A Wearable Platform for Research in Augmented Hearing

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Objectives

Develop wearable, real-time, open-source speech processing platform (OSP) to:

- Facilitate audiological studies—diagnosis, treatment, and assessment—in the lab and in the field
- Allow for development of new audio/speech signal processing algorithms
- Give hearing aid (HA) users far more control over their devices
- Lower the barrier of entry to assisted hearing research
- Enable new discoveries in hearing healthcare

Introduction

We previously [1] presented OSP on a lab-based system: Macbook, audio interface, breakout box, and custom analog BTE-RICs. The same real-time master hearing aid (RT-MHA) software—with additional functionality—now runs on an embedded system in a wearable form factor. In addition, the wearable system hosts an embedded web server (EWS) that serves web apps to any browser-enabled device, which may control and monitor the state of the RT-MHA system in real time. We present a characterization of several aspects of the performance of the wearable system, and an overview of the EWS and new features in the RT-MHA.

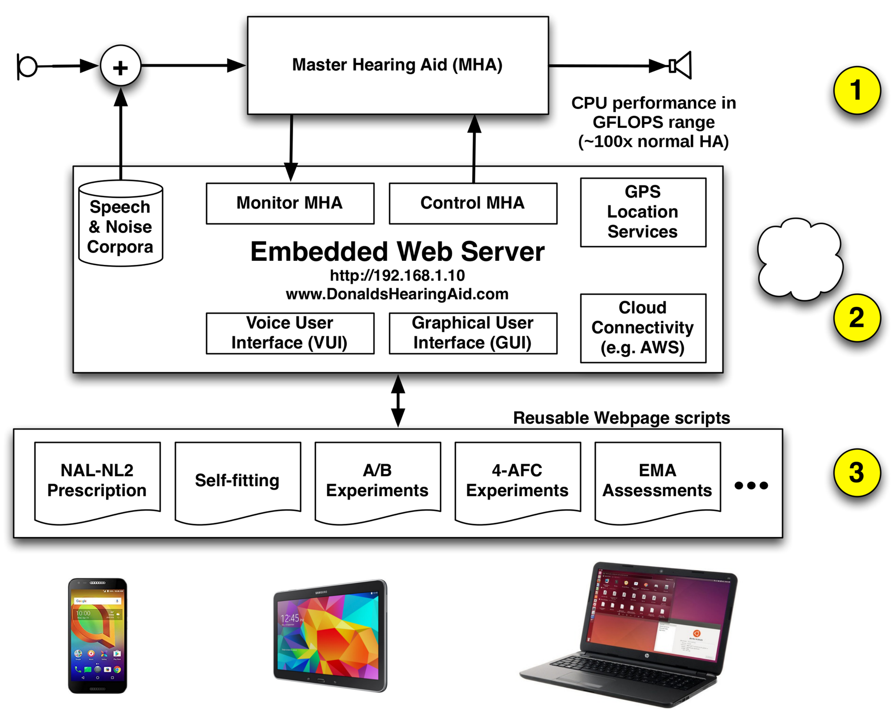

Figure 1: System architecture. (1) = RT-MHA. (2) = EWS. (3) = Web apps.

