Tools for assessing efficacy of hearing loss compensation

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Hello, everyone. Before I start I just want to thank you for this opportunity for giving us some time to discuss the tools we've been working on.
My name is Sergio Luna…*
[Louis introduction]
We are excited to show you our open speech platform, but before we get started I just wanted to let you know that the person who was originally going to give this presentation was unable to make it due to unforeseen circumstances, and so we're filling in for them. Personally I've been working on the embedded web server API, and Louis has been primarily involved with the embedded hardware, but there are portions of the platform which neither of us have really worked on and which we're not as knowledgeable about. So if there are any questions at the end, of course we will try to answer them to the very best of our ability, but if for whatever reason we don't know the answer we would be more than happy to take your information down or point you to the appropriate people.
Anyways, with all that said, lets begin.

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Open Speech Platform for Hearing Aids Research

- http://openspeechplatform.ucsd.edu
- Supported by National Institute of Health, NIH/NIDCD grant R01DC015436, "A Real-Time, Open, Portable, Extensible Speech Lab"
- See https://github.com/nihospr01/OpenSpeechPlatform-UCSD for latest software and documentation

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We're here to discuss our Open Speech Platform for Hearing Aid Research, also called OSP.
Briefly, OSP is a hardware and software platform for facilitating hearing-related clinical trials in the lab and in the field, and for the development of new algorithms in hearing and speech processing.
Our most recent release, 2018a, is available for download at openspeechplatform.ucsd.edu. 2018b, which contains the content discussed in this presentation, will be released this summer.

The system is currently deployed at these institutions, and we'd like to thank them all for the valuable guidance and feedback we have received.

Vision and Goals

- Provide a research tool with rapid prototyping capabilities for audiological and speech science investigations.

- A platform comprising of hardware, software, and speech processing algorithms to advance HAs, hearables, etc. and real-time signal processing algorithms

Through our platform, our goal has been to provide a research tool with rapid prototyping capabilities for audiological and speech science investigations.

We envision this through a platform comprising of hardware, software, and speech processing algorithms to support research to advance the state of the art in hearing aids and real time signal processing algorithms.

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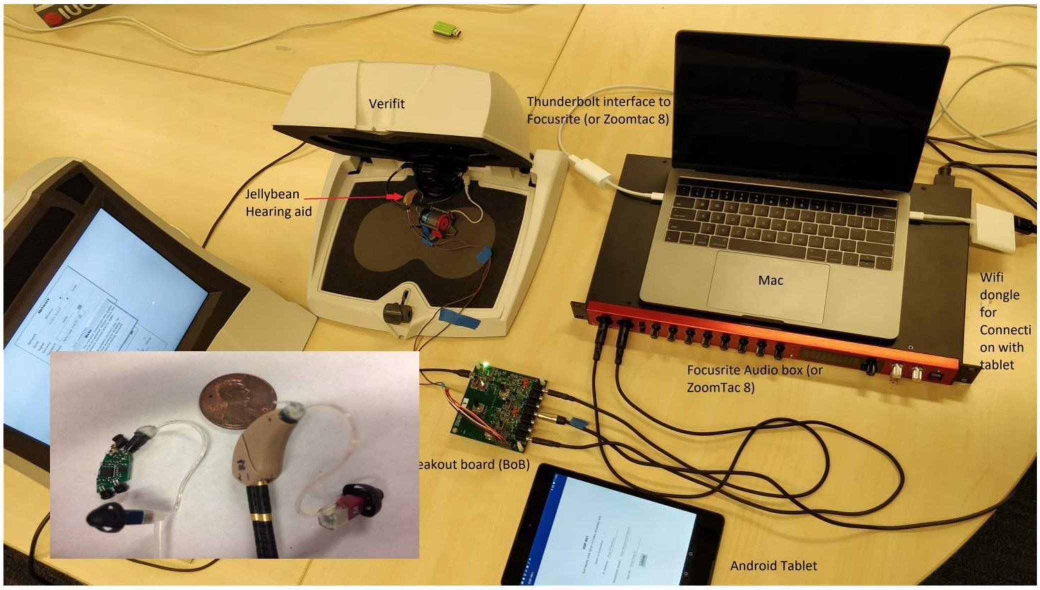

