Audio Processing With An Open Development System

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Overview

- MEMS MICROPHONES
  - Inside a digital MEMS microphone
  - Interface
  - PDM format

- FROM COMPONENTS TO SYSTEMS
  - MCU based devices and microphones acquisition
  - Microphones Arrays
  - Decimation, filtering and the PCM format

- FROM SYSTEMS TO APPLICATIONS
  - The Open.Audio framework
    - BlueVoice
    - Echo cancellation
    - Source localization
    - Beamforming

PCM Digital Audio

Dynamic Range

- 16 bit per sample,
- linear quantization
- $20\log_{10}(2^{16}) \cong 96\ dB$

Frequency

- Bandwidth: 20 Hz – 20 kHz
- Most common sampling rates:
  - 8 kHz, 16 kHz, 44.1 kHz, 48 kHz,
  - High Quality Audio: 88.2 kHz, 96 kHz, 192 kHz.

MEMS Microphones

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Digital MEMS Microphones: inside view

Digital MEMS microphones:

- ultra-compact, low-power, omnidirectional
- built with a capacitive sensing element and an IC interface

A/D and Digital i/f

Sensing

Pulse Density Modulation interface:

- 1 to 3 MHz
- 1-bit resolution
- Fully digital

- Capacitive membrane
- Omnidirectional
- Analog

Digital MEMS Microphones: outside view

- Digital MEMS microphone interface: relevant signals
  - Input Clock (CLK) @ 1 to 3 MHz (high frequency)
  - Data Out (DOUT): PDM, 1 bit per sample, @ input CLK frequency
  - Left Right (LR)
    - Enables connection of 2 microphones on the same data line
    - LR selects the CLK edge where data is valid

Bottom port

Top port

PCM Vs. PDM data format

- PCM Audio Signal – 16 bit per sample

- Pulse Density Modulation
  - The relative density of the pulses corresponds to the analog signal's amplitude

Noise Shaping property of PDM

PDM(*s(t)*): Single-Sided Power Spectrum

Audio Signal *s(t)*: Power Spectrum

From components to systems

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STM32 MCU for audio acquisition and processing

- STM32 Micro Controller Unit
  - Family of 32-bit Flash microcontrollers based on the ARM® Cortex®-M core
  - Very broad family of MCU featuring from very high performance to ultra low-power, supporting system optimization via on-chip peripherals and communication interfaces while maintaining ease of development.

- Example: STM32F446
  System-level performance
  - ARM® Cortex™-M4 32-bit RISC
  - 225 DMIPS and 608 CoreMark® @ 180MHz
  - 128 Kbyte RAM, 512k Flash
  - Floating Point Unit, DSP instructions
  - Up to 20 communication interfaces, including 4x USARTs 2x UARTs @ 11.25 Mbit/s, 4x SPI @ 45 Mbit/s, 3x I²C, 2x CAN, SDIO, HDMI CEC and camera interface.

- Example: STM32L4
  Ultra low power, dedicated PDM interface
  - ARM® Cortex™-M4 32-bit RISC
  - 100 DMIPS w. FPU and ST ART Accelerator™ @ 80MHz
  - 128 Kbyte RAM, up to 1024k Flash
  - Ultra Low-power mode, wake-up time 5 µs
  - DFSDM microphone interface with PDM to PCM conversion

STM32 MCU and audio acquisition

Direct acquisition of digital MEMS microphones via:

- Serial: SAI/I2S/SPI: 1 or 2 microphones share CLK and data line
- Parallel: GPIO: Up to 16 (or 32) microphones
- DFSDM dedicated interface – only on STM32L4
  - DFSDM = Digital Filter for Sigma Delta Modulator
  - Hardware acceleration for PDM to PCM conversion, for up to 4 microphones

Example of Microphone acquisition via I2S

STM32FXX

Single microphone - Mono

Dual microphone - Stereo

SPI Acquisition of a single microphone

SPI as a slave
external Quartz

- *12.288 MHz quartz is divided by 12 (timer)*
- *Microphone clock = 1024000 Hz*
- *Target Fs = 16 KHz*
- *Decimation factor = 64*

