



"Nakshatra" #90-91 AECS 2nd Stage, Opp. Ramana Maharshi
Heritage Center, RMV 2nd PO, Nagashettyhalli,
Bangalore -- 560094

Ph: +91 80 57672396, +91 80 23517063
<http://www.esqube.com>

PRODUCTS BROCHURE

1. Voice-ID Engine

Applications:

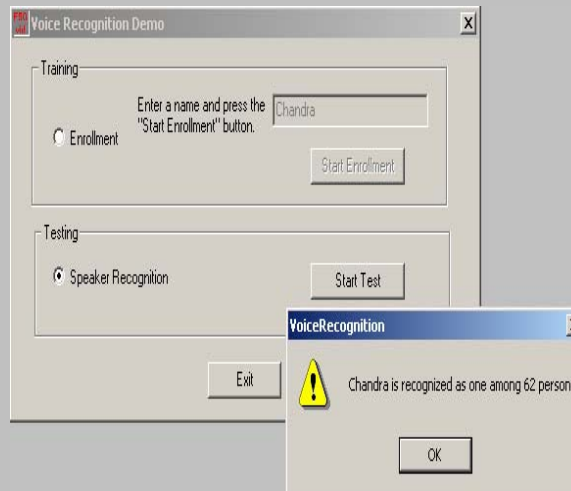
Voice-ID is a biometric useful in many user authentication systems, such as premises entry/log, web data access, banking, tele-check etc. Unlike several other biometrics, such as finger print, Voice-ID is non-invasive and user friendly.

Features:

1. Text independent.
2. Language independent.
3. Minimum enrolment speech (60 sec) and test speech (5 sec).
4. 99% correct recognition for population of 50-100 speakers.

Demo software is available for trial enrollment. Enrolment accepts the name of the person (without spaces) and the speech data for about 60 seconds.

After enrolment, the speaker can test his/her voice by selecting the speaker recognition button and inputting a test voice of any text in any language in 5sec.



2. Telephony Voice Dialer

Esqube Voice dialer is an isolated word speech recognition system which can render a PABX to be intelligent. Each user can maintain a personal phone book and dial-out numbers by own voice without having to struggle with the keys of telephone; this is particularly useful in a poor light or cordless phone type of usage.

Features:

1. Speaker dependent isolated word recognizer based on acoustic sub word models.
2. Voice tag rare independent of language, accent and robust to mood and other artifacts of a speaker, unlike the typical cell phone solutions.
3. Supports single and multi-channel recognition.
4. Unique feature of speaker aliasing makes the enrolment process simple.
5. Less memory requirements.
6. Robust to environment noise.

3. Language Identification

The language identification solution is to recognize a language, independent of what has been spoken and who has spoken. This is a proprietary solution that can be realized for any set of languages based on training speech data, without any knowledge of the linguistics of different languages.

Features:

1. Number of different languages: 5-10.
2. Training voice data: 30 sec/spkr, 10-20 spkrs.
3. Test voice data: 5-10 sec.

4. Variable Rate Speech Coder

This is a proprietary speech coding algorithm based on switched CELP technology. The coder is variable rate adapted to the voice source (not channel as in AMR) to provide an optimum performance in packet communication networks such as VoIP.

Channel based adaptation is also included for a set of average bitrates, viz., 4kbps, 3kbps, 2kbps where the speech perceptual quality is optimized for each bitrate.

Variable bitrate and channel adaptation is well suited for congestion control in VoIP, i.e., 25% source bitrate reduction is far better than even 5% random packet loss.

5. Speech Enhancement

This is a proprietary solution for improving the listenability of noisy speech; both stationary and time-varying noise is addressed. The speech enhancement algorithm ensures that the information content of speech is retained maximally, unlike other algorithms in the literature.

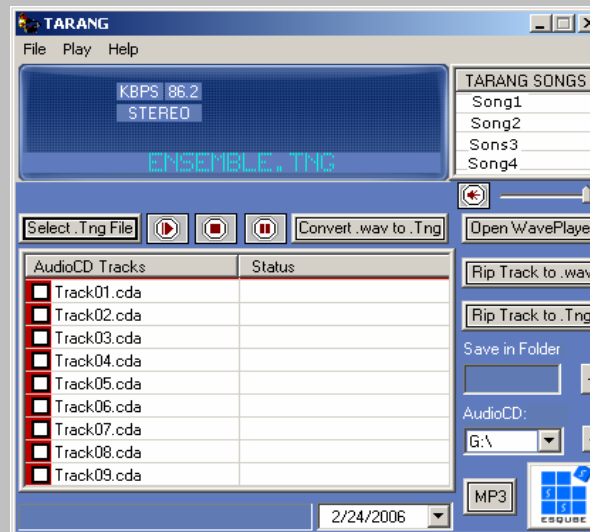
6. TARANG Audio Compression

ESQUBE audio compression technology aims at solving one of the toughest problems that the music industry is facing, that of music piracy. Music files in MP3 format have become very popular and are freely made available on the internet. As a result, though the consumer gains by way of acquiring 'free music', the music distributors are at a big loss! TARANG audio compression helps to solve this problem because it is a proprietary solution. The music encoded by TARANG can be decoded only by TARANG decoder. A consumer pays a nominal fee for each song or album encoded through TARANG and hence owns music legally and can enjoy high quality digital music (success of iTunes). A similar approach can be used for distributing any other audio content such as lectures/discourses etc. The use of TARANG eliminates the need for expensive Digital Rights Management software (saves licensing fee, cost etc.)

Features:

1. Low complexity encoding and decoding compared to MP3 or AAC.
2. Amenability to mixed signal processing in hardware

3. Compression factor: For CD quality music ~ 6-8:1 compression with near indistinguishable quality of signal reconstruction.
4. Proprietary algorithm not available publicly.
5. User-specified custom metadata integration into the audio bit stream.



QTARANG: QTARANG adds a new feature to TARANG. Pleasant sounding stereo sound is synthesized at the decoder during the playback process. In terms of complexity and speed, QTARANG is twice as fast as TARANG. Compression factor: ~12-16.

7. QStereo Surround

ESQUBE surround is an integration of TARANG, QTARANG and a proprietary multi-channel surround sound production technique. The TARANG format audio can be connected to a multi-channel surround sound system such as home-theatre playback systems, for a rich listening experience. For headphone users, we use yet another novel technique for providing the rich multi-channel listening experience on headphones. In usual headphone listening, the listener may experience fatigue over long duration of listening because the sound appears to come from inside his head. We process the signal in such a way that the sound is

perceived to be coming from outside of the head and that the user is immersed in a sound field. This gives a richer and enjoyable listening experience to the listener.

8. ESQUBE Music streaming solution

This is a new proprietary software streaming solution using the TARANG audio coding technology. The technology is potentially useful for internet or mobile music distribution.