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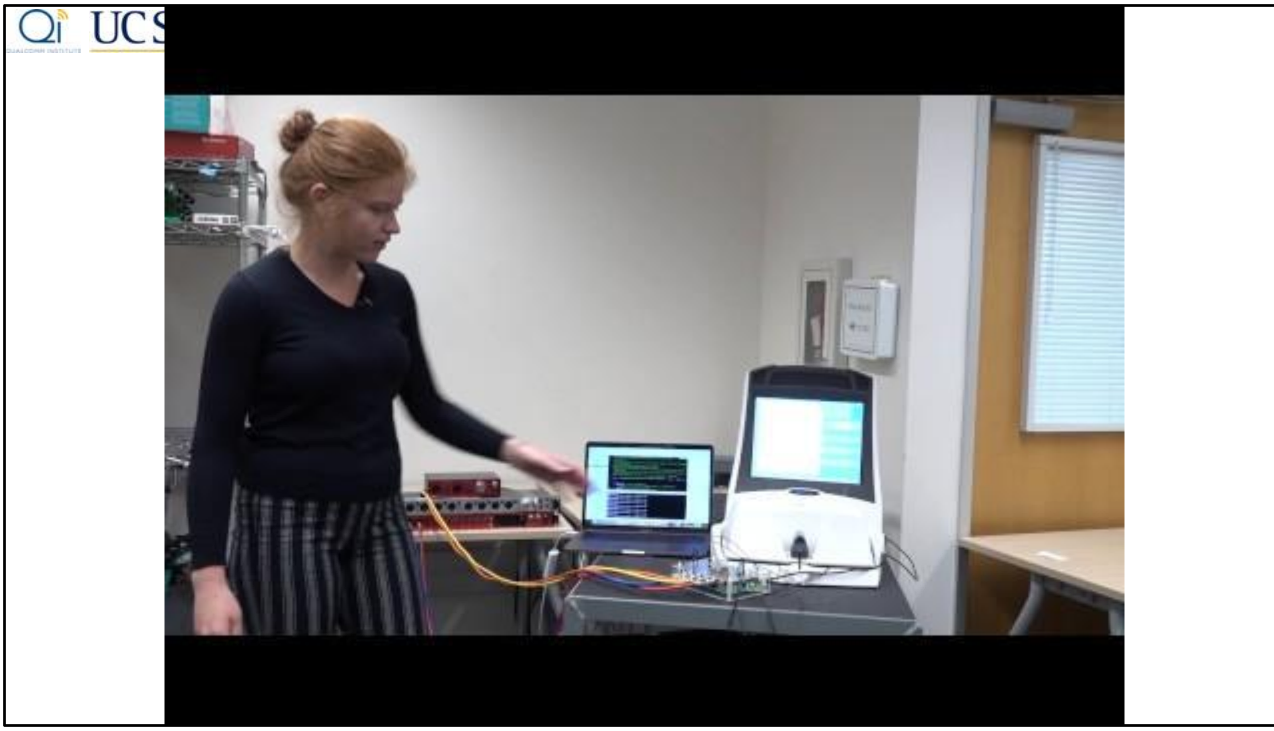
This is an image of the lab system for our open speech platform. It has enabled lab studies, and provided a pathway for implementing a wearable system for field studies.

We have a video explaining the lab setup in more detail:



[Play video and then]:

https://youtu.be/t3JT4G0HgU8

As a reminder, this is an open platform--so if you already have a Mac and an audio interface such as the Zoom Tac 8 or Focusrite Clarett, and you want to use your own mics and receivers or analog ear-level assemblies, you can start using OSP today, for a total cost of zero!

