



An n-order hold approach for fractional-delay interpolation in auralization

Andrew Christian

Structural Acoustics Branch
NASA Langley Research Center

Randall Ali

Dept. of Electrical Engineering (ESAT)
STADIUS Center for Dynamical
Systems, Signal Processing and Data
Analytics, KU Leuven

Eric Greenwood

Assistant Professor
Dept. of Aerospace Engineering
The Pennsylvania State University

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Motivation



- We're interested in making artificial sounds for vehicles that we have high-resolution aeroacoustic predictions of.
 - The process is “auralization”
- If we can do this for vehicles that don't yet exist, perhaps we can take the human response to their noise signatures into account during the design phase.



Motivation

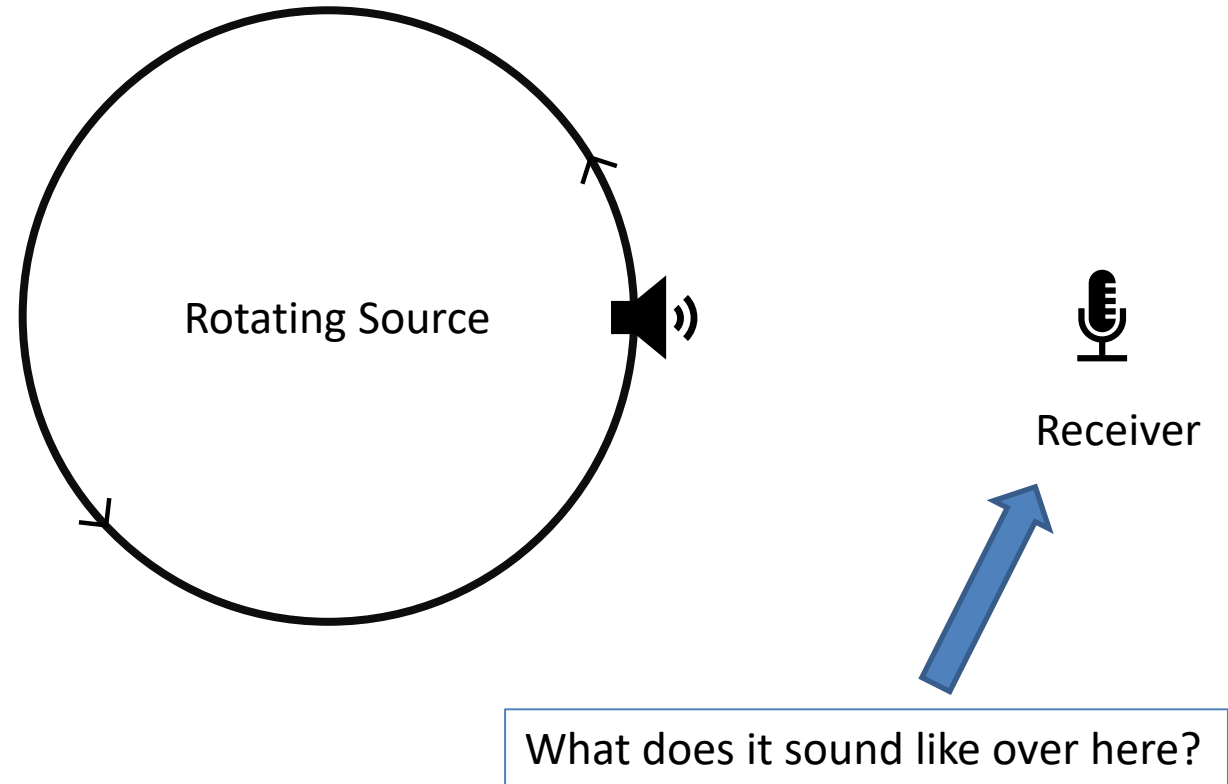


- Many current “advanced air mobility” concepts involve multi-rotor configurations for package delivery, air taxi, etc.
- We can create very high-resolution predictions for these types of vehicles, but could there be pitfalls in bringing these computer-generated sounds to life?



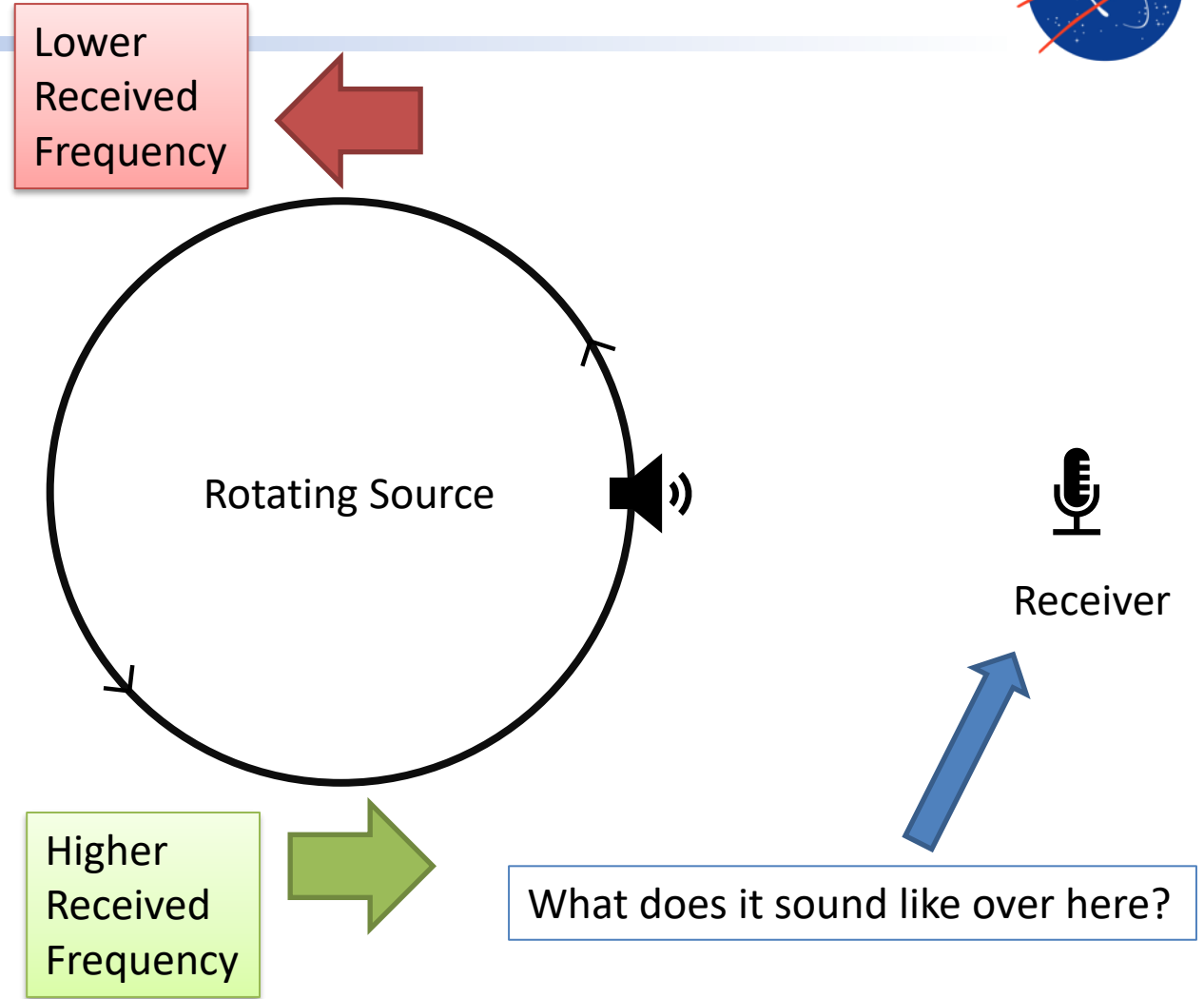
A Spinning Contrivance

- We are going to consider a very exaggerated scenario in this presentation:
 - 650 Hz monopole pure-tone source
 - Velocity of source = 45 m/s
 - About 2 RPS
 - Sound speed = 80 m/s
 - Circle radius = 3m
 - Receiver 5m away from circle center
- What we are not including:
 - Spherical Spreading
 - Atmospheric Absorption
 - Ground Reflections
 - 3D Effects, etc.



