

TeamSpirit® Conferencing

HD Multipoint Voice and Video Conferencing for Cross-platform Real-Time Communications

Overview

TeamSpirit® Voice&Video Conferencing is a software development kit (SDK) for cross-platform multipoint HD voice and HD video over IP (V2oIP) conferencing on PCs and Mobile endpoints including Windows PC, Mac, Linux, Android, iOS and Windows Phone. TeamSpirit® Conferencing enables the development of both enterprise and consumer solutions to meet the latest trends of voice & video over IP telephony.

The TeamSpirit® Conferencing platform includes a set of fully interoperable voice and video Engines for converged communication applications available for PCs, mobile terminals and Web applications. To power multipoint videoconferencing applications, SPIRiT utilizes the TeamSpirit® Conferencing Server Engine - a software server unit boasting an unprecedented level of scalability, enabling 1,000 concurrent video channels on a standard PC.

TeamSpirit® Conferencing relies on the most advanced Scalable Voice and Video coding technologies – SPIRiT proprietary patent-free wideband codec IP-MR™ (IETF RFC 6262) and H.264 SVC (Scalable Video Coding). Together with SPIRiT's proprietary AEC, ARS, NetJet, stream control and routing algorithms, it delivers to each endpoint the highest signal quality that the terminal and its bandwidth can handle right now, with no quality degradation and no heavy transcoding on the server side. SPIRiT software constantly monitors the channels and automatically, on-the-fly, adapts each incoming and outgoing stream to the currently available bandwidth and processing power of each particular user.



Picture 1. Carrier-grade cross-platform TeamSpirit® Voice&Video Conferencing on Android, Windows Phone and OS X

TeamSpirit® Voice&Video Conferencing components:

- Voice&Video Client-side Engines for PCs & Mobile platforms
- Voice&Video Conferencing Server Engine

TeamSpirit® Voice&Video Conferencing applications include:

- Innovative all-IP communications solutions, starting with consumer V2oIP and multi-party business collaboration to VoLTE carriers standards and industrial V2oIP
- Various applications for business and private collaboration, videoconferencing, VoIP communications, web portals, social networks, gaming, education and government, IMS-based services, unified communications and fixed-mobile convergence

Benefits

- Launch high-margin real-time collaboration products and services for businesses
- Differentiate with best in class scalable, error-resilient, adaptive cross-platform HD audio/video quality from SPIRiT DSP
- Avoid expensive engineering resources required for development of voice and video processing subsystem
- Enlarge your offering with support of all popular terminal types and platforms including tablets/smartphones
- Economize on server infrastructure with unprecedented performance of Scalable coding (1000 video channels per \$4K server)

Key features

- Premium quality of HD audio and video
- Any terminal type - standard PC with standard mic/camera, smartphone, tablet
- Unique voice and video processing technology with stream protection and adaptation to network, execution and physical environment
- Scalable Voice and Video for best quality across all terminal types and extremely high server performance
- Business-level security and compatibility with 3rd party systems
- Support by Voice & Video Engine Experts with 15+ years in VVoIP

Server capacity*

- Up to 8000 voice channels
 - Up to 1000 voice & video channels
- *Intel, quad core 3GHz

Product Description

Experiencing the High Definition Quality of Voice and Video

Communicate in HD with TeamSpirit® Conferencing

TeamSpirit® Voice&Video Conferencing is tailored to enrich your communications solutions with superior sound and visual intensity of high definition audio and video. Professional V2oIP performance and carrier-grade quality is provided by high-precision DSP processing and smart multi-component and multi-mode protection against major network communication impairments such as echo, noises, device clock drift, packet loss, jitter, delays and audio/video synchronization issues - inherent across all IP networks.

SPiRiT's unique Automatic Rate Selection (ARS) module employs various PSNR, resolution and FPS combinations to control outgoing bitrate. With more ways to control the bitrate, the Engine secures fluid video, even under adverse network conditions. With the SPiRiT CPU load control adjustments, the performance of the voice and video engine can adapt to various PC profiles and varying execution environments.

