Why to Buy a Complete Voice and Video Engine?

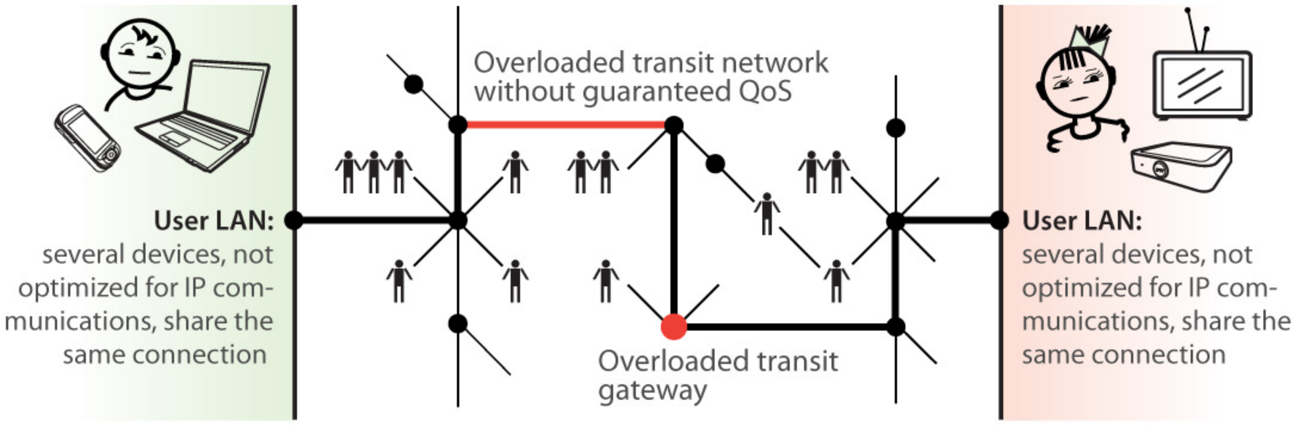
IP-COMMUNICATIONS HAVE BECOME AN EVERYDAY REALITY

IP technologies lay a solid foundation for multiple carriers' services, such as Unified Communication, Enterprise Mobility, IMS, SoIP, IPTV, etc. These services allow users always stay on and enjoy the convenience of voice, data, video, and mobile service in one package, offering cost efficiency and service and feature parity.

IP-services include a real-time voice and video over IP communication feature via a PC or mobile terminal application, as today we communicate constantly at home, in the office, even on the road. The quality of IP communications has become the key to the overall IP-services' success, as users are unwilling to put up with the degradation of the PSTN quality and reliability, even with multiple additional features offered.

However packet networks are not well suited for real-time applications like voice and video communications. They offer only the "best-effort" service scenario. To secure multimedia communication quality inside their IP-networks carriers have implemented network-specific QoS (Quality of Service) tools, which were supposed to resolve all network issues and guarantee perfect multimedia communication experience. Such "smart" networks could have worked with all kinds of low-end user-terminals with the same quality level.

But with the growing globalization an isolated "smart" network has become insufficient. The service is often accessed not within carrier's own network: for example when carrier's subscriber travels abroad and wants to use the habitual service, the traffic travels through a range of networks: partner networks, even satellite channels, using different technologies, with varying QoS. The problem of network overload only worsens the situation. As a result a range of common issues inherent to VoIP occurs.



For end-to-end IP communication quality user terminals themselves must include "smart" voice and video processing software solutions to deal with all critical quality issues at the terminal side, complementing and amplifying network QoS.

SPIRIT's carrier-grade client-side VoIP software under TeamSpirit® brand successfully conquers the problems of IP communications, working in line with network QoS tools and delivers HD voice and video over packet networks: IP, WiFi, WiMAX, WiBro and 3G networks.

Let's review the basic principles and the most common issues of IP communications.

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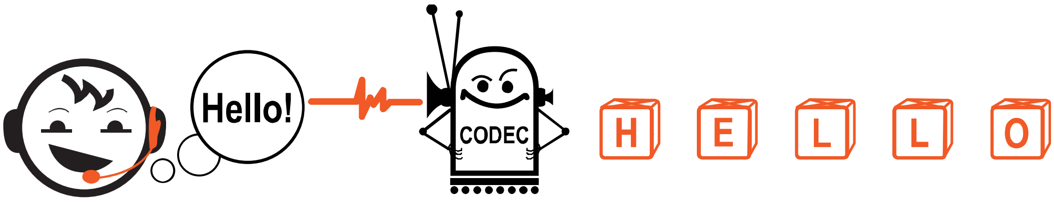
VOICE CODING

Traditional voice communication is analog, while data networking is digital. A speech codec converts voice into digital form. The codec splits the data stream into packets (this is called voice "packetizing") and sends it across the network to the listener, where data packets are converted back into voice. Speech codec must be capable of dealing with packet loss. The quality of speech produced by the codec is critical for total quality provided by VoIP-systems.