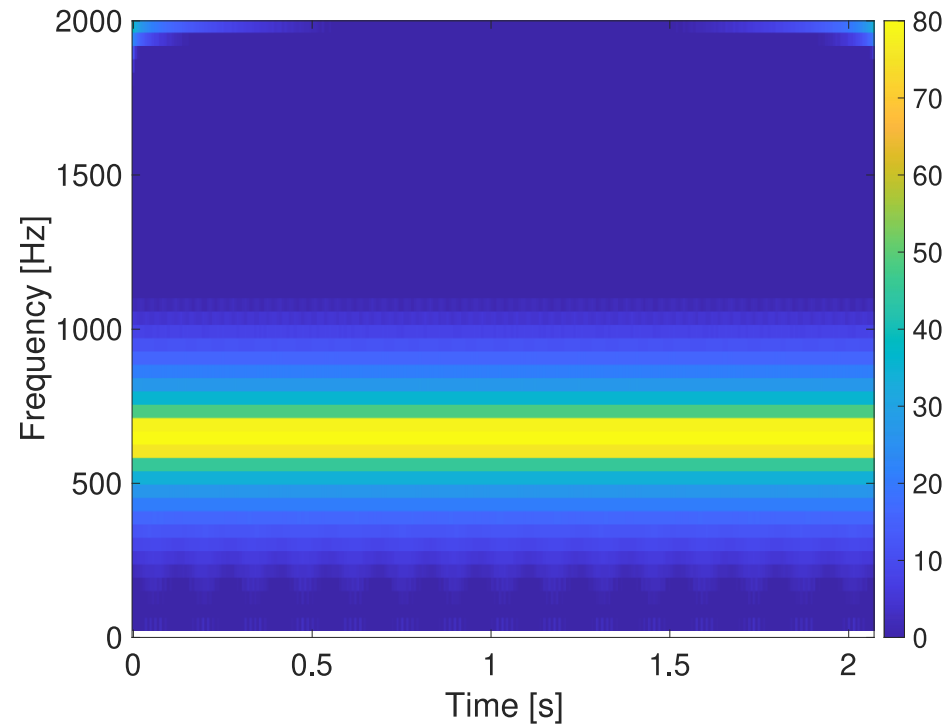
A Spinning Contrivance

- The primary effect of this scenario is to induce a significant Doppler shift on the tone at the receiver.
- This is caused by the time-stretching of the sound when the source is moving towards/away from the receiver

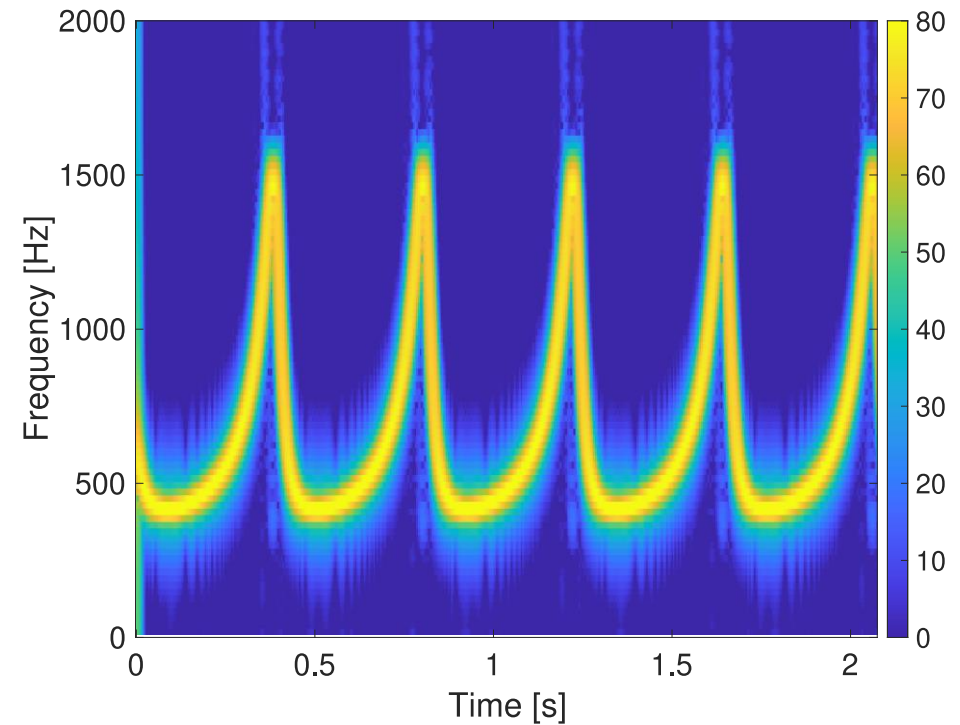


If it is done correctly...

Source Signal

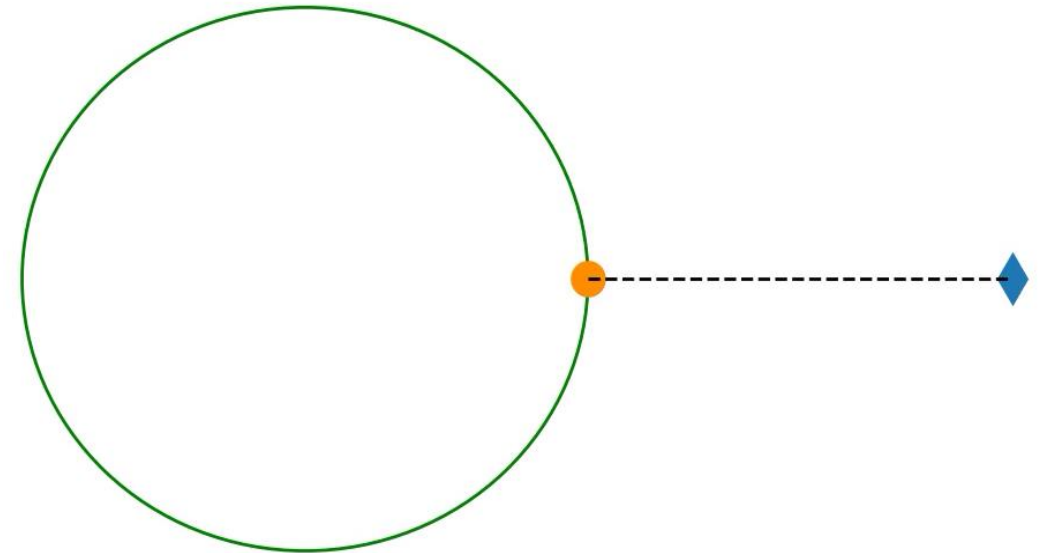


Received Signal



Auralization Processing Approaches

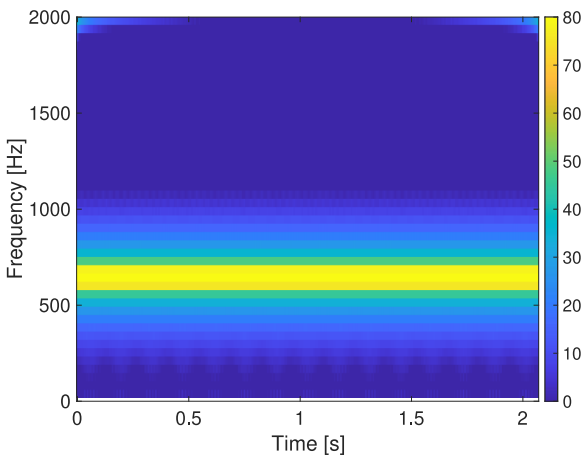
- The delay between the source and receiver varies continuously and will in general not fall on a source sample, so this requires some sort of “*Fractional Delay*” processing.
 - Other categories of DSP methods are based on the same underlying math: “Signal Reconstruction,” “Bandlimited Interpolation”
 - There are many ways to implement these DSP schemes (source-time vs. receiver-time dominant paradigms, etc.).
- The Doppler shift is a result of this processing – it arises naturally from a delay line that changes with length over time – and the quality of the simulation is a direct result of the quality of the interpolation used.