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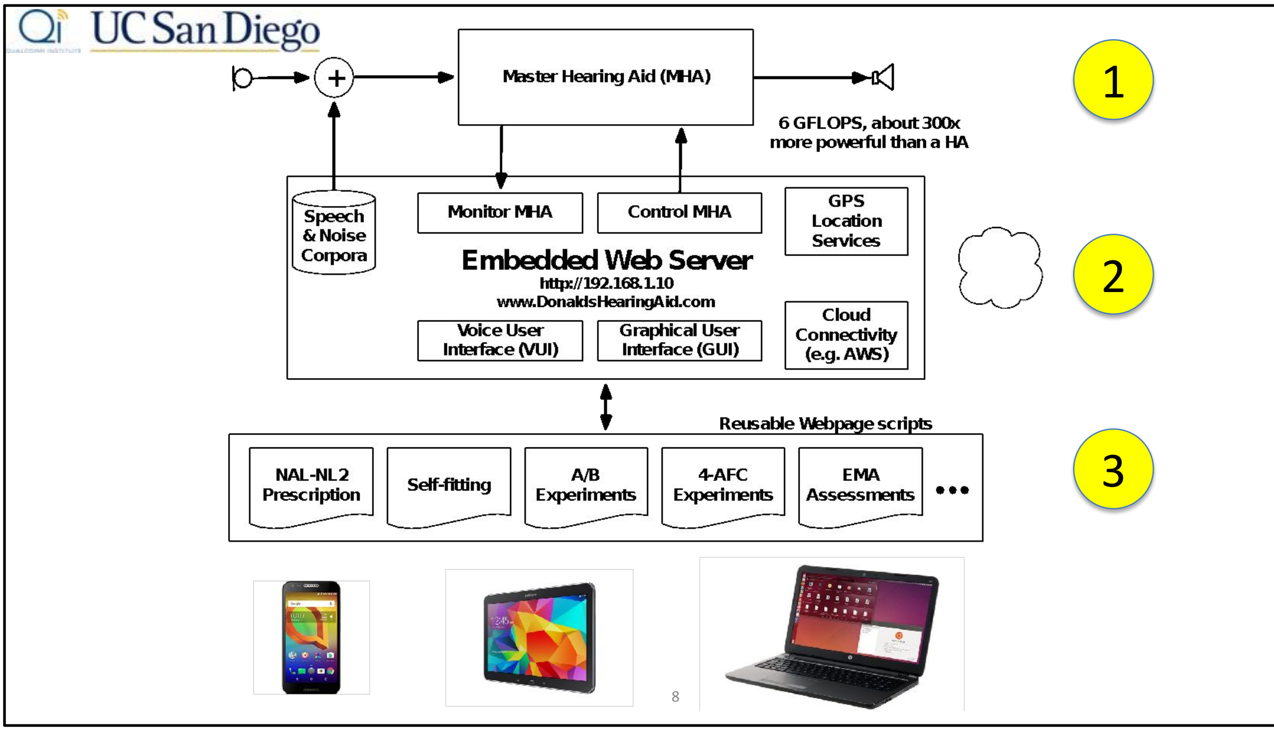
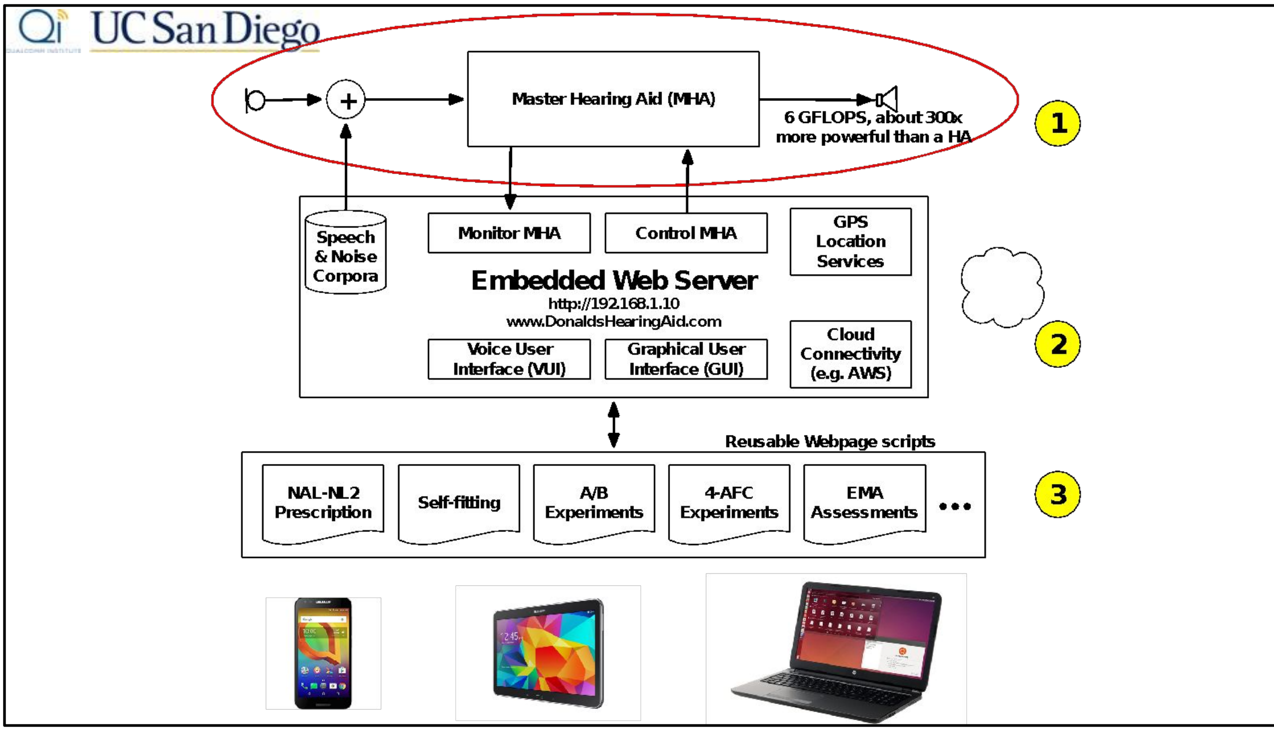
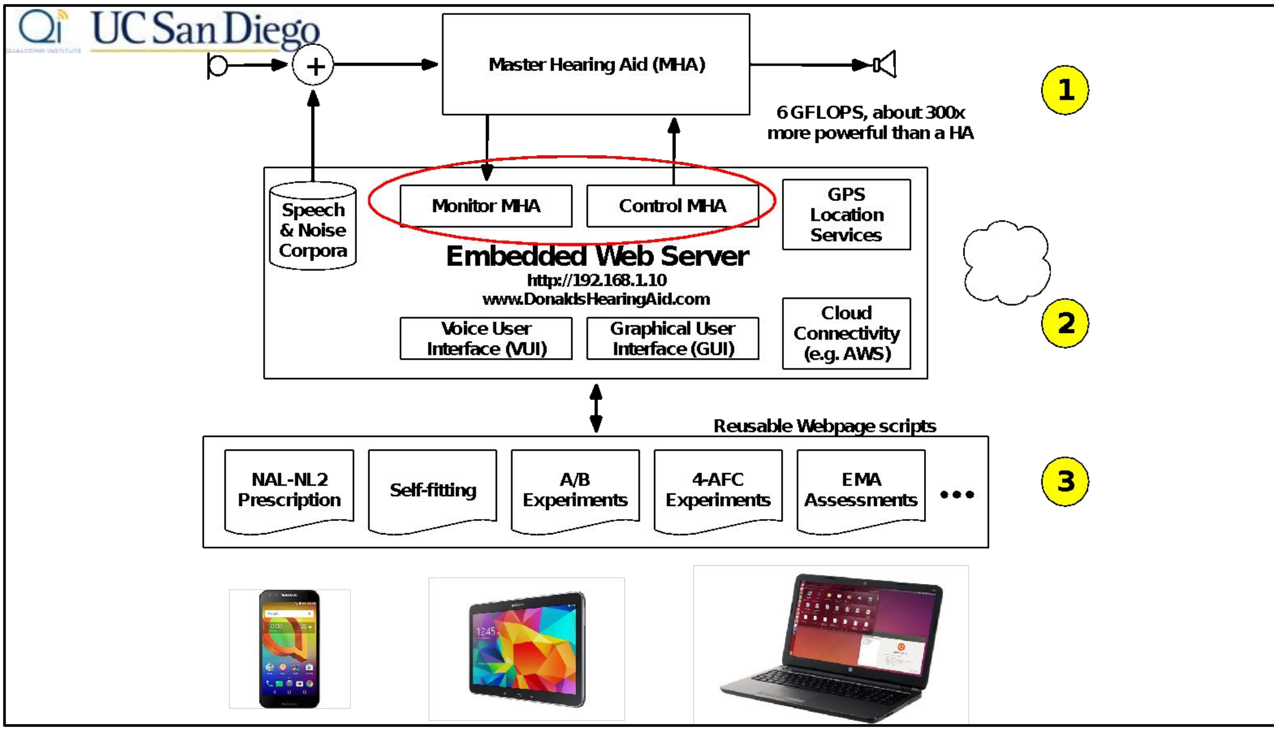
We also have been working hard on an embedded version of the hardware, but we will be launching this
[CLICK]
at IHCON 2018. For today, we're here primarily to discuss the embedded web server, as you will see. But, first, an overview of the platform.



