Voice Communication Package v7.0

of front-end voice processing software technologies

General description and technical specification

(Revision 1.0, May 2012)

General VCP information

Voice Communication Package (VCP) is a suit of digital signal processing technologies enabling high quality voice communication for a variety of applications including automotive hands-free, mobile phones, Bluetooth headsets, audio and video conferencing systems, intercom systems and others. VCP was designed to enable the highest possible voice quality in various acoustic environments while consuming relatively low MIPS and memory resources.

VCP provides natural support for narrowband (8KHz) and wideband (16KHz) speech. For wideband speech VCP implements special mechanisms providing significant reduction of MIPS and memory requirements compared to doubling them with the brute-force approach without a noticeable degradation of output speech quality.

The current, 7th VCP generation accumulates many years of practical Alango experience providing a scalable, highly optimized solution for voice communication applications. Besides the software DSP technologies Alango has developed an unique set of auxiliary software and hardware tools facilitating development, debugging, testing, acoustic tuning, problem identification and reporting.

VCP components and structure

VCP structure and signal flow is shown in Figure 1 .

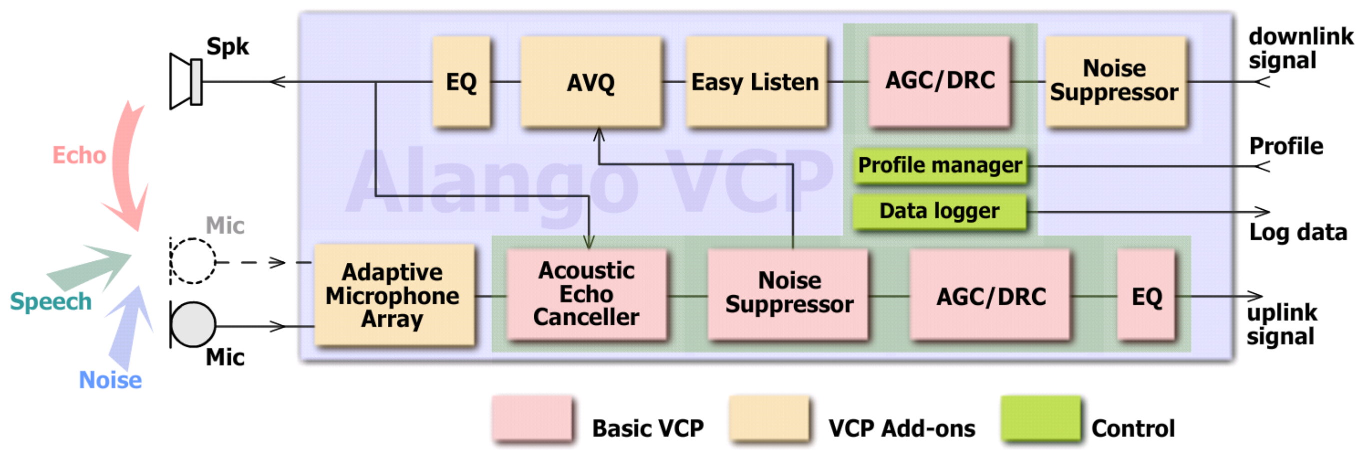

Figure 1 VCP structure and signal flow

VCP consists of basic, control and add-on blocks. Basic VCP comprises technologies that are "must have" for virtually any full-duplex, hands-free voice communication system. It also includes control blocks adding flexibility and ease of system debugging, testing and acoustic tuning. Add-on blocks include technologies providing additional benefits to end users.

Basic VCP includes the following signal processing blocks:

- Uplink: Acoustic Echo Canceller – cancels acoustic echoes caused by speaker-microphone acoustic coupling enabling full-duplex communication
- Uplink: Noise Suppressor – reduces stationary, transient and tonal noises in the microphone signal
- Uplink: Automatic Gain Control and Dynamic Range Compressor – compensates for rapidly changing distances between the user and the microphone
- Uplink: Multiband equalizer – allows precise adjustment of microphone frequency response
- Downlink: Automatic Gain Control and Dynamic Range Compressor – compensates different network signal levels and other

The following advanced VCP blocks may be included:

- Uplink: Adaptive Microphone Array – reduces all types of ambient noises (including wind noise)
- Downlink: Noise Suppressor – reduces stationary, transient and tonal noises in the downlink signal
- Downlink: Automatic Volume & Equalization control - amplifies and equalizes the loudspeaker signal according to the ambient noise providing perceptually equal loudness and intelligibility in variable conditions
- Downlink: Multiband Equalizer – allows precise adjustment of loudspeaker, communication channel or room frequency response
- Downlink: EasyListen – slows down the incoming speech in real time improving intelligibility in noisy environments, in mobile conditions or when having a conversation in foreign language

On the customer request the package can be easily configured to include only those features required for a specific device to meet the customer resource constraints.