Wearable Unit

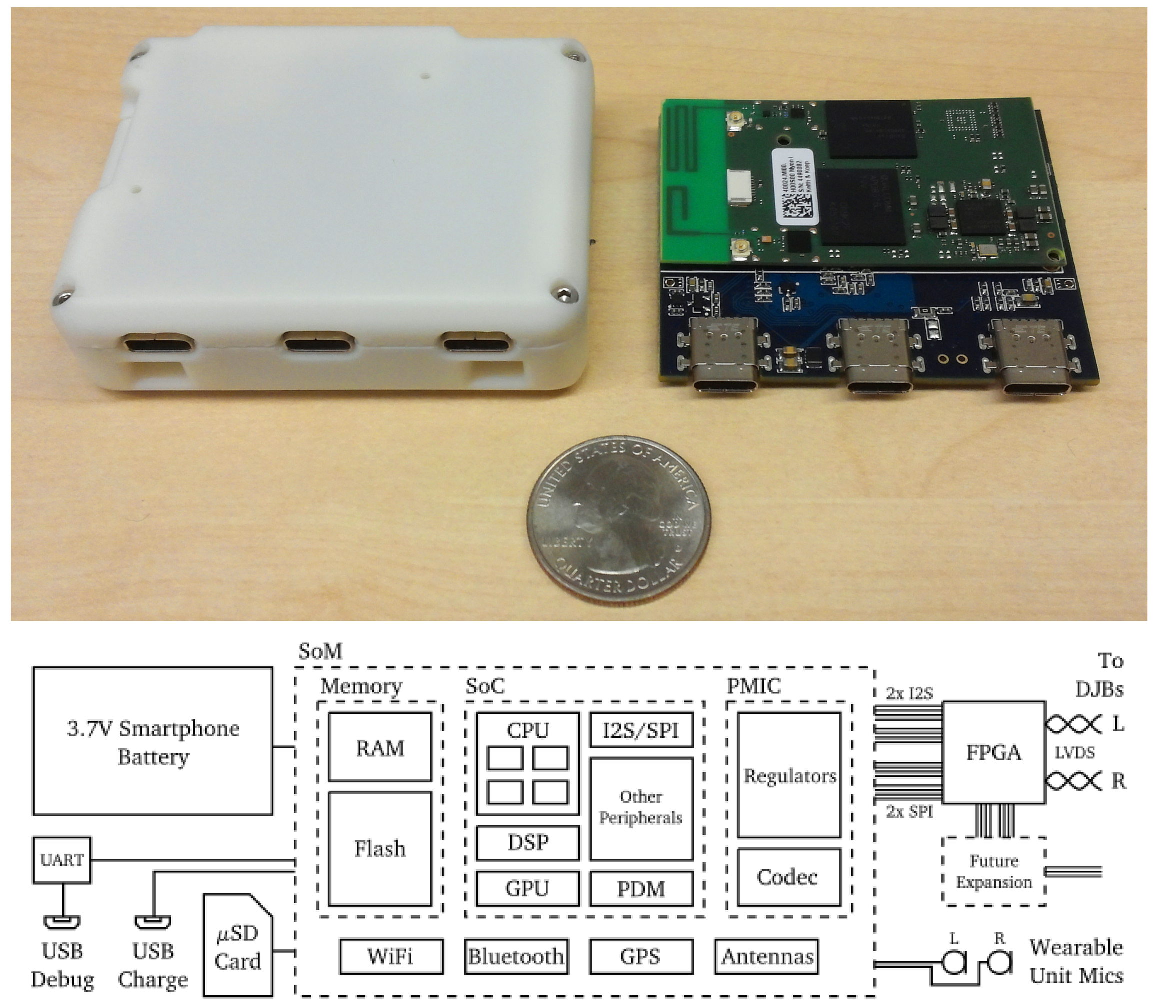

Figure 2: Wearable unit photo and block diagram.

- SoC: Qualcomm Snapdragon™ 410E (APQ8016E)
- Quad-core 64-bit ARM Cortex A53 @ 1.2 GHz; also DSP, GPU available
- Connectivity: WiFi, Bluetooth, GPS
- OS: off-the-shelf ARM64 Debian Linux
- FPGA for interface to BTE-RICs and an analog connection for future expansion
- Replaceable smartphone-type battery

Operation	Times (μs)		
	Minimum	Average	Maximum
Overall	626	694	4635
Downsampling	32	32	91
Filtering	282	282	822
Noise management	12	12	408
Peak detection	24	30	384
WDRC	18	18	384
AFC	180	190	3661
Upsampling	32	33	91
Total channel:	580	597	5841

Table 1: Performance statistics for real-time processing of 1ms audio buffers, over $\approx$ 3 minute sample. `/sbin/init` is bound to core 0 and the RT-MHA process is bound to cores 1 - 3, with highest process priority.

State	Avg. Current (mA)	Battery life (h)
Idle (no WiFi)	0.243	8.23
Idle	0.284	7.04
RT-MHA (HA process disabled)	0.317	6.31
RT-MHA (all processing enabled)	0.440	4.55

Table 2: Current draw (at nominal 3.7VDC) and battery life (assuming 2000mAh Li-ion battery) for common system use cases.

