



TeamSpirit® Transcoding for Interoperability

HD Voice and Video Transcoding for Cross-platform Real-Time Communications

Overview

TeamSpirit® Transcoding Server enables interoperability of both enterprise-grade and mass market products to make IP networks / PSTN calls and video calls from different software and hardware video conferencing products based on popular codecs.

TeamSpirit® Transcoding delivers high-performance bitrate, frame rate and frame size streaming adaptation. Video transcoding H.264 / VP8 is provided, as well as voice transcoding with popular codec pairs. TeamSpirit® Transcoding boasts an unprecedented level of scalability, enabling up to 400 concurrent voice calls and up to 50 VGA concurrent video calls (sessions, each session occupying two server ports) on a standard PC server, and decreases patent royalties for software end-points. With TeamSpirit Transcoding telcos and service providers avoid most of standard codec patent payments for (millions of) end-points and pay for standard codec patents only for (thousands of) server channels.

TeamSpirit® Transcoding has been specifically designed for voice and video interoperability in the following applications and services:

- All-IP communications products, both consumer V2oIP and multi-party conferencing
- Various applications for business and private collaboration, V2oIP communications, web portals, social networks, gaming, education and government, IMS-based services, unified communications and fixed-mobile convergence

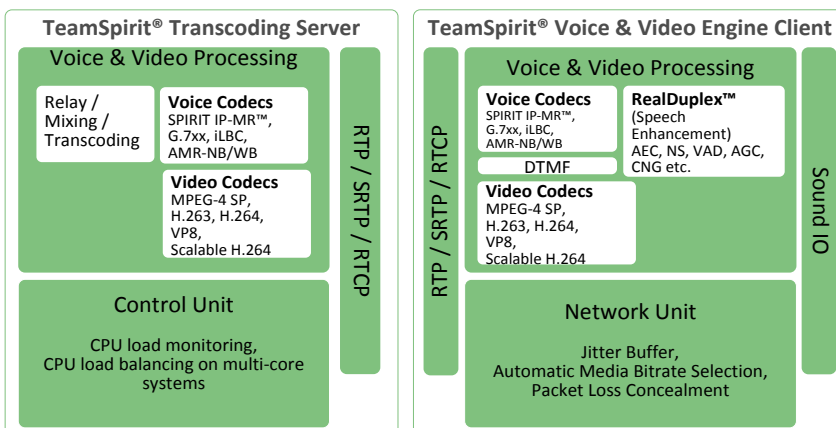
Product Description

Experiencing HD Quality of Voice and Video

TeamSpirit® Transcoding is tailored to enrich your communications solutions with HD voice and video.

SPiRiT's unique Automatic Rate Selection (ARS) module employs various codec parameters combinations to control outgoing bitrate. With more ways to control the bitrate, SPiRiT software delivers HD voice and video, even under adverse network conditions. With the SPiRiT CPU load balancing control on multi-core systems, the performance of the server can adapt to various hardware profiles and varying execution environments.

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



Benefits

- Interoperate with real-time voice and video communication products and services
- Expand your offering with support of all popular terminal types and platforms
- Economize on server infrastructure with unprecedented performance of scalability

Key features

- Premium quality of HD audio and video
- Unique voice and video processing technology with stream protection and adaptation to the network 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

Performance

- Up to 400 voice calls on a regular PC server based on 8-core 3GHz Intel CPU
- Up to 50 VGA or 5 full HD video calls on a regular PC server based on 8-core 3GHz Intel CPU

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Unifying Real-time Communications

TeamSpirit® Transcoding supports all IP network types including wireline, WiFi, 3G, LTE and provides superior real-time adaptation to various network conditions.

Focus on Cost-effectiveness

TeamSpirit® Transcoding ensures up to 10 times lower server infrastructure costs in comparison with hardware-based conferencing systems. TeamSpirit® Transcoding has the lowest requirements for client-side CPU and network bandwidth.

Acknowledged Reliability

SPiRiT DSP recognized for its voice and video processing expertise serves more than 250 global IP communications providers and equipment manufacturers. SPiRiT applies the latest innovative approaches and sophisticated DSP mathematics methods to improve the quality of voice and video processing and achieve an unprecedented level of conferencing experiences.

Specifications

Video Codecs	<ul style="list-style-type: none"> ▪ H.265 ▪ H.264 ▪ H.263(+) ▪ VP8
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 ▪ SILK
Video Formats	<ul style="list-style-type: none"> ▪ Up to Full HD (1080p) at 30fps ▪ Flexible video layout control on each terminal
Speech Enhancement	<ul style="list-style-type: none"> ▪ VAD/CNG/DTX (<i>minimizes channel payload during silence periods</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"> ▪ CPU load balancing control on multi-core systems ▪ CPU load monitoring
Network unit	<ul style="list-style-type: none"> ▪ Adaptive Jitter Buffer ▪ Automatic Media Bitrate Selection ▪ Packet Loss Concealment (up to 30%)
Supported transcoding options/modes	<ul style="list-style-type: none"> ▪ Voice announcement ▪ Voice call recording
Telephony Algorithms	<ul style="list-style-type: none"> ▪ DTMF over RTP out-of-band (RFC 2833)
Media Transport	<ul style="list-style-type: none"> ▪ RTP/RTCP (RFC 3550/3551 (IETF SIDD0064/0065)) ▪ SRTP (RFC 3711) ▪ Support for an external transport
Signaling (**)	<ul style="list-style-type: none"> ▪ SIP, MGCP ▪ Any proprietary signaling protocol
Supported OS	<ul style="list-style-type: none"> ▪ Server: Windows / Linux (2.6 core) ▪ PC client: Windows PC / Mac OS X ▪ Mobile client: iOS / Android / Windows Phone 8(***)

*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.

**Functionality available through partners

***Call for availability

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