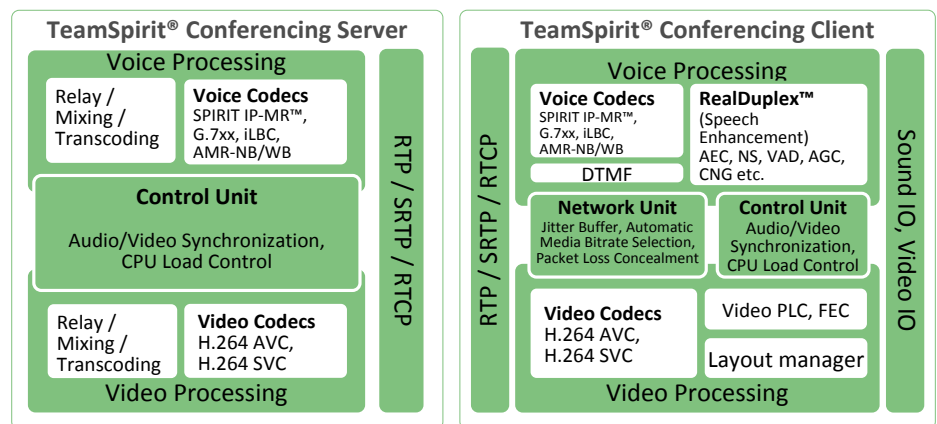
By transmitting the details of a speaker's tone, inflection, mimics, face expression and remote ambience, HD calling provided by TeamSpirit® Conferencing ushers in a great communication experience for natural human interaction.

Unifying Real-time Communications

Enhance the communication experience and mobility of your IP communications equipped with TeamSpirit® Conferencing.

TeamSpirit® Conferencing provides cross-platform massively multi-point videoconferencing technology with deep platform optimization capabilities, including support for diverse IP endpoints such as:

- Windows, Mac or Linux-based desktop PCs (x86 and x64)
- Plugins for all major web browsers
- Android, iOS or Windows Phone powered handsets (ARM)
- Custom/embedded STP and video terminals (ARM, TI)



TeamSpirit® Voice&Video Conferencing supports all IP network types including wireline, WiFi, 3G, 4G (WiMAX/LTE) and provides superior real-time adaptation to various network conditions.

Aiming at Greater Performance

Increase the capability and value of your solution with TeamSpirit® Conferencing.

Thanks to SPiRiT's in-depth ARM/x86/x64 optimization and new multi-core processing support, TeamSpirit® Voice&Video Conferencing provides premium voice and video (HD 1080p, 30 fps) on a standard PC and extended battery life on mobile endpoints.

TeamSpirit® Voice&Video Conferencing ensures high user capacity of your solution by supporting 1000+ video channels and 8000+ audio channels on one standard \$4K PC hardware server, and increasing ARPU and optimizing hardware, server bandwidth and power resource usage.

Focus on Cost-effectiveness

Capitalize on SPiRiT's leadership in voice and video quality and efficiency.

A turn-key software-based solution, TeamSpirit® Voice&Video Conferencing ensures up to 10 times lower server infrastructure costs in comparison with hardware-based conferencing systems. TeamSpirit® Conferencing has the lowest possible requirements for client-side CPU and network bandwidth.

Flexible plug-n-play TeamSpirit® Conferencing is developed to minimize the time-to-market of your products and jumpstart profits in record time.

Acknowledged Reliability

Reinvigorate your solution with real-time voice and video quality acknowledged by more than 200 global IP communications providers and equipment manufacturers.

SPiRiT, recognized for voice and video processing expertise, is driving innovations in real-time IP communications. SPiRiT applies the latest scientific approaches and sophisticated DSP mathematics methods to improve the quality of voice and video processing and achieve an unprecedented level of conferencing experiences.