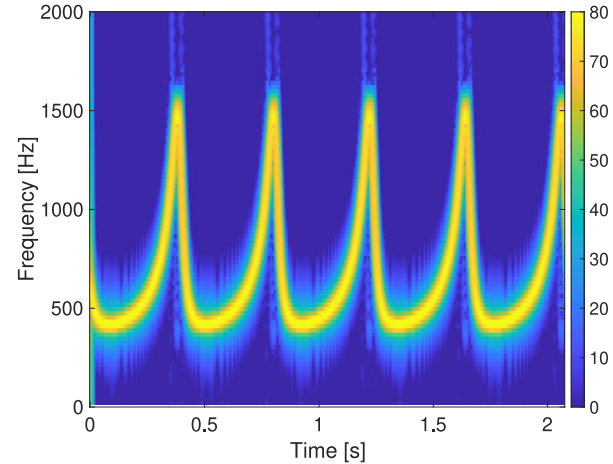
What If We Lower the Receiver Sampling Frequency?



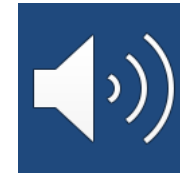
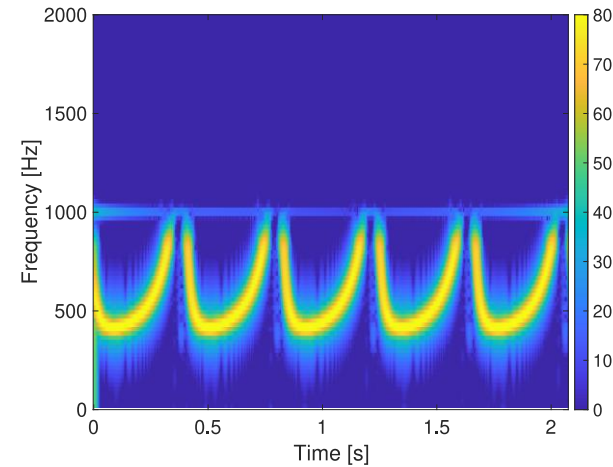
Source: 650 Hz sine



Receiver $f_s = 8$ kHz



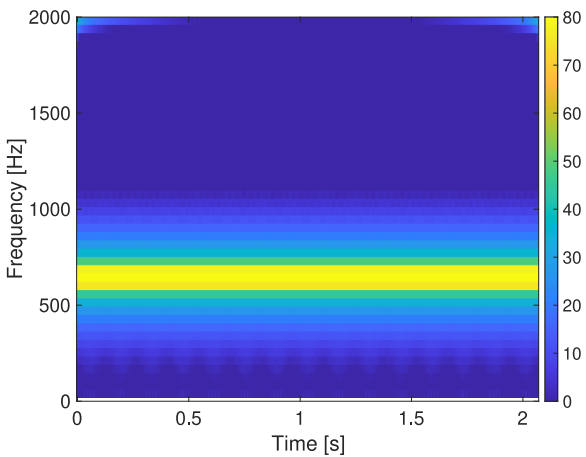
Decimate to $f_s = 2$ kHz



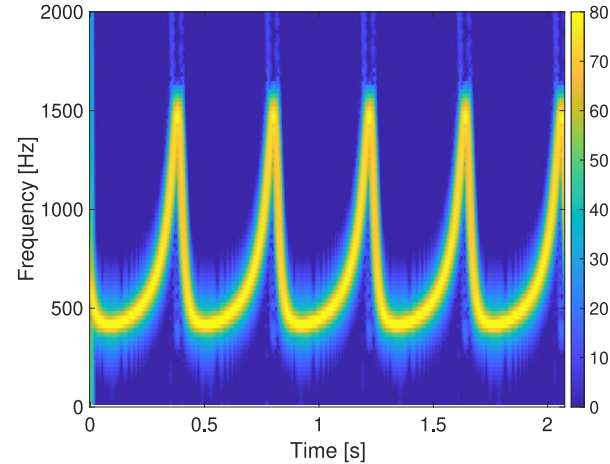
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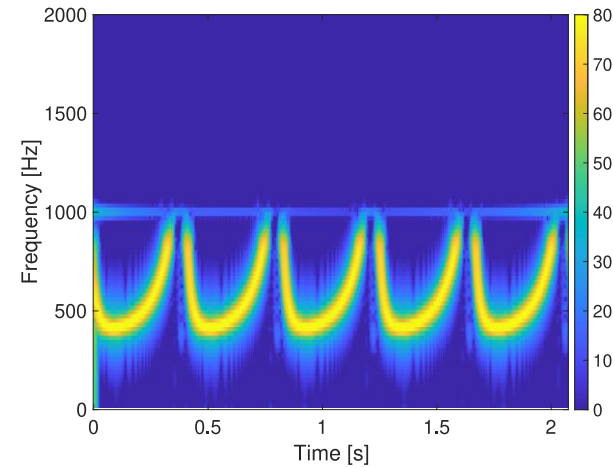
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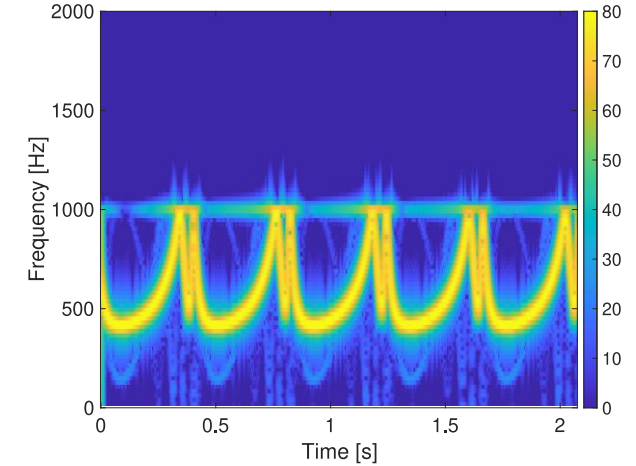
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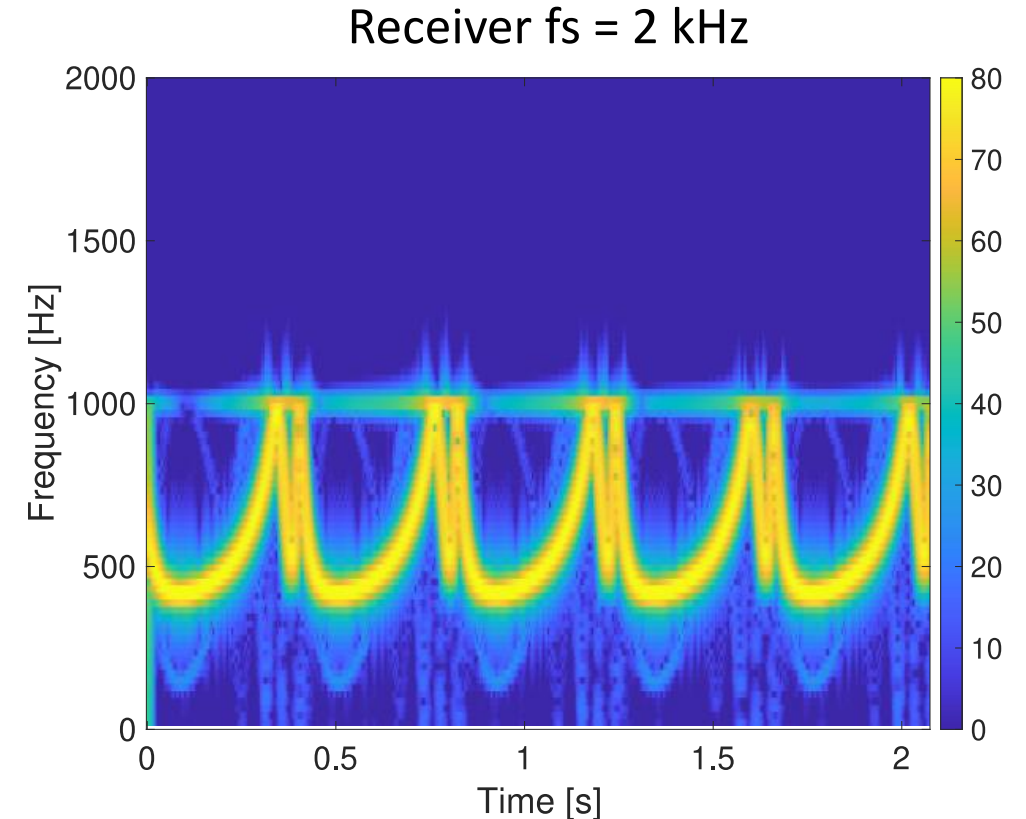
Receiver $f_s = 2$ kHz