SPI as a slave
Clock generated by the timer

- *Timer clock input = 48000000*
- *Microphone clock = 1024000 Hz*
- *Target Fs = 16 KHz*
- *Decimation factor = 64*

Multi-Microphone Acquisition

SPI – Stereo Mode With External Quartz

GPIO acquisition of up to 16 microphones

External oscillator generates PCM Audio Sample rates
De-interleaver stage is required (can be SW):

Before de-interleaver:

After de-interleaver:

Before de-interleaver:

After de-interleaver:

PDM to PCM Conversion library

- Target: 16 bit PCM Format
- PDM to PCM conversion
  - PDM signal: 1 bit, @ 1 to 3 MHz
  - PDM 2 PCM Key processing stages:
    - Sinc3 low-pass filter with decimation
    - IIR signal conditioning
    - Gain control

PDM2PCM library structure

Initialization – once at system startup

- Audio Sampling Frequency
- Audio Channels Input / Output
- LP filter cutoff frequency
- HP filter cutoff frequency

PDMFilter_InitStruct

Runtime – once every 1 ms

- PDM Buffer (in)
- PCM Buffer (out)
- Microphone Gain
- PDMFilter_InitStruct

PDM2PCM library performance

- Library configuration
  - Low pass filter (24000Hz) enabled
  - High pass filter (10Hz) enabled

- Values expressed in MIPS for typical configurations (*)

PCM sampling rate	16 KHz	32 KHz	48 KHz
PDM_Filter_64	~5	~10	~15
PDM_Filter_80	~6	~12	N/A
PDM_Filter_128	~9	N/A	N/A

(*) microphone clock = PCM sample rate x decimation factor

- Total amount of RAM = 780 bytes
- Audio frame corrisponds to 1 ms of PDM audio.
- Profiling done using IAR Embedded workbench, version 7.30 in simulation mode; optimization set to High Speed

MEMS Microphone development systems examples

- *Single-board* solutions:

- Modular architectures:

+

+

=

X-NUCLEO-CCA02M1

- MEMS microphone evaluation board
  - STM32Nucleo Expansion, compatible with STM32 ODE
- 2X MP34DT01–M microphones
- 1x miniUSB FS connector:
  - USB audio data streaming
- Up to 4 microphone synchronized acquisition and streaming
- 6X ST MEMS Microphone coupons housing:

STEVAL-MKI129V1
STEVAL-MKI129V2
STEVAL-MKI129V3

X-NUCLEO-CCA02M1 Acquisition Architecture

- 2 microphones sharing the data line
  - Enabled by L/R PIN configuration

- 1X I2S
  - Master mode: precise timing
  - Acquires data from 2 microphones
- 1X SPI
  - Slave mode: uses I2S clock
  - Acquires data from 2 microphones
- 1X Timer
  - Halves the frequency of the I2S clock to feed the microphones

X-NUCLEO-MEMSMIC1

- SDK to build applications using digital MEMS microphones
  - Microphones acquisition source code
- Audio input class USB driver
  - Standard USB microphone
  - Audio streaming to a Host PC

User may just focus on audio processing development

```
/**
* @brief  User function that is called when 1 ms of PDM data is available.
*         In this application only PDM to PCM conversion and USB streaming
*                  is performed.
*         User can add his own code here to perform some DSP or audio analysis.
* @param  none
* @retval None
*/

void AudioProcess(void)
{
  BSP_AUDIO_IN_PDMToPCM((uint16_t * )PDM_Buffer,PCM_Buffer);
  Send_Audio_to_USB((int16_t *)PCM_Buffer, AUDIO_SAMPLING_FREQUENCY/1000*AUDIO_CHANNELS);

}
```

Single entry point for user's Audio Signal Processing code

From systems to applications

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Low power, low resources design

- IoT typical constraints can be met also by embedded Audio designs based on MEMS microphones and Microcontrollers:

- Low Power constraints
  - Battery powered designs
  - Wearable
  - Always on systems

- Low resources constraints
  - Limited MIPS/MHz count
  - Limited RAM and FLASH

- Low cost constraints

A system-level solution should deploy an optimal balance of HW and SW resources

Open Software eXtension free SW licensing program

Open.Audio
free SW licensing program

Open Software Expansion Environment

Libraries

- Open. MEMS
- Open. RF
- Open. Audio

Open.Framework: Implementation Ex

- BlueVoiceLink
- BlueMicroSystem

STM32 ODE

- STM32 Nucleo development boards
- STM32 Nucleo expansion boards
- STM32 Cube software
- STM32 Cube expansion software

