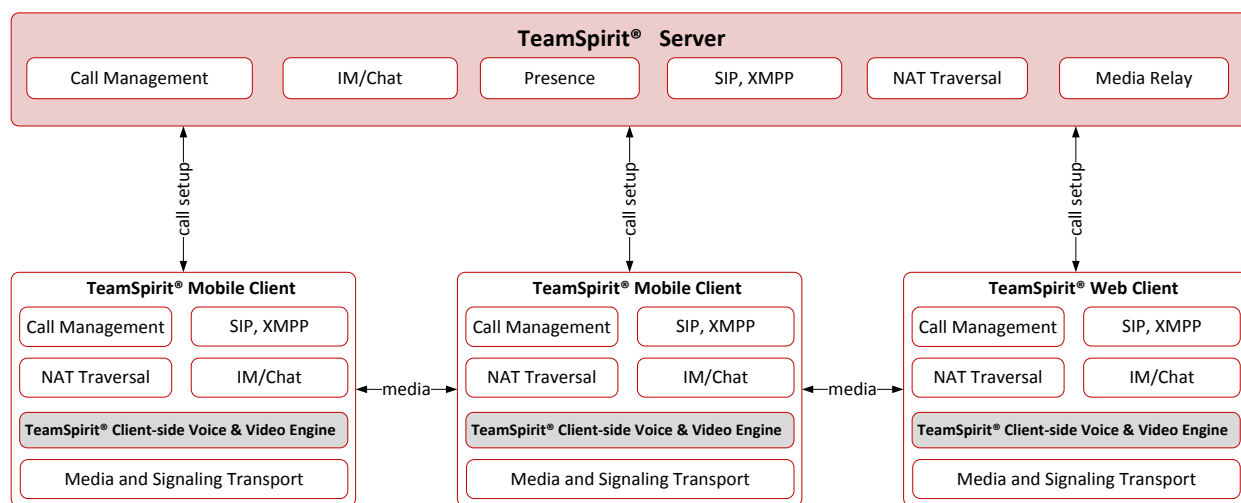# TeamSpirit Mobile Video Call SDK

for blitzkrieg development of mobile, HD, multi-platform VVoIP applications by service providers and software vendors

- Leverage 1000+ person-years experience of SPIRIT with VoIP technology

- Fast time to market

- No extra engineering efforts for development team

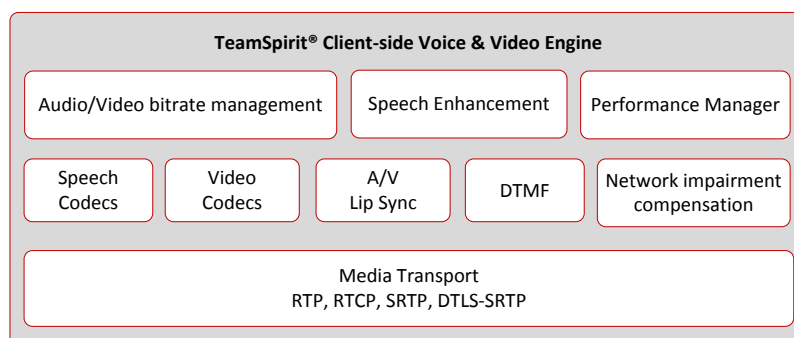- Premium cross-platform HD voice and video quality

TeamSpirit® Mobile Video Call SDK is media processing and network impairments compensation technologies required for the rapid development of peer-to-peer video calling apps. The SDK includes an exhaustive set of software components for both server and client side logic.

Smooth and clear video call over impaired mobile networks is delivered using both standard and state-of-the-art technologies developed in-house by SPIRIT DSP:

- Resource-efficient audio and video codecs, including SPIRIT IP-MR® (IETF RFC 6262, patented but free from patent royalty adaptive wideband audio codec specifically optimized for robust high- quality voice transmission over IP networks), as well as H.264 and VP8 video codecs along with native support of hardware and software video acceleration.

- Dedicated module for IP network impairments compensation, such as latency, jitter and packet loss with various voice and video packet loss robustness and quality improvement algorithms.

- Cutting-edge speech enhancement component with tightly integrated Acoustic Echo Cancellation (AEC), Noise Suppression, Automatic Gain and Anti-Howling Controls and smart Sound IO module for interaction with any device's IO subsystem.

TeamSpirit®-based mobile softphones allows superior HD voice and video communications on a wide variety of WiFi/3G/4G-enabled mobile devices. With TeamSpirit® Mobile Video Call SDK, service providers and software developers can focus on their own applications and quickly bring new products to market that include the best IP calling experience.

Specifications

TeamSpirit® Server:	▪ Call Management ▪ IM/Chat ▪ Presence	▪ SIP, XMPP ▪ NAT Traversal ▪ Media Relay

TeamSpirit® Mobile Client:

Speech Codecs[3]	▪ SPIRIT IP-MR™ (IETF RFC 6262)[1] ▪ G.722, G.722.1, GSM AMR WB ▪ G.711, G.711 App.II, G.723.1, G.729AB, G.726, GSM EFR, GSM AMR NB, iLBC
Video Codecs[3]	▪ H.263, H.263+ ▪ MPEG-4 SP ▪ H.264 AVC ▪ VP8 ▪ Hardware Video Accelerators Support
Video Formats	▪ Up to 4K
Video Framerate	▪ Up to 30 fps
Speech Enhancement	▪ Acoustic Echo Cancellation ▪ Noise Suppressor (tightly integrated with AEC to provide superior voice quality) ▪ Automatic Gain Control (adjusts speaker and microphone gains) ▪ Non-linear Processing ▪ Voice Activity Detection, Comfort Noise Generation ▪ Anti-howling ▪ Clock Drift Control
Control unit	▪ Lip Synchronization (audio/video sync) ▪ Voice & Video Automatic Rate Selection ▪ CPU Load Control
Service unit	▪ Adaptive Jitter Buffer ▪ Multi-component Voice and Video Packet Loss Concealment (up to 30%) ▪ RTP Packetizer
Telephony Algorithms	▪ DTMF over RTP in-band (ITU-T Q.23), out-of-band (RFC 4733, RFC 4734)
Media Transport	▪ RTP/RTCP (RFC 3550/3551 (IETF STDD0064/0065), RFC 4585, RFC 5506, RFC 5104, RFC 3611, RFC 6263) ▪ SRTP (RFC 3711) ▪ DTLS-SRTP (RFC 5764)
Signaling	▪ SIP, XMPP
Supported OS	▪ iOS ▪ Android ▪ Windows Phone[2] ▪ Linux Mobile

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

[2] *Call for details*

[3] *Codec set is configurable: SPIRIT delivers only those codecs which match a specific type of the customer's application.*