (2/2)

- Selection of serial control mode
  - The serial control interface may be selected to operate in either 2 or 3-wire modes. This is achieved by setting the state of the **MODE** pin.

MODE	INTERFACE FORMAT
0	2 wire
1	3 wire



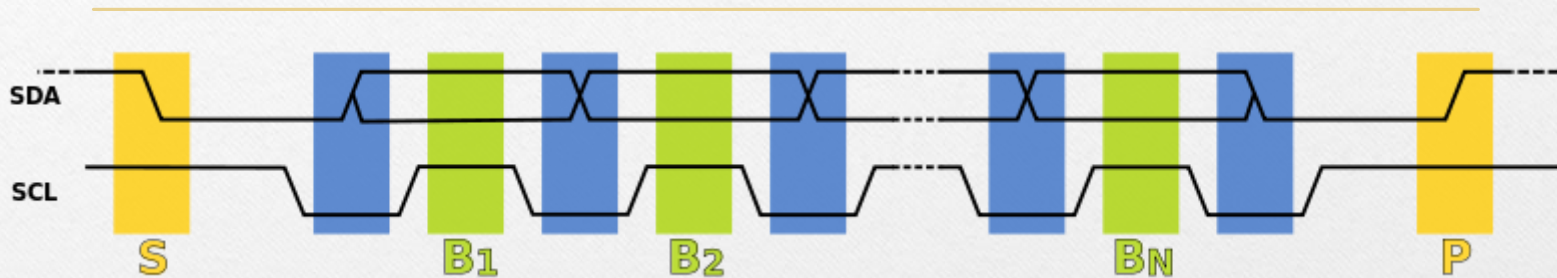
# 2-Wire Serial Control Mode

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- The WM8731/L supports a 2-wire MPU (Microprocessor unit) interface, which is compatible with I<sup>2</sup>C protocol.
  - I<sup>2</sup>C (Inter-Integrated Circuit, referred to as I-squared-C) uses only two bidirectional open-drain lines, Serial Data Line (SDA) and Serial Clock (SCL).

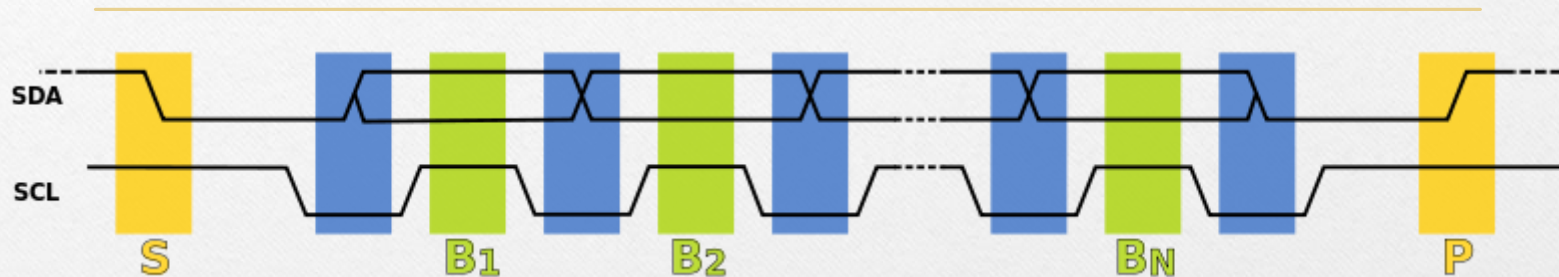


# I<sup>2</sup>C protocol (1/4)



- Data transfer is initiated with the START bit (**S**) when SDA is **pulled low** while SCL stays high.  
SDA=1'b0
- Then, SDA sets the transferred bit while SCL is low (**blue**) and the data is sampled (received) when SCL rises (**green**).

## I<sup>2</sup>C protocol (2/4)

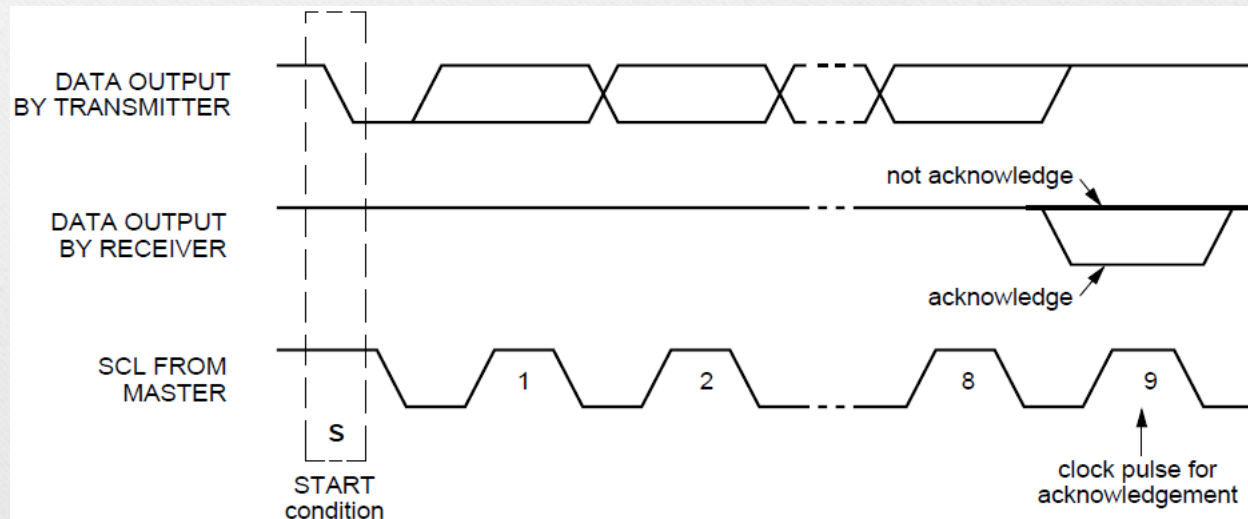


- When the transfer is complete, a STOP bit (**P**) is sent by **releasing** the data line to allow it to be pulled up while SCL is constantly high. *SDA=1'b1 or SDA=1'bz*
- In order to avoid false marker detection, the level on SDA is changed on the **falling edge** and is captured on the **rising edge** of SCL.