Embedded Audio Software building blocks

Open.AUDIO BlueVoice: Voice streaming over BLE

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BlueVoiceLink application overview

Central Unit

Microphone

BLE

X-NUCLEO-IDB04A1
X-NUCLEO-CCA02M1

STM32 Nucleo

USB

USB out
@16kHz

Half-duplex
audio
@64kbps

Bluetooth SMART

Peripheral Unit

BLE

Microphone

X-NUCLEO-IDB04A1
X-NUCLEO-CCA02M1

STM32 Nucleo

USB

USB out
@16kHz

www.st.com/bluevoicelink

Open Software Expansion

Open.FRAMEWORK

OSXBlueVoiceLink
The BlueVoice implementation example

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BlueVoiceLink SW and HW Architecture

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Open.AUDIO OSXBlueVoice library

Voice over BLE Vendor Specific Profile library

www.st.com/bluevoice

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MEMS MICs SW libraries soon to join the Open.AUDIO family

- SOUND SOURCE LOCALIZATION
  - using a MEMS microphone array, it estimates the angle of arrival of audio signal

- BEAMFORMING
  - using a MEMS (omnidirectional) microphone array, it creates a virtual directional microphone

- AEC
  - Acoustic Echo Cancelation (Speex based)

Open.AUDIO Beamforming

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Open.AUDIO Beamforming Library

- Input: 2 PDM signals from ST digital MEMS microphones
  - PDM to PCM conversion integrated for high efficiency
  - PDM domain processing achieves very high temporal resolution (no PCM sample interpolation)

- Based on Differential Microphone Array configuration, with microphones spaced *up to* 21 mm.
  - Compatible 'by default' with the X-NUCLEO-CCA02M1 STM32Nucleo Expansion board
  - Easily customizable using plugin Microphones Coupon boards
- The library supports 4 algorithm options:
  - "Cardioid basic",
  - "Cardioid Denoise"
  - "Strong"
  - "ASR ready"

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X-Nucleo based Beamforming: basic configurations

2-microphone configuration based on the onboard MP34DT01-M

4-microphone configuration based on MEMS microphone coupon such as STEVAL-MKI155V1

Linear

Squared

Beamforming Library – Algorithm options

- Cardioid basic: 1st -order Differential Microphone Array (DMA)

- Cardioid denoise: a denoise filter is added to the end fire beam forming output

- Strong: back to back cardioid and adaptive noise removal filter

- ASR ready: conceptually equivalent to the Strong optimization, but specifically optimized for Automatic Speech Recognition applications.

Beamforming Profiling

- MIPS footprint has been computed when input signal is in PDM format. Both generic distance and the special case when d = 2.1 cm are shown.

	d < 2.1 cm	d = 2.1 cm
Level	CPU (MIPS)	CPU (MIPS)
Basic Cardioid	11.5	9.7
Cardioid Denoise	41.2	39.5
Strong	84.3	73.4
ASR Ready	54.3	43.4

- Audio frame corrisponds to 128 PCM sample, equal to 8 ms.
- RAM footprints varies with the level of optimization, as follows:

Level	RAM (Bytes)
Basic Cardioid	2580
Cardioid Denoise	20892
Strong	41212
ASR Ready	22900

- Profiling done using IAR Embedded workbench, version 7.30; optimization set to High Speed

Open.AUDIO
Sound Source Localization

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Sound Localization

- Signals are acquired by one or two *couples* of microphones in order to estimate the sound direction of arrival
  - 360 degrees range with 4 microphones
  - 180 degrees range with 2 microphones

- Up to 1 degree theoretical resolution
  - Selectable resolution
- Two algorithms implemented
  - **XCORR:**

    low-MIPS and low-resolution
    Supports cm-sized microphone arrays

  - **GCC-PHAT:**

    Supports mm-sized Differential Arrays

- Simple Voice activity detector included, based on energy threshold

Sound Direction of Arrival

X-CORR: Cross correlation algorithm
- estimates the acoustic delay Δ
- Derives angle ϑ (DoA) from Δ