Technical details

VCP features and performance

- Supported sampling rates: 8 KHz, 16 KHz
- Fast filter convergence (< 300ms) with no initial echo
- Convergence in double talk and intense noise
- Robustness to speaker signal distortions
- Echo canceller filter length up to 500ms
- Residual echo level (echo suppression level): < -70dB
- Noise suppression up to -30dB
- Noise adaptation time: 100-500ms (depending on the noise type)
- Maximal AGC gain: 24dB
- Equalizer band width: 125Hz

Supported DSP cores and resources requirements

According to customers' requests VCP has been ported and optimized for a number of DSP and MCU cores:

ARM, CEVA, Renesas, Tensilica, ARC, Blackfin, Kalimba.

Support for additional cores will be provided in a future.

The resources requirements given below correspond to **ARM Cortex A8** and they are provided for reference only. Real numbers may vary depending on the DSP core, target application, memory configuration, operating system, VCP version and processing parameters. All numbers below are for 8KHz sampling rate and the Basic VCP (all blocks enabled) with 64ms acoustic echo cancellation adaptive filter, 9dB stationary noise suppression, automatic gain control, dynamic range compressors (uplink, downlink)

MCPS: 26; Code (ROM): 45KByte; Data (RAM): 16KByte

VCP technical advantages

VCP integrates the largest number of front-end voice processing technologies scalable for different applications. Overall, VCP component technologies provide multiple advantages:

Basic package advantages

1. Low computational and memory resources. This is achieved by:

   a. Tight integration of processing blocks sharing computations and memory.
   b. Special Assembly language optimization for particular DSP cores
2. Highly efficient echo canceller
   a. Sub-band scheme with a large number of frequency sub-bands (125Hz sub-band width) reducing the computational complexity of adaptive filters. This makes VCP very efficient on wideband voice and long adaptive filters
   b. Proprietary, complex LMS adaptive filtering with very fast adaptation time and robust convergence in double talk.
   c. Sub-band echo suppressor blocking only those spectral parts of the uplink signal where echo is distorted and cannot be completely cancelled by adaptive filters
3. Noise suppressor with fast noise adaptation time.
   a. Utilization of a proprietary, very reliable voice activity detector reducing the adaptation time on "noise only" sections (voice is not detected)
   b. Suppression of fast changing, transient noises (e.g. noises of passing cars). A proprietary detector of transient noises further reduces noise adaptation time when transient noise is detected.
4. Automatic Gain Control with almost instantaneous amplification of low level voice signals without being confused by ambient noises. Such noise robustness is enabled by the voice activity detector.
5. Narrow band equalizers with 125Hz bandwidth allow precise compensation of loudspeaker or microphone frequency response irregularities.

Advanced blocks advantages

1. Sub-band adaptive dual microphone array with:
   a. Very fast adaptation time allowing efficient cancellation of noises in changing environments.
   b. Support for different configurations (end-fire, broadside, close talk, echo cancellation)
2. Automatic, ambient noise dependent speaker volume and frequency equalization technology with minimal computation requirements (borrowing noise estimation from uplink noise suppressor)
3. EasyListen™ - highly efficient, real time slowing down the incoming speech for better intelligibility.

Development and tuning tools

VCP configurator

VCP Configurator is a PC Windows graphical application allowing controlling VCP functionality. Figure 1 illustrates VCP screenshot. VCP blocks and their parameters are shown according to their real position in the signal processing chains. Parameters are provided with short prompts as well as detailed help. VCP configurator generates an acoustic profile structure that can be uploaded into a device under tuning in real time via UART or other available interface.