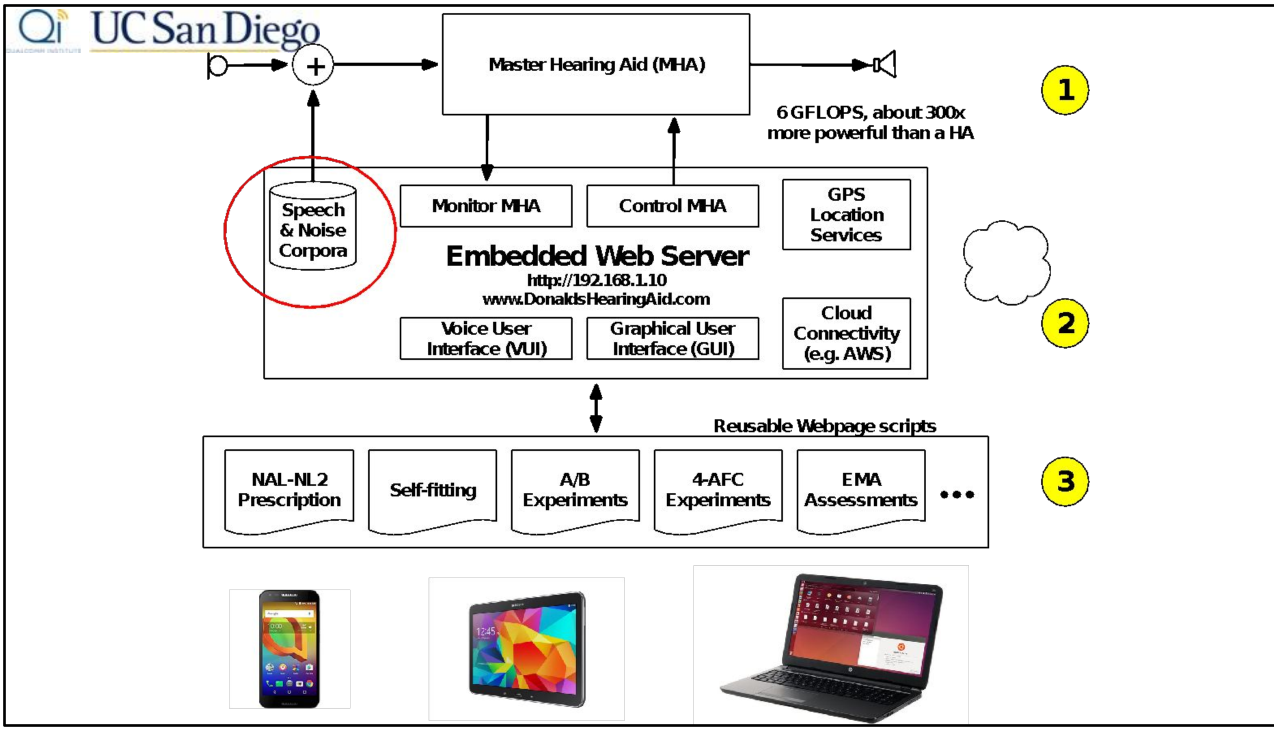
This graphic shows the overall structure of the platform.
Here you'll see we have labeled three regions.
The first region is the real time sound processing subsystem, the second region is the embedded web server subsystem, and the 3rd third region are the web apps.



