# OPEN SPEECH PLATFORM TOOLS:
# A REALTIME MASTER HEARTING AID SOFTWARE

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**Abstract**

In this work, we present status of the **Open Speech Platform** (OSP) at UC San Diego under a NIH R01 grant. The overarching goal of our research is to develop an open source, wearable, speech processing platform that can be configured at compile time and at run time for audiologists and hearing aid researchers to investigate new hearing aid (HA) algorithms in lab and field studies for improving hearing healthcare.

## 1 Open Speech Platform

The system level block diagram of OSP is shown in Figure 1.

1. Ear level assemblies, pre-amps, ADC, DAC, power amp
2. A dedicated digital signal processor (DSP) for all realtime ($< 10$ ms) HA processing
3. A general purpose CPU for low latency ($50-500$ ms) processes and miscellaneous functions
4. Remote connectivity for user/researcher devices to configure run-time parameters

Figure 2. shows an example set up for using "Release 2017a" of the software described in the following sections.

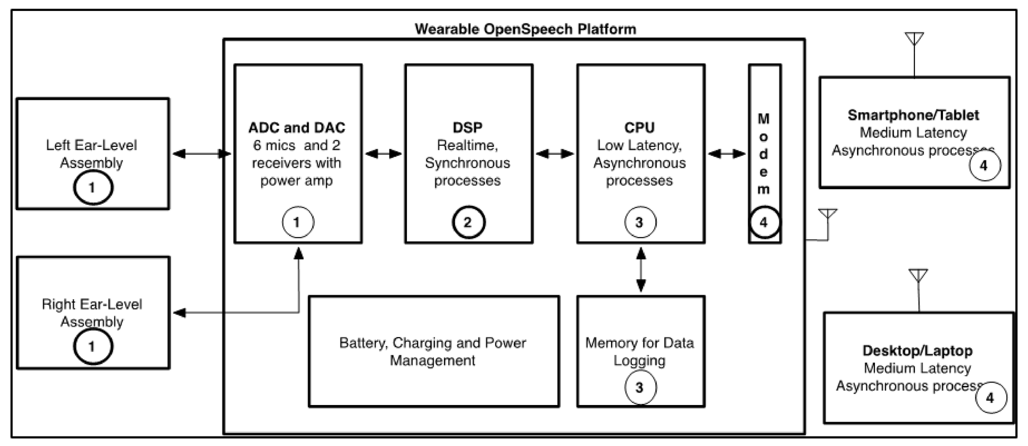


**Figure 1:** Open Speech Platform Block Diagram

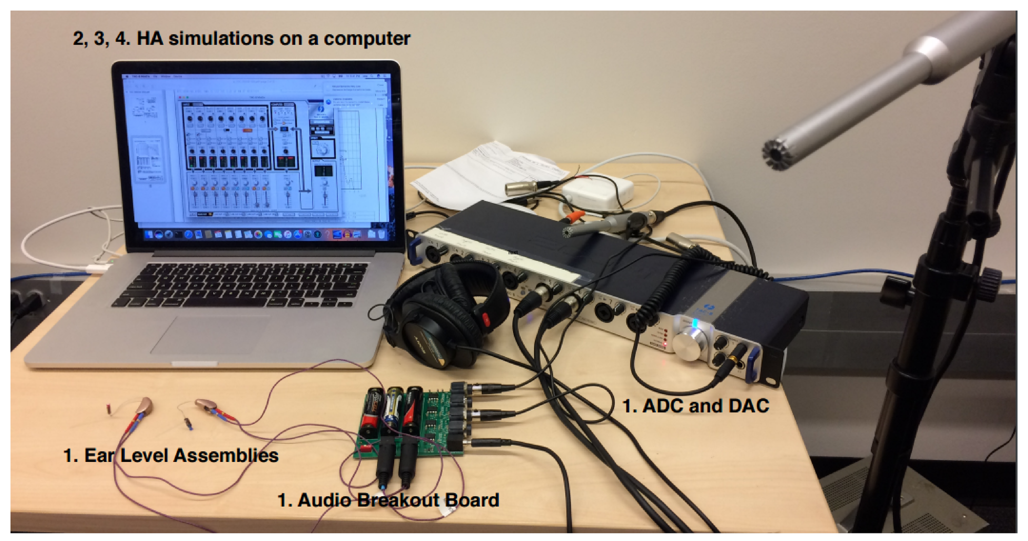


