Comprehensive recording of auditory evoked potentials by projecting over a base of functions

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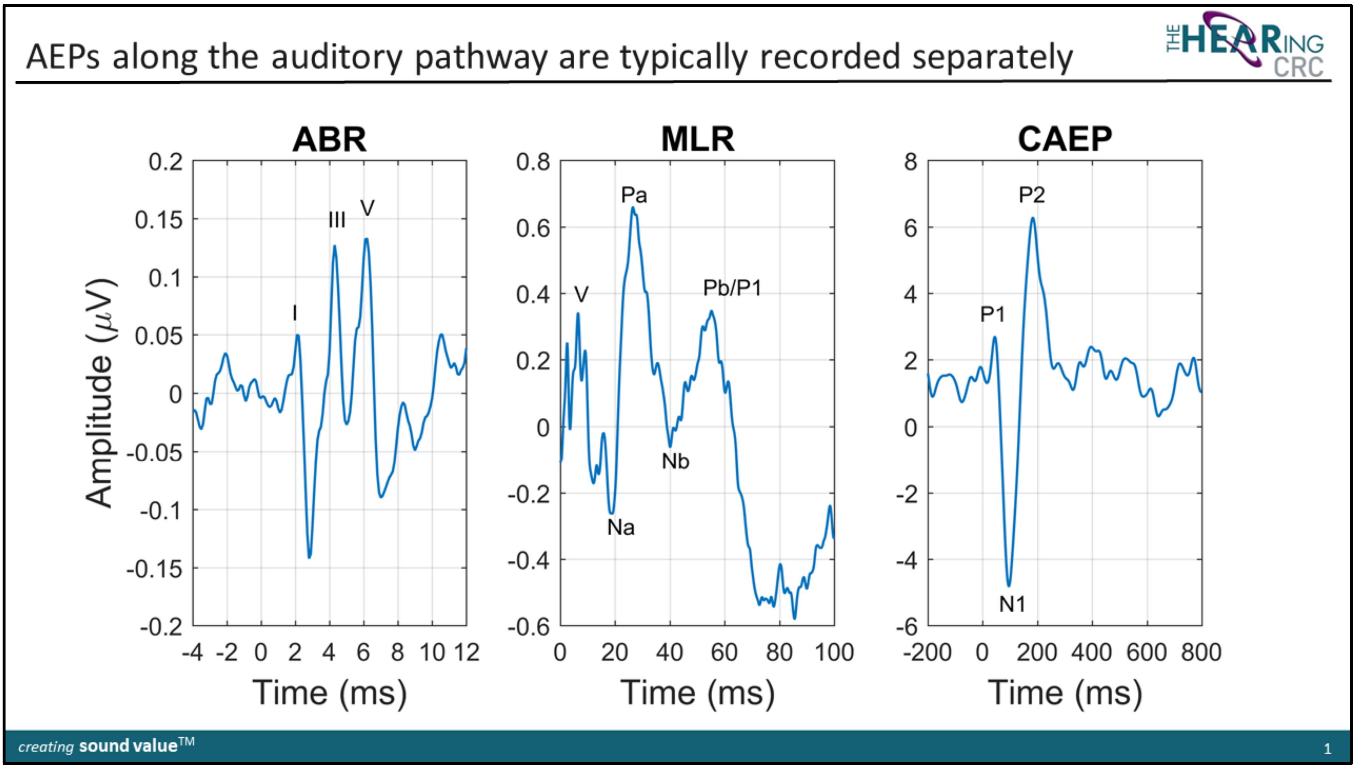
- Good morning, I am Joaquin Valderrama, I am with the National Acoustic Laboratories and Macquarie University.

- It is for me a pleasure and an honour to present here at this conference the main results of our current research.

- In this presentation, I will talk about a latency-dependant filtering that allows, among other applications, the comprehensive recording of AEPs along the auditory pathway, from the ABR wave I to the auditory cortex.



AEPs along the auditory pathway are typically recorded separately

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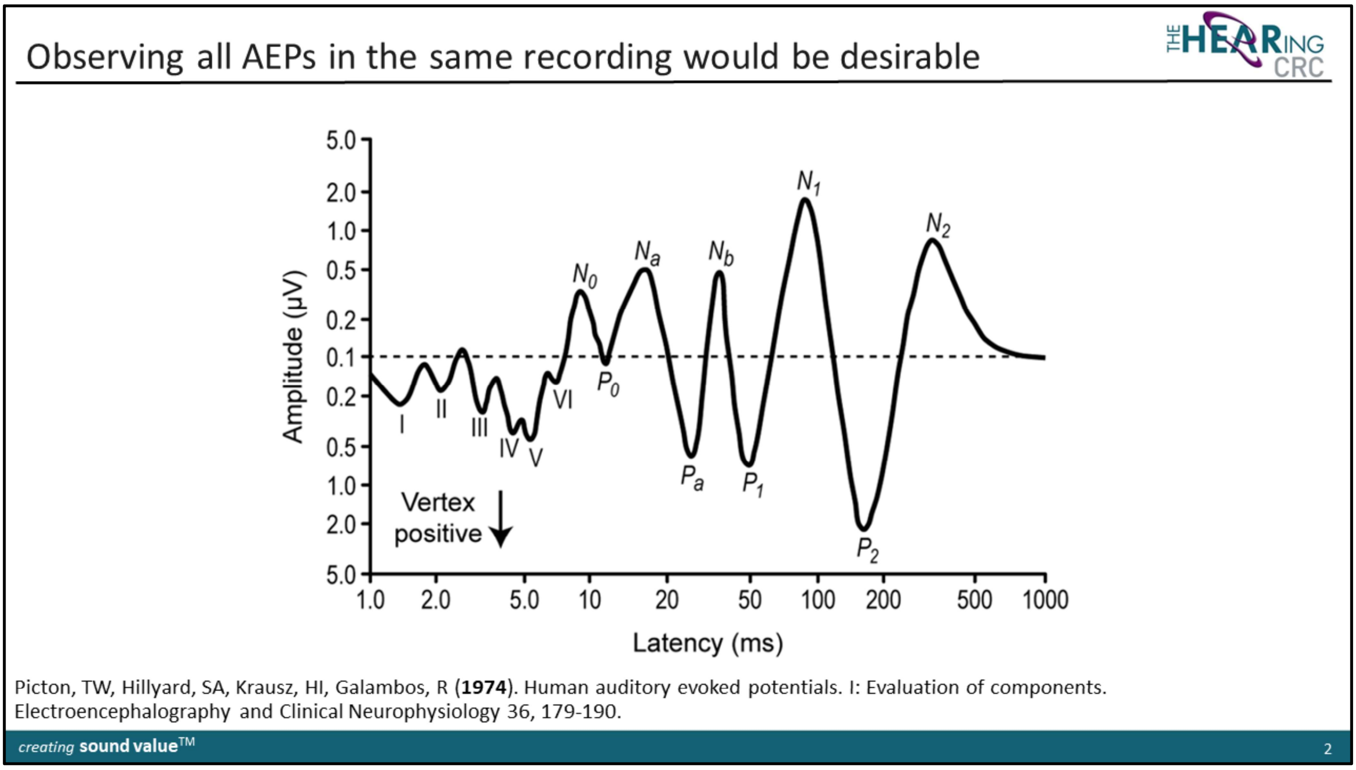
- AEPs along the auditory pathway are typically recorded separately.

- ABR signals are obtained by presenting several brief stimuli, using filters with cut-off frequencies between [100 - 3000] Hz; while CAEP are obtained by presenting fewer stimuli at a lower rate, with filters between [1 - 30] Hz.

- This is a problem, because it prevents all AEPs to be recorded in the same recording. We cannot analyse CAEP in ABR signals, neither the ABR in CAEP signals.



Observing all AEPs in the same recording would be desirable

Picton, TW, Hillyard, SA, Krausz, HI, Galambos, R (**1974**). Human auditory evoked potentials. I: Evaluation of components. Electroencephalography and Clinical Neurophysiology 36, 179-190.