So again, the first region is the real time sound processing subsystem called the Master Hearing Aid, and we're not going to go into too much detail on this today.
For the purposes of this presentation, all you need to know is that it supports mostly basic with some advanced hearing aid processes including: Amplification, Adaptive Feedback Cancellation, Noise Management and Directional microphones.
The current version of the MHA software is already publicly released and can be freely downloaded from openspeechplatform.ucsd.edu for use with the laptop-based hardware system.
As far as latency, for the current, laptop-based system, the analog-to-analog latency is about 8ms, though for the upcoming embedded version it's about 5 ms.



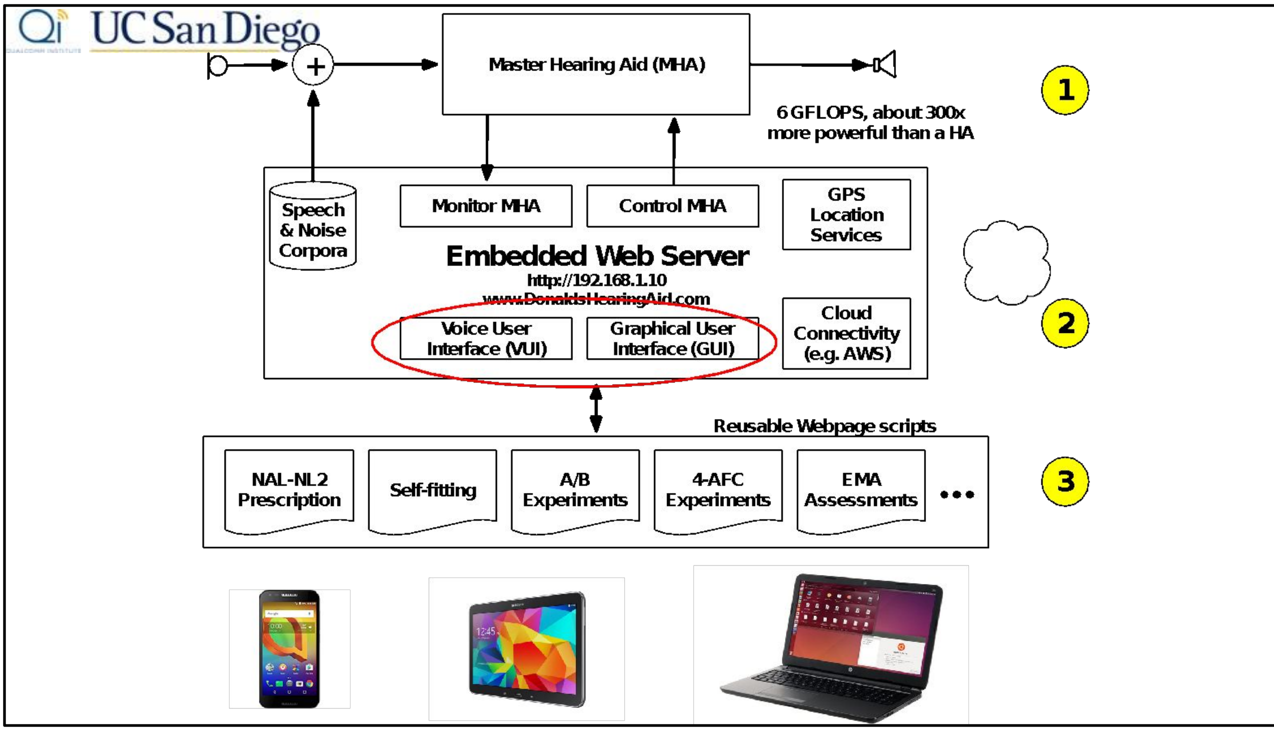
Moving on we can discuss region 2, the embedded web server.
You may be wondering what exactly an embedded web server entails. In short, it's an application you can connect to via a web browser like you would any other website, but supports real time monitoring and control of the Master Hearing Aid.
This means that the device running the HA processing--in other words the laptop, or the embedded unit in the upcoming version--is actually running a web server. You can go to an IP address or a URL from any device with a web browser, and the embedded web server will serve pages to you which allow you to monitor and control the HA processing. There's no external server or web hosting required.
In a few slides, we will discuss some examples of setups where the embedded web server is controlling HA processing parameters. But for now, we will finish the overview of the system.