SPIRIT IP-MR™ (IP Multi-Rate) codec has been developed specifically for packet networks. SPIRIT IP-MR overcomes packet loss up to 30% and provides top speech quality.

ECHO

Echo (line (electric) and acoustic) is a common problem for telecommunication systems. Line echo is caused by the interface between the two-wire "local subscriber loop" and the four-wire transmission system of telecommunications trunk lines. Acoustic echo originates in a local audio loop when a microphone picks up an audio signal from speakers. This signal is then sent back to the peer side along with the user voice. Echo is an annoying side effect which directly influences voice quality and user satisfaction.

TeamSpirit includes cutting-edge speech enhancement algorithm – RealDuplex™. It filters out all echo and noise effects making user experience as natural as face-to-face communication even when delay is long.

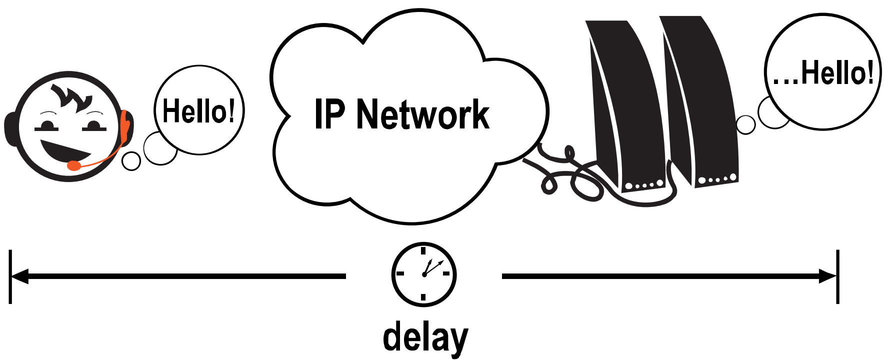
DELAY

Latency (or delay) is the time that it takes a packet to make its way through a network end-to-end. In telephony terms, latency is the measure of time it takes the talker's voice to reach the listener's ear. The delay causes two major problems:

> initial echo & noise become more audible

> hesitations in the speaker' interactions arise

Generally if the overall delay is more than about 40 ms, an irritating echo becomes clearly audible.



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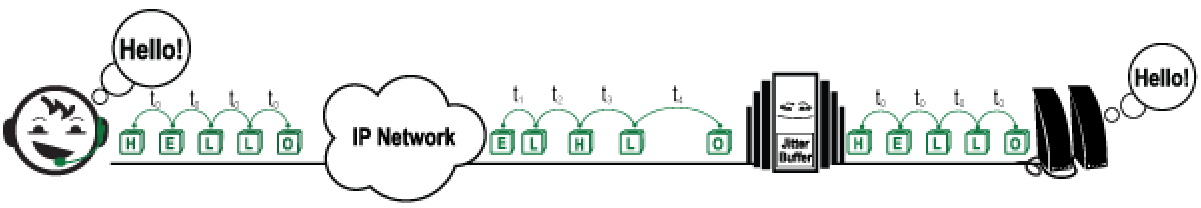
JITTER

Jitter is the measure of time between when a packet is expected to arrive to when it actually arrives.

Let's review an example: with a constant packet transmission rate of 20 ms, packets are expected to arrive at the destination at exact 20 ms intervals. But it is not always the case: packets arrive at uneven intervals, some of them arrive too late, sometimes packets arrive in bursts. Jitter introduces additional difficulties for the packet data decoding. A decoder needs packets to arrive at even intervals to reproduce the speech signal without distortions. If the packets are delivered irregularly, voice-breaks arise.

