SPIRIT DSP
Voice & Video Engine Experts

TeamSpirit® Voice&Video Engine Embedded

Internet-based communication steals telcos' minutes more and more. To retain users world's leading carriers are forced to roll out IP-based voice and video communication services. Hence, IP-based terminal devices must be VVoIP capable. The challenging factor for OEMs is to meet carriers' high voice and video quality requirements and secure it over congested and overloaded networks.

TeamSpirit® Voice&Video Engine Embedded complements and amplifies network-specific QoS to guaranty positive user experience with IP voice and video services.

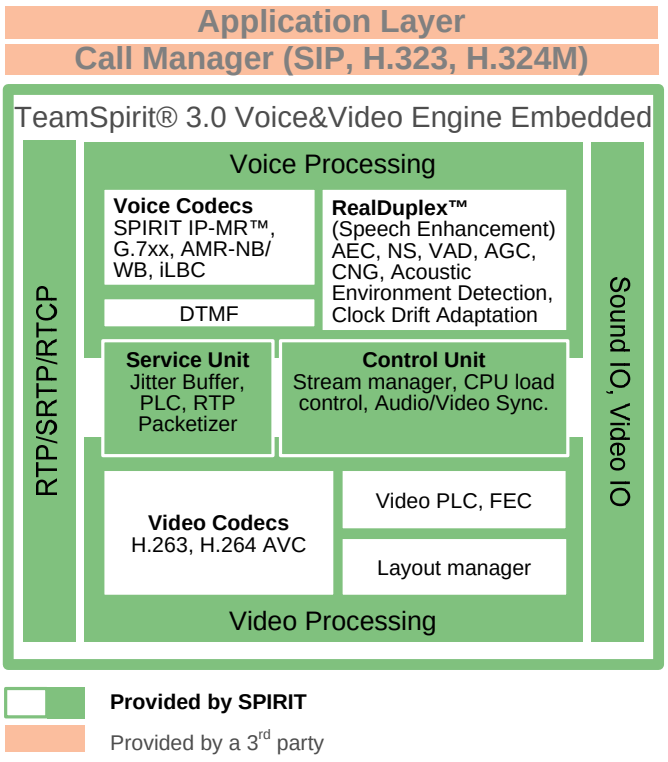
Overview

TeamSpirit® Voice&Video Engine Embedded is voice and video SDK for IP terminal equipment, such as media phones, video IP-phones, video conferencing terminals, IP set-top boxes, residential gateways, multimedia terminal adapters, etc.

TeamSpirit® Voice&Video Engine Embedded includes:

- Highly optimized, low-MIPS speech codecs, such as G.711, G.723, G.729, etc. and H.264 video codec with video packet loss robustness and other video quality improvement algorithms
- Cutting-edge speech enhancement module that includes echo canceller (AEC), noise suppresser and other must-have features
- Special module for network impairments compensation, such as latency, jitter and packet loss.

TeamSpirit® Voice&Video Engine Embedded mitigates all critical quality issues to secure positive IP voice and video experience for end-users. It is optimized for different processors including ARM, TI DaVinci (DM64xx), OMAP, etc, providing voice and video communication capabilities even on resource-constrained platforms. High-level API makes the integration of the TeamSpirit® Voice and Video Engine Embedded quick and easy.



Telecom providers should create more compelling communities by delivering a cross-platform, fully integrated communication and collaboration experience across mobile, fixed and IPTV services. This will enable to stem loss-of-share of communication time, as well as reduce churn.
"The changing face of communication" Analysis IBM Communications Sector,

Benefits

- HD voice and video to guaranty excellent communication experience
- High voice and video quality over wireless and wireline IP networks
- Compliance with international quality standards to help OEMs pass carriers' acceptance tests
- Resource-efficient solution to lower processor load
- Minimized integration efforts for fast time-to-market

Key Features

- Wideband and ultra-wideband voice processing
- High-resolution video – up to VGA, 30 fps
- Channel adaptation for reliable performance in managed and unmanaged networks
- Tightly-integrated solution with high level API
- Low memory and CPU consumption
- Support for IMS and traditional VoIP

Applications

- Media phones
- Video IP phones
- Video conferencing terminals
- Video telephony set-top boxes
- Residential VoIP gateways
- Multimedia terminal adapters

Platforms

- ARM9E/11, Cortex A8
- TI OMAP3
- MIPS32
- Intel Atom™

Specifications

Speech Codecs	▪ SPIRIT IP-MR* ▪ G.722, G.722.1, AAC LD, GSM AMR-WB ▪ G.711, G.711 App.II, G.723.1, G.729AB, G.729.1, GSM EFR, GSM AMR-NB, iLBC
Video Codecs	▪ H.263 *(up to 30 fps)* ▪ H.263+ *(up to 30 fps)* ▪ MPEG.4 *(up to 30 fps)* ▪ H.264 AVC *(up to 30 fps)* ▪ Hardware video accelerators support
Video Formats	▪ QCIF to VGA
Video Framerate	▪ up to 30 fps
Speech Enhancement -	▪ Acoustic Echo Cancellation *(operates in full duplex mode, consumes 30 MIPS)* ▪ Noise Supressor (*tightly integrated with AEC to provide superior voice quality)* ▪ Automatic Gain Control *(adjusts speaker and microphone gains)* ▪ Voice Activity Detection ▪ Comfort Noise Generation ▪ Line Echo Cancellation G.168 – 2004 ▪ Clock Drift Control
Control unit	▪ Audio/Video Synchronization ▪ Lip synchronization (audio/video sync) ▪ Video ARS ▪ CPU Load Control
Service unit	▪ Adaptive Jitter Buffer ▪ Video packet loss concealment including forward error correction ▪ Voice Packet Loss Concealment (up to 30%) ▪ PTP Packetizer
Telephony Algorithms	▪ DTMF over RTP in-band *(ITU-T Q.23),* out-of-band *(RFC 2833)*
Media Transport	▪ RTP/RTCP *(RFC 3550/3551 (IETF SIDD0064/0065))*
Supported OS	▪ Symbian ▪ iPhone ▪ Android ▪ Linux: Moblin, Midinux, Ubuntu ▪ Windows CE 4.x ▪ Windows CE 5.0

** The SPIRIT IP-MR™ codec, which payload is currently being standardized by the Internet Engineering Task Force (IETF), has been developed specifically for packet networks and ensures maximum speech quality on both the LAN and global IP networks such as the Internet*

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