# I<sup>2</sup>C protocol (3/4)

- After every 8 data bits in one direction, an "acknowledge" bit (0) is transmitted in the other direction.



# I<sup>2</sup>C protocol (4/4)

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- About inout port:

```
module inout_port(oe, clk, SDA)

input  oe; // output enable
input  clk;
inout  SDA;

wire a; // output data
reg b; // input data

assign SDA = oe? a: 1'bz;

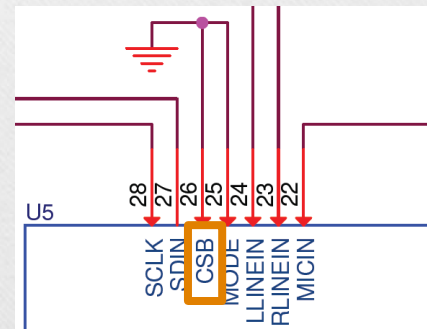
always @(posedge clk) begin
    b <= SDA;
end
```



## 2-Wire Interface (1/2)

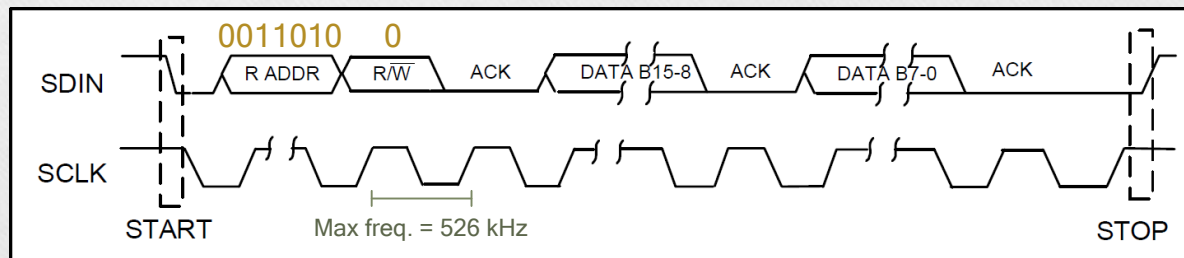
- The device operates as a **slave device** only.
- The WM8731/L has one of two slave address that are selected by setting the state of the **CSB** pin.

CSB STATE	ADDRESS
0	0011010
1	0011011



# 2-Wire Interface (2/2)

- 2-wire serial interface
  - ADDR[6:0] (7 bits) are Slave Address Bits
  - R/W is '0', indicating a write
  - B[15:9] (7 bits) are Register Address Bits
  - B[8:0] (9 bits) are Register Data Bits





# Register Map

REGISTER	BIT[8]	BIT[7]	BIT[6]	BIT[5]	BIT[4]	BIT[3]	BIT[2]	BIT[1]	BIT[0]	DEFAULT
<b>R0 (00h)</b> Left Line In	LRINBOTH	LINMUTE	0	0	LINVOL[4:0]					0_1001_0111
<b>R1 (01h)</b> Right Line In	RLINBOTH	RINMUTE	0	0	RINVOL[4:0]					0_1001_0111
<b>R2 (02h)</b> Left Headphone Out	LRHPBOTH	LZCEN	LHPVOL[6:0]						0_0111_1001	
<b>R1 (01h)</b> Right Headphone Out	RLHPBOTH	RZCEN	RHPVOL[6:0]						0_0111_1001	
<b>R4 (04h)</b> Analogue Audio Path Control	0	SIDEATT[1:0]		SIDETONE	DACSEL	BYPASS	INSEL	MUTEMIC	MICBOOST	0_0000_1010
<b>R5 (05h)</b> Digital Audio Path Control	0	0	0	0	HPOR	DACMU	DEEMPH[1:0]		ADCHPD	0_0000_1000
<b>R6 (06h)</b> Power Down Control	0	POWEROFF	CLKOUTPD	OSCPD	OUTPD	DACPD	ADCPD	MICPD	LINEINPD	0_1001_1111
<b>R7 (07h)</b> Digital Audio Interface Format	0	BCLKINV	MS	LRSWAP	LRP	IWL[1:0]		FORMAT[1:0]		0_1001_1111
<b>R8 (08h)</b> Sampling Control	0	CLKODIV2	CLKIDIV2	SR[3:0]				BOSR	USB/ NORMAL	0_0000_0000
<b>R9 (09h)</b> Active Control	0	0	0	0	0	0	0	0	Active	0_0000_0000
<b>R15 (0Fh)</b> Reset	RESET[8:0]									not reset

# Left Line In

REGISTER	BIT[8]	BIT[7]	BIT[6]	BIT[5]	BIT[4]	BIT[3]	BIT[2]	BIT[1]	BIT[0]	DEFAULT
R0 (00h) Left Line In	LRINBOTH	LINMUTE	0	0	LINVOL[4:0]					0_1001_011 1

- Just use the default setting if we do not use the line input.
  - 000\_0000\_0\_1001\_0111



# Right Line In

REGISTER	BIT[8]	BIT[7]	BIT[6]	BIT[5]	BIT[4]	BIT[3]	BIT[2]	BIT[1]	BIT[0]	DEFAULT
R1 (01h) Right Line In	RLINBOTH	RINMUTE	0	0	RINVOL[4:0]					0_1001_011 1