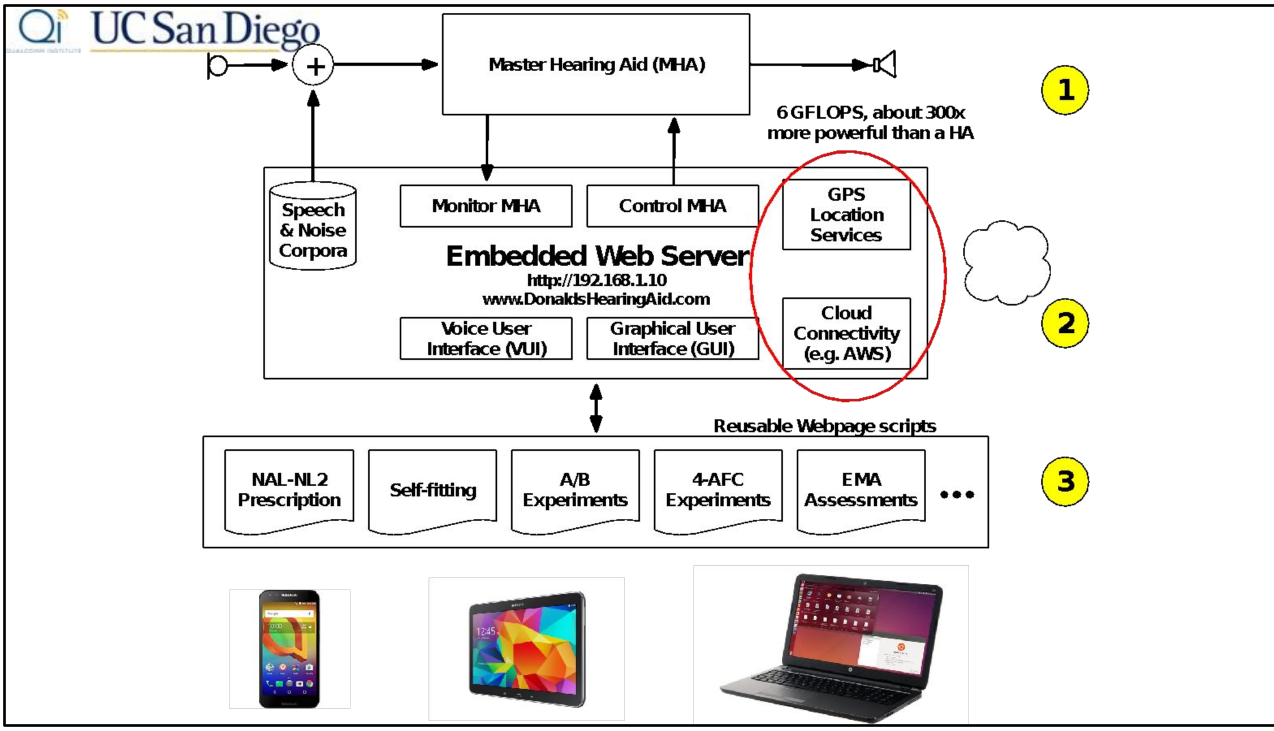
One common requirement of many audiological and speech science investigations is to present stimuli in a repeatable and reproducible (consistent) manner.

The embedded web server will come with a selection of commonly used speech and noise corpora, and adding sound files is made to be very simple.
The system can also log input and output audio for extended periods of time and store it internally (it can be 24 hours or more depending on the platform, compression, and some other factors). Then, the researcher can for instance examine the input audio during a time when the user reported that the hearing aid sounded bad, and even take that same input audio and replay it to the user with different speech enhancement parameters or other settings.

The takeaway here is that the embedded web server isn't just changing individual parameters of the MHA processing, it can actually direct the overall MHA Input/Output.

