VideoMost SDK for PC Client

Carriers nowadays are considering strategies to roll out profitable IP-services. And IP calls are becoming an inalienable part of their service offerings. With IP infrastructure available today carriers may span their communication services to the domain of personal computers offering its clients more features like presence, IM, desktop video conferencing and more. Quality expectations for IP voice and video communication services are very high now. IP communication promises HD quality voice and video. But congested and overloaded IP links make the task of securing excellent user experience a real challenge.

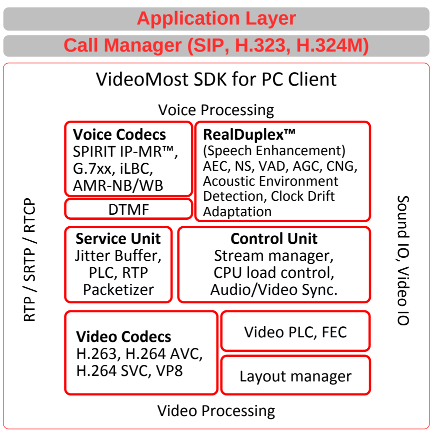
To guaranty high user-to-user voice and video communication quality, SPIRIT offers its VoIP software – VideoMost SDK for PC Client. The SDK complements (and remarkably amplifies!) network-specific QoS for positive user experience with voice and video communication services.

Overview

VideoMost SDK for PC Client is a voice and video SDK for real-time communication over IP networks. It implements HD voice and video processing and has excellent performance characteristics.

VideoMost SDK for PC Client includes:

- Highly optimized standard voice codecs, such as: G.711, G.723, G.729, etc. and a patent-free wideband SPIRIT IP-MR™ (IETF RFC 6262) codec optimized for voice transmission over IP networks and robust to packet loss.
- H.264 SVC video codec, video packet loss robustness and other video quality improvement algorithms
- Sophisticated speech enhancement module with acoustic echo cancellation (AEC), noise suppression and other must-have features
- Network adaptation module that compensates network jitter and packet loss



Benefits

- HD voice and video for life-like communication experience
- Compliant with major ITU-T and TIA international voice quality standards
- High voice and video quality over any wireless and wireline IP network
- Complete, reliable, pre-integrated solution – already deployed in IP services of leading telecom operators

Key Features

- Wideband and ultra-wideband voice processing
- HD video support (1280x720, up to 30 fps)
- Scalable voice and video codecs
- Channel adaptation for reliable performance in both managed and unmanaged networks
- Integrated engine framework
- IMS and traditional VoIP architectures are supported

Applications

- Carriers' IP services:
  Unified Communications, SoIP, basic IP voice and video calls
- Video softphones
- Instant messengers
- IMS clients

VideoMost SDK for PC Client is a part of full **VideoMost SDK server with PC and Mobile Clients**, a software development kit (SDK) for cross-platform multi-point voice and video over IP conferencing on any PC and mobile endpoint. To power multipoint videoconferencing applications SPIRIT provides **VideoMost SDK server** - a software server unit featured by unprecedented level of scalability and enabled to support up to a 1,000 concurrent video channels on a standard $4,000 PC hardware.

Specifications

Speech Codecs	▪ SPIRIT IP-MR™ (IETF RFC 6262)[1] ▪ G.711, G.722.1, G.723.1, G.726, G.729AB ▪ GSM AMR-NB, GSM AMR-WB ▪ iLBC
Video Codecs	▪ H.263 (up to 30 fps) ▪ MPEG-4 (up to 30 fps) ▪ H.264 AVC (up to 30 fps) ▪ H.264 SVC (up to 30 fps) ▪ VP8[2]
IMS Features support	▪ 3gp video sharing ▪ Voice and video recording ▪ 3gp playback/record ▪ RTSP, SDP protocols ▪ Push-to-talk
Speech Enhancement	▪ Acoustic Echo Cancellation (operates in full duplex mode, consumes 30 MIPS) ▪ Notebook Wizard for automatic AEC configuration (secures high speech quality in hands free mode) ▪ Noise Suppressor (tightly integrated with AEC to provide superior voice quality) ▪ Automatic Gain Control (adjusts speaker and microphone gains) ▪ VAD/CNG/DTX (minimizes channel payload during silence periods) ▪ Clock Drift Control
Control unit	▪ Lip synchronization (audio/video sync) ▪ Video ARS ▪ CPU Load Control
Service unit	▪ Adaptive Jitter Buffer ▪ Voice packet loss concealment (up to 30%) ▪ RTP Packetizer
Signaling	▪ SIP (RFC 3261)/SIMPLE ▪ XMPP/Jingle ▪ Connectivity Pack ▪ Any proprietary signaling protocol
Telephony Algorithms	▪ DTMF over RTP in-band (ITU-T Q.23), out-of-band (RFC 2833)
Media Transport	▪ RTP/RTCP (RFC 3550/3551 (IETF STDD0064/0065)) ▪ SRTP (RFC 3711)
Supported OS	▪ Windows 98/2K/XP/Vista/7/8 ▪ Mac OS X ▪ Linux 2.6 core

[1] *The SPIRIT IP-MR™ codec has been developed specifically for HD voice over IP networks and ensures maximum speech quality on both LAN and Internet. IP-MR™ is used in many commercial business-grade systems deployed by dozens of Tier 1 manufacturers since 2004. IP-MR ™outperforms most of modern wideband coding standards in terms of packet loss robustness and speech quality. IP-MR™ features Scalable Speech Coding technology providing adaptive quality and ultra-low server CPU consumption in conferencing scenarios.*

[2] *Functionality is available through partners*

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