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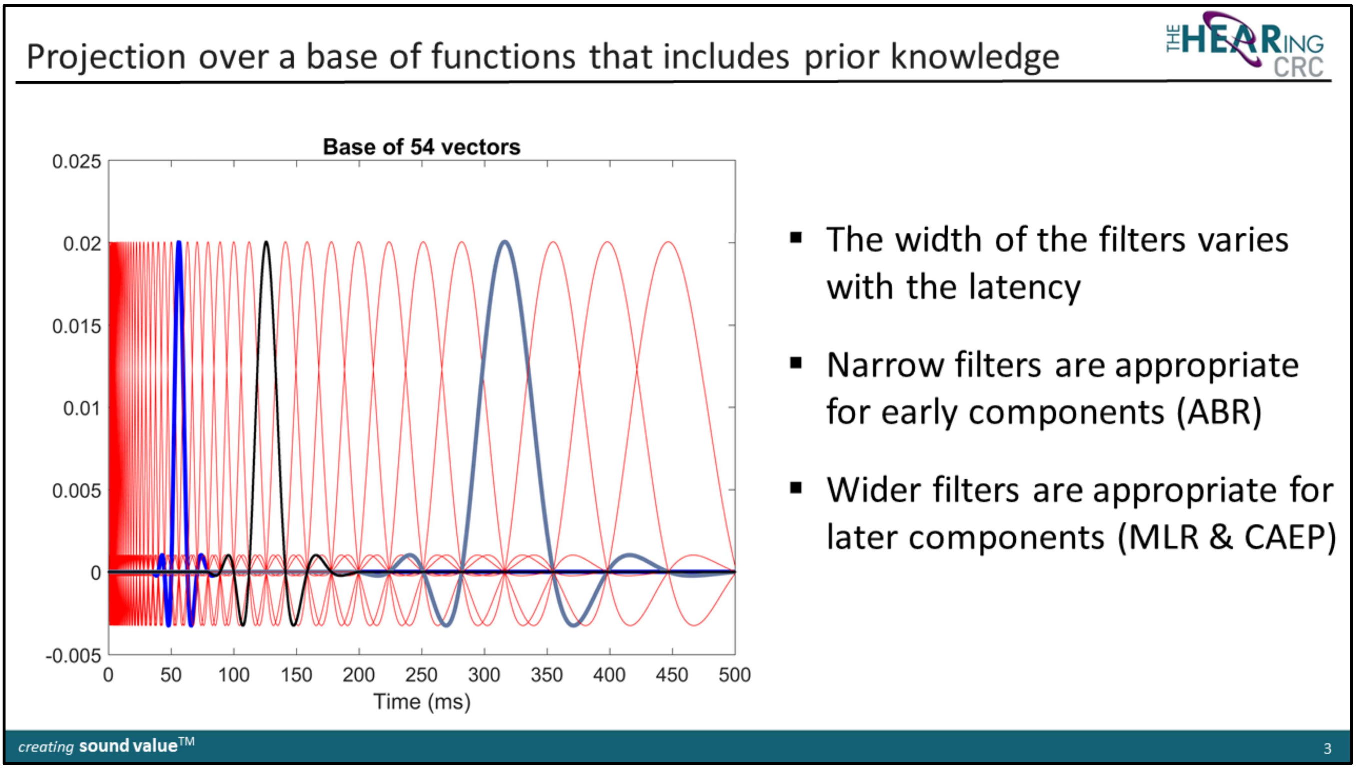
- However, observing all AEPs in the same recording would be desirable.

- This idea is not new. This is a diagram [NOT A REAL RESPONSE] which was presented in 1974.

- Please note the logarithmic scale of the time axis, which allows analysis of every component. The use of a linear scale would make the analysis of the early components difficult, because they would be very close one to each other.

- As mentioned before, this is a diagram, not a real response, because obtaining a signal like this is not straightforward. If we filtered the raw data with a wide filter, then the ABR would be contaminated by low-frequency noise, and the CAEP would be contaminated by high-frequency noise.

- What we need is some kind of filtering stage with a variable cut-off frequency, dependant on the latency of the response. In such a way that we could filter high-frequency components in the early response (ABR) and lower-frequency components in the late response (CAEP). This is precisely what we have implemented.



Projection over a base of functions that includes prior knowledge

- The width of the filters varies with the latency
- Narrow filters are appropriate for early components (ABR)
- Wider filters are appropriate for later components (MLR & CAEP)

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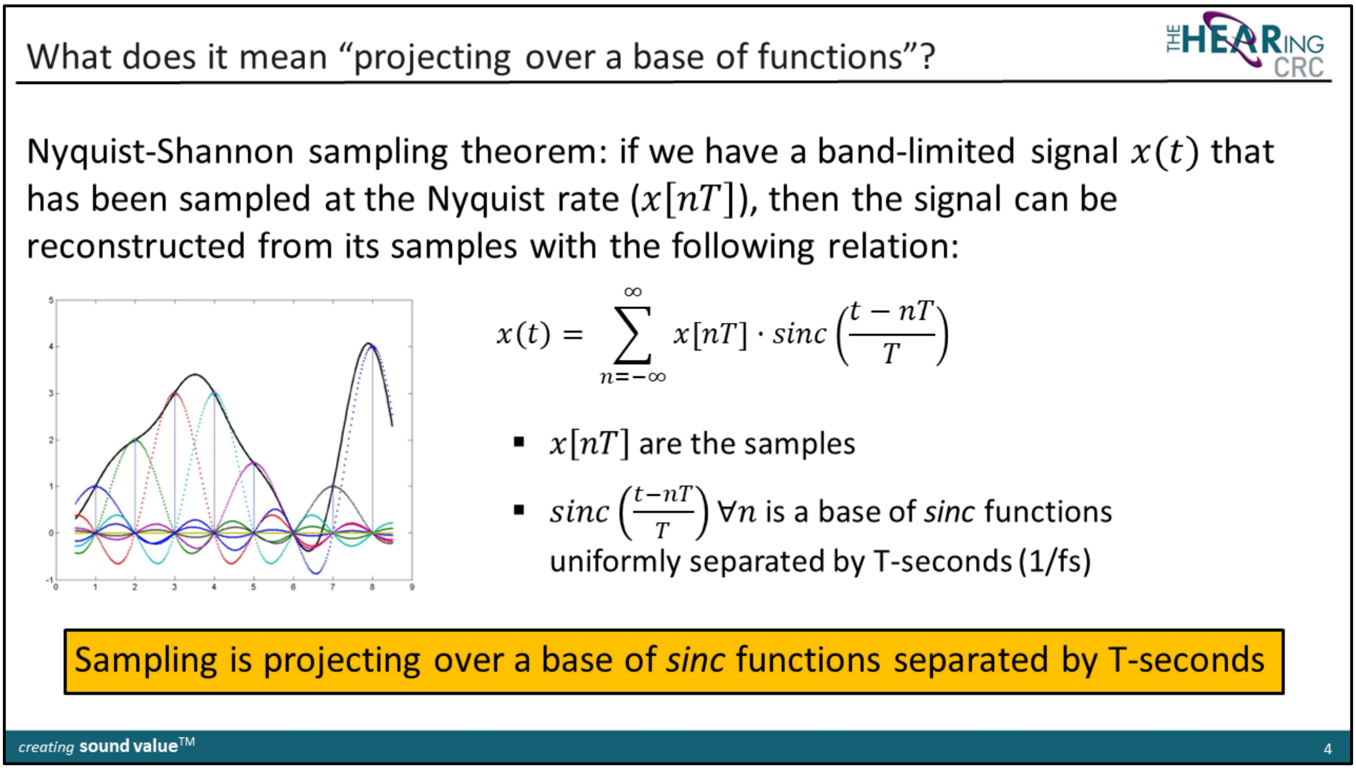
- How? Projecting the recordings over an orthonormal base that includes prior knowledge of the response.

- Let’s go slowly in this section. We have built a base of functions in which the width of the filters is latency-dependant. Narrow filters are positioned in the beginning of the response, where high-frequency components are expected (ABR), while wider filters are positioned as the latency of the response increases, appropriate for MLR and CAEP.

- These filters prevent low-frequency components in the early part of the response and high-frequency components in the late response.

- Why have we chosen this filter morphology and not a Gaussian function for example?



What does it mean “projecting over a base of functions”?