- Just use the default setting if we do not use the line input.
  - 000\_0001\_0\_1001\_0111

# Left Headphone Out

REGISTER	BIT[8]	BIT[7]	BIT[6]	BIT[5]	BIT[4]	BIT[3]	BIT[2]	BIT[1]	BIT[0]	DEFAULT
R2 (02h) Left Headphone Out	LRHPBOTH	LZCEN	LHPVOL[6:0]							0_0111_100 1

- Here we can just use the default setting.
  - 000\_0010\_0\_0111\_1001



# Right Headphone Out

REGISTER	BIT[8]	BIT[7]	BIT[6]	BIT[5]	BIT[4]	BIT[3]	BIT[2]	BIT[1]	BIT[0]	DEFAULT
R3 (03h) Right Headphone Out	RLHPBOTH	RZCEN	RHPVOL[6:0]							0_0111_100 1

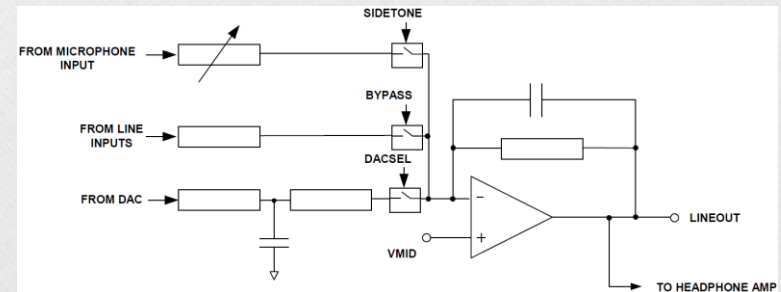
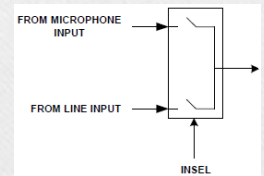
- Here we can just use the default setting.
  - 000\_0011\_0\_0111\_1001

# Analogue Audio Path Control

REGISTER	BIT[8]	BIT[7]	BIT[6]	BIT[5]	BIT[4]	BIT[3]	BIT[2]	BIT[1]	BIT[0]	DEFAULT
R4 (04h) Analogue Audio Path Control	0	SIDEATT[1:0]		SIDETONE	DACSEL	BYPASS	INSEL	MUTEMIC	MICBOOST	0_0000_101 0

- Enable boost, disable mute, choose microphone input, disable bypass, and select DAC.

- 000\_0100\_0\_0001\_0101





# Digital Audio Path Control

REGISTER	BIT[8]	BIT[7]	BIT[6]	BIT[5]	BIT[4]	BIT[3]	BIT[2]	BIT[1]	BIT[0]	DEFAULT
R5 (05h) Digital Audio Path Control	0	0	0	0	HPOR	DACMU	DEEMPH[1:0]		ADCHPD	0_0000_100 0

- Disable soft mute
  - 000\_0101\_0\_0000\_0000

# Power Down Control

REGISTER	BIT[8]	BIT[7]	BIT[6]	BIT[5]	BIT[4]	BIT[3]	BIT[2]	BIT[1]	BIT[0]	DEFAULT
R6 (06h) Power Down Control	0	POWER OFF	CLKOUTPD	OSCPD	OUTPD	DACPD	ADCPD	MICPD	LINEINPD	0_1001_111 1

- Choose power on and disable all the power down options.
  - 000\_0110\_0\_0000\_0000



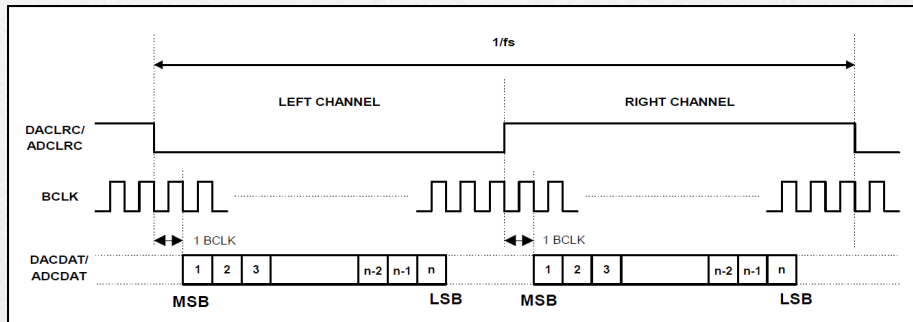
# Digital Audio Interface Format (1/2)

REGISTER	BIT[8]	BIT[7]	BIT[6]	BIT[5]	BIT[4]	BIT[3]	BIT[2]	BIT[1]	BIT[0]	DEFAULT
R7 (07h) Digital Audio Interface Format	0	BCLKIVE	MS	LRSWAP	LRP	IWL[[1:0]		FORMAT[1:0]		0_0000_101 0

- Choose I<sup>2</sup>S format, 16-bit length, and master mode.
  - 000\_0111\_0\_0100\_0010

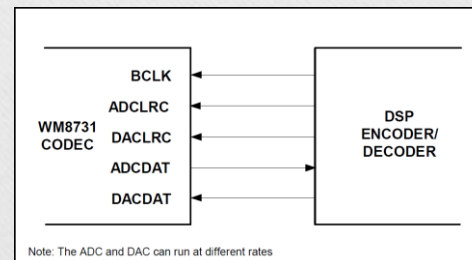
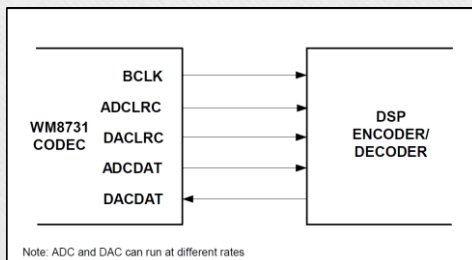
# Digital Audio Interface Format (2/2)