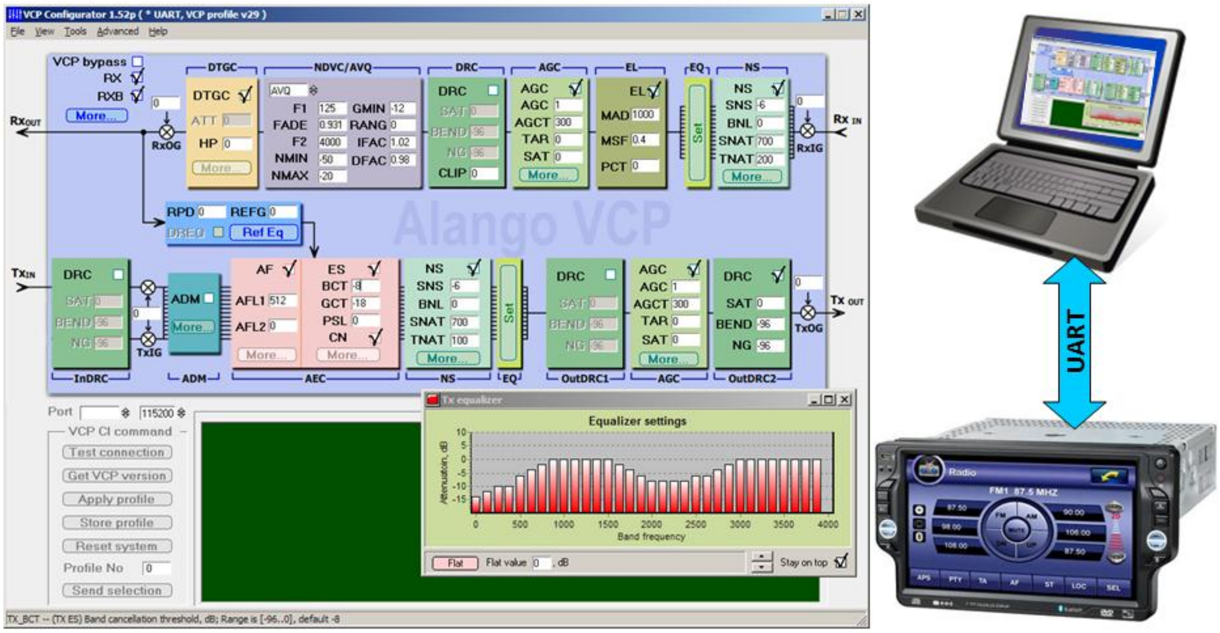

Figure 2 VCP Configurator application

Signal and data logger

Alango Logger and the corresponding PC Windows application allows real time monitoring and storing of VCP input/output signals. Logger functionality helps during the device development stage, accelerates acoustic tuning and simplifies problem identification and reporting. Figure 3 illustrates the Logger functionality.

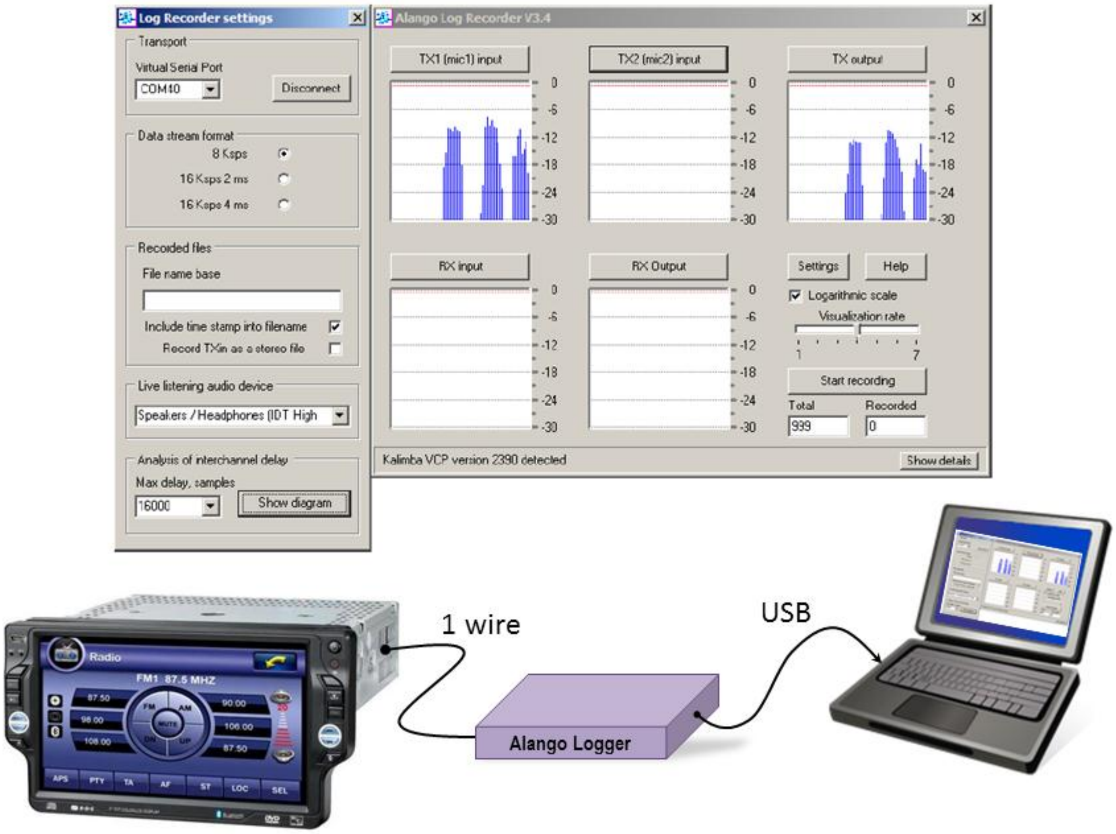

Figure 3 Alango Logger

The device under development/testing/tuning is connected via 1 wire interface to Alango Logger that is transferring accumulated data into a Windows computer via USB. Logger PC application parses the incoming stream separating input/output signals and auxiliary data. The input/output signal levels are shown in real time. It is possible to store the signals as well as to listen one of the signals via headphones.

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