Embedded System for Audio Source Separation

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Overview

- Purpose
- Hardware Architecture
- Source Separation Algorithm
- Experimental Results

Introduction

- PURPOSE

  To provide an embedded solution to real time audio signal processing, such as in source localization and separation applications.

- ALGORITHM

  Beamforming algorithm - preserves the signal of

  a single source, coming from a selected direction, while

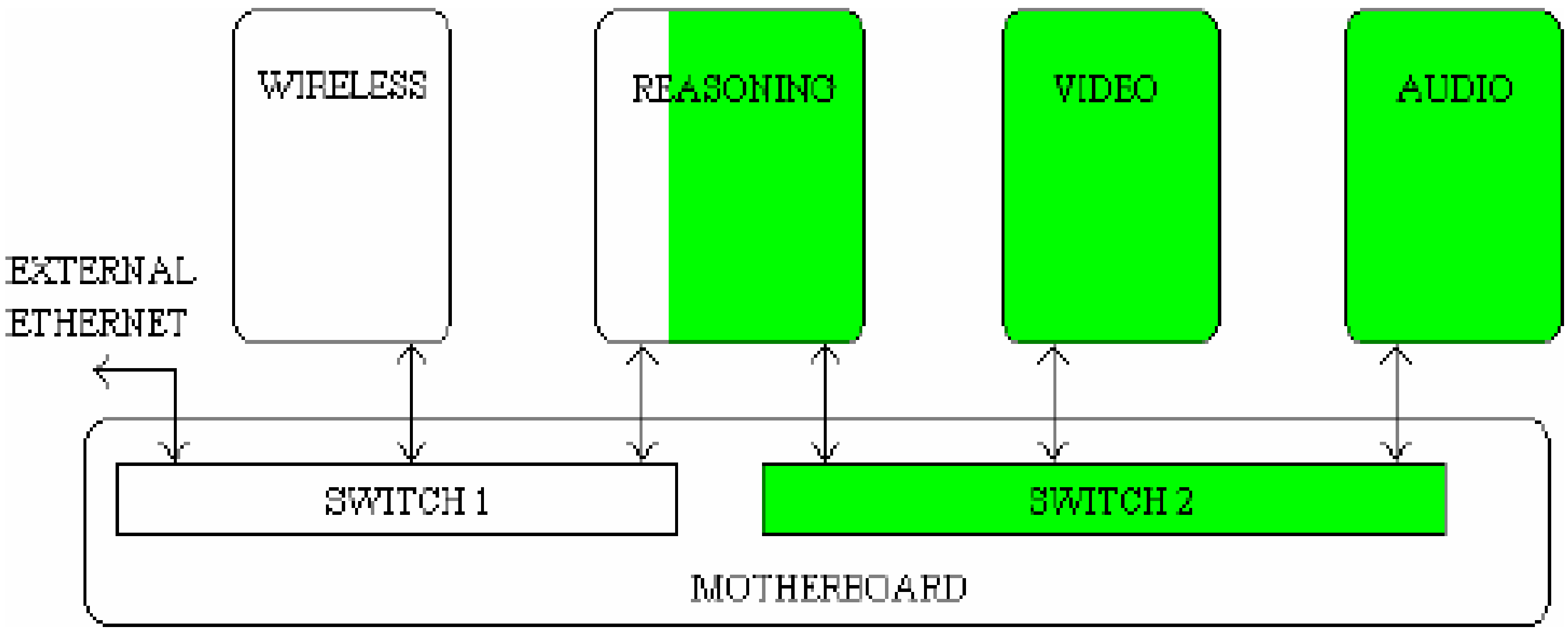
  rejecting the signals coming from other directions.

- APPLICATIONS

  source localization, audio events recognition, speaker recognition, etc.