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Nyquist-Shannon sampling theorem: if we have a band-limited signal $x(t)$ that has been sampled at the Nyquist rate ($x[nT]$), then the signal can be reconstructed from its samples with the following relation:

$$x(t) = \sum_{n=-\infty}^{\infty} x[nT] \cdot sinc\left(\frac{t - nT}{T}\right)$$

- $x[nT]$ are the samples
- $sinc\left(\frac{t-nT}{T}\right) \forall n$ is a base of *sinc* functions uniformly separated by T-seconds (1/fs)

Sampling is projecting over a base of *sinc* functions separated by T-seconds

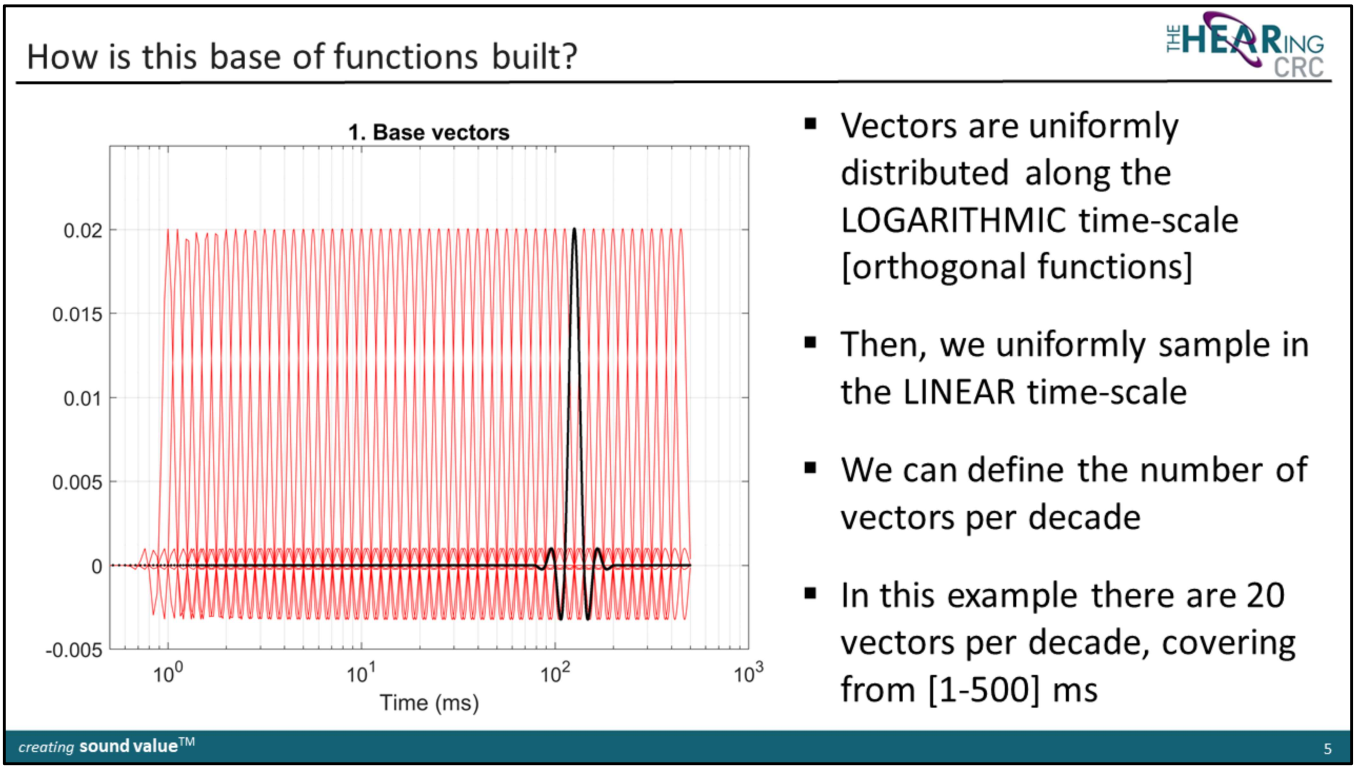
- The basis is on the Nyquist Theorem, which states that… [read]

- In this equation, x[nT] are the samples, and sinc() for all n is a base of sinc functions linearly separated by T-seconds (1/fs).

- This means that sampling can be considered as projecting the raw signal over a base of sinc functions linearly separated by T-seconds (sampling period or the inverse of the sampling frequency).

- This base of vectors is orthogonal, since the scalar product of any pair of vectors of the base is 0.

- This is why we have chosen sinc functions as functions of the base.



How is this base of functions built?

1. Base vectors

- Vectors are uniformly distributed along the LOGARITHMIC time-scale [orthogonal functions]
- Then, we uniformly sample in the LINEAR time-scale
- We can define the number of vectors per decade
- In this example there are 20 vectors per decade, covering from [1-500] ms

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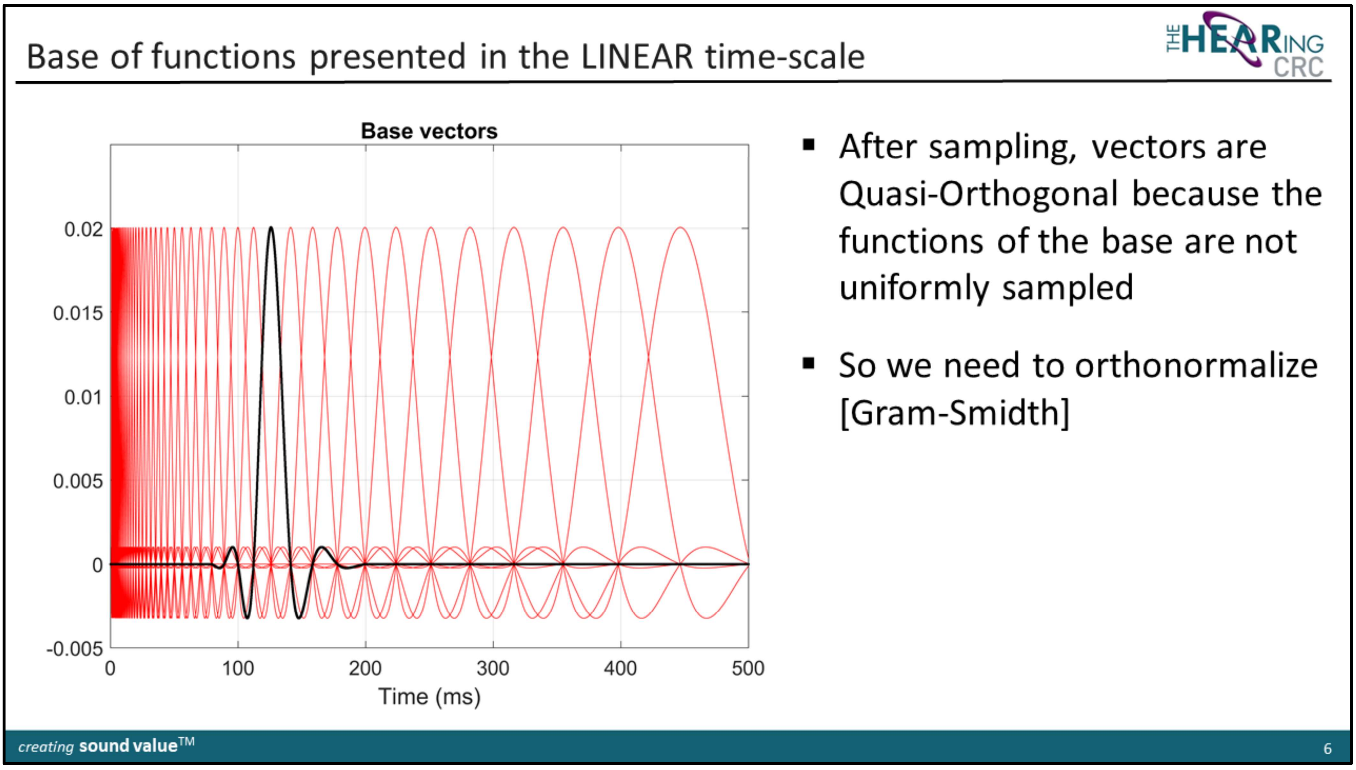
- How is this base of functions built?

- What we have done is to apply the sampling theorem, but instead that presenting the sinc functions linearly in the time domain, we have presented the sinc functions uniformly in the logarithmic time domain.