**Figure 2:** Lab Set Up for "Release "2017a"

## 2 The Real-time Master Hearing Aid (RT-MHA)

The RT-MHA block diagram with signal flows is depicted in Figure 3. This architecture with different sampling rates has the benefit of minimizing hardware latency and improving spatial resolution of beam forming with multiple microphones. The basic HA functions include (i) Sub-band decomposition, (ii) Wide dynamic range compression and (iii) Adaptive feedback cancellation. These algorithms are provided in source code and compiled libraries.

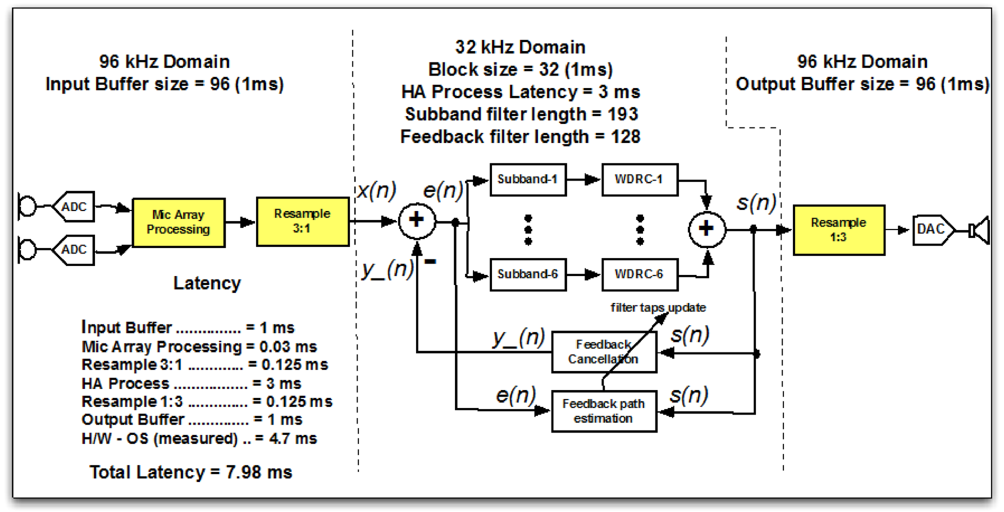


**Figure 3:** RT-MHA Software Block Diagram with Signal Flows

### 2.1 Sub-band Decomposition

Sub-band decomposition is implemented as a bank of 6 FIR filters whose frequency responses are shown in Figure 4. The filters are designed in MATLAB and saved as .flt files for inclusion with RT-MHA. MATLAB scripts are provided for changing number of bands and filter lengths.

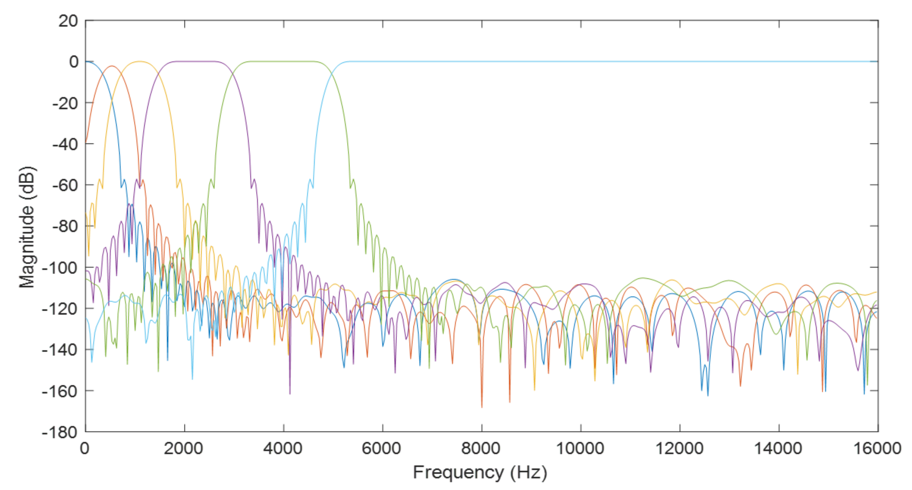


**Figure 4:** Frequency Response of the 6-Channel Filter Bank

### 2.2 Wide Dynamic Range Compression (WDRC)

The WDRC algorithm is based on envelope detection and non-linear amplification, developed by Jim Kates. The compression ratios, attack times, release times and knee points can be specified at compile time and changed at run time using the user device.