HARDWARE PLATFORM

Structure of the Embedded System

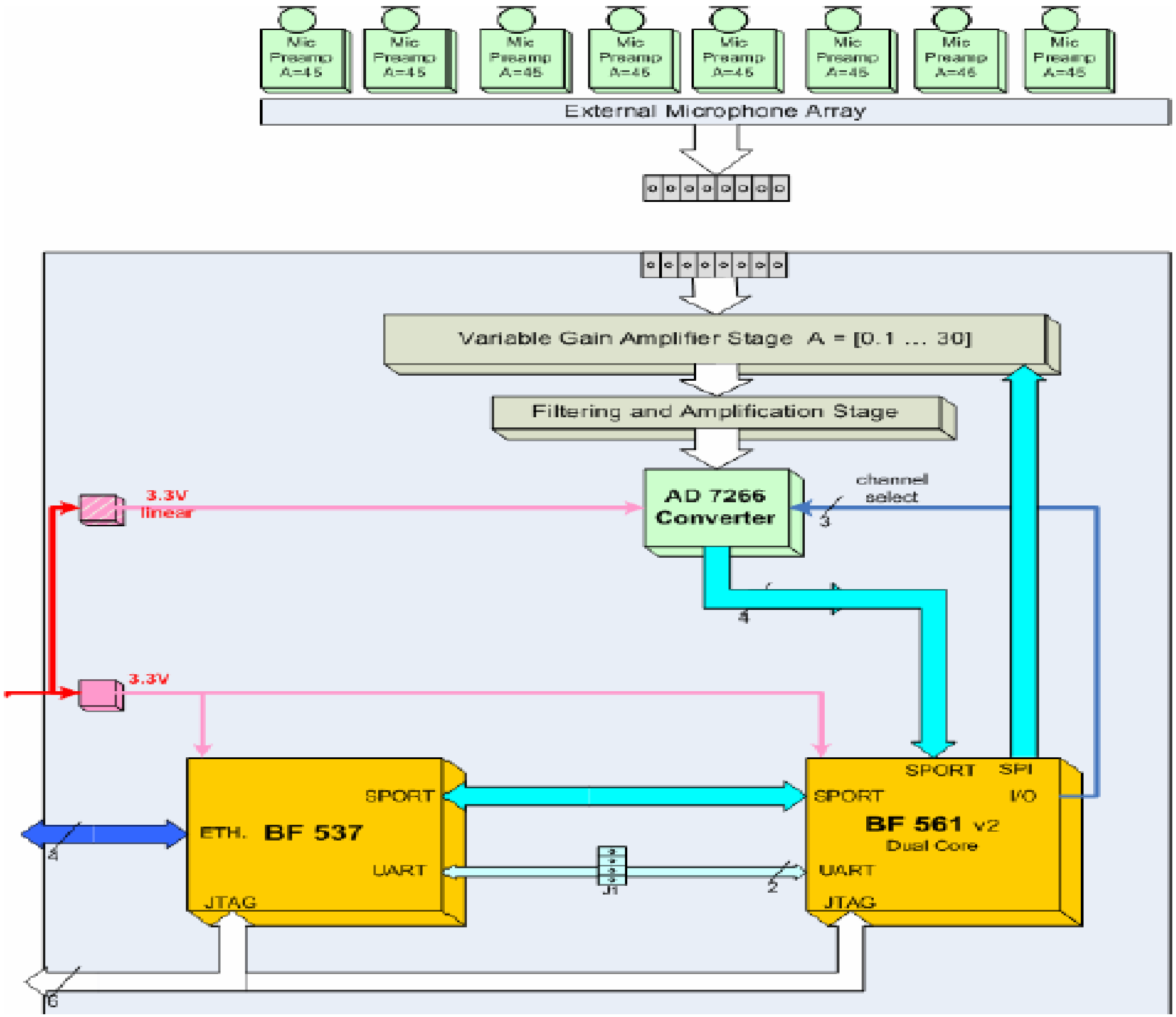


AUDIO BOARD

HARDWARE RESOURCES

- Microphone array
- Signal conditioning circuits
- AD converter
- 2 DSP modules
- Communication interfaces

BLOCK DIAGRAM OF THE AUDIO BOARD



MICROPHONE ARRAY

- 8 electret microphones and the corresponding low noise amplifiers placed on a microphone array
- uniformly placed in a linear array, all of them having the axis in the same direction, normal to their line

SIGNAL CONDITIONING CIRCUITS

- allow filtering (cutoff frequency is 22 kHz) and programmable gain (the range is 10-2700)

A/D CONVERTER

- AD 7266, 12 bits, 8 input channels, 2 simultaneous samplings, up to 375 kHz sampling frequency

DSP MODULES

- responsible for performing data acquisition, data buffering, signal processing and communication with the other boards
- One of the modules contains the processor Blackfin BF537, the other one contains the dual-core Blackfin BF561.

BF561

BF537

- The processor BF561 performs configuration of the data channels, data acquisition, data buffering and audio signal processing.
- BF537 performs signal processing and Ethernet communication

COMMUNICATION INTERFACES

- Between processors - SPORT , UART

  SPORT - synchronous serial port communication - allows data flow up to 60 Mbps The processors communicate to each other via a fast serial

  UART - asynchronous serial interface - lower speed - up to 7 Mbps

- AD 7266 -> BF561

  SPORT interface
- BF561 -> AD 7266

  SPI
- AUDIO BOARD <-> OTHER BOARDS

  Ethernet interface on the BF537
- JTAG INTERFACE

  used for program loading in the flash memory and debugging.