Base of functions presented in the LINEAR time-scale

- After sampling, vectors are Quasi-Orthogonal because the functions of the base are not uniformly sampled
- So we need to orthonormalize [Gram-Smidth]

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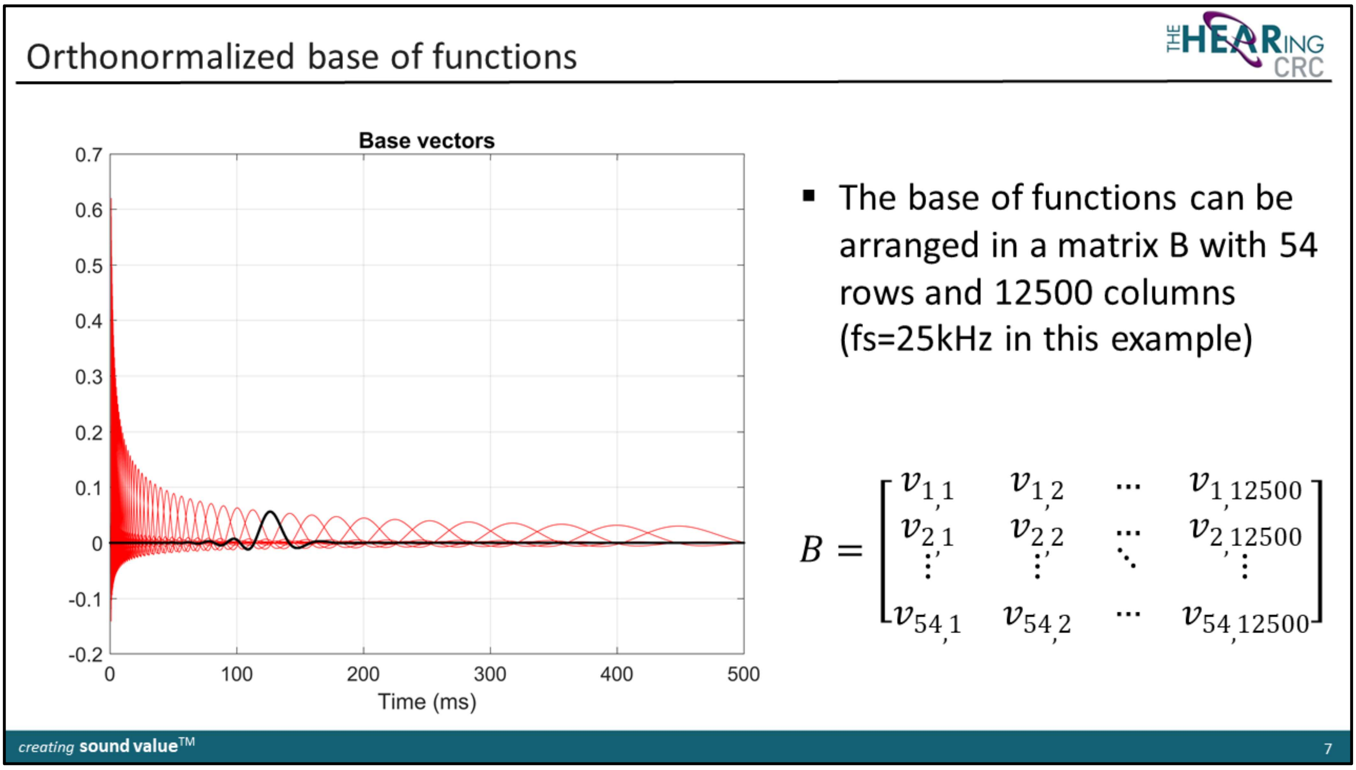
- Text



Orthonormalized base of functions

- The base of functions can be arranged in a matrix B with 54 rows and 12500 columns (fs=25kHz in this example)

$$B = \begin{bmatrix} v_{1,1} & v_{1,2} & \cdots & v_{1,12500} \\ v_{2,1} & v_{2,2} & \cdots & v_{2,12500} \\ \vdots & \vdots & \ddots & \vdots \\ v_{54,1} & v_{54,2} & \cdots & v_{54,12500} \end{bmatrix}$$

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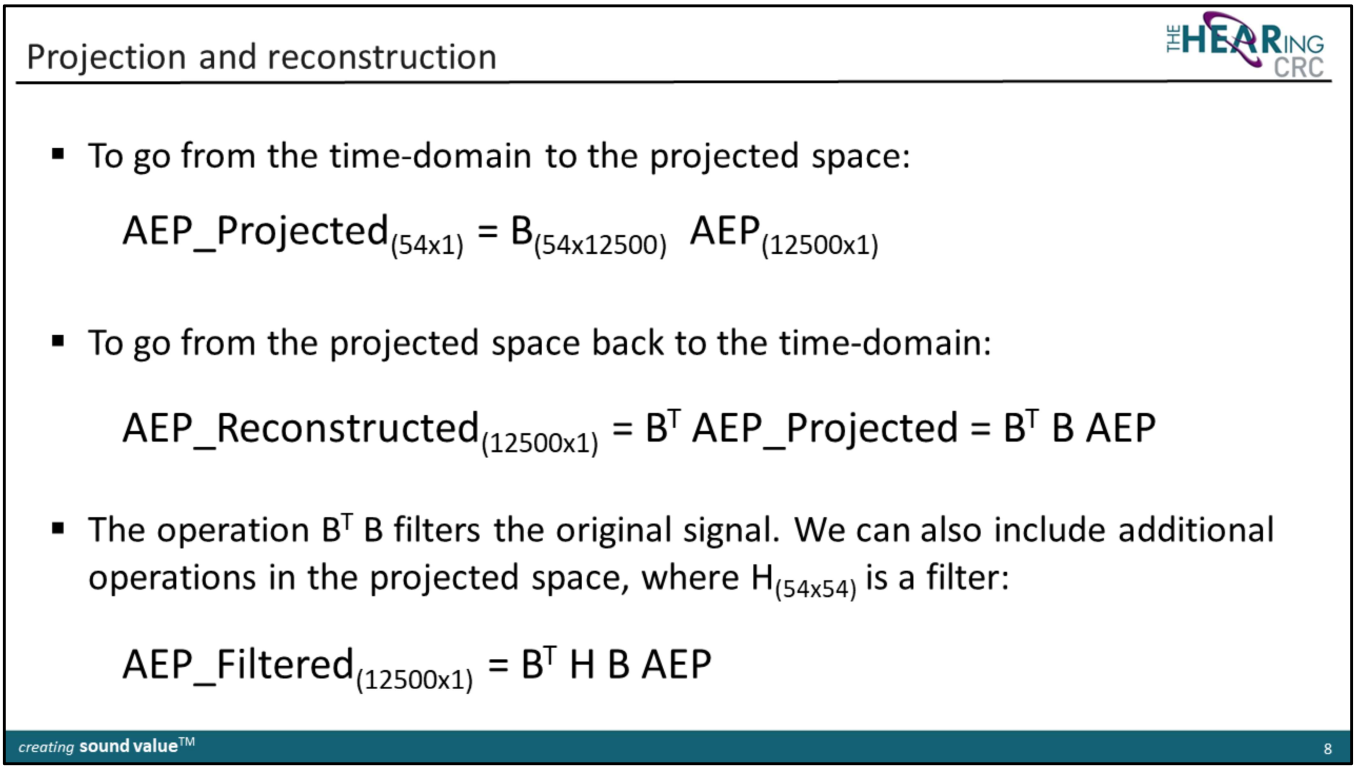
- Text



Projection and reconstruction

- To go from the time-domain to the projected space:

$$\text{AEP\_Projected}_{(54x1)} = \text{B}_{(54x12500)}\ \ \text{AEP}_{(12500x1)}$$