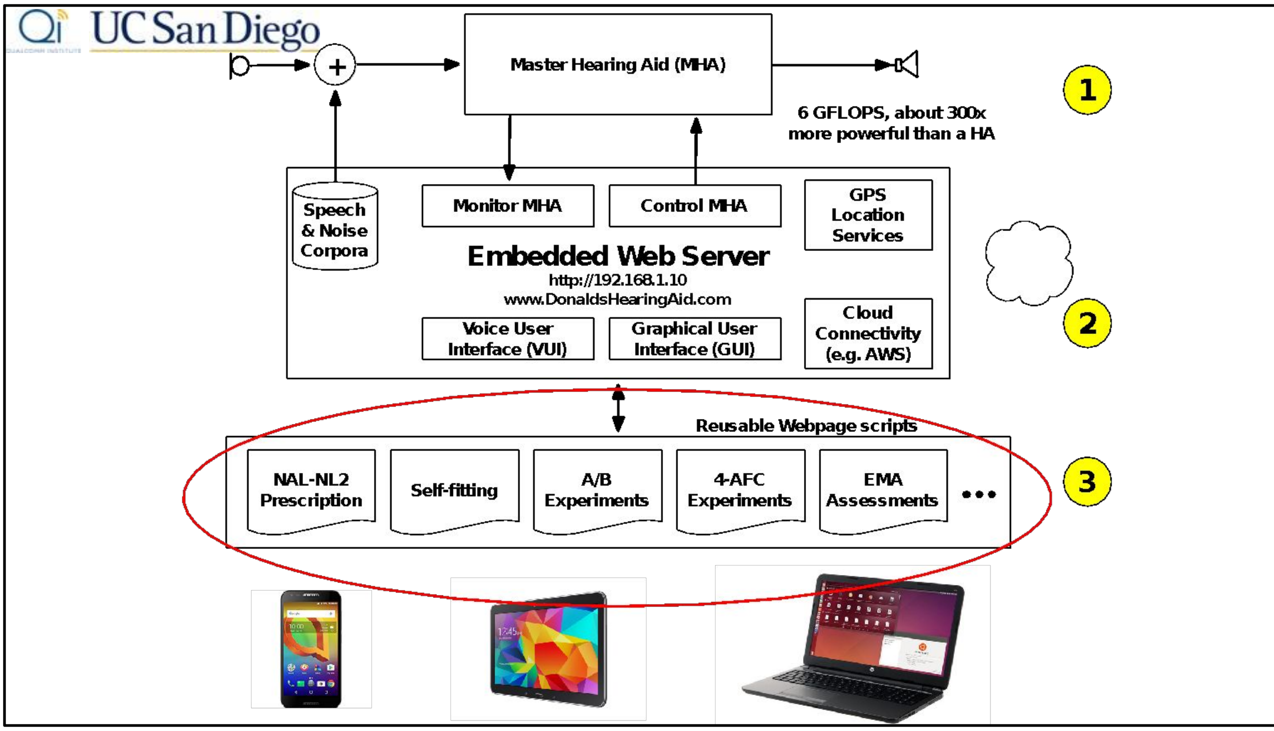


Graphical interfaces within the web pages served by the embedded web server access a straightforward API to obtain and send the hearing aid parameters.
This gives the researcher the option to create a simple web page in plain HTML for rapid prototyping, or a feature-rich web page with JavaScript for more custom interfaces.
The great thing about that is that it's all up to the specific needs of the researcher.
We will show some examples shortly.
Additional APIs for voice user interfaces--as opposed to graphical interfaces--are planned for the near future.



For the embedded version of the platform, since it is based on a chipset from commercial smartphones, we have GPS for location based capabilities. With this we can change the state of the hearing aid based on location, and assess the state of the user using Ecological Momentary Assessments.
However this will be discussed at greater length when we discuss the final embedded platform.

The fact that communication from the user to the hearing aid is done through an API also makes this tool easy to use with cloud-connected services like Amazon Web Services. An example of this could be using AWS's S3 as a service for storing large amounts of recordings for later use.



Lastly, we have come to the part that is most important and perhaps most interesting for this audience. We've asked ourselves: how do all of these parts come together to support typical investigations you run? In the third region of the chart above, our platform includes a set of reusable and editable web page scripts that we think would be useful to researchers.