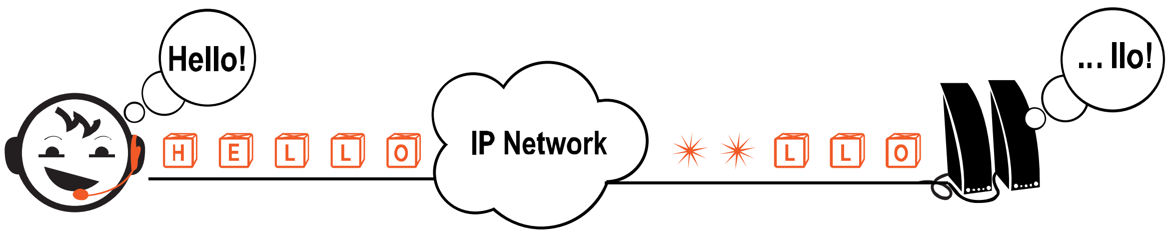
Adaptive Jitter Buffer by SPIRIT collects incoming packets and buffers them, sorting them in correct order. Even if packets arrive at uneven intervals or in bursts, it doesn't affect voice and video quality.



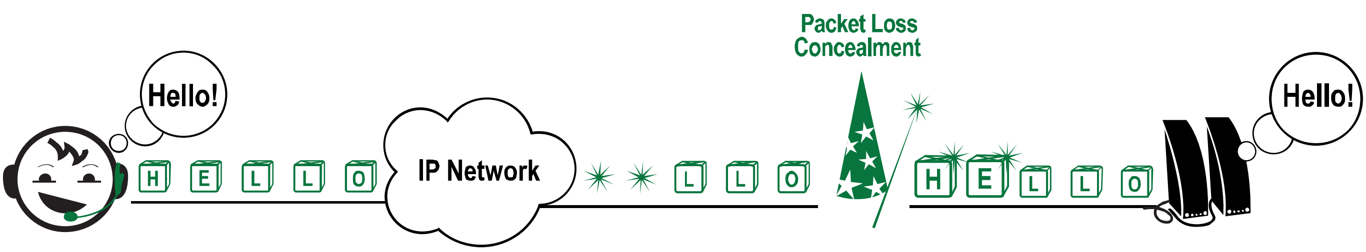
Adaptive Jitter Buffer smoothes variations caused by packets re-ordering. It buffers packets to restore their original order.

PACKET LOSS

Some packets simply do not arrive at the destination at all. Lost packets create gaps in conversation, destroy communication dynamics and severely affect voice quality.

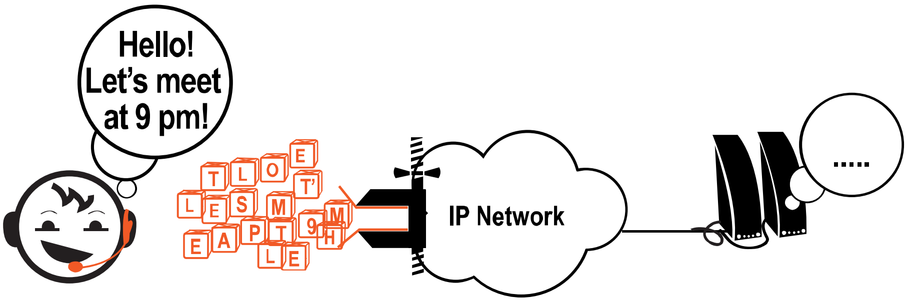


Packet loss concealment (PLC) algorithm by SPIRIT hides transmission losses in audio systems. Moreover TeamSpirit solution is capable of recovering lost packets and minimizing information loss.

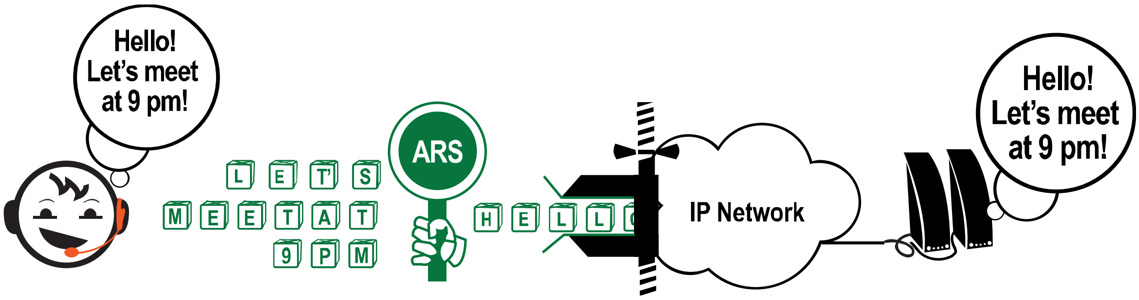


TRANSMISSION RATE

Network conditions vary permanently, sometimes there are too many packets sent at the same time, which causes buffer overflow and transmission breaks.