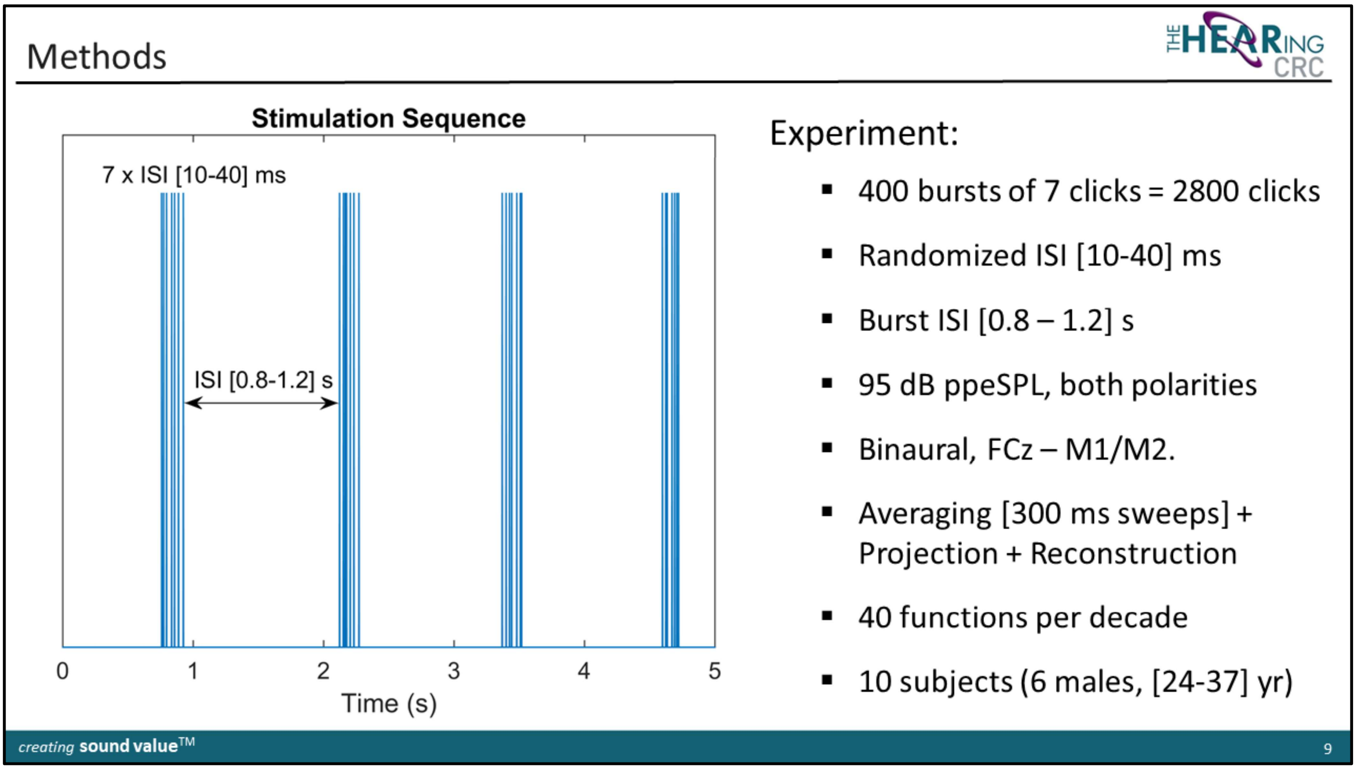
- To go from the projected space back to the time-domain:

$$\text{AEP\_Reconstructed}_{(12500x1)} = \text{B}^{\text{T}}\ \text{AEP\_Projected} = \text{B}^{\text{T}}\ \text{B AEP}$$

- The operation B^{T} B filters the original signal. We can also include additional operations in the projected space, where $\text{H}_{(54x54)}$ is a filter:

$$\text{AEP\_Filtered}_{(12500x1)} = \text{B}^{\text{T}}\ \text{H B AEP}$$

- Text



Methods

Experiment:

- 400 bursts of 7 clicks = 2800 clicks
- Randomized ISI [10-40] ms
- Burst ISI [0.8 – 1.2] s
- 95 dB ppeSPL, both polarities
- Binaural, FCz – M1/M2.
- Averaging [300 ms sweeps] + Projection + Reconstruction
- 40 functions per decade
- 10 subjects (6 males, [24-37] yr)

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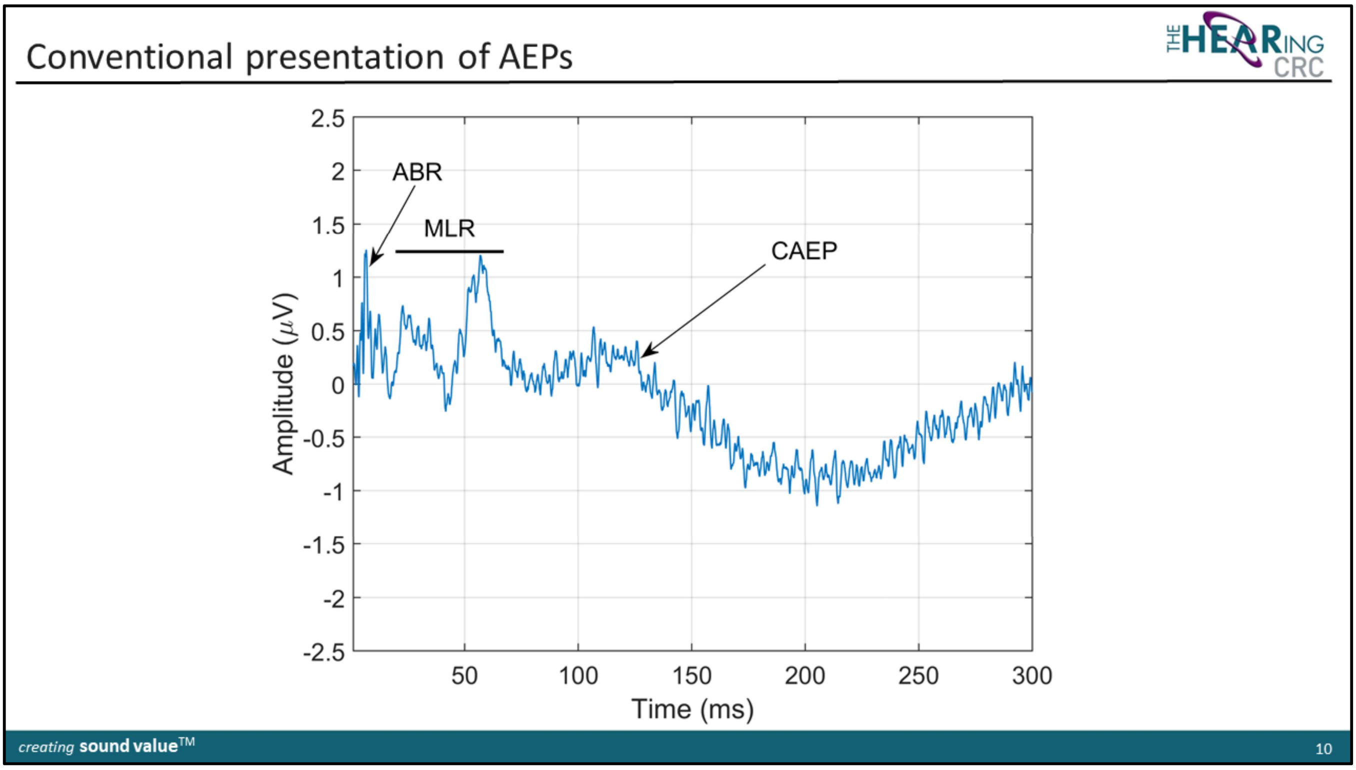
- The stimulation paradigm that we used to test the performance of the proposed latency-dependent filtering was a stimulation sequence consisting of 400 bursts of 7 clicks, which leads to a total amount of 2800 clicks.

- The ISI within the bursts of clicks was randomized between 10 and 40 ms, and the ISI between each burst of clicks was randomized between 800 ms and 1.2 seconds.

- The clicks were presented binaurally at 95 de ppeSPL, in the two polarities, with an electrode setup FCz-combined mastoid (re-referenced offline).

- Processing of data consisted of averaging 300 ms from each stimulus onset, and projecting and reconstructing from a base of functions consisting of 40 functions per decade.

- We recorded data from 10 normal hearing subjects selected from research personnel from the NAL and MQ University, 6 males, aged between 24 and 37 years.



