Speech Technology Center

Voice of the Future

Complete Solutions

PROFESSIONAL CALL RECORDING
NOISE CANCELLATION
POLICE AUDIO EQUIPMENT
DISTRIBUTED TRANSCRIPTION

Speech Technology Center Limited (STC) develops, manufactures and markets software and hardware solutions, standalone devices, DSP boards and software toolkits.
STC's main areas of activities are:

- professional digital voice recording;
- speech processing and enhancement;
- audio signal analysis;
- voice and music recognition;
- speaker verification and identification.

Founded in 1990, STC has an excellent reputation in the high-tech market, designing algorithms for high performance speech processing applications. STC provides end-user solutions as well as SDK for PC and DSP platforms.

STC scientific team focuses on developing unique know-how in the field of speech/signal technology. All STC products are based on proprietary patented technologies. Speech Technology Center offers solutions primarily for the following customer groups:

- Security, police and law enforcement;
- Government;
- Telecommunication and call-centers;
- Parliaments and congress halls;
- Financial institutions;
- Broadcasting.

STC develops internationally each year, and we are looking forward to doing mutually beneficial business.

PROFESSIONAL DIGITAL VOICE RECORDING

SMART LOGGER

Multi-channel call recording and monitoring system.

Smart Logger records voice conversations from analog and digital telephone lines and other sources, and gives instant and comfortable access to recorded data. The operator interface is designed to be very simple and to avoid additional staff training expense.
The recording quality is extremely high, which allows records to be used for speaker verification and identification and to achieve better results in case of further noise filtering.
Smart Logger is available as a scalable distributed Network system for call-centers or monitoring centers and as a portable system for field applications.
Smart Logger is featured with: Fax transmission decoding, Noise filtering, Speech transcription.

Smart Logger is vital for any of the following:
Police and other Law Enforcement Agencies: Legal recording and monitoring of conversations with mobile or fixed monitoring centers.
Emergency services (fire, ambulance, rescue): Recording of incoming voice messages for effective follow up of accidents.
Dispatcher sites of organizations with a dependency on verbal communications (power, telephone, oil, water, gas suppliers): Recording of messages of internal and external communication for subsequent control and possible investigations.
Call-centers of financial institutions (securities traders, banking, and insurance): Recording of important verbal communications for financial safety purposes.
Call-centers of any organizations: Recording of conversations for safety reasons (e.g. against terrorist calls), tracking customer issues and for staff performance control.

Highlights:
- Interface with all major telephone systems including analog, ISDN S0, E1 trunks, 4- and 2- wire digital extensions of PBXes, microphones.
- Fax transmission decoding with optional Fax Reader module supporting a range of protocols (V.27, V.29, V.17 and some non-standard ones).
- Rich database functionality with a number of fields, filtering, sorting and export. The database is protected with a 4-level access restriction.
- Simultaneous recording and playback. Real-time audio channel monitoring.
- Embedded transcription module can be used to transcribe recorded conversations into text for further use.
- Possible bit rates are: 128, 64, 32, 16 Kbit/s modes. SNR (signal-to-noise ratio) at 128 Kbit/s is more than 75 dB.
- Recognizes DTMF codes and other service signals, including pulse-dialled numbers and CallerID. Activation triggers include intelligent voice activation, on/off hook detection, and activation by external sensor.
- Reduces the amount of noise in recorded conversations making the speech clearer and more intelligible for the listener.
- Smart Logger is available in several languages and can be easily localized for others.

PROFESSIONAL DIGITAL VOICE RECORDING

GNOME-2

Professional pocket digital stereo voice recorder.

Gnome-2 is designed to achieve the best performance possible when recording in real-life noisy environments. It records conversations with extremely high quality with or without compression. Equipped with two high-quality internal microphones or small size external microphones, Gnome-2 provides rich functionality with lots of settings available via an easy-to-use interface.
Gnome-2 has an 8-digit PIN-code access restriction and strong audio data encryption. Rugged metal case insures durability of the unit as well as protects it againts detection and jamming.