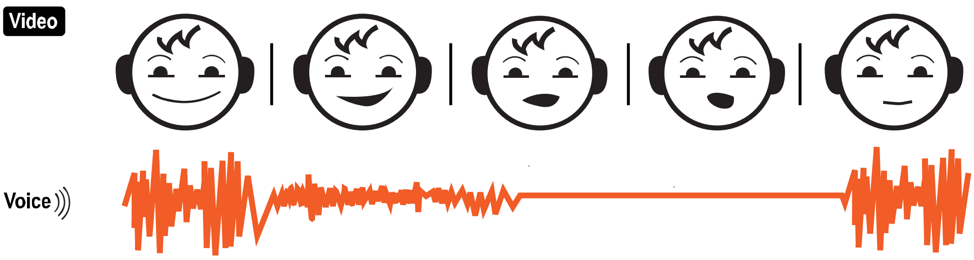


Automatic Rate Selection by SPIRIT constantly monitors network conditions and dynamically adjusts the codec's bitrate. ARS calculates statistics such as flow bandwidth, packet loss, jitter, etc. and manages the generated media traffic accordingly.



VOICE AND VIDEO SYNCHRONIZATION

By simultaneous transmission of voice and video, the user may experience mistiming of speech and picture, as the transmission rate of the packets varies.



To secure consistency of voice and video and to fight mistiming SPIRIT has developed cutting-edge lip-synchronization algorithm.

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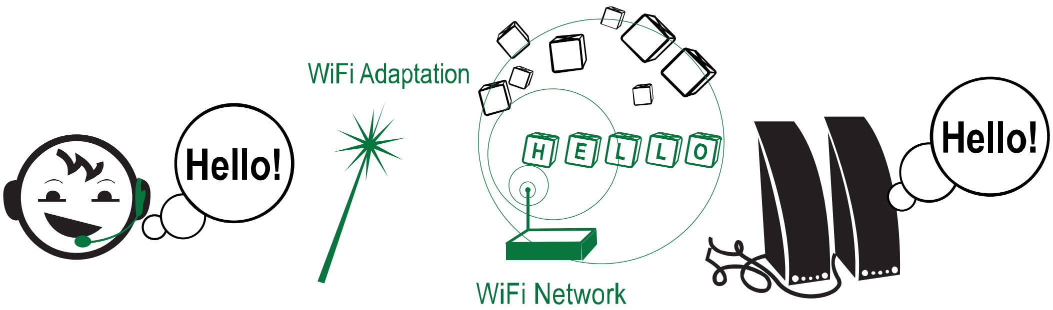
WIFI ADAPTATION

Wireless networks are more susceptible to delays, jitter and packet loss. Congestion often occurs when several users are connected to the same access point.

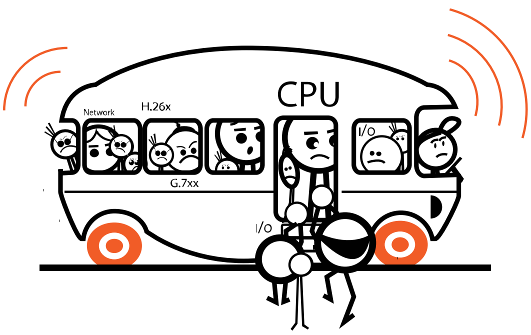
TeamSpirit includes a special WiFi adaptation module which handles WiFi-specific network impairments and makes the voice and video transmission over WiFi more stable and reliable. As a result the voice and video quality is not affected.

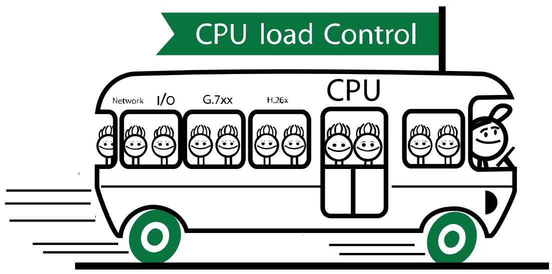


INTEGRATION

Lots of components are required for positive end-user experience with IP communication. If they all are run on a single processor-platform, they consume too many resources, preventing effective performance of each other and generally causing system overload.