Conventional presentation of AEPs

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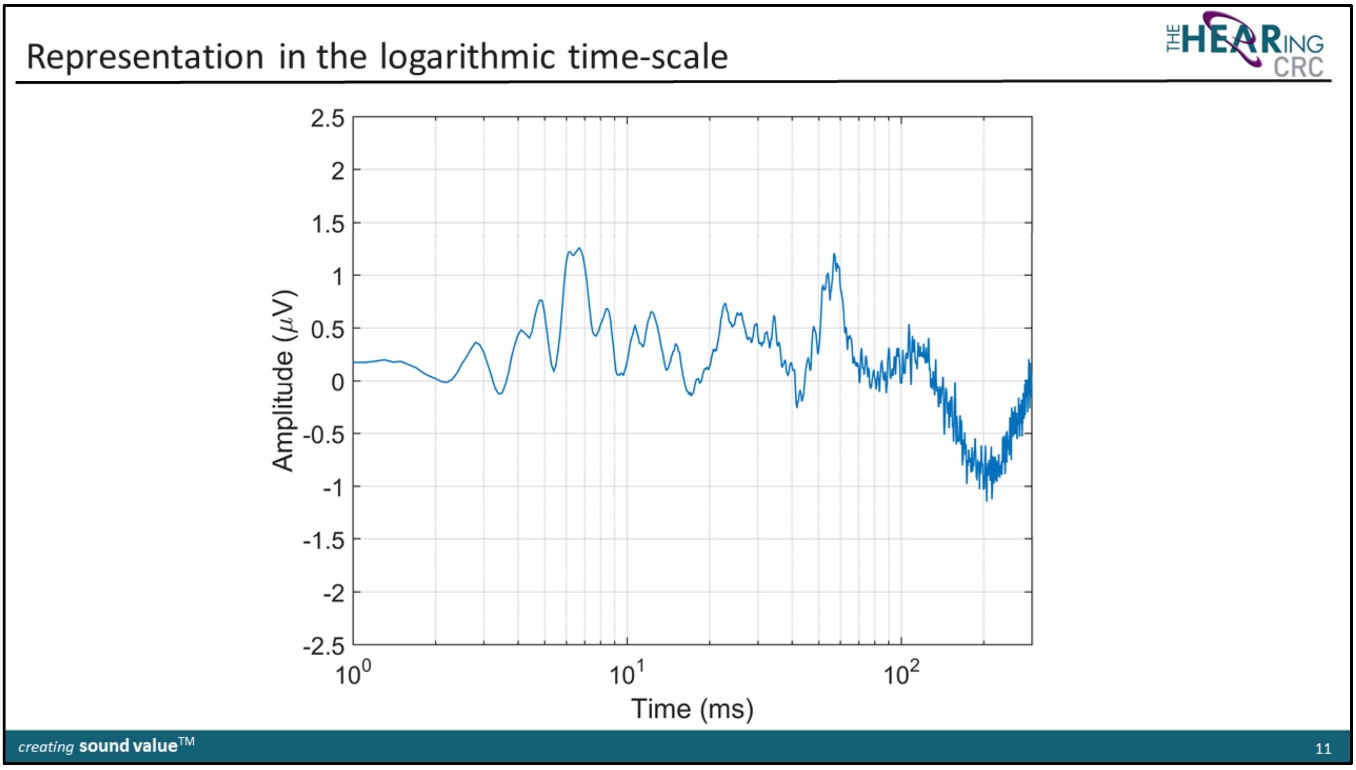
- This figure shows the average of the 2800 sweeps corresponding to the first 300 ms from stimulus onset.

- This figure shows the components of the ABR, MLR and a small-amplitude component of the CAEP.

- However, this figure also shows that the linear time scale is not appropriate to show all these components.



Representation in the logarithmic time-scale

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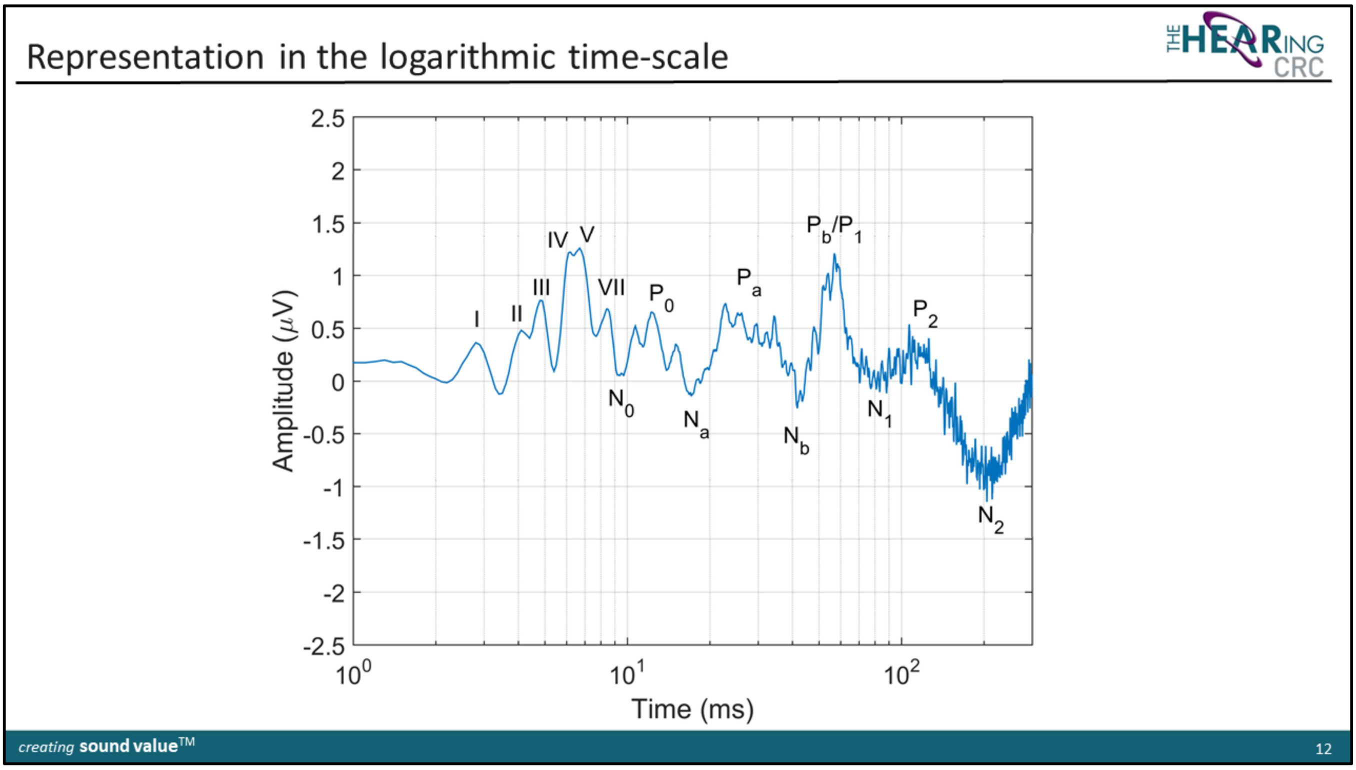
- Plotting the same signal in the logarithmic time scale is more appropriate.



Representation in the logarithmic time-scale

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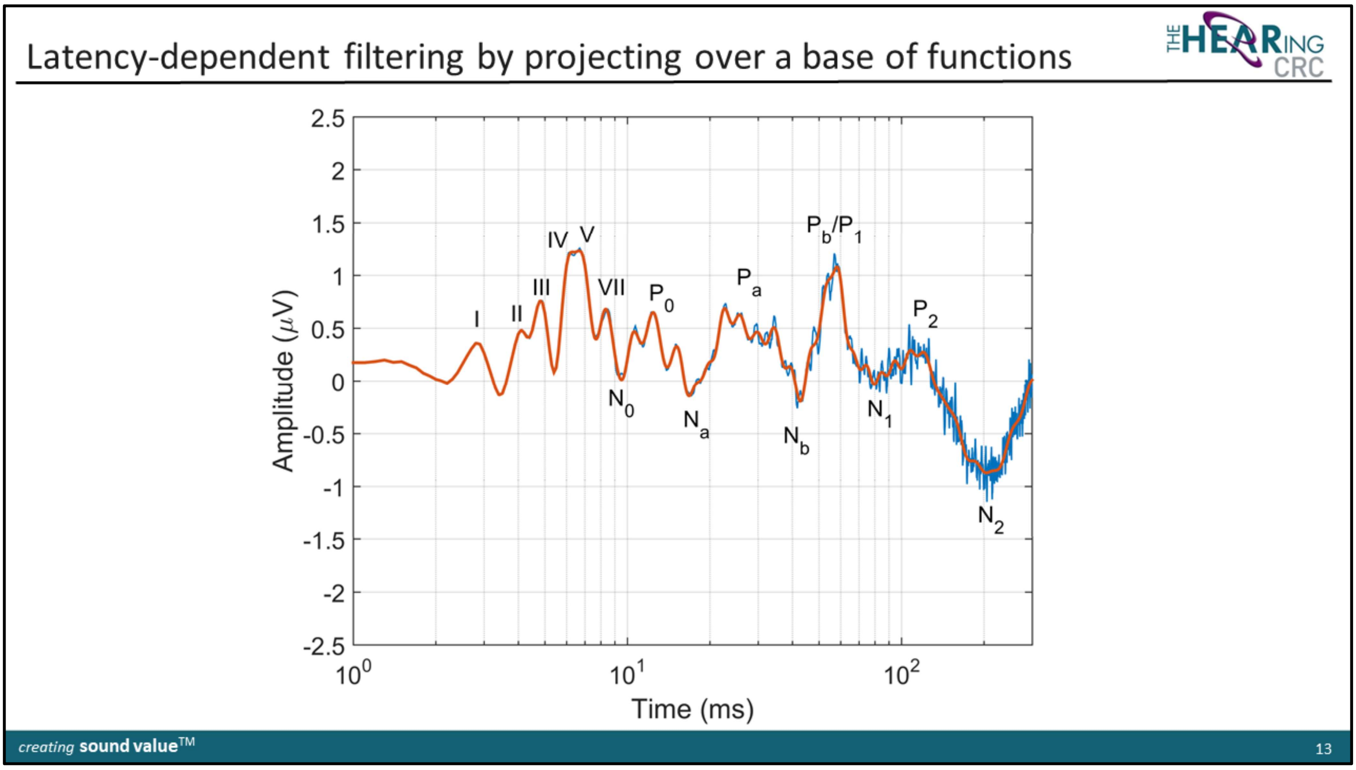
- Here, we can identify all components more clearly.