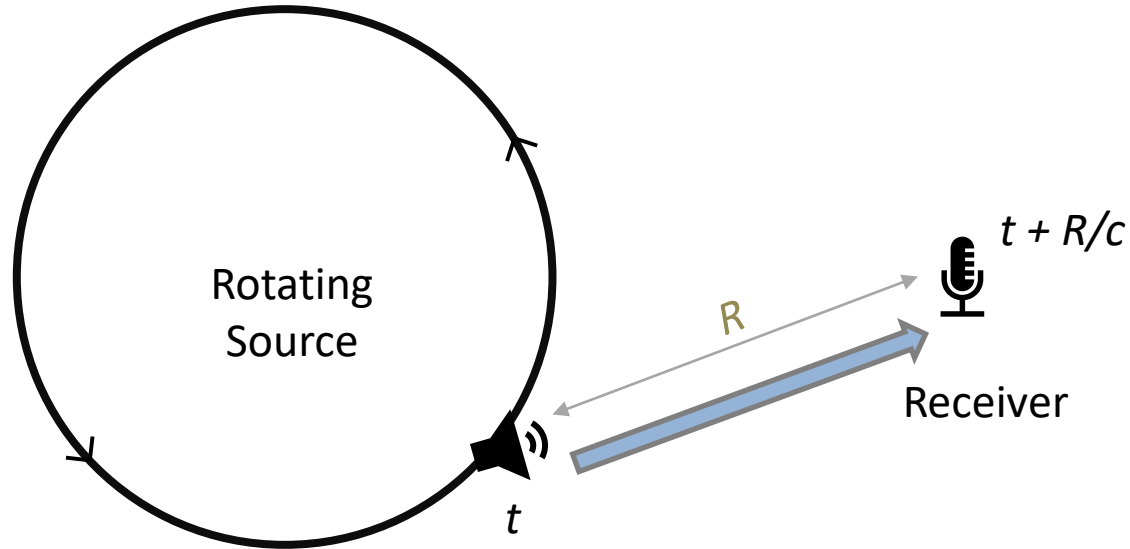
Aliasing!



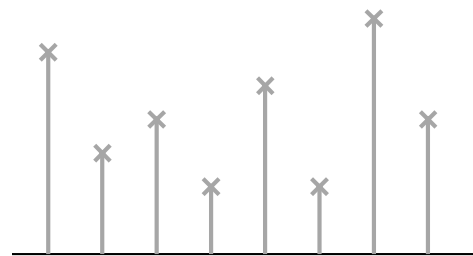
- There are ways to deal with this issue within existing auralization schemes:
 - Oversample at receiver
 - Necessarily slow, and requires knowledge about the entire source/transmission path
 - Filtering at source
 - Also requires us to evaluate the entire transmission path, and then communicate information back to the source
 - In this example, we'd get nothing...
 - Bandlimited interpolation
 - Same drawbacks as filtering
 - Computationally expensive



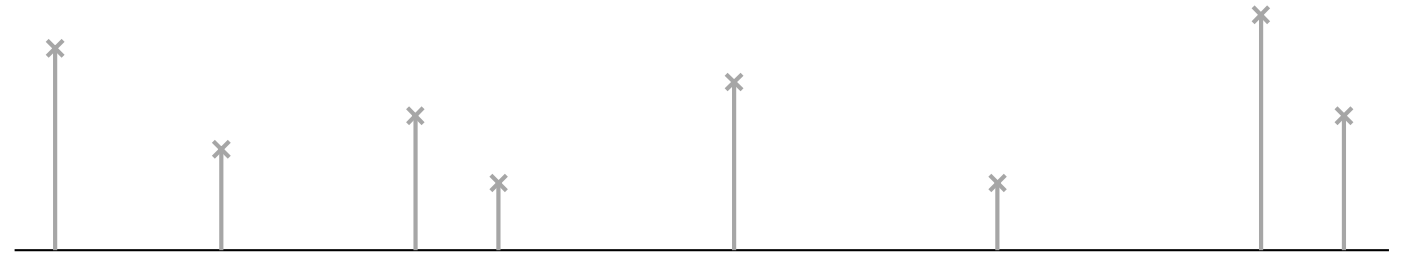
Can We Imagine A New Way?



- ‘Transmit’ the source samples to receiver using propagation delay applied to the samples of the source from their ‘natural’ times.
- A.k.a. ‘source-time’ dominant