- I<sup>2</sup>S format



➤ 16-bit length

- Master mode v.s. Slave mode





# Sampling Control (1/2)

REGISTER	BIT[8]	BIT[7]	BIT[6]	BIT[5]	BIT[4]	BIT[3]	BIT[2]	BIT[1]	BIT[0]	DEFAULT
R8 (08h) Sampling Control	0	CLKODIV2	CLKIDIV2	SR[3:0]				BOSR	USB/ Normal	0_0000_000 0

- Choose USB mode (fixed MCLK 12MHz) and sampling rate = 32 kHz.
  - 000\_1000\_0\_0001\_1001

# Sampling Control (2/2)

SAMPLING RATE		MCLK FREQUENCY	SAMPLE RATE REGISTER SETTINGS					DIGITAL FILTER TYPE
ADC	DAC		REGISTER SETTINGS					
kHz	kHz	MHz	BOSR	SR3	SR2	SR1	SR0	
48	48	12.000	0	0	0	0	0	0
44.1 (Note 2)	44.1 (Note 2)	12.000	1	1	0	0	0	1
48	8	12.000	0	0	0	0	1	0
44.1 (Note 2)	8 (Note 1)	12.000	1	1	0	0	1	1
8	48	12.000	0	0	0	1	0	0
8 (Note 1)	44.1 (Note 2)	12.000	1	1	0	1	0	1
8	8	12.000	0	0	0	1	1	0
8 (Note 1)	8 (Note 1)	12.000	1	1	0	1	1	1
32	32	12.000	0	0	1	1	0	0
96	96	12.000	0	0	1	1	1	3
88.2 (Note 3)	88.2 (Note 3)	12.000	1	1	1	1	1	2



# Active Control

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REGISTER	BIT[8]	BIT[7]	BIT[6]	BIT[5]	BIT[4]	BIT[3]	BIT[2]	BIT[1]	BIT[0]	DEFAULT
R9 (09h) Active Control	0	0	0	0	0	0	0	0	Active	0_0000_000 0

- Activate interface
  - 000\_1001\_0\_0000\_0001

# Reset Register

REGISTER	BIT[8]	BIT[7]	BIT[6]	BIT[5]	BIT[4]	BIT[3]	BIT[2]	BIT[1]	BIT[0]	DEFAULT
R15 (15h) Active Control	RESET[8:0]									not reset

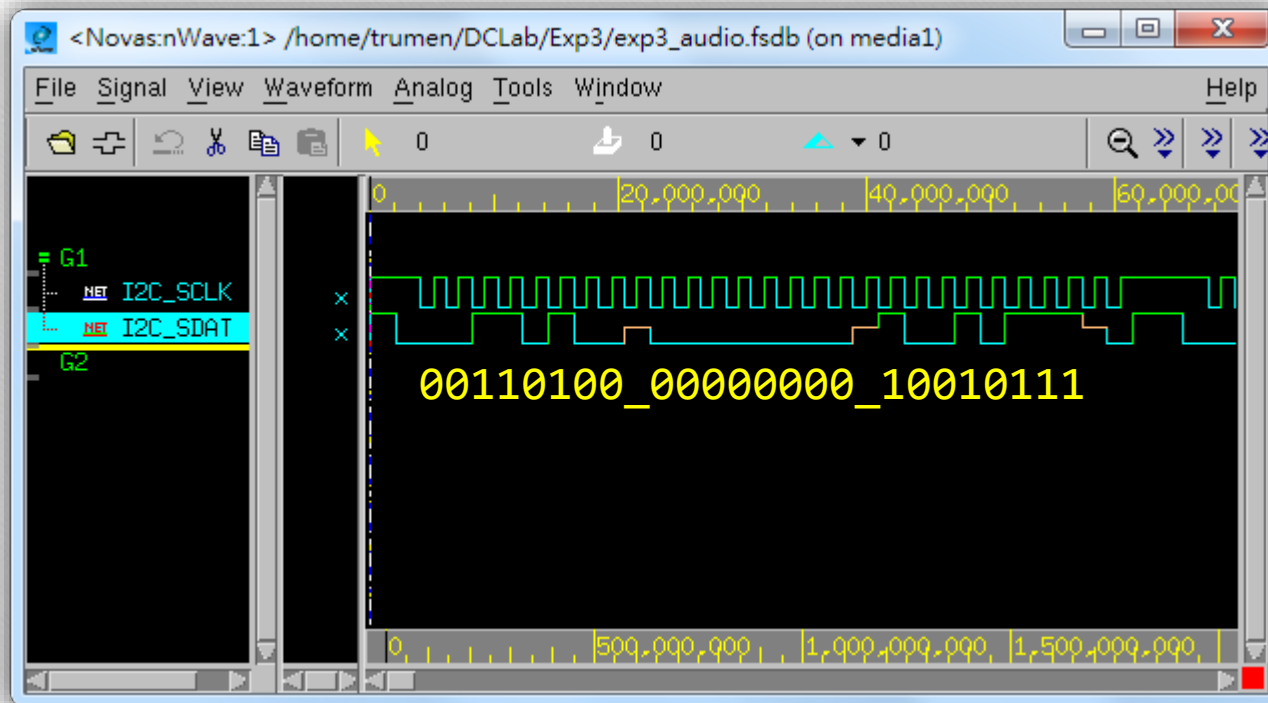
- You can try to reset the device to a known (?) state.
  - 000\_1111\_0\_0000\_0000 (?)