Gnome-2 can be used by any organization needing high quality mobile digital stereo recording, in particular such as:
Police and other Law Enforcement Agencies: Recording during special operations and inside police patrol cars.
Security companies: Recording inside cash-in-transit vans and other recording applications for safety reasons.
Emergency services (fire, ambulance, rescue): Rescue operations tracking and recording inside rescue vehicles.
Legal practices and institutions: Recording of important conversations.

Highlights:
- High recording quality ensured by a 24 bit ADC (analog-to-digital converter) and highly sensitive microphones.
- SNR (signal-to-noise ratio) is more than 72 dB, which allows Gnome-2's recordings to be used for forensic examinations.
- Stores data on Compact Flash memory cards. Easy and fast PC download using a CF adaptor.
- Recording time up to 23 hours for a 512 MB card, depending on the recording mode.
- Gnome-2 software for MS Windows (9x/ME/NT/2000/XP) allows setting of data protection and PIN codes.
- Recording can be activated by voice, manually or with 5 embedded real-time timers. Voice activation is easily adjusted by means of graphical user interface displaying current signal levels. It is possible to set different threshold levels for every channel.
- Slim and strong metal case of 115 x 55 x 15 mm. The weight is 132 g.

NOISE CANCELLATION AND SPEECH ENHANCEMENT

Compact mobile device ANF STC-H156 and Sound Cleaner Premium software are ideal for anyone interested in sound improvement and noise reduction but particularly for:
Police, Forensic labs and other Law Enforcement Agencies: Processing records for further use in crime investigation. Live sound quality improvement when recording or listening in a field environment.
Audio Engineers : Preparing sound for CD mastering & broadcasting. Mobile processing of recorded audio data.
Broadcasting Companies : Mastering of “live” interviews and reports for air. Live speech quality improvement when recording in field conditions.
Universities and Audio Laboratories : Professional speech enhancement.

ANF (AUTOMATIC NOISE FILTER) STC-H156

Professional digital noise filtering device.

ANF is a compact mobile device, which filters noise enabling the listener to hear speech clearly. It is essential for live processing in field conditions. ANF has 7 types of automatically adapting noise filters, the settings of the filters are adjustable via LCD screen and keyboard.

Highlights:

- On-the-fly adjustment of noise filtering parameters during signal input and playback;
- Available filters:
  - Single channel adaptive, distortion compensation;
  - Dual channel reference adaptive, distortion compensation;
  - Adaptive inverse filtering;
  - Harmonics (tone) filtering;
  - Adaptive broadband filtering;
  - Composite (inverse and broadband adaptive) filtering;
  - Pulse-like filtering.
- Additional processing types include:
  - Harmonics suppression;
  - Timbre correction;
  - Pseudo-stereo;
  - Loop playback from internal memory of 60 sec of audio.
- Effective against the following noise sources: communication channel interference, office equipment, industrial and vehicle engines, street traffic, environmental noise, slowly varying music, hiss and rumbling, reverberations and echo effects.
- Mobile and robust construction makes ANF ideal for use by military or law enforcement agencies. It can be powered by standard power supply, car or other external battery.

SOUND CLEANER PREMIUM

Professional real-time, noise cancellation and speech enhancement software.

Sound Cleaner is a unique software solution for filtering of noise-corrupted sound recordings and improvement of speech intelligibility. The user can adjust filtering parameters and hear the resulting changes in real time.

Highlights:

- The signal is processed by a sequence of modules. There are 14 typical schemes of processing available; you may also build customized noise filtering schemes.All modules work simultaneously, and all options are adjustable in real time while listening to the output.Each module is represented by a separate window with a number of basic and professional controls.