- However, we also observe that in the logarithmic time scale the later components of the signal are particularly contaminated by high-frequency components.



Latency-dependent filtering by projecting over a base of functions

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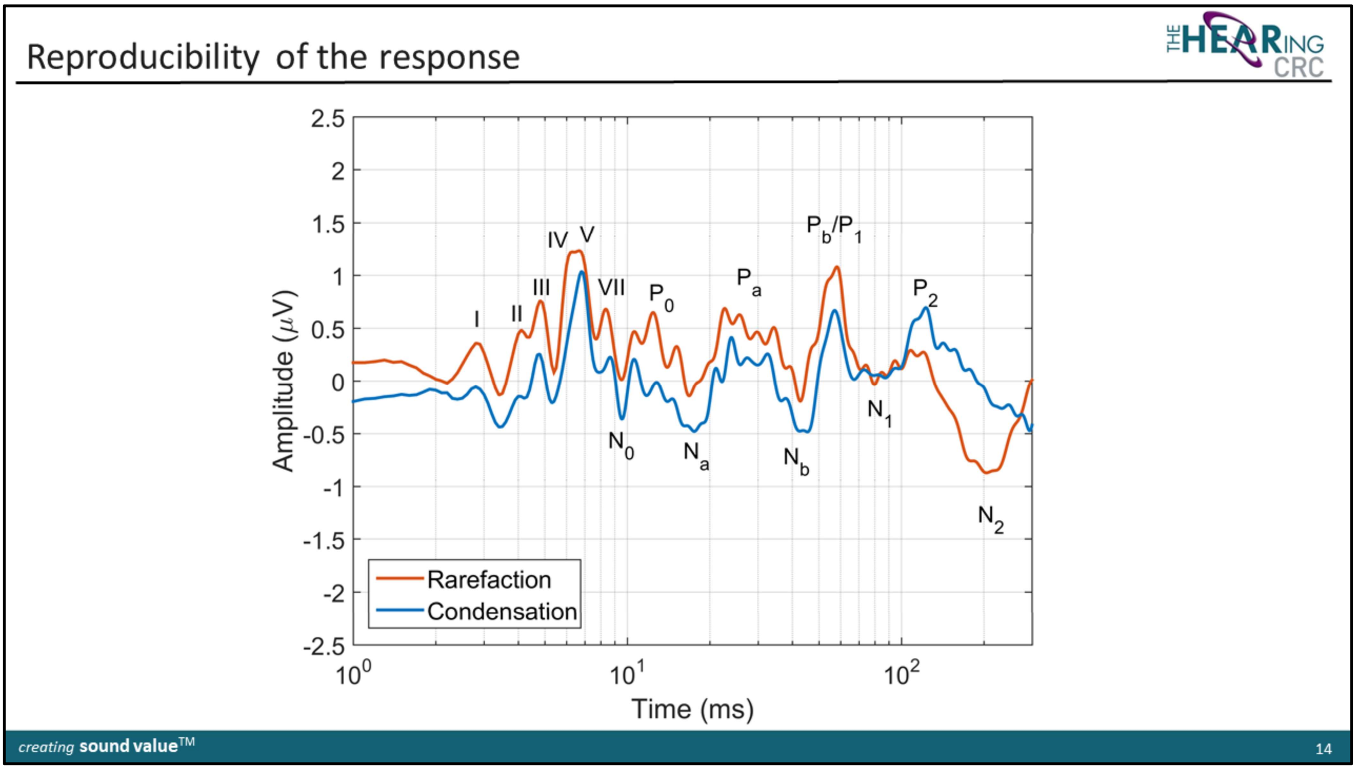
- The orange line shows the result of projecting and reconstructing from the proposed based of functions.

- This signal visually shows the performance of the latency-dependent filtering technique.



Reproducibility of the response

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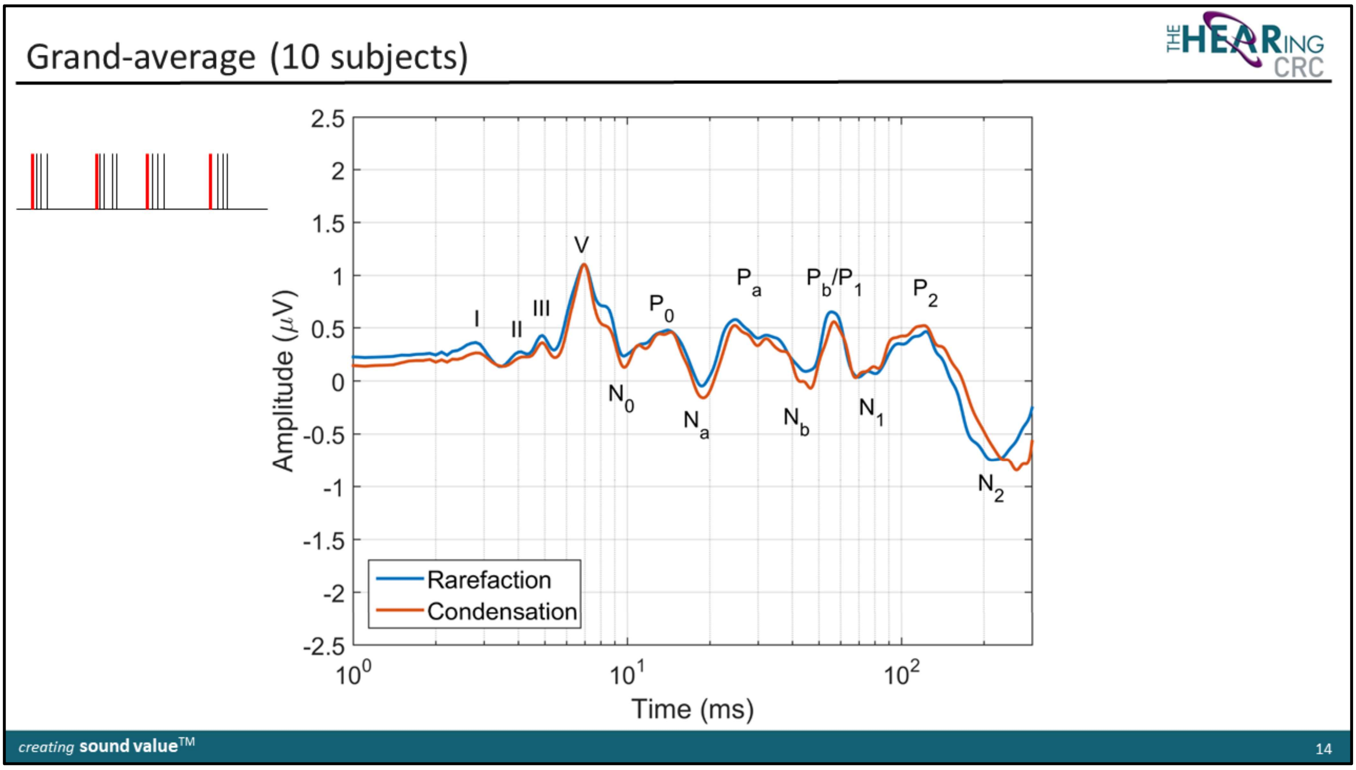
- These two signals show that all components of the signal are reproducible in the two polarities.