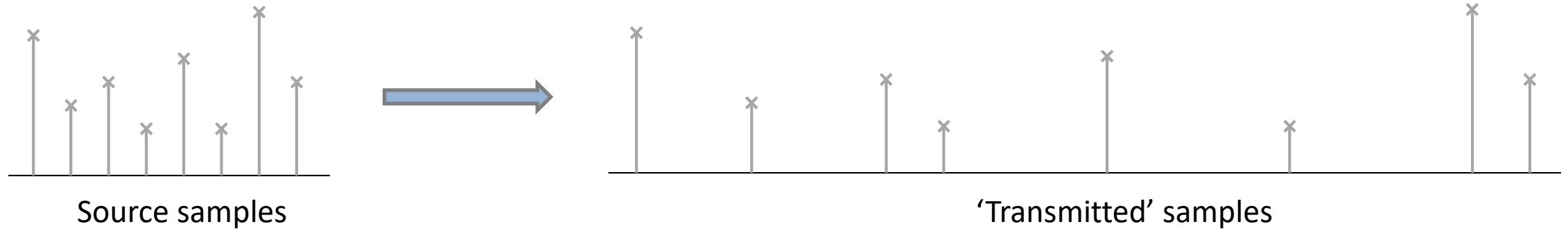
Uniformly-spaced samples at source



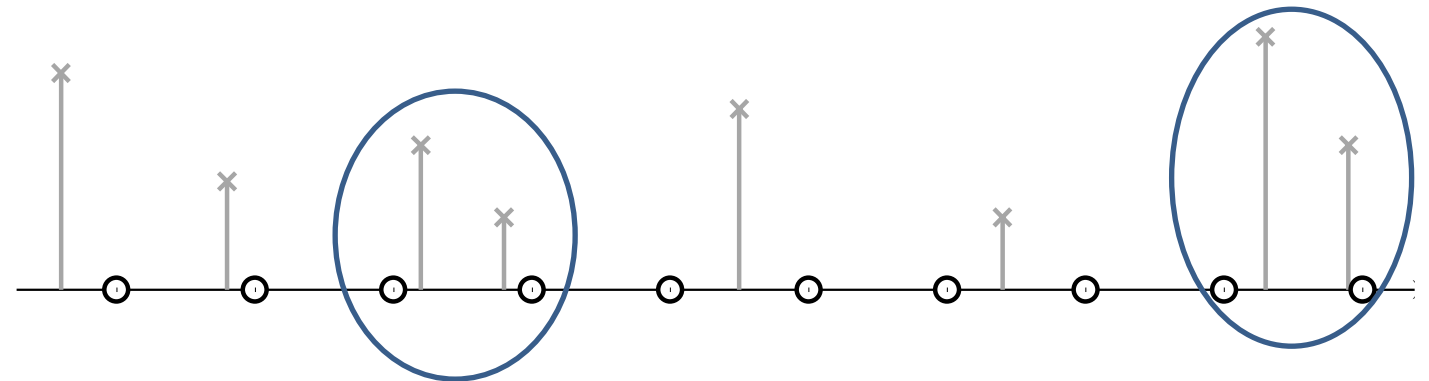
Irregularly-spaced samples at the receiver

We can't listen to this...

Can We Imagine A New Way?



Define the receiver f_s and interpolate into this uniform grid of samples



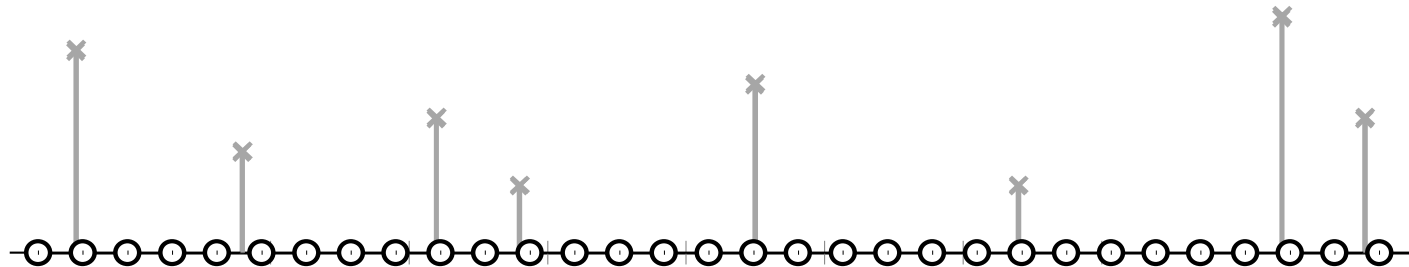
If the time spacing of the receiver samples $>$ than the smallest time difference between any two consecutive 'transmitted' samples, aliasing *can* occur!

Our proposal: NoTAP

N-oversampled Transmitted Acoustic Pressure



- Define the desired receiver sampling frequency
- Define a highly-oversampled domain to create a dense grid
- Interpolate the transmitted samples over this dense grid (zero, first, or n-order hold)

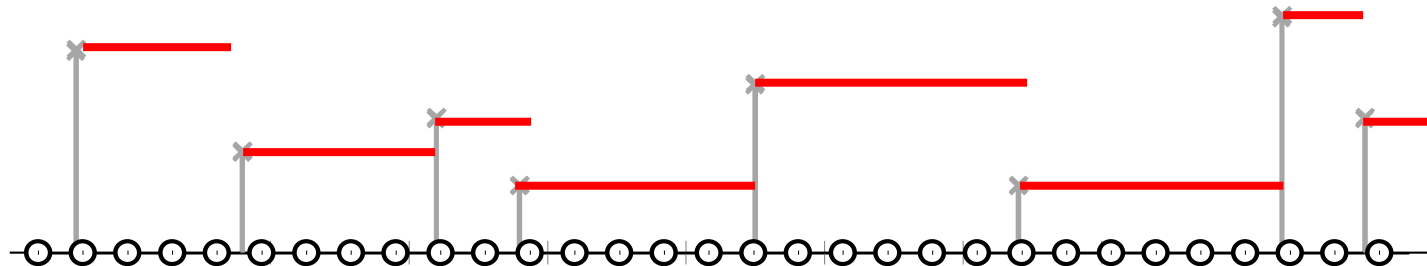


Our proposal: NoTAP

N-oversampled Transmitted Acoustic Pressure



- We can't use zero-insertion upsampling here due to amplitude scaling problems
- But we don't have to use very complex interpolation methods
- **Zero-order Hold → Create a step function**; First-order hold → Linear interpolation

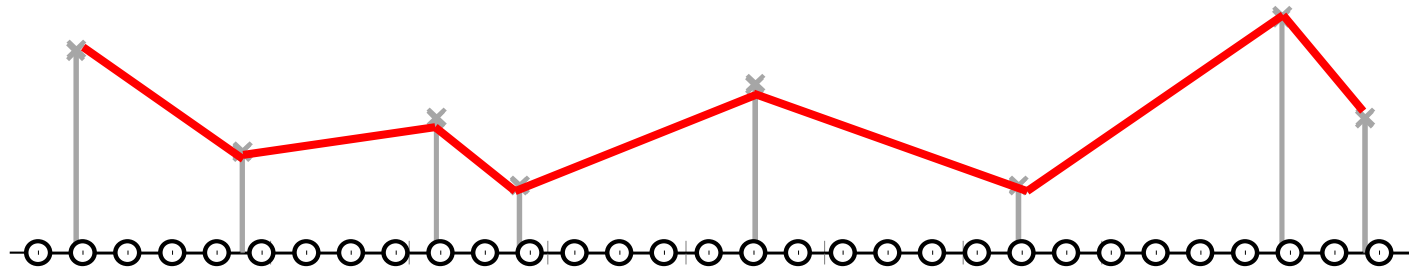


Our proposal: NoTAP

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Our proposal: NoTAP

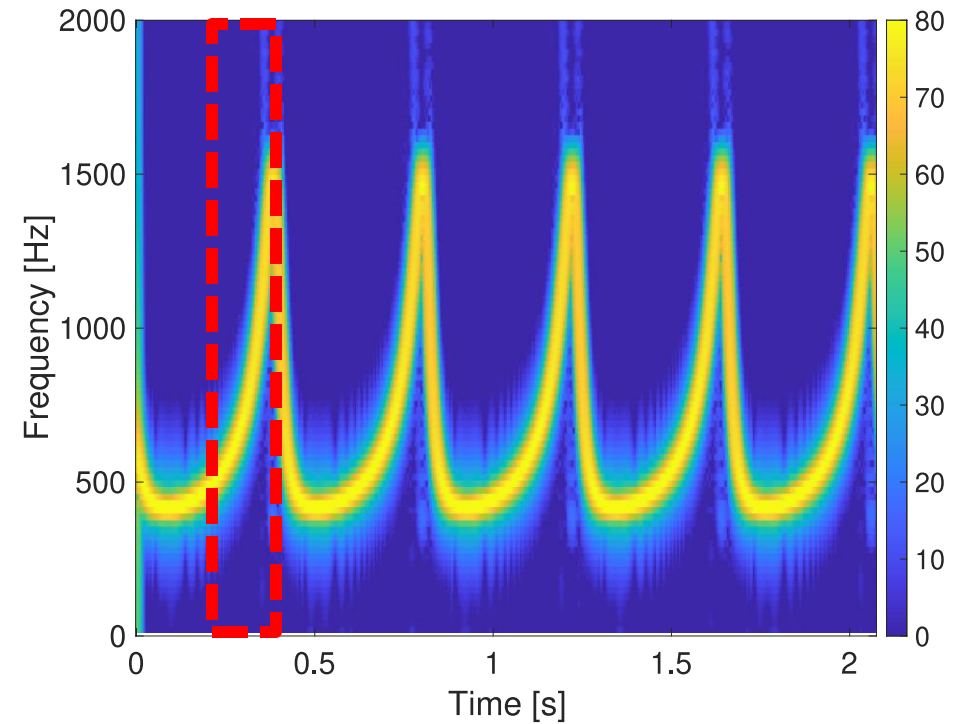
N-oversampled Transmitted Acoustic Pressure



- Lowpass filter and downsample to your desired receiver sampling frequency
 - This can be done with successive IIR decimations – a very fast process
- This idea of zero-order hold and then filter has been used for a long time in digital to analog conversion.
- In the example we will also upsample back to 44100 Hz in order to listen to the result...