Digital BTE-RICs

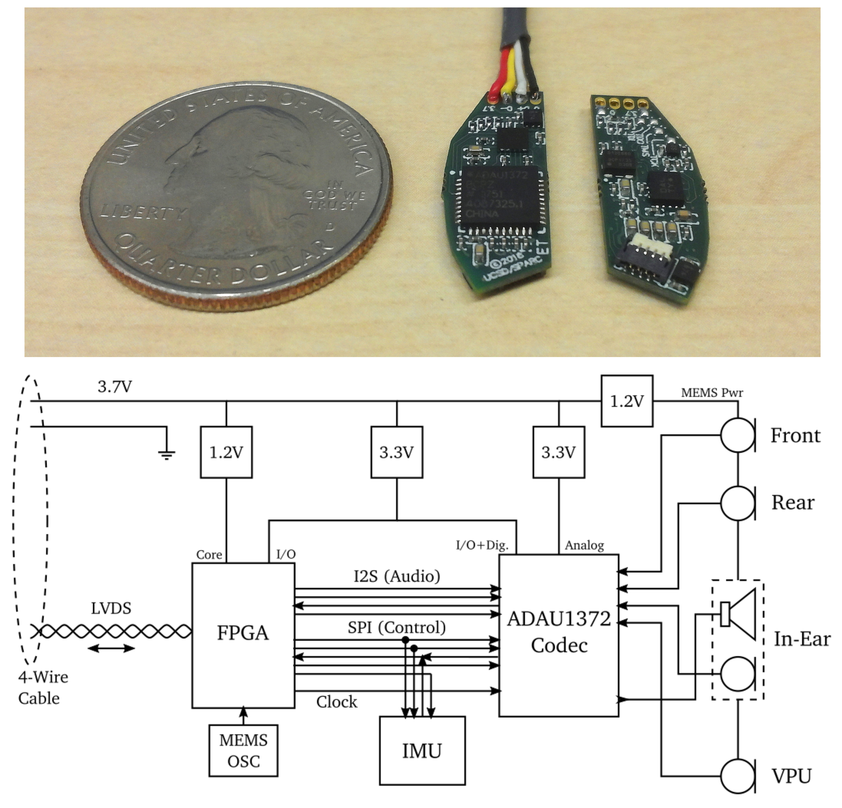

Figure 3: Digital BTE-RIC main PCB photo and block diagram.

- Up to four mics per ear: front, rear, in-ear, and a bone conduction microphone
- Analog Devices ADAU1372 codec; $F_s = 96$kHz; 24bits/sample
- IMU (six axis accelerometer + gyroscope)
- FPGA for data interface
- Isolated analog & digital power supplies
- Round-trip (analog to analog) latency: $\approx$2.3 ms hardware latency + $\approx$3.3 ms for RT-MHA processing = overall **$\approx$5.6 ms**

Metric	Units	Commercial HAs				OSP Lab System		OSP Wearable	
		A	B	C	D	X	Y	X	Y
Average Gain	dB	40	40	25	35	40	40	35	38
Max OSPL90	dB SPL	107	112	110	111	121	130	119	129
Average OSPL90	dB SPL	106	109	108	106	112	126	111	125
Average Gain @ 50 dB	dB	37	39	25	35	35	41	34	38
Low Cutoff	kHz	0.2	0.2	0.2	0.2	0.2	0.2	0.2	0.2
High Cutoff	kHz	5	6	5	6.73	8	6.3	8	6.73
Equivalent Input Noise	dB SPL	27	26	30	27	29	28	28	27
Distortion @ 500 Hz	% THD	1	1	0	0	2	1	5	1
Distortion @ 800 Hz	% THD	1	1	0	0	3	2	5	1
Distortion @ 1600 Hz	% THD	0	0	0	0	1	1	2	0

X: with high-bandwidth Sonion receiver
Y: with high-power Sonion receiver

Table 3: ANSI 3.22 test results for OSP system configurations as measured by Audioscan Verifit 2, as compared to results from four commercial HAs.

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Speech Enhancement

A baseline speech enhancement (SE) module has been added to RT-MHA. The module performs (1) peak and valley detection, (2) noise power estimation, and (3) Wiener filtering on the output of each of the six subbands. We provide three options for noise power estimation: (1) power averaging, (2) weight averaging, and (3) minima controlled recursive averaging methods. After obtaining the noise power estimate using any of the above techniques, classic Wiener filtering is performed to suppress the noise contents for enhancing the speech. A control (0...1) adjusts between no SE and aggressive SE.

Embedded Web Server (EWS)

The Embedded Web Server (EWS) hosts a suite of web apps, which in turn can be used to control the RT-MHA. This allows users to be able to manipulate the RT-MHA directly from any browser-enabled device. The apps are implemented in HTML, CSS, JavaScript, and PHP or similar tools. EWS enables researchers to easily adapt apps for unique experiments, and with a much lower barrier to entry than Android or iOS programming. Example web apps include: an intuitive self-fitting app called "Goldilocks" [2], an AB/ABX task, and a researcher page, which can be used for amplification, noise management and feedback management.

References

[1] H. Garudadri *et al.*, "A realtime, open-source speech-processing platform for research in hearing loss compensation," *2017 51st Asilomar Conference on Signals, Systems, and Computers*, no. 51, pp. 1900–1904, 2017.

[2] C. Mackersie, A. Boothroyd, and H. Garudadri, "Research on hearing-aid self-adjustment by adults," *The Journal of the Acoustical Society of America*, vol. 143, no. 3, pp. 1743–1743, 2018.

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