- Sound Cleaner consists of 19 different processing modules including:
  - 8192 independent band graphical equalizer with advanced inverse spectrum calculation capabilities and real-time spectrum visualisation;
  - 2048 manually adjustable equalizer sliders, it is also possible to operate sliders by group;
  - adaptive broadband noise filter;
  - adaptive frequency compensation filter;
  - 4 adaptive stereo filters in time and frequency domain;
  - adaptive inverse filter;
  - pulse-like interference filter;
  - dynamic processing automatic gain control module;
  - waveform visualisation module;
  - playback speed alteration module.
- Improves SNR (signal-to-noise ratio) up to 50 dB for tonal noise, up to 20 dB for broadband noise.
- Effective for both restoration and enhancement of poor quality recordings, recordings made in noisy environments and real-time sound loaded from external sources.
- Sound Cleaner can generate a report of modules and settings used to process a particular recording.

IKAR LAB

Professional hardware and software set for advanced audio/speech signal analysis.

IKAR Lab is an excellent environment for speech signal analysis. It consists of visualization, analysis, processing, speech enhancement and editing tools. IIKAR Lab was designed especially for the extraction and in-depth analysis of voice parameters.

IKAR LAB is widely used in:
Forensic Audio Laboratories: IKAR Lab is used for speech analysis, speech enhancement, speaker identification, identifying traces of editing, audio equipment and tape authentication.
Scientific Audio Research: IKAR Lab is a universal toolkit for speech signal analysis in speech research, audio sciences and acoustics.
Medical Applications: Speech and hearing diagnostics.
Equipment Certification Labs: Precise measurement of technical characteristics of electronic audio equipment.

Highlights:

- Powerful visualisation tools – allowing viewing the audio signal in the tiniest detail and comparing audio signals. Sophisticated speech signal analysis with rich options:
    - Simultaneous access to different analysis types.
    - Pre-set standard options lists as well as manual/automatic fine tuning.
    - Refinement of spectral parameters by overlaying formant tracks on the spectrogram.
    - Refinement of pitch curves through direct overlay of the pitch pattern on the cepstrogram.
- Numerous editing tools:
    - Load as many signals into one window and open as many windows as needed.
    - Manual redraw, linear and non-linear transformations.
- Powerful environment noise cancellation and speech signal enhancement:
    - Adaptive filters in time and frequency domain.
    - Precise digital equalizer with automatic and manual adjustment.
- Audio equipment and media authentication procedures.
- Testing of electronic audio equipment.

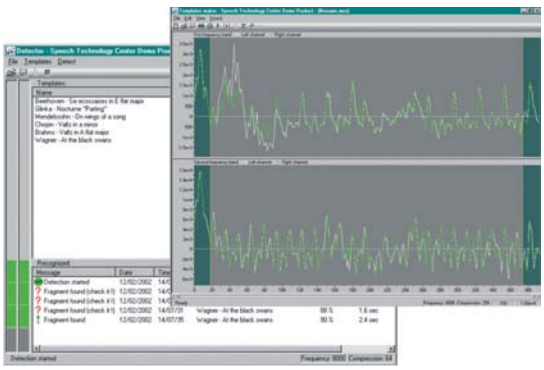
Training: STC experts provide basic and advanced training in speech enhancement, audio analysis, speaker identification and other forensic audio subjects. Consultancy and training can be tailored to customer needs.

JINGLE TRACKER

Software for identification of music samples and advertising jingles in live sound streams.

Jingle Tracker captures how many times and when exactly jingles or music samples are played on a given broadcasting channel. It is based on STC's Music Spotting Technology.

Jingle Tracker is used by:
Advertising Agencies: Captures time of broadcast of commercial advertisement.
Broadcasting Companies: Captures and logs the time music or jingles were played.
Music Copyright Owners: Checks and logs the occurrences and time a music jingle was broadcasted.