Far Field Sound Source

Acoustic Delay

Angle ϑ is the Direction of Arrival

Angular resolution depends on:

- Sampling frequency Fs
- Distance *d* between microphones

$$\Delta = d \cos \vartheta$$

Source Localization Profiling

- RAM, FLASH and MIPS footprint has been computed for a continuous execution of the source localization library at every input frame
- In typical applications a simple and effective optimization can be implemented by reducing the frequency of localization update, leading to a lower MIPS count.
- Audio frame corresponds to 512 PCM sample at 16 KHz (32 ms).
- Profiling was done using IAR Embedded Workbench, version 7.30; optimization set to High Speed, based on target STM32F401 @ 84MHz

Algorithm	Microphones	Resolution	CPU (MIPS)	FLASH (Bytes)	RAM (Bytes)
XCORR	2	30 degrees (distance = 0.08 m)	2.2	3424 + 24714 (*) (*) for CMSIS DSP software library by ARM, code and tables.	1161+ 2048* Channel Number
XCORR	4	30 degrees (distance = 0.08 m)	4.5		
XCORR	2	12 degrees (distance = 0.16 m)	4.0		
XCORR	4	12 degrees (distance = 0.16 m)	7.9		
GCC-PHAT	2	1 degree	58.4		9236 + 4096* Channel Number
GCC-PHAT	4	1 degree	116.2		
GCC-PHAT	2	2 degrees	31.8		
GCC-PHAT	4	2 degrees	63.0		
GCC-PHAT	2	6 degrees	14.1		
GCC-PHAT	4	6 degrees	27.6		

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Beamforming and Localization development system

- BEAMFORMING
  - using a MEMS microphone (omnidirectional) array, the library enables to create a virtual directional microphone

- SOUND SOURCE LOCALIZATION
  - using a MEMS microphone array, it can estimate the angle of arrival of the audio signal

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Open.AUDIO
Acoustic Echo Cancellation

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Acoustic echo cancellation

REMOVE FAR END ECHO IN NEAR END TRANSMISSION

- ✓ VOIP
- ✓ Teleconference systems
  - Business
  - Residential
  - Consumer
  - Automotive

Acoustic echo cancellation

- Acoustic Echo Cancellation removes a *far end* signal from the audio acquired by a microphone.
- Typical use: when a microphone and a loudspeaker coexist in the same environment and are driven by the same microcontroller.
- The Open.AUDIO AEC library is based on the Open Source Speex suite and includes:
  - AEC filter
  - Adjustable filter tail length
  - White noise filter
  - Simple automatic gain control

AEC - profiling

- Profiling has been done in order to estimate MIPS, RAM and FLASH consumption.
- The tool chain used for the tests is IAR Embedded Workbench 7.30, with optimization set to High, speed. Target adopted is a STM32F407 running at 168 MHz, with a total amount of 210 MIPS.
- For the test, filter tail length has been set to 128

Options	CPU (MIPS)	FLASH (Bytes)
AEC	54.7	35507
AEC + Denoiser	92.3	
AEC + Denoiser + AGC	97.7	

- RAM utilization depends on the chosen tail length and on the activated options. For example, using a tail length of 128 with all the options activated the library needs about 40 KB.

Advanced Demonstrators

BlueVoice + Speech Recognition

cloud-based ASR

Voice Capture - Tx

Audio
Over
ble

Central Unit - Rx

BlueVoice processing brings no degradation of performance for 3rd party ASR

Beamforming and Localization Demo system

- Hardware
  - Nucleo-F401RE (STM32 board)
  - X-NUCLEO-CCA02M1 (AUDIO IN expansion board)
  - µ-4 array hosting 4 MP34DT01 digital MEMS microphones
- Firmware
  - Microphone acquisition
  - Embedded Beamforming
  - Embedded Source Localization
  - Composite USB output:
    - Audio class (audio streaming)
    - Virtual COM Port (configuration)
- Software
  - PC control software to configure:
    - Optimizations
    - Beam direction
    - Source localization
    - ASR (3rd party embedded solution)

Thank you!

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