Speech Recognition and Voice Separation for the Internet of Things

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Discussion Outline

- **Motivations and contributions**
- **Background**
- Proposed voice-enabled IoT prototype
- Reconstruction lowpass filter for a voice-enabled IoT prototype
- Results
- Summary and conclusion

Motivation


- Ways of communicating with IoT devices
  - *Graphical User Interface (GUI)*
  - *Speech Interfaces*
- **Limitations** of the current smart home IoT devices (e.g. a smart speaker)
  1. Devices are not customizable: static functionality (voice commands and accuracy)
  2. Smart home speakers cannot handle complex scenarios such as:
     1. They fail processing combined commands separated by "**and**".
     2. They fail processing two **concurrent** commands


Contributions

- **Contributions** of this paper are two folds:
  1. Prototype: A customizable voice-enabled IoT system

  2. Model and Implementation: A model for handling two concurrent voice commands to a voice-enabled IoT device.
     - For example, the case a person says, “Dim the lights.” and at the same time the other person says, “Turn on the TV.”

Background

- Smart home speakers
  - Voice-enabled device widely use *speech processing* and *natural language processing* to create a
    - Recording is done by the device
    - Processing is done in the Cloud
- Blind Source Separation (BSS)
  - The Cocktail party effect
  - The problem of processing multiple concurrent voice commands by a voice-enabled IoT device
- BSS solution: Independent component Analysis
- Low-pass filters in signal processing (we use the Butterworth filter)

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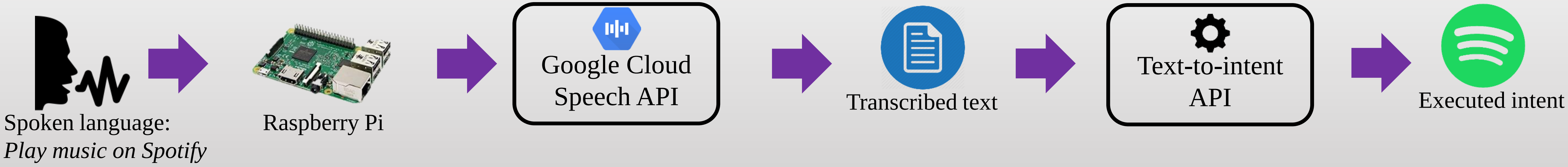
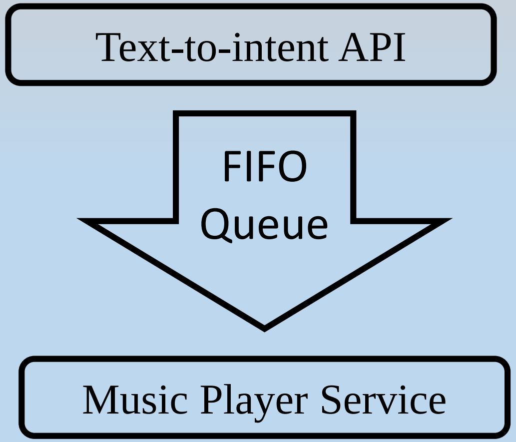
Proposed voice-enabled IoT Prototype



- The proposed model consists of the following components:
  1. The Raspberry Pi records voice and sends it to the Google Cloud speech-to-text API
  2. The Google Cloud speech-to-text API transcribes the voice into text
  3. The text-to-intent API receives the text and converts it to an intent and target device.

Proposed voice-enabled IoT prototype – Text-to-intent API

- Text-to-intent API receives the transcribed text from the Google Cloud speech-to-text API and extracts the followings using a simple language model:
    1. The *intent* of the voice message
    2. The target *device* that the command is intended to be executed on.
- The intents that are currently supported by our proposed prototype are
    - *Play music*
    - *Pause music*
    - *Resume music*
    - *Stop* music
- Device
    - *An open-source command-line music player*



Proposed voice-enabled IoT Prototype – Hardware

- Inexpensive prototype! **$68.42**
- The main hardware components are:
  - Raspberry Pi 3 Model B Motherboard, $35.80
    - Quad core Cortex A53 @ 1.2GHz
    - 1GB SDRAM
    - Wireless 802.11
    - Bluetooth 4.0
  - Kinobo USB 2.0 Mini Microphone, $4.65
  - Samsung 64GB Micro SD Card, $19.99
  - Raspberry Pi Case, $7.98
  - Other hardware: keyboard, cables, etc.
- Sofware: Raspbien, Python, Cloud API, …

