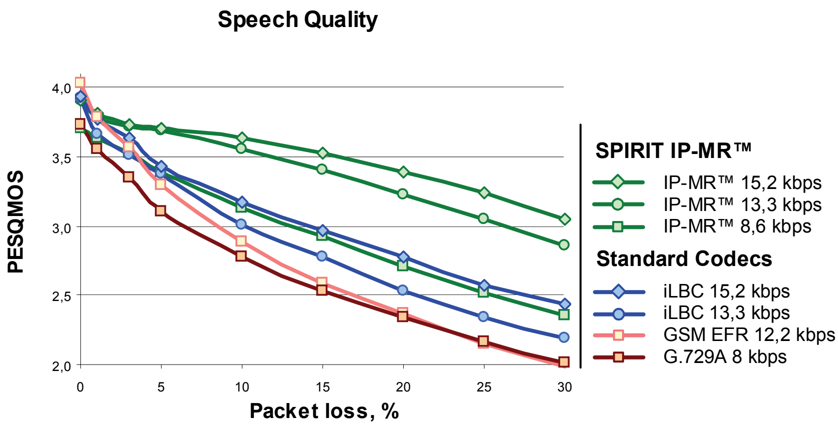
SPIRIT IP-MR™ – Dedicated VoIP Codec

The SPIRIT IP-MR™ (IP Multi-Rate) codec, which payload is currently being standardized by the Internet Engineering Task Force (IETF), has been developed specifically for packet networks. SPIRIT IP-MR ensures maximum speech quality on both the LAN and "best effort" based IP network conditions such as the Internet. It is designed to overcome the adverse network conditions of packet loss up to 30%, variable network errors and delay. SPIRIT IP-MR ensures that the bandwidth consumed is no more than the amount of useful instantaneous information content in the speech, thus ensuring optimal traffic and channel usage during the whole conversation.



Benefits

- Proprietary patent-free codec (is not a subject for patent fees like the most standard codecs)
- High speech quality
- Adaptive multi rate
- Variable bit rate
- Robustness to packet loss
- Tandem-free conferencing

Key Features

- Scalable quality for different channels
- Compatibility
- Ability to reduce voice traffic
- Layers prioritization

Key Solution Advantages

High speech quality
SPIRIT IP-MR delivers same or better speech quality as standard wide band and narrow band speech codecs at the same bit rate

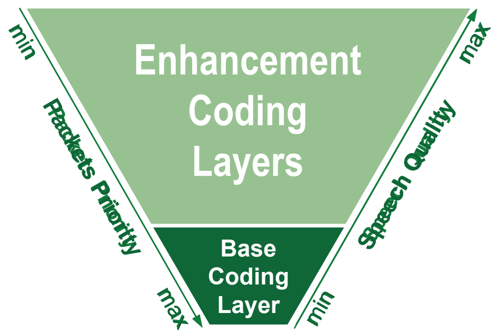
Adaptive multi rate
The SPIRIT IP-MR codec's bit rate is constantly changing in compliance with the current network conditions (capacity, packet loss ratio, etc.) avoiding channel overload and excessive platform/processor resource usage.

Variable bit rate
SPIRIT IP-MR provides constant high speech quality using channel capacity in the most effective way. Encoder bit rate varies in accordance with the actual speech content (voiced/unvoiced, pauses, stationary/non-stationary voiced, etc.). SPIRIT IP-MR dramatically optimizes and reduces traffic while keeping the quality, as encoding is adaptive to the actual prosodic characteristics of the speech.

Robustness to packet loss
SPIRIT IP-MR successfully passes speech through poor networks and provides over 30% packet loss robustness. Speech quality is enhanced while error propagation is decreased due to the proprietary solution with limited usage of the interframe correlation without speech quality loss. Based on current channel condition SPIRIT IP-MR automatically selects one of the source-channel coding modes, a priori optimized for different channel conditions, and provides the best available speech quality.

Tandem-free conferencing
Tandem-free mixing is available with no transcoding and no speech quality degradation even in case of different connection quality. So SPIRIT IP-MR allows all speakers to enjoy the maximum channel capacity. Low bitrate participants have no influence on others, thanks to scalable coding - the volume of received information is restricted only by client's channel characteristics.

SPIRIT IP-MR Unique Features



Scalable quality for different channels SPIRT IP-MR can smoothly increase speech quality due to fine granularity bandwidth and bit rate scaling in a wide range – from 7kbps to 34kbps. There are multiple coding layers – base layer and several enhancement layers that are coded independently. This approach allows to parse the bit-stream to multiple bit-streams with different bit rates and to decode each stream separately.

Compatibility between the IP-MR codecs operating at different bit rates and bandwidths. Encoder and decoder could effectively work under varying network conditions without transcoding. So there's no speech quality degradation and no need in additional resources usage.

Ability to reduce voice traffic in poor network capacity during transmission and deliver necessary number of layers to the receiving side that guarantees meaningful speech decoding without packet loss concealment procedure.

Layers prioritization - using protocols with packet prioritization makes codec extremely robust to packet losses. Data from each layer could be included in separate packets that ranged in accordance with the importance of contained information and transmitted to the client side independently. Obviously, high-priority packets are transmitted first of all, so that information-of-interest is always delivered without artifacts. On the other side, lower priority packets are also delivered if the connection quality is good enough. For example, if speech is encoded with highest quality (48 kbps bit rate) more than 85 % of the bitstream can be lost (or discarded), but residual still can be decoded meaningfully providing toll speech quality.

Technical Specifications

Frame Size	Algorithmic Delay	Signal Input	Sampling Freq	Number of Coding Layers
20 ms	25 ms	Linear PCM 16 bit	16 kHz	6

Number of coding layers	1	2	3	4	5	6
Average bitrate for active speech	7.7	9.9	14.5	21.1	28.3	34.5
Speech quality PESQ MOS	3.75	3.85	3.99	4.12	4.21	4.27

SPIRIT IP-MR™ is available on PC and embedded platforms (including Tensilica HiFi2 and ARM-cores). Resource requirements are available upon request.

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