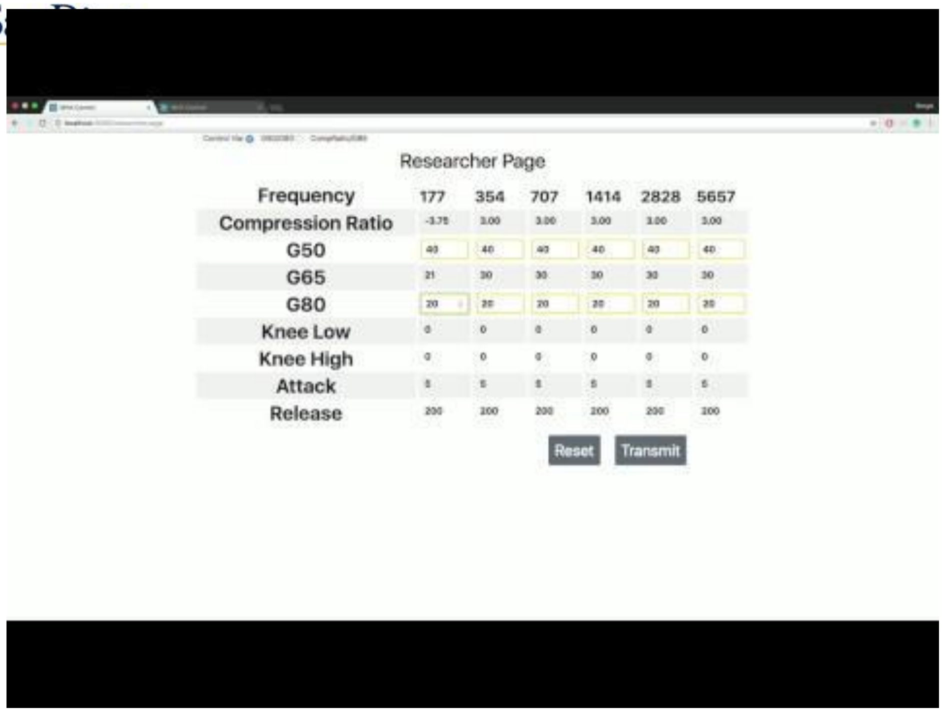
We include these as a way to jump start prototyping for researchers. As shown here, we've started with the assumption that users of this platform are interested in:

1. Initial settings prescription for the HA parameters, such as NAL-NL2
2. Investigating intelligibility improvements as a function of various HA parameters in various background conditions
3. Providing stimuli and getting user responses
4. Having the ability to do various Ecological Momentary Assessments

These are only a few of the possible scripts that could be useful to your research, and we plan to constantly be making additions to this list, and making them available to you.

For this part of our platform, we'd love to hear your input and forward it to the appropriate people to see it implemented. You can contact us through our website at openspeechplatform.ucsd.edu, or ask us questions during the questions period or any time during the conference.

Now, we have a couple videos demonstrating the operation of these web pages and the embedded web server.



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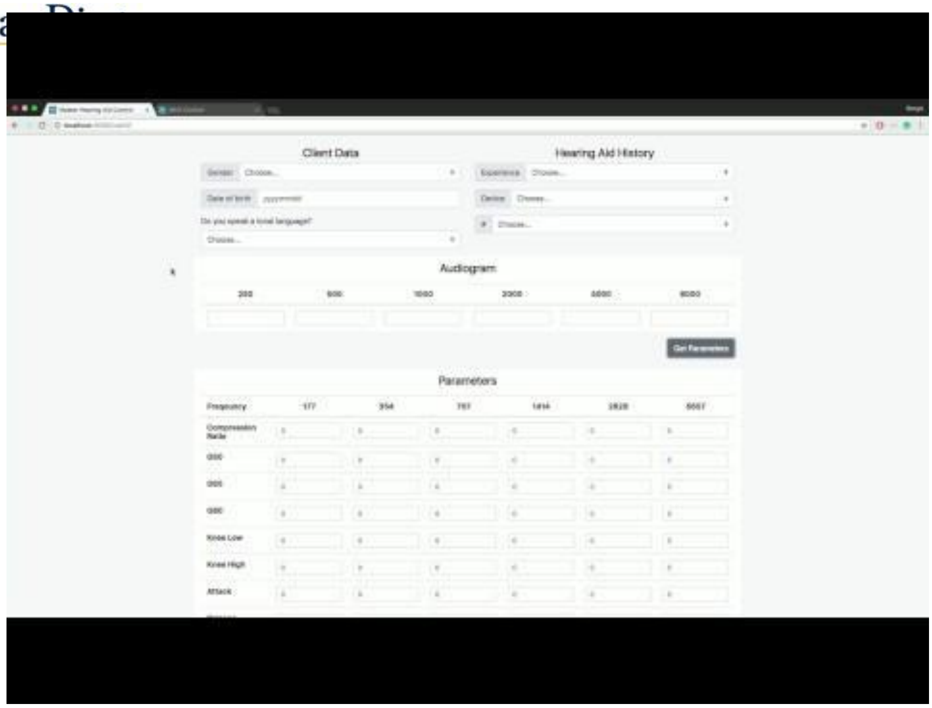

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That concludes our presentation for today; and we’ll now move onto questions, if there are any.