ReconVox

Product Information

ReconVox is our high performance **Continuous Speech Recognition** product. Thanks to its ability to work both in ***Word-Spotting*** or continuous speech mode with custom language models and grammars than can be created from scratch, it fits into a wide range of applications, from controlling electronic devices with voice commands to interacting with automatic call-centers using natural language or getting the full transcription of a radio broadcast.

ReconVox is designed to be easily integrated in almost any operating environment because it's available as a ***SDK*** *(Software Development Kit)* which delivers all its functionality via a complete ***API*** (*Application Programming Interface*).

Thanks to its efficiency and flexibility, a wide range of **applications** can be delivered:

- **IVR** (Interactive Voice Response): conversations close to natural language in automatic call-centers.
- **Automatic search by content**: spotting **keywords** or sentences in audio/video recordings or live streams.
- **Automatic transcription** services from radio broadcasts, trials, meetings…
- **Alarms and domotics**: electronic devices controlled by voice commands, (de)activation of alarms…
- **Voice commands in cars**: GPS, hands-free phone calls…

- **Media clipping** and **automatic tagging of contents.**
- **Education**: scoring of **pronunciation** for every single word in language learning or in some speech pathologies like dyslexia or aphasia.

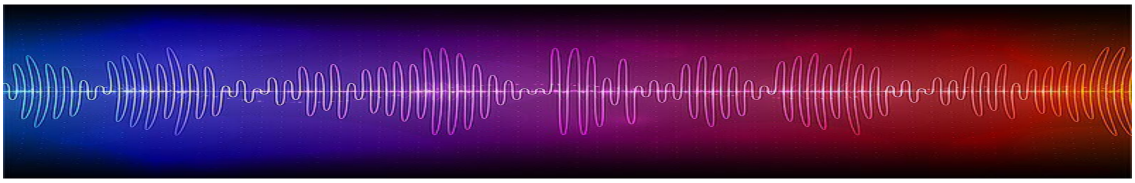
In addition, if security is a factor for the application, a ***ReconVox*** based Speech Recognition system can work together with our Voice Biometrics technology, ***BioVox***. This way, it's possible to perform continuous speaker authentication along all the interaction of the user with the system, always in the background and in transparent fashion, for example for secure transactions by phone in banking applications.

PRODUCT

- Speaker Independent Continuous Speech Recognition system.

KEY FEATURES

- Recognition task can be fine tuned: **isolated words** or **continuous speech**.
- **Speaker independent**: doesn't need to be trained for a specific speaker.
- **AutoLearn**: automatic adaptation on the fly for a specific speaker, dialectic region or noisy environment.
- **WordSpotting**: detection of keywords or short sentences within unrestricted audio.
- **Vocabulary can be customized**: from a few commands to thousands of words.
- **Grammar and language models can be customized from scratch**. Two different types of language models: **rules based grammars** vs **flexible, open language models**.
- Currently available in **Spanish, English** and **Korean** languages. New languages in preparation.
- Acoustic models available in **8 KHz** (phone channel) and **16 KHz** (PC, radio, domotics, apps).
- **Hardware efficient** recognition engine: can be integrated into embedded systems.



TECHNICAL SPECIFICATIONS

- Speech signal preprocessing: automatic activity detection and signal filtering.
- Recognition speed in continuous speech mode[1]: 2x – 4x faster than real time (typical, depends of vocabulary size).
- Recognition speed in Word-Spotting mode[1]: 7x faster than real time.
- Supported audio formats: PCM linear 16 bits 8/16 KHZ (recommended), A-Law, µ-Law, MP3.
- Memory requirements: 3 MB (engine) + 9 MB (per language) + 20 MB *AutoLearn*.
- Disk space: 5 MB / language.
- Minimum recommended CPU: Intel i5, 2.5 GHz w/4 CPU cores or equivalent.

SUPPORTED PLATFORMS

- Windows® 7, 8, 10.
- Linux, several distributions.

1 With minimum recommended CPU.

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