Grand-average (10 subjects)

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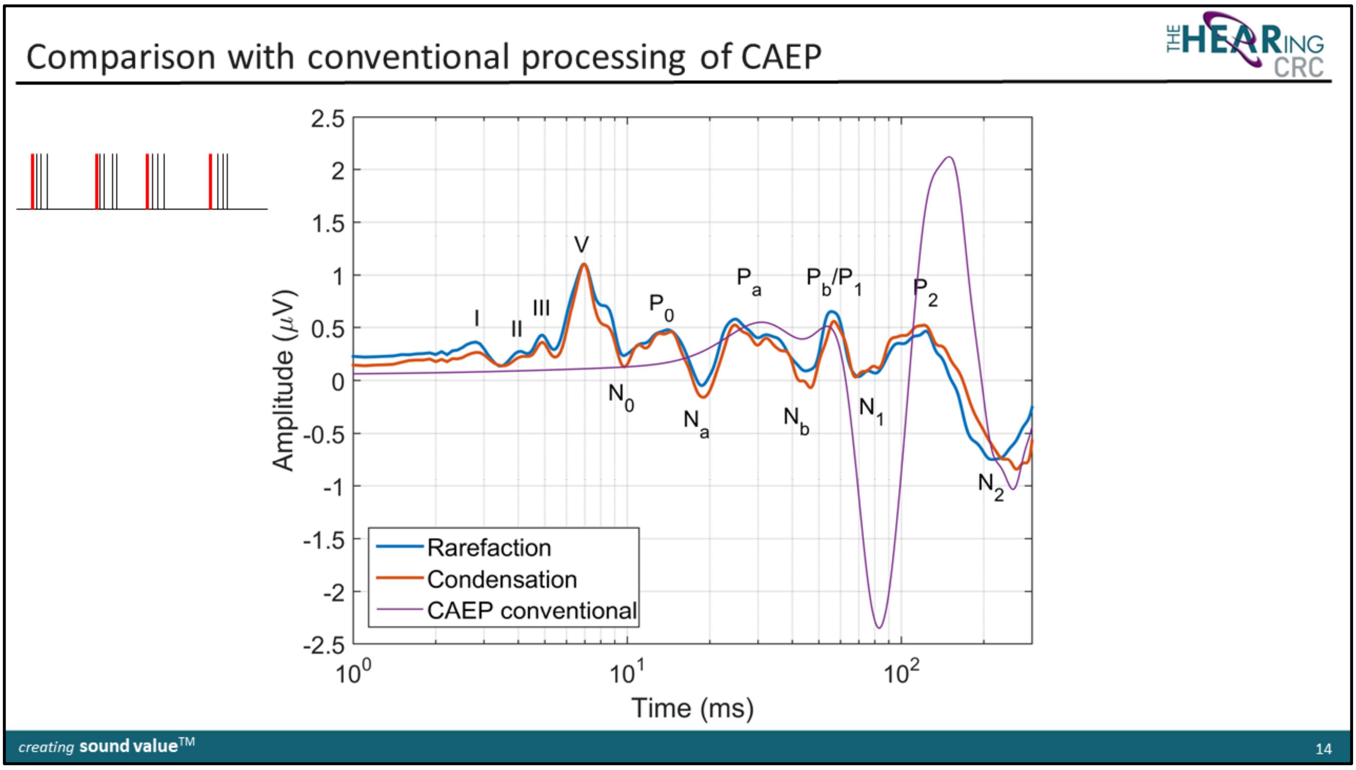
- The grand-average signals obtained with the full dataset (10 subjects) show these signals with a greater quality.

- We can observe that the amplitudes of the CAEP components are particularly reduced.

- This is probably a consequence of contributing to the CAEP only the first click of each burst [top-left figure], in other words, the CAEP is elicited by the full burst, rather than individually by each click.

- Therefore, the average of all clicks leads to a lower-amplitude CAEP component.



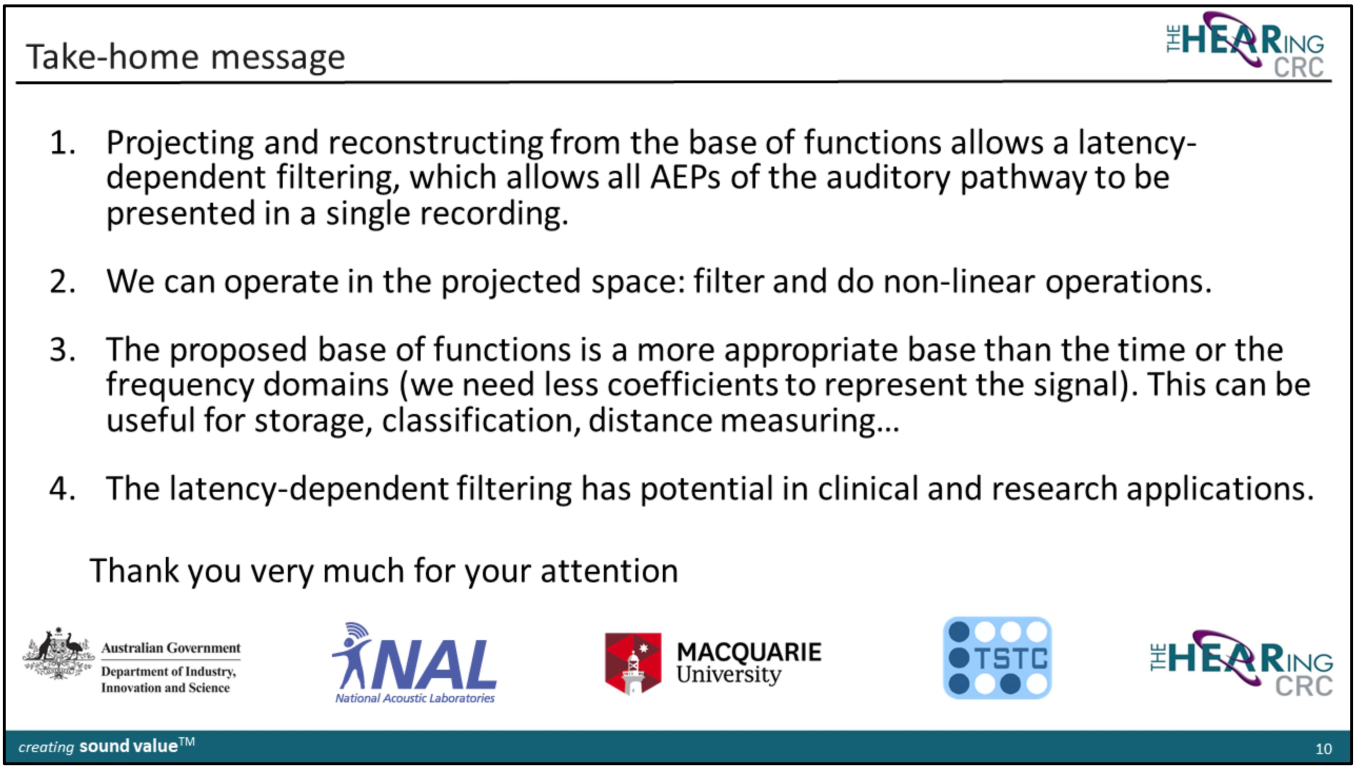
Comparison with conventional processing of CAEP

- This effect can be observed if we compare the projected signals with the CAEP obtained conventionally considering only the responses corresponding to the first click of each burst [red clicks on the top-left figure].

- Here we see that the amplitude of the conventinal CAEP is much larger (approximately 7 times larger) than the one obtained with the latency-dependent filtering.

- However, we also observe that in the conventional CAEP, the ABR and MLR components cannot be identified.

- Future work will aim to improve the processing of the signal to compensate for this effect.



Take-home message

1. Projecting and reconstructing from the base of functions allows a latency-dependent filtering, which allows all AEPs of the auditory pathway to be presented in a single recording.
2. We can operate in the projected space: filter and do non-linear operations.
3. The proposed base of functions is a more appropriate base than the time or the frequency domains (we need less coefficients to represent the signal). This can be useful for storage, classification, distance measuring…
4. The latency-dependent filtering has potential in clinical and research applications.

Thank you very much for your attention

- I would like to finalise by highligting the three main “take-home messages” of this talk.

- I hope you have enjoyed this presentation. Thank you very much for your attention.