# Recommended settings

Left Line In	000_0000_0_1001_0111
Right Line In	000_0001_0_1001_0111
Left Headphone Out	000_0010_0_0111_1001
Right Headphone Out	000_0011_0_0111_1001
Analogue Audio Path Control	000_0100_0_0001_0101
Digital Audio Path Control	000_0101_0_0000_0000
Power Down Control	000_0110_0_0000_0000
Digital Audio Interface Format	000_0111_0_0100_0010
Sampling Control	000_1000_0_0001_1001
Active Control	000_1001_0_0000_0001

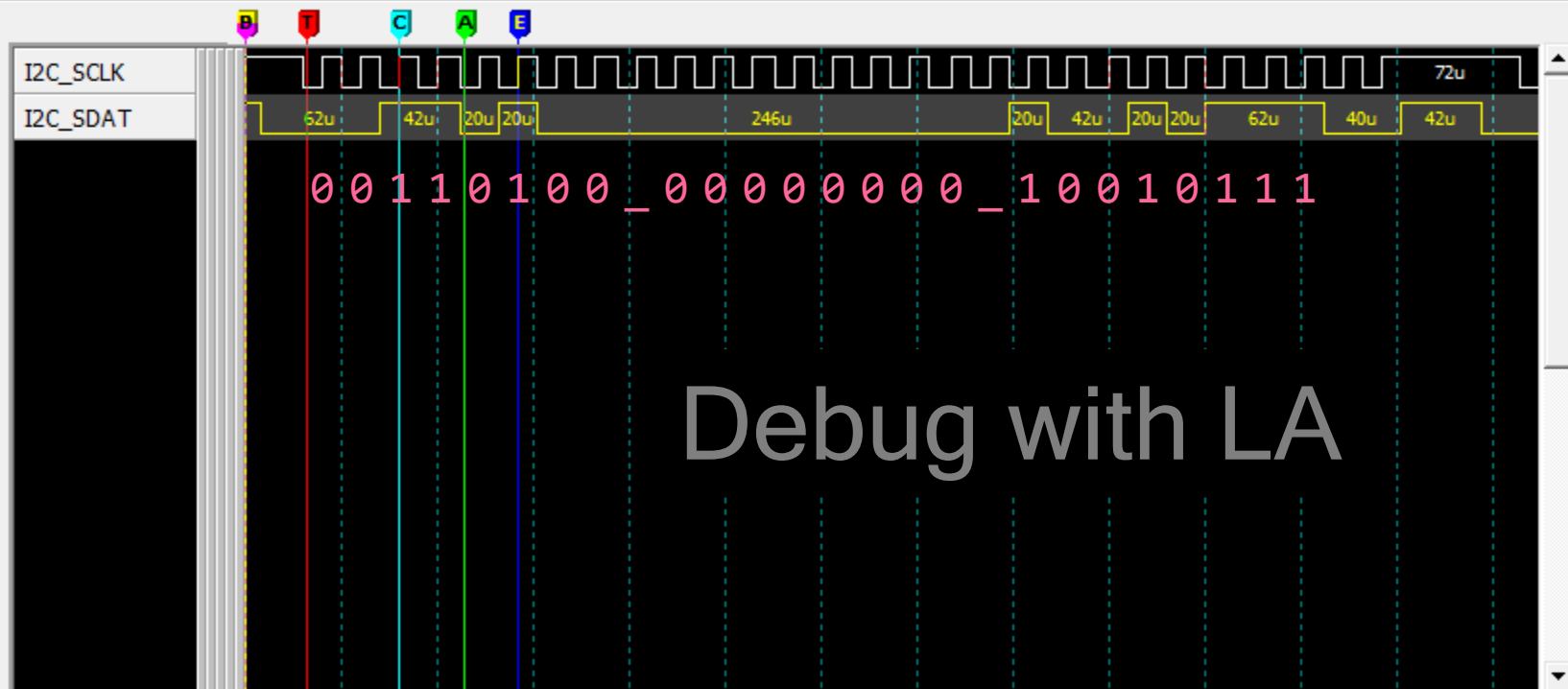
# Check the Waveform





邏輯分析儀 - 未命名

檔案(F) 訊號(L) 波形(W) 檢視(V) 硬體(S) 工具(T) 說明(H)