SPiRiT provides its valued customers with detailed technical documentation and comprehensive support throughout the SDK integration

SPiRiT boasts:

- 15+ years of voice and video R&D expertise
- 15 PhDs in R&D
- 250+ successful projects with leading worldwide carriers, service providers, OEMs and software development companies

Specifications

Speech Codecs	<ul style="list-style-type: none"> ▪ SPIRIT IP-MR™ (IETF RFC 6262)* ▪ G.711, G.722, G.723.1, G.726, G.729AB, G.722.1 ▪ GSM AMR-NB, GSM AMR-WB ▪ iLBC ▪ Speex 						
Video Codecs	<ul style="list-style-type: none"> ▪ MPEG-4 SP ▪ H.263 ▪ H.264 AVC ▪ SPIRIT Scalable H.264 ▪ VP8 						
Video Formats	<ul style="list-style-type: none"> ▪ Up to Full HD (1080p) at 30fps ▪ Flexible video layout control on each video terminal 						
Speech Enhancement	<ul style="list-style-type: none"> ▪ Acoustic Echo Cancellation (<i>full duplex mode</i>) ▪ Noise Suppressor (<i>tightly integrated with AEC to provide superior voice quality</i>) ▪ Automatic Gain Control (<i>adjusts speaker and microphone gains</i>) ▪ VAD/CNG/DTX (<i>minimizes channel payload during silence periods</i>) ▪ Clock Drift Control (<i>deceases influence of hardware clock drift to AEC performance</i>) 						
MCU Logic	<ul style="list-style-type: none"> ▪ Smart Routing of scalable media, Relay, Mixing, Transcoding 						
Control Unit	<ul style="list-style-type: none"> ▪ Audio/Video Synchronization ▪ CPU Load Control 						
Network unit	<ul style="list-style-type: none"> ▪ Adaptive Jitter Buffer ▪ Automatic Media Bitrate Selection ▪ Packet Loss Concealment (up to 30%) 						
Supported conference options/modes	<ul style="list-style-type: none"> ▪ Support for multiple layouts and scenarios ▪ Support for audio-only participants ▪ Presentation modes ▪ Voice activated switching ▪ Moderator features supported (layout control, speaker selection, listen only mode) ▪ Video recording and image snapshots ▪ Standard telephony features (mute/hold) 						
Telephony Algorithms	<ul style="list-style-type: none"> ▪ DTMF over RTP in-band (<i>ITU-T Q.23</i>), out-of-band (<i>RFC 2833</i>) 						
Media Transport	<ul style="list-style-type: none"> ▪ RTP/RTCP (<i>RFC 3550/3551 (IETF SIDD0064/0065)</i>) ▪ SRTP (RFC 3711) ▪ Support for an external transport 						
Signaling (**)	<ul style="list-style-type: none"> ▪ SIP ▪ Any proprietary signaling protocol 						
Supported OS	<table border="1"> <tr> <td>▪ Server:</td> <td>Windows / Linux(2.6 core)</td> </tr> <tr> <td>▪ PC client:</td> <td>Windows PC / Mac OS X</td> </tr> <tr> <td>▪ Mobile client:</td> <td>iOS / Android 2.x, 4.x / Windows Phone 7(***)</td> </tr> </table>	▪ Server:	Windows / Linux(2.6 core)	▪ PC client:	Windows PC / Mac OS X	▪ Mobile client:	iOS / Android 2.x, 4.x / Windows Phone 7(***)
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*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(tm) is used in many commercial business-grade systems deployed by dozens of Tier 1 manufacturers since 2004. IP-MR(tm) outperforms most of modern wideband coding standards in terms of packet loss robustness and speech quality. IP-MR(tm) features Scalable Speech Coding technology providing adaptive quality and ultra-low server CPU consumption in conferencing scenarios.

**Functionality available through partners

***Call for availability

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