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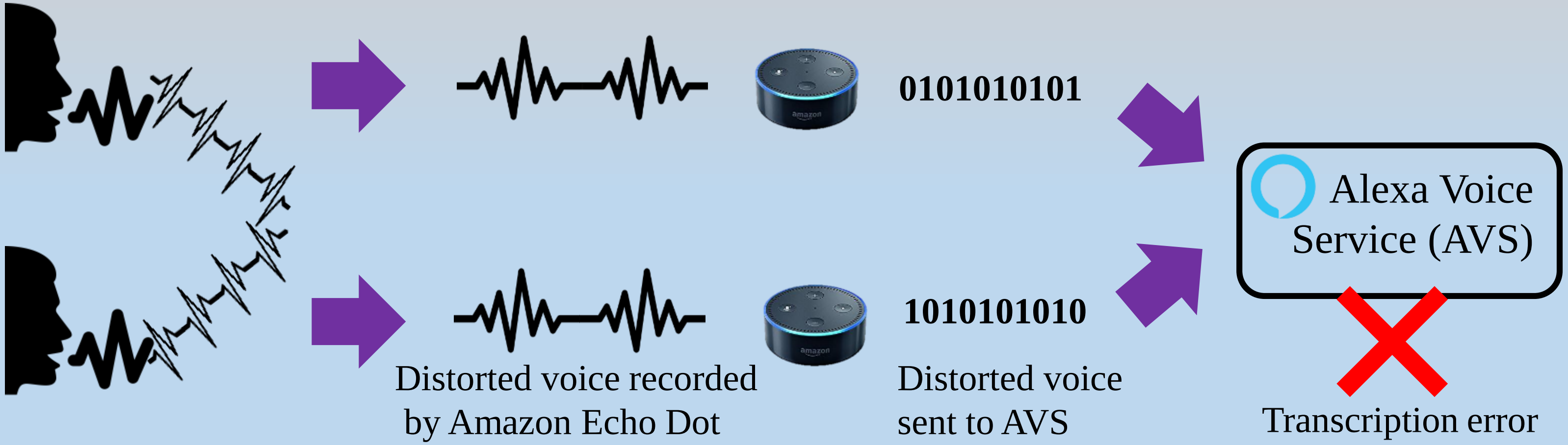
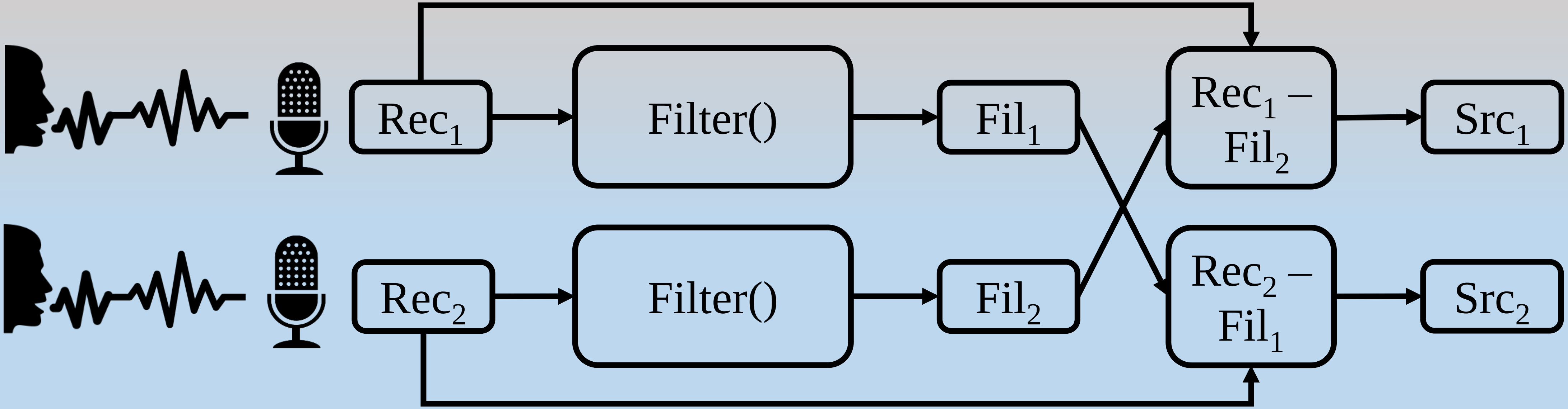
Reconstruction Low-pass Filter for a Voice-enabled IoT Prototype

- **Problem**: Two Echo Dots are placed at the proximity of each other and two persons simultaneously talk with their proximate Dot, the voice recorded by each Echo Dot is distorted by a low frequency voice of the other party.
- **Goal**: Process both recordings recorded by the Echo Dots and then extract and execute both issued commands.