Debug with LA

Label

Wait for Trigger — Retrieve LA Waveform... 0.0%



T  
A

41

T  
B

16

A  
B

57



45

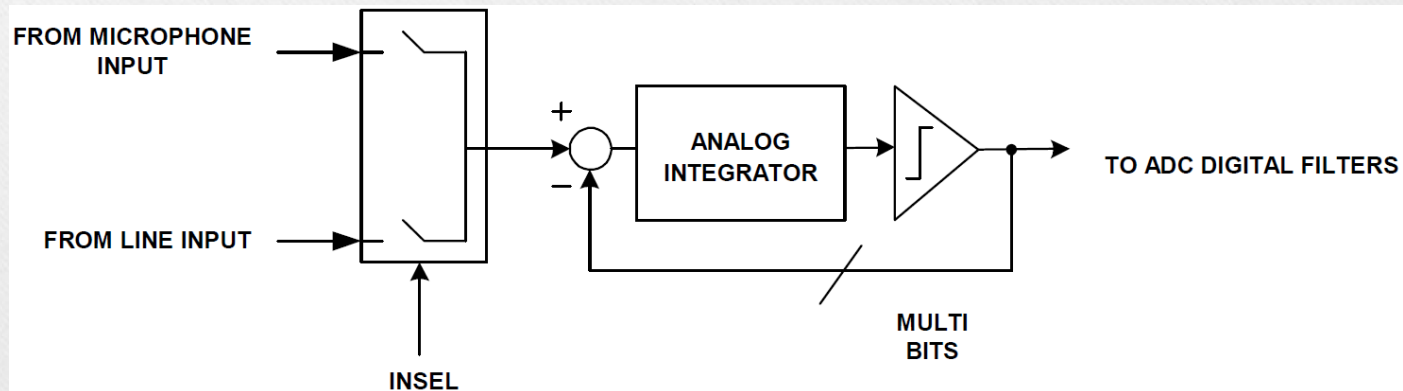
# Device Operation

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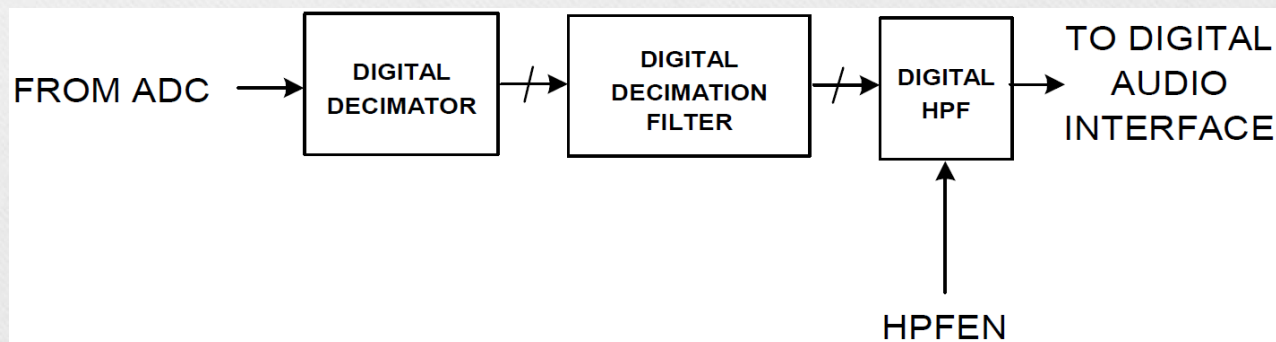
# ADC

- The WM8731/L uses a multi-bit oversampled sigma-delta ADC. A single channel of the ADC is illustrated below.



# ADC Filters

- The ADC filters perform true **24 bit signal processing** to convert the raw multi-bit oversampled data from the ADC to the correct sampling frequency to be output on the **digital audio interface**.





# Digital Audio Interfaces (1/5)

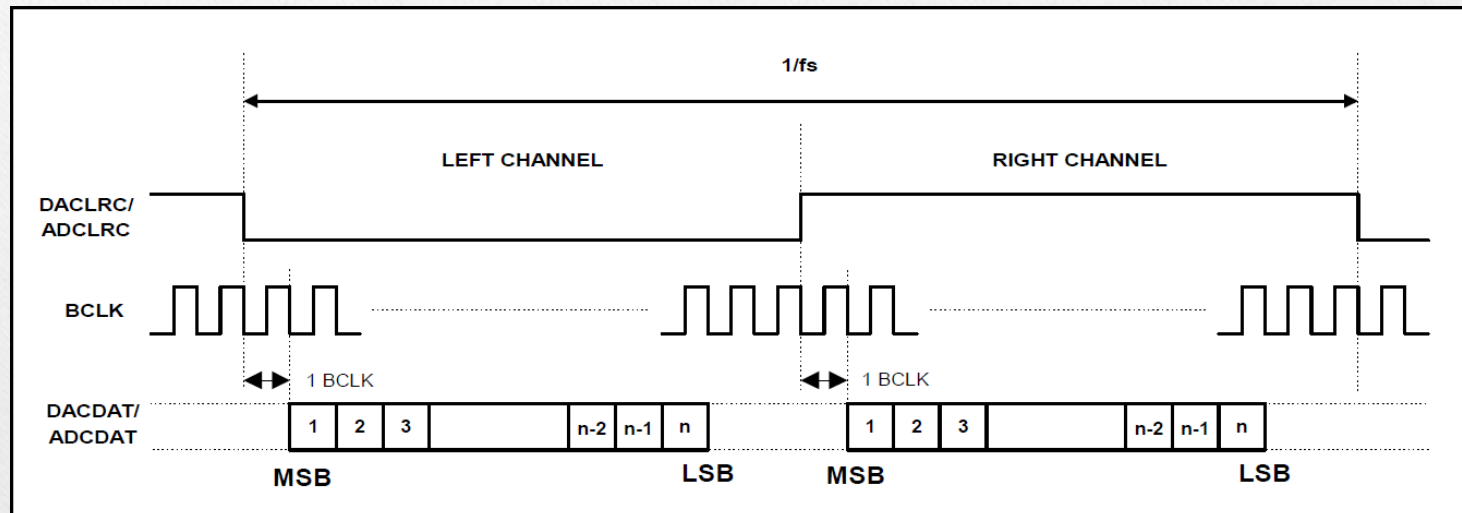
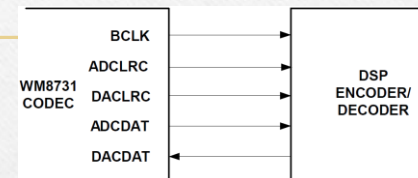
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- WM8731/L may be operated in either one of the 4 offered audio interface modes. These are:
  - Right justified
  - Left justified
  - I<sup>2</sup>S
  - DSP mode
- All four of these modes are MSB first and operate with data 16 to 32 bits.