CPU load control module of TeamSpirit Voice&Video Engine constantly monitors the processor load level and dynamically adjusts the engine resource consumption to prevent CPU overload.





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PERFORMANCE

Running a set of 3rd party components without system overload requires high-end processors, especially for mobile handsets. This makes the final device more expensive and reduces the potential product market.

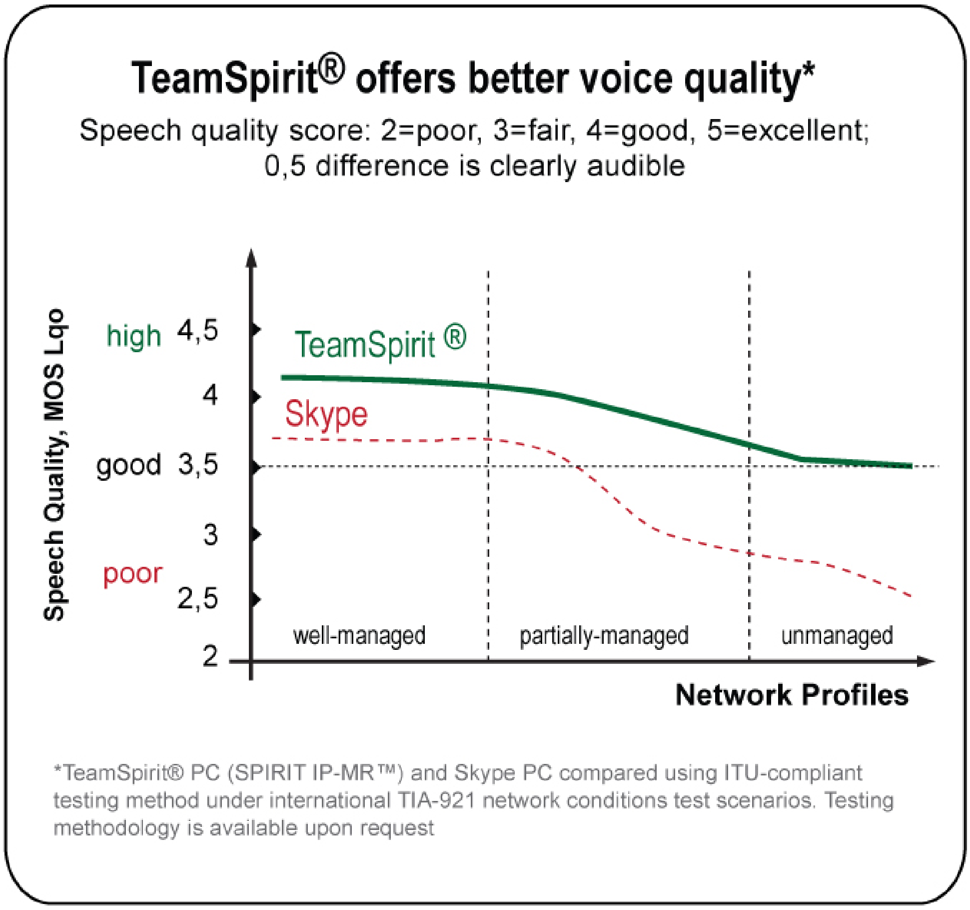
But this is not the case using a complete solution: all the components are tuned to work together within an optimized framework, which results in unprecedented synergy effects. Thanks to the highest degree of integration and optimization TeamSpirit® Voice&Video Engine brings significant resource savings and proves much more power-efficient than any set of 3rd party components.

TeamSpirit 3.0 Voice&Video Engine Mobile provides up to 10 fps video on 200 MHz processors (other solutions need 400MHz for voice processing only) and at the same time supports both software and hardware accelerators inside popular application processors to deliver 30 fps VGA video on mobile devices.

TeamSpirit 3.0 Conferencing Engine secures up to 5000-8000 concurrent voice connections per standard Intel-based server.

SUMMARY: IDEAL SOLUTION FOR CARRIERS' IP-SERVICES

TeamSpirit® Voice&Video Engine is a complete, pre-integrated, resource-efficient solution which complements and remarkably amplifies network QoS to deliver high-quality voice and video over IP, 3G and WiFi networks. The Engine complies with major international voice quality standards (ITU-T and TIA), helping device manufacturers pass operators' acceptance procedures and carriers increase user-satisfaction with their IP-services.



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