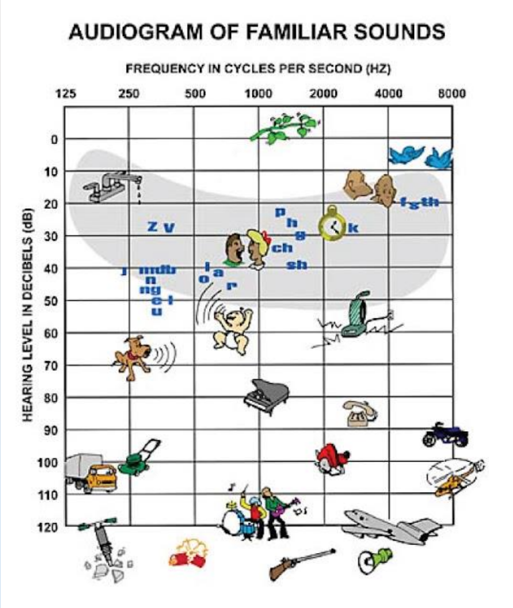
Proposed Reconstruction Lowpass Filter (RLF)

- The Butterworth filter is used to build the proposed Reconstruction Lowpass Filter (RLF)



Proposed Reconstruction Low-pass Filter (RLF)

- Consider the recorded voice from each microphone rec_i is a mixture of source signals src_i, noise signals $noise_i$, where $i \in \{0, 1\}$ and filtered voice fil_j is an approximation of the noise:



  - $rec_i = src_i + noise_{(i+1 \bmod 2)}$
  - $src_i = rec_i - noise_{(i+1 \bmod 2)}$
  - $src_i = rec_i - fil_j \qquad i \neq j$

- In this work we used a 6th order Butterworth filter with the cut-off frequency of 500 Hz.

Dataset for Blind source Separation

- Two Persons are participated in the study
  - Voices are stored as *wav* audio format
  - Available online: https://github.com/hmofrad/viota
- Different proximities to the microphones (Person$_i$, microphone$_i$)
- Common smart speaker commands are used.

Dataset	Number of sentences	Microphone proximity
Dataset 1 (near)	30	Near
Dataset 2 (far)	44	Far

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Results

- Performance metric we use is Word Error Rate, WER = (S + D + I)/N
  - #**S**ubstitutions
  - #**D**eletions
  - #**I**nsertions
  - #**N**umOfWords
- WER is widely used in speech processing and NLP
- Algorithms are:
  - Baseline model which uses the raw recording files
  - Reconstruction Independent Component Analysis (RICA)
  - The proposed Reconstruction Lowpass Filter (RLF)

Results

- RICA performs the worst because it overfits the input recordings.
- The proposed RLF has overall improvement of 2-3% compared to the Baseline model
- Our results are always better for both datasets.

Dataset	Microphone	Baseline	RICA	RLF
Dataset 1 (near)	Mic_1	0.96 ± 0.11	0.91 ± 0.22	**0.99 ± 0.03**
	Mic_2	0.95 ± 0.13	0.35 ± 0.37	**0.96 ± 0.12**
	$(Mic_1+Mic_2)/2$	0.95 ± 0.12	0.63 ± 0.29	**0.97 ± 0.08**
Dataset 2 (far)	Mic_1	0.96 ± 0.10	0.95 ± 0.13	**0.98 ± 0.04**
	Mic_2	0.44 ± 0.39	0.18 ± 0.39	**0.47 ± 0.40**
	$(Mic_1+Mic_2)/2$	0.70 ± 0.24	0.56 ± 0.26	**0.73 ± 0.22**

Discussion

- The 2-3% improvement may not be a groundbreaking improvement at the first glance but
  - Our results are better than both Baseline and RICA models
  - At scale it significantly contributes to the Cloud throughput, availability, and utilization by reducing the number of commands send by users.
- Avoid potential Cloud upgrades and expansion
  - Reduce number of retries due to accuracy
  - Keep the number of requests low
  - Requests are now less noisy ➔ will result in intended action

Summary and Conclusion

- A customizable voice-enabled IoT prototype is proposed which can be used as a preprocessing step to the speech-to-text API
  - Raspberry Pi
  - Google Cloud speech-to-text API
  - Text-to-intent API
- Devising a method for voice separation in IoT environment.
  - Reconstruction Lowpass Filter (RLF)
- Takeaways
  - A good preprocessing can eliminate potential retries on the Cloud
  - This is achievable with a inexpensive hardware.