# Digital Audio Interfaces (2/5)

- I<sup>2</sup>S mode

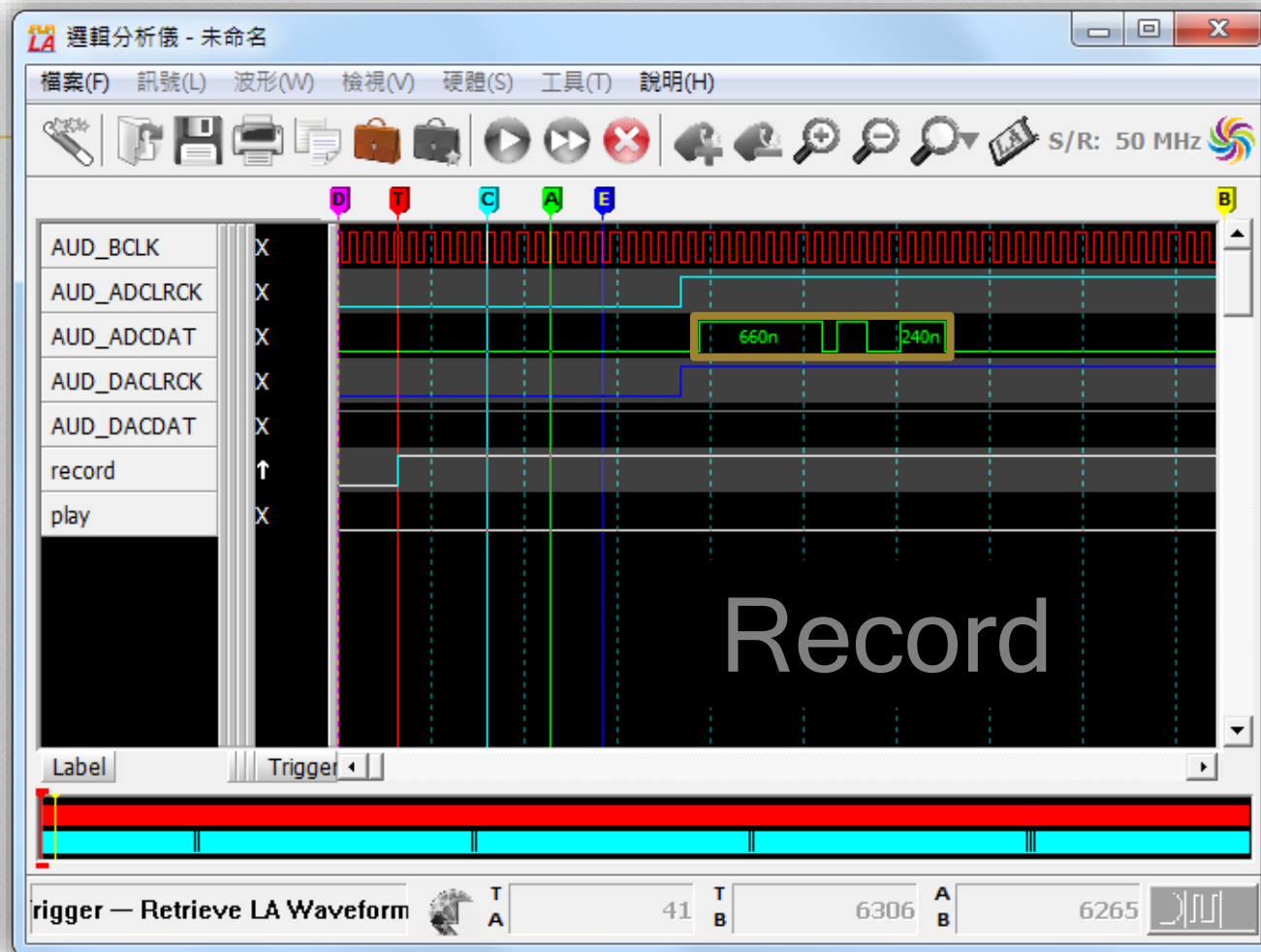
## Master mode



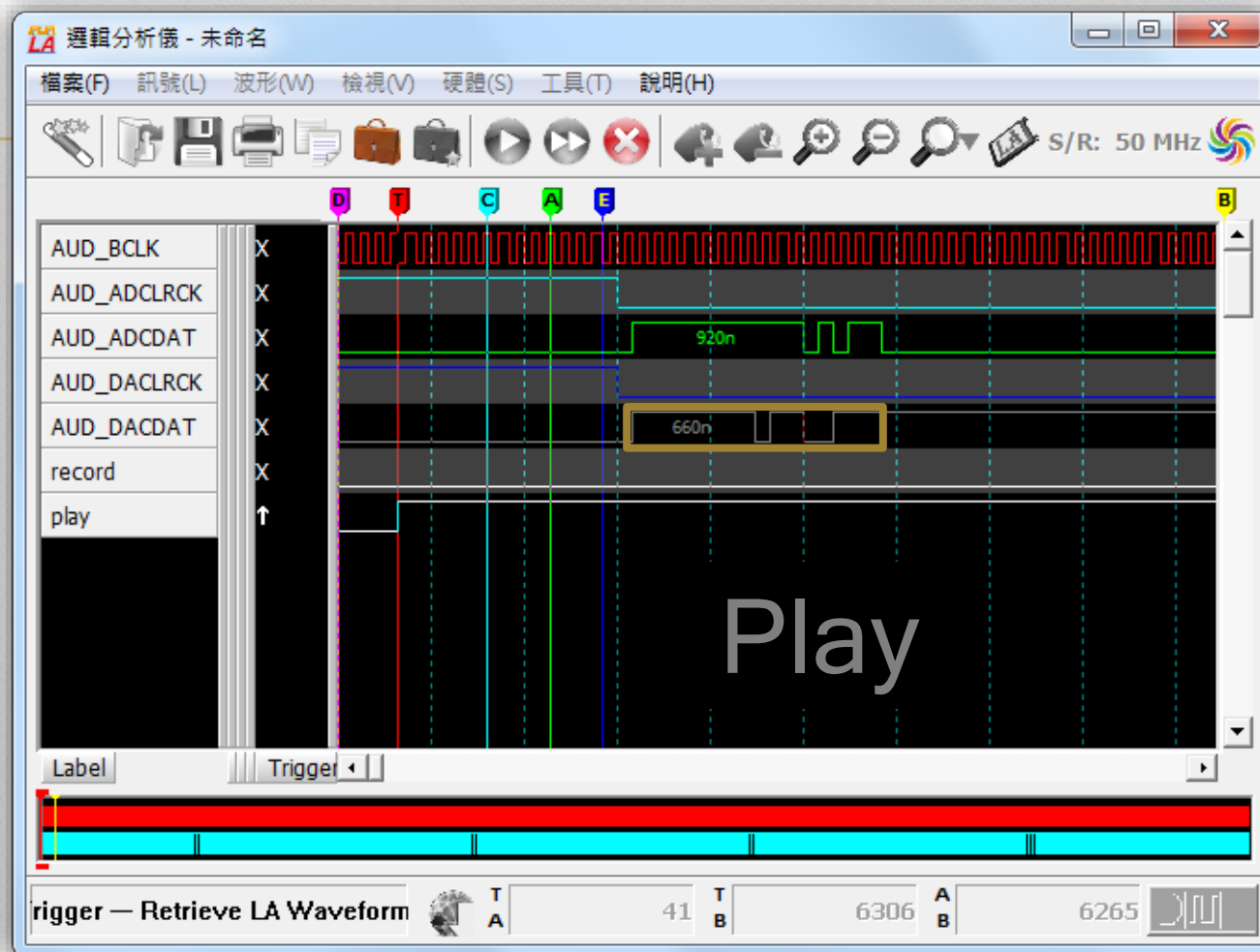
$n = 16, 20, 24, \text{ or } 32$



# Digital Audio Interfaces (3/5)



# Digital Audio Interfaces (4/5)





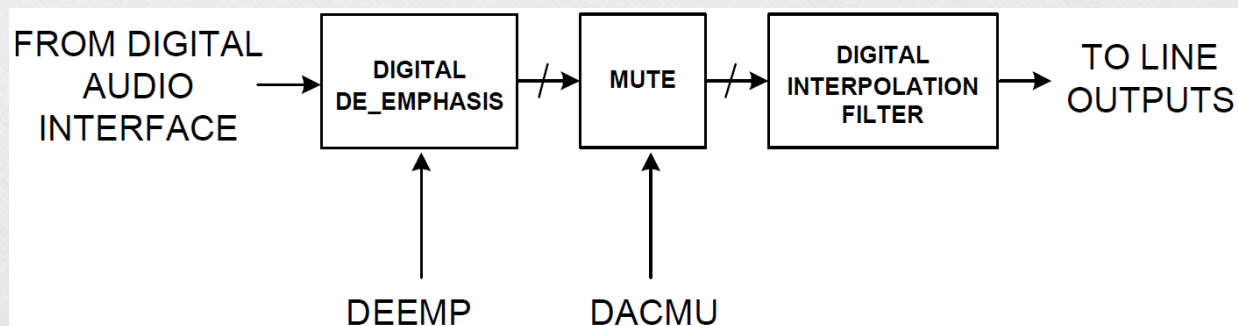
# Digital Audio Interfaces (5/5)

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- The length of the digital audio data is programmable at 16/20/24 or 32 bits.
- The data is signed 2's complement.
  - If the ADC is programmed to output 16 or 20 bit data then it strips the LSBs from the 24 bit data.
  - If the ADC is programmed to output 32 bits then it packs the LSBs with zeros.
  - Similar adjustments in DAC.

# DAC Filters

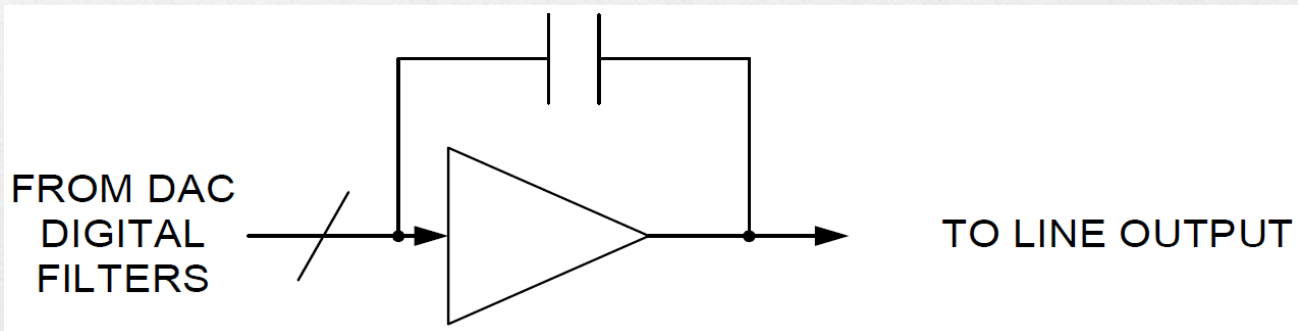
- The DAC filters perform true **24 bit signal processing** to convert the incoming digital audio data from the digital audio interface at the specified sample rate to multi-bit oversampled data for processing by the analogue DAC.





# DAC

- The WM8731/L employs a multi-bit sigma delta oversampling digital to analogue converter.



# The End.

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Any question?



# Reference

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1. [http://en.wikipedia.org/wiki/Audio\\_signal](http://en.wikipedia.org/wiki/Audio_signal)
2. <http://en.wikipedia.org/wiki/I%C2%B2C>
3. "THE I 2C-BUS SPECIFICATION VERSION 2.1" by Philips.
4. "DE2-115 User Manual" by Terasic.
5. "DE2-115\_MB.pdf" by Terasic.
6. "WM8731.pdf" by Wolfson Microelectronics.