SOURCE SEPARATION THROUGH BEAMFORMING

Beamforming

- a signal processing technique used in sensor arrays for directional signal transmission or reception

'Delay and Sum' - conventional beamforming algorithm

- makes use of the delayed versions of the recorded signals and modify the characteristic of the microphone array, so as to emphasize the source emitting from the selected direction
- farther the audio sources are from this characteristic, more attenuated they are

(1) Delay and sum:

$$y(k) = (x_1(k) + x_2(k-z) + x_3(k-2z) + \ldots + x_8(k-7z))/8$$

- *y(k)* - output signal
- *x* variables denote the signals received by each microphone
- *z* - a positive delay

(2) DOA – Direction of Arrival:

$$\alpha = \arcsin(\frac{z \cdot c}{d})$$

α - value of the preferred DoA

d - distance between two consecutive microphones

c - speed of the sound in the propagation environment

z - value of *z* equals the interval between the moments of arrival of the sound coming from the selected DoA to two consecutive microphones

- For preserving the signal arrived from the selected direction in (2):

  - the 8 sequences of the signal coming from the direction expressed in (2) are properly delayed, with multiples of *z*, as in (1) and the resulted sequences become identical.

- For attenuating the signal arriving from other directions:

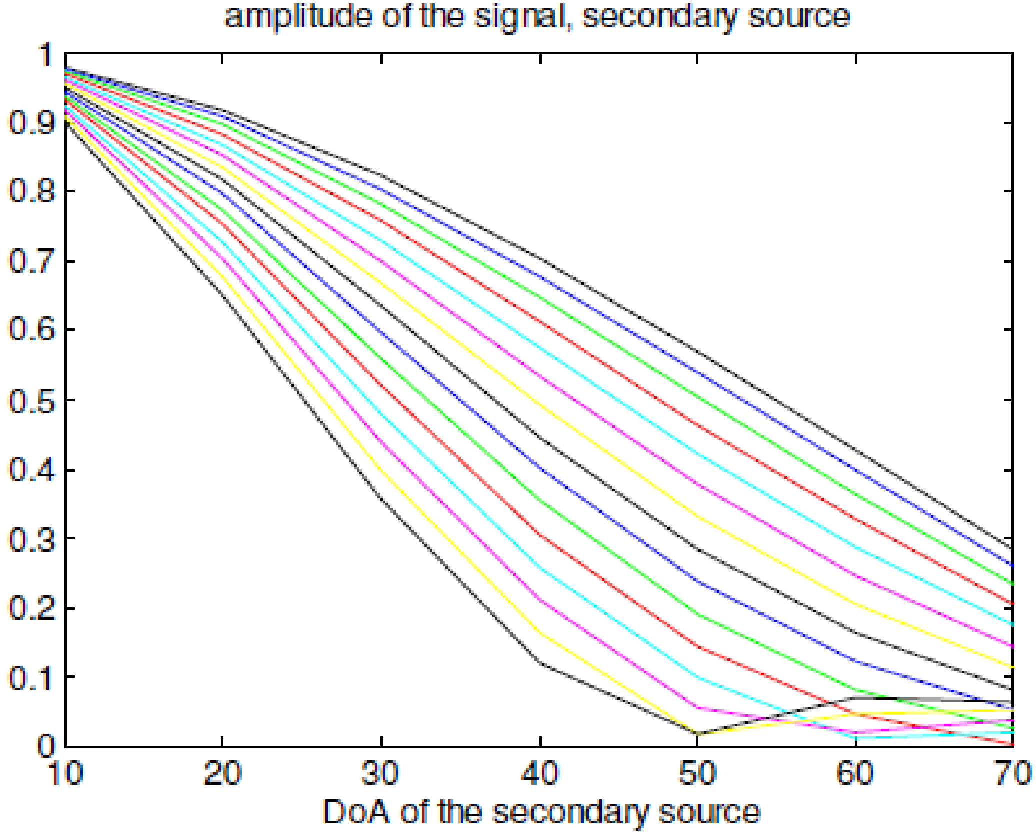
  - for the sources emitting from other directions than (2),the intervals between the moments of arrival to the microphones do not equal the delays used in (1).

  - As a result, the sound coming from other directions will be attenuated, with respect to the sound coming from the preferred DoA

  - attenuation effect depends on the preferred DoA, the DoA of each secondary source, the frequency of the component, the distance between the microphones (*d*) and the number of microphones.

Result of attenuation of the secondary source

(the selected direction is considered along the normal, while the secondary sound comes from a different angle)



PROCESSING THE SIGNAL

1. 8 microphone signals are sampled, AD converted and stored, for a specified time
2. The signals are pass-band filtered
3. The signals are delayed, then added as in (1). The value of z depends on the selected DoA and is computed according to (2).
4. The sequence resulting from (1) is transmitted to other systems, in order to be DA converted or recorded;
5. The processing sequence is resumed, using the next sequence of sampled signals.

EXPERIMENTAL RESULTS

- The source emitting from the preferred DoA was a human voice, while the secondary source was a periodic signal (wide band audio noise)
- The chosen distance between the microphones was 30mm
- difference between the DoAs of the two sources was set to 60 degrees
- the signal of the secondary source was attenuated by 12dB or more, with respect to the signal coming from the selected source.

NO QUESTIONS??????

THANK YOU!!!!