### 2.3 Adaptive Feedback Cancellation (AFC)

The AFC module is based on filtered-X least mean squares (FXLMS) and provided in source code. In future, we anticipate additional AFC modules as libraries.

## 3 User Device

The user device software is implemented on an Android device above TCP/IP layer in a software stack called OSPLayer. Release 2017a enables real-time control of compression profiles in all the sub-bands, including location of the knee-points, attack time and release time in each of the sub-bands. Figure 5 shows the screen shots of examples of the user device software.


**Figure 5:** Left panel shows RT-MHA parameters that can be controlled in real-time. Middle panel shows an example of self-fit software. Right panel shows example of a word recognition test to evaluate HA performance.

## 4 Performance Evaluation

Table 1 shows performance of RT-MHA compared with 3 commercial HAs from Opticon.

| | Oticon Opn | | Oticon Alta2 | | Oticon Sensei | | UCSD RT-MHA | | |
|---|---|---|---|---|---|---|---|---|---|
| | Specs | Measured | Specs | Measured | Specs | Measured | Specs | Measured | Units |
| SPL90 peak | 127 | 115 | 118 | 116 | 117 | 124 | na | 120 | dB SPL |
| SPL90 avg | 121 | 112 | 114 | 112 | 113 | 118 | 116 | 118 | dB SPL |
| Gain Full | 55 | 47 | 47 | 43 | 44 | 51 | 42 | 41 | dB |
| Gain Ref | 45 | 36 | 38 | 34 | 36 | 41 | na | 41 | dB |
| Eq Input Noise | 25* | 18 | 18* | 25 | 18* | 26 | <30 | 29.1 | dB SPL |
| Distortion: 500 Hz | <2 | 1 | <2 | 0 | <2 | 1 | <5 | 2 | % THD |
| Distortion: 800 Hz | <3 | 2 | <2 | 1 | <2 | 0 | <5 | 2 | % THD |
| Distortion: 1600Hz | <2 | 1 | <2 | 0 | <2 | 0 | <5 | 1 | % THD |
| End-2-end Delay | na | 8.0 | na | 5.7 | na | 6.0 | <10 | 7.7 | ms |

**Table 1:** Performance of RT-MHA, compared with commercial HAs

Table 2 compares two AFC algorithms implemented in RT-MHA in terms of the maximums stable gain (MSG), added stable gain (ASG) and computational complexity.

| Input File | MSG w/o AFC | ASG (FXLMS) | ASG (Advanced) | Runtime (FXLMS) | Runtime (Advanced) | Additional Comp. |
|---|---|---|---|---|---|---|
| TIMIT_male_sa1 | 18.79 | 10.75 | 17.33 | 1.0000 | 1.0427 | 4.27% |
| TIMIT_male_si743 | 18.49 | 9.81 | 15.83 | 1.0000 | 1.0369 | 3.69% |
| TIMIT_male_si1306 | 18.79 | 9.84 | 17.06 | 1.0000 | 1.0344 | 3.44% |
| TIMIT_female_sa2 | 18.69 | 8.16 | 11.94 | 1.0000 | 1.0471 | 4.71% |
| TIMIT_female_sx68 | 18.89 | 9.07 | 14.17 | 1.0000 | 1.0336 | 3.36% |
| TIMIT_female_sx397 | 18.79 | 8.06 | 14.46 | 1.0000 | 1.0565 | 5.65% |
| Brahms | 18.89 | 11.48 | 16.38 | 1.0000 | 1.0299 | 2.99% |
| Trilogy | 18.89 | 11.21 | 19.17 | 1.0000 | 1.0296 | 2.96% |
| salvation | 18.99 | 11.12 | 16.86 | 1.0000 | 1.0217 | 2.17% |
| Railroad Xing | 18.89 | 6.69 | 10.65 | 1.0000 | 1.0396 | 3.96% |
| Average | 18.81 | 9.62 | 15.39 | 1.0000 | 1.0372 | 3.72% |

**Table 2:** Comparison of the Advanced AFC Algorithm with the FXLMS

## 5 Open Speech Platform Release 2017a

Figure 6 depicts salient software tools included in Release 2017a.