Highlights:

- The software puts music sample templates into a database and then searches the inputted sound stream and logs the occurrence of identical music samples.
- Large amounts of sound data can be processed without user intervention.
- Low resource consumption. Can be used on standard personal computers.
- Adjustable to any specific application. Depending on the application, the false acceptance/false rejection balance can be adjusted.
- High accuracy on various kinds of music and jingles.
- Adjustable speed & accuracy.

NESTOR

Distributed Speech Transcription System.

Nestor is a Network based distributed transcription and documenting system, designed to make transcription faster, more efficient and of better quality. It provides automated workflow to multiple transcriptionists for fast documenting of large volume of speech. Sound recorded by recording stations and stored on a separate sound server is subsequently automatically distributed to operators. Each operator then types the transcription of one short sound fragment. All fragments of sound and text are connected into the whole document. The whole text including linked sound is available at the editor's workstation to check, correct, print and archive.

Nestor is used by:
Parliaments, Congress Halls, Documentation Departments: Transcription of speeches, meetings, negotiations.
Journalists and News Agencies: Transcription of interviews.
Academic Institutions: Transcription of lectures, congresses, workshops.

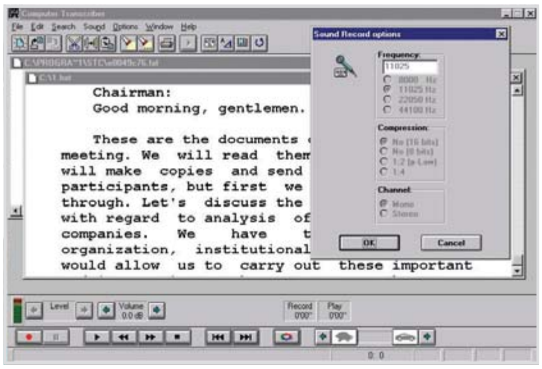
Highlights:
- All the workflow is automatically controlled by Nestor and managed from the editor's and administrator's workstations.
- Workplaces of transcriptionists are optimized for maximum comfort and efficiency. The operators enjoy various time- and labour-saving features like Sound Stretcher and frequency adjustment.
- Nestor is fully scalable for large implementations.
- Fast transcription - full text of an event can be obtained in several minutes after the event is finished.
- Information protection - operators can hear only small portions of speech and don't have access to the whole information.
- Time marks allow checking of transcription quality and make correction an easy task.
- Nestor has performance management, statistics and archiving features.
- Nestor's workflow engine can process speech coming from several channels simultaneously by different groups of transcriptionists in several different languages. Nestor can be easily localized for any language.

CAESAR

Transcription software.

Caesar is a PC application that greatly increases efficiency of speech transcription.
It combines the functionality of a text editor with audio playback and recording.
Caesar saves time and effort to any organization for production of transcription.

Caesar is widely used by:
Documentation departments, assistants and secretaries: transcription of speeches, lectures, meetings, negotiations, memos.
Police, security authorities, private security organizations: Interview transcription.
Journalists and news agencies: Interview and speeches transcription.
Courts and Legal institutions: Transcription of court sessions.
Medical organizations: Transcription of medical notes.



Highlights:
- Full control over sound playback by hot keys.
- STC's Sound Stretcher technology is used to change speech playback speed without distorting voice pitch. Speech rate acceleration/slow-down is up to 3 times from the original. Smooth rate adjustment.
- Loop mode allows working with hard to understand speech fragments.
- Caesar automatically links speech and text providing an easy way to find corresponding speech fragments.
- Substantial improvement in the work rate even for experienced transcriptionists.

Speech Technology Center

www.speechpro.com

Headquarters: 4 Krasutskogo, St.Petersburg 196084, Russia
Phone: +7 812 3310665, +7 812 3258848, **Fax**: +7 812 3279297
e-mail: info@speechpro.com

European office: Speech Tech GmbH
Saargemünder str. 211, Saarbrücken 66119, Germany
Phone: +49 681 9655709, **Voicemail/fax**: +49 69 25577077
e-mail: info@speech-tek.com
www.speech-tek.com