Applying NoTAP to the Contrivance

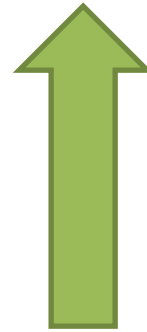
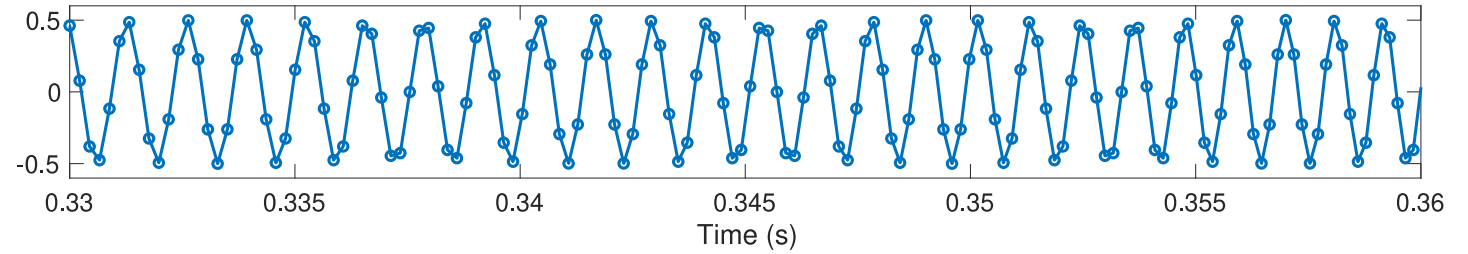
- Recall the full spectrum of the signal when $f_s = 8k$.
- Let's first focus on the section in time where the frequency goes from below 1k to above 1k (i.e., where aliasing would occur).



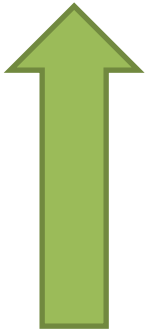
Applying NoTAP



Transmitted samples – irregularly spaced



Lower Frequency

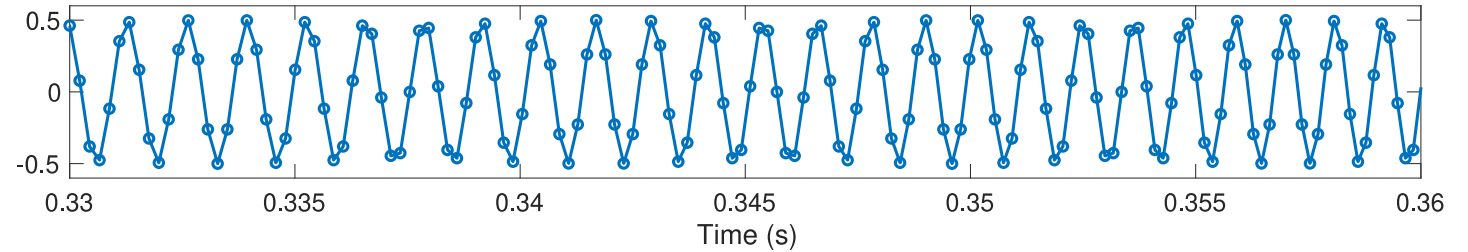


Higher Frequency

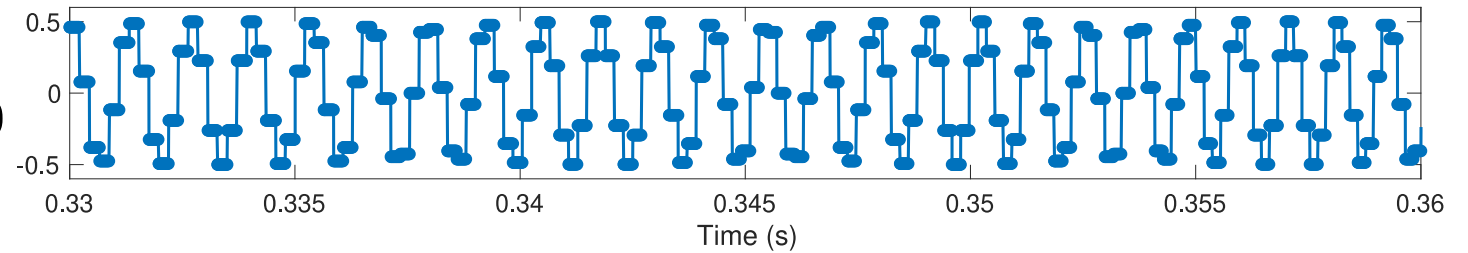
Applying NoTAP



Transmitted samples – irregularly spaced



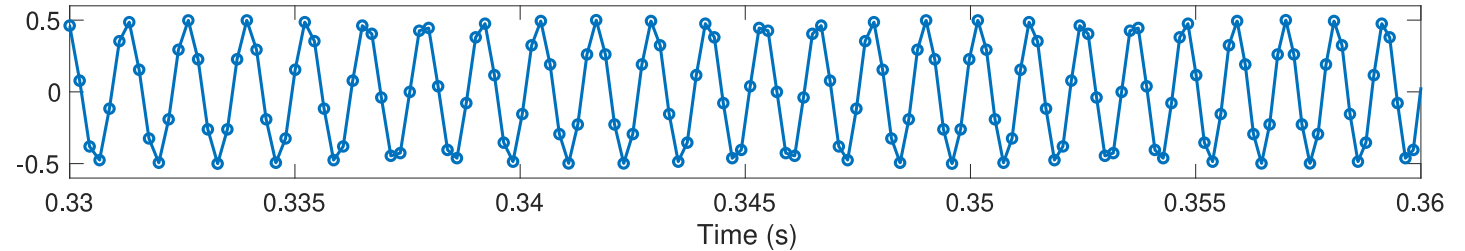
Zero-order hold interpolation over the newly defined oversampled grid for $N = 400$



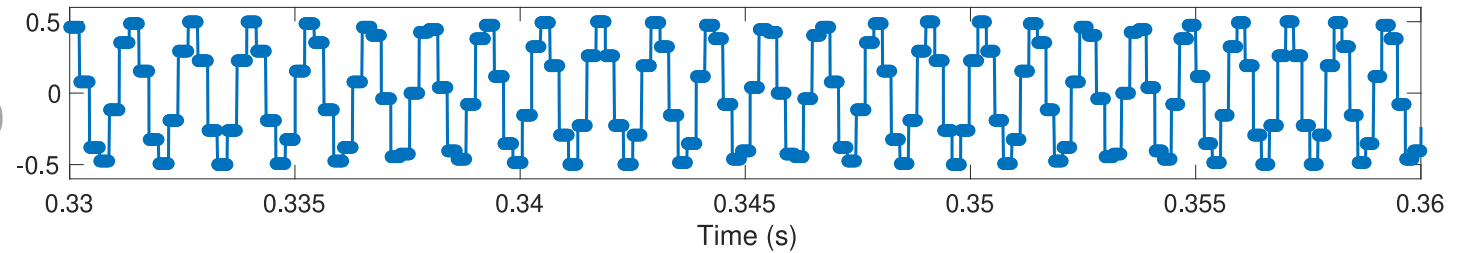
Applying the NoTAP to the previous scenario