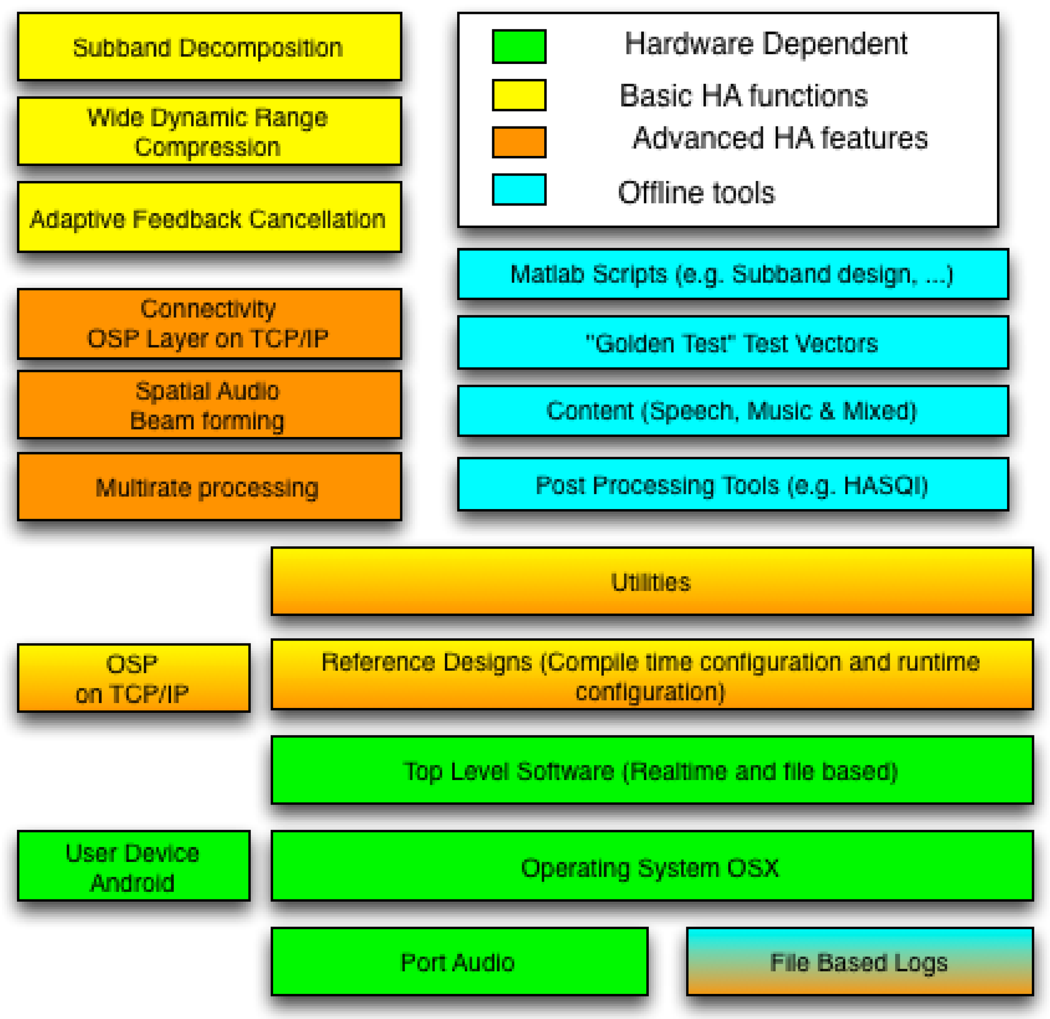


**Figure 6:** Software Stack Diagram of RT-MHA

## Takeaway Message

Calling Audiologists and Hearing Aid Scientists to provide feedback and feature requests at `hgarudadri@ucsd.edu`

## Acknowledgements

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