2Hz

Real Time Voice Processing SDK

Our Voice Processing SDK provides best in class, AI powered **Noise Suppression** and **HD Voice** capabilities for Real Time Voice Communications in a form of **Media Server Plugin**. The SDK plugs into Media Server (e.g. Asterisk), expects Voice Audio Stream as input and returns enhanced Audio Stream in Real Time.

Noise Suppression

- Noise reduction of up to 43 dBs
- No impact on Human Voice
- Language Independent
- Advanced Voice Activity Detection
- Supports Many Noise Types:
  - Street, Crowd, Restaurant, Babble, Siren, In-Car, Stationary, Factory, Music, etc
- Supports dynamic range of audio sampling rates, 8kHz-44.1kHz

HD Voice

- Real Time Narrowband to Wideband expansion with Voice enhancement powered by Machine Learning
- From 8kHz to up to 44.1kHz

Media Servers

- Asterisk 13+, FreeSwitch 1.6+

Hardware Needs

- Ubuntu 16+ or Centos 7
- Multi-CPU or GPU
- 8GB RAM minimum
- 1GB disk

Performance

- 10 concurrent streams per 1 CPU
- 500 concurrent streams per 1 GPU
- Processing Latency - 1ms
- End to End Voice Latency - 10ms

Licensing

- Flexible licensing model

Storage

- No audio or user data is stored within the server

Contact us for more information on features and pricing

contacts@2hz.ai | +1 4084390019 | www.2hz.ai
2150 Shattuck Ave Penthouse 1300, Berkeley, CA 94704