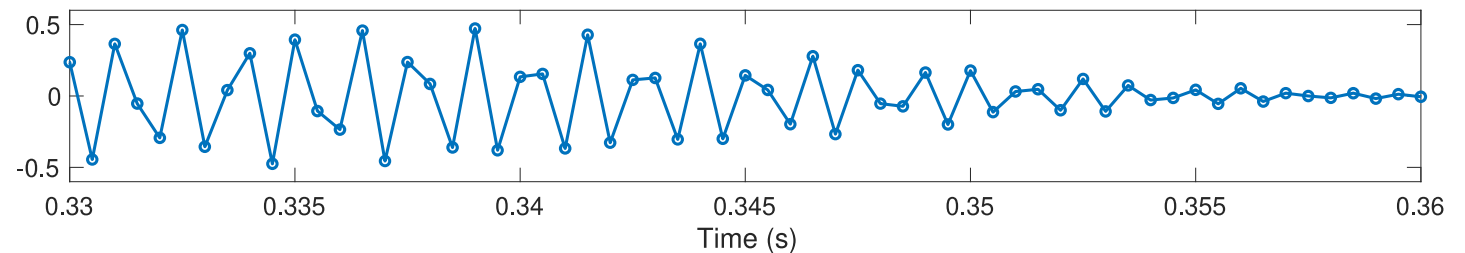
Transmitted samples – irregularly spaced



Zero-order hold interpolation over the newly defined oversampled grid for $N = 400$



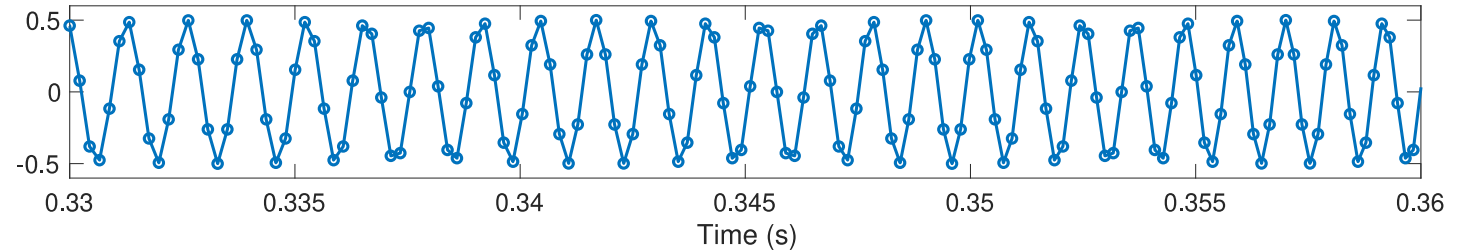
Low-pass filtered and downsampled to $f_s = 2$ kHz



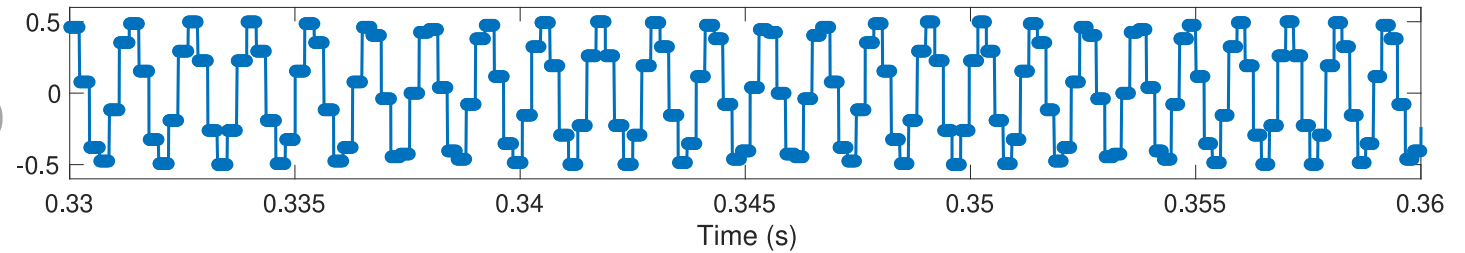
Applying the NoTAP to the previous scenario



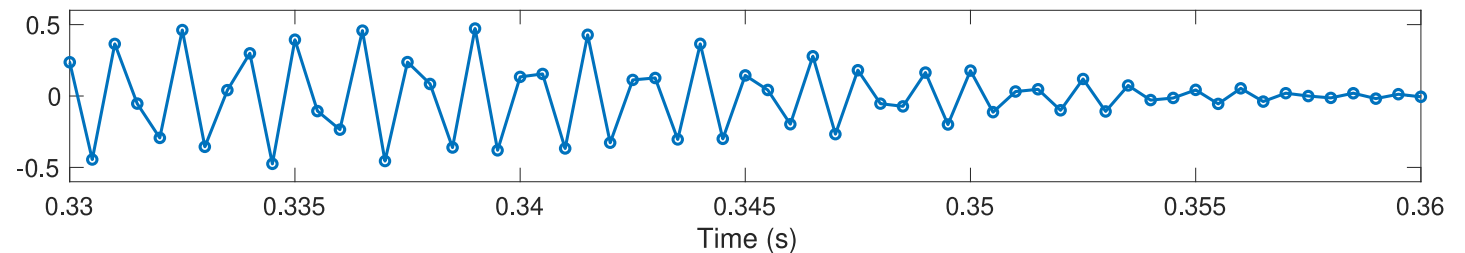
Transmitted samples – irregularly spaced



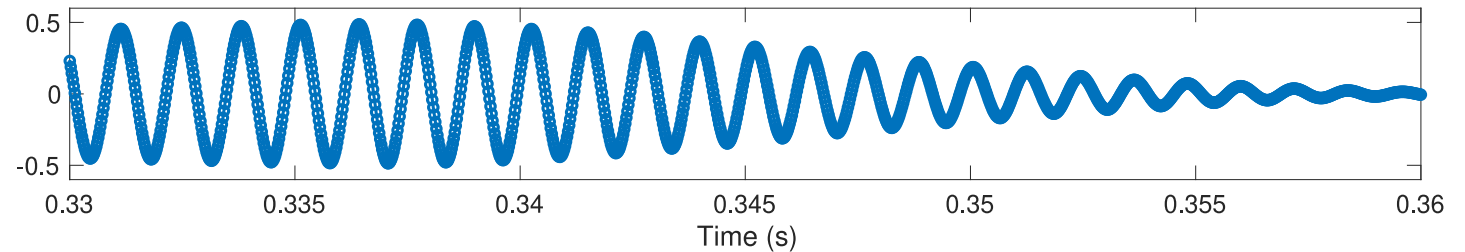
Zero-order hold interpolation over the newly defined oversampled grid for $N = 400$



Low-pass filtered and downsampled to $f_s = 2$ kHz



Upsampled to $f_s = 44.1$ kHz for listening.
Effect of anti-aliasing filter evident.

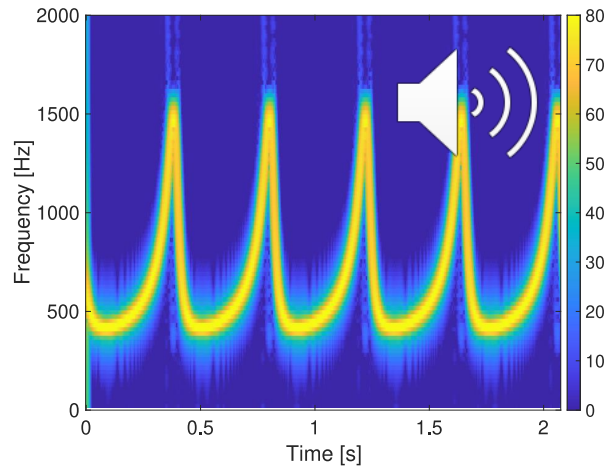


Some Results

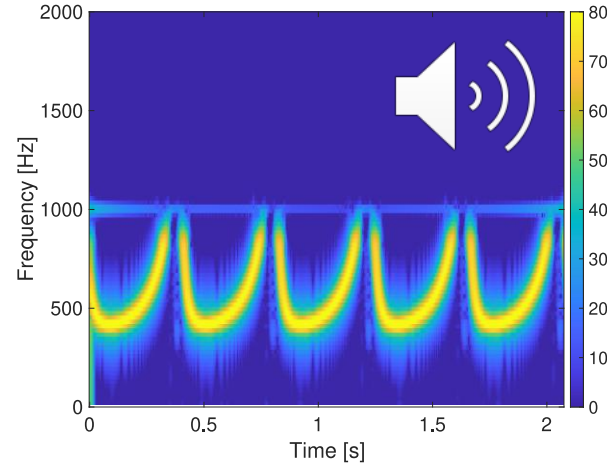


Conventional
Methods

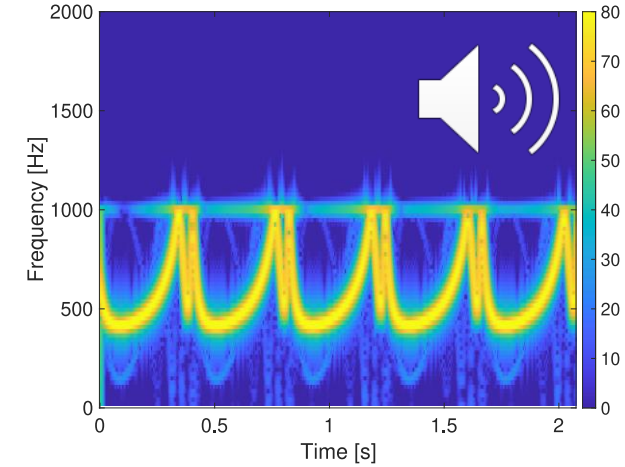
Receiver $f_s = 8$ kHz



Decimated to $f_s = 2$ kHz

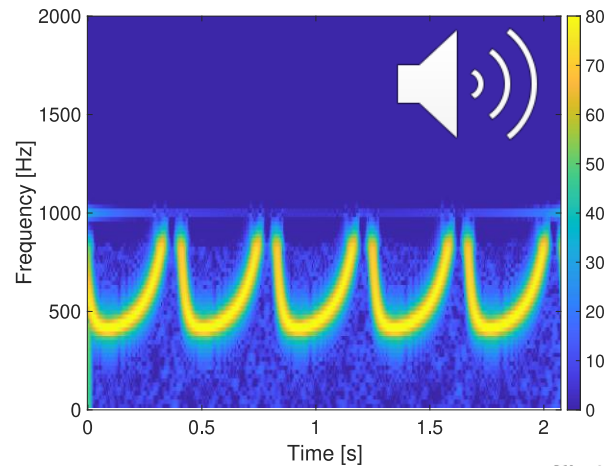


Receiver $f_s = 2$ kHz

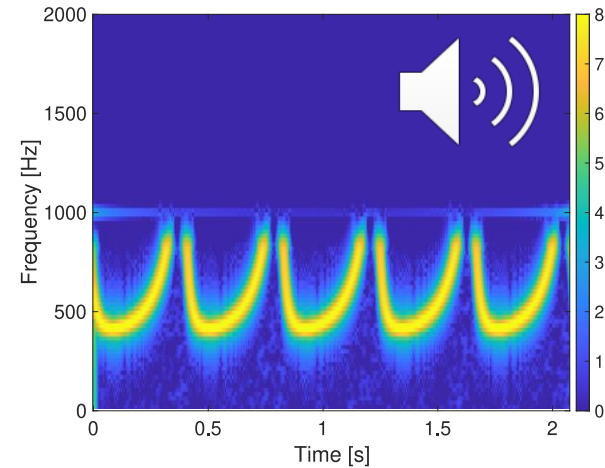


NoTAP
Method

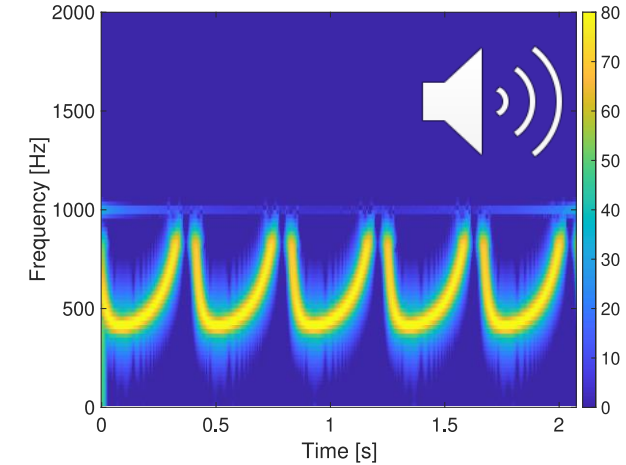
Zero-Order Hold ($N = 400$)



Zero-Order Hold ($N = 800$)



First-Order Hold ($N = 400$)



Receiver $f_s = 2$ kHz

Colorbar: dB re 20 μ Pa

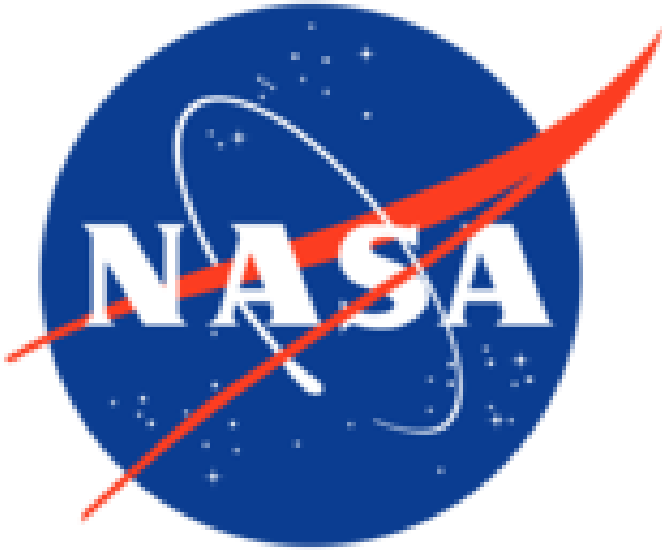
Christian, Ali, Greenwood; ASA Spring 2022, 3pSP7



Conclusions

- The benefits of not tapping:
 - The basis of this approach is fundamentally distinct from extant auralization processing methods, and so has properties that set it apart in strategic ways:
 - Built in antialiasing properties.
 - Very fast in general, especially wrt. receiver-time dominant approaches.
 - It does not require transmission of information from the receiver back to the source.
 - The receiver can be (almost completely) ignorant of the properties of the source/transmission path.
 - This can run in truly real time (for zero-order hold) – processing samples from the source as they arrive.
- This presentation has covered the basics but has not gone into enough of the details to make a usable implementation.
 - One needs to track the “effective” received sampling frequency to avoid inducing distortions...

Questions?



Andrew Christian – andrew.christian@nasa.gov

Randall Ali – randall.ali@esat.kuleuven.be

Eric Greenwood – eric.greenwood@psu.edu