SPIRIT IP-MR® - HD Voice Codec for IP-networks

IP-MR® (IP Multi-Rate) is a patented wideband, error resilient, HD voice codec designed by SPIRIT for mobile VVoIP and web videoconferencing software products. IP-MR® which payload is standardized by the Internet Engineering Task Force (IETF RFC 6262) provides multilayer, scalable (layered), packet loss-resilient speech coding technology for communications in IP networks.

The codec's variable bit rate and ability to adapt on the fly to ever changing network bandwidth and other network impairments secures the best voice quality available at the moment under specific network conditions. The codec's capabilities are enhanced by its robustness to packet loss and round trip delays. Built-in proprietary techniques help to provide high speech quality even when the network is heavily overloaded and/or has insufficient resources. Transmitted voice layers are prioritized and high-priority packets are subject to guaranteed delivery (in QoS networks).

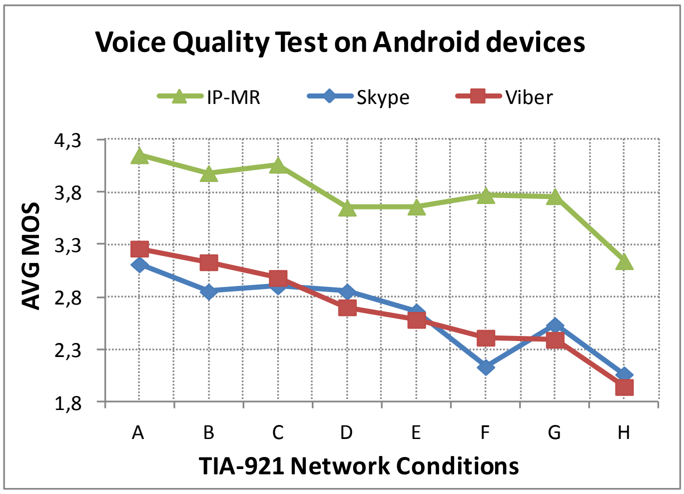
Progressive coding algorithm used in IP-MR® makes it possible to remarkably improve voice quality without needless network load due to excessive redundancy. As a result, IP-MR® has the ability to connect multiple participants that use network channels with different bandwidth without transcoding to prevent delays and voice quality degradation. IP-MR® automatically adjusts bit rates and the codec scalability provides maximum voice quality for all participants adjusted for their current channel bandwidth. The patented coding algorithm and built-in network impairments compensation mechanisms allow for optimization of voice traffic to ensure smooth and high-quality voice communication. The optimized algorithmic complexity of IP-MR® allows the codec to run just fine on mobile devices – see comparison tests on Android and iOS devices below.

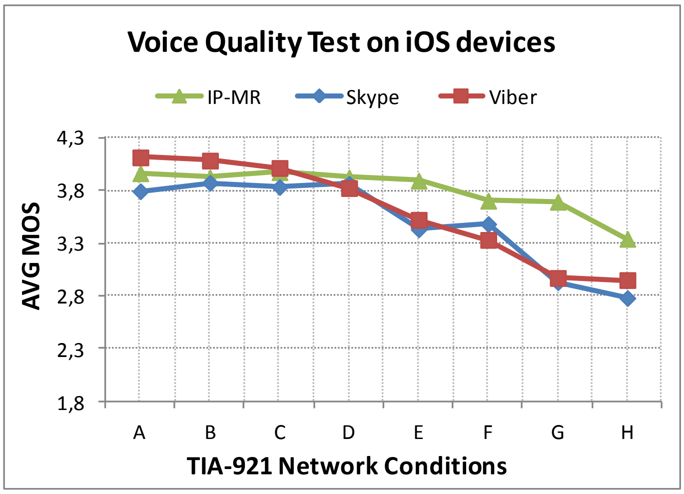
Benefits

- 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

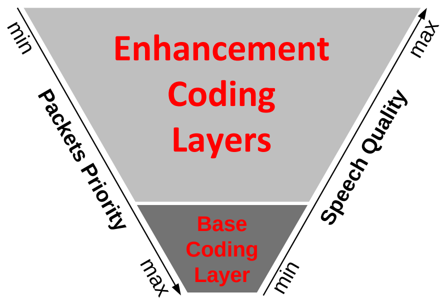
- Scalable quality for different channels
- Compatibility
- Ability to reduce voice traffic
- Layers prioritization





The best results are achieved when IP-MR® runs inside VideoMost® SDK – a scalable client-server software-only SDK offered by SPIRIT. VideoMost® SDK is a software-only videoconferencing server with mobile and PC clients for Windows, OS X, Linux, Android and iOS, complete with SIP and XMPP signaling, firewall/NAT traversal, doc sharing, white board, conference recording, transcoding and more. It allows for up to 1000 concurrent VGA (or up to 500 HD) video channels per a single and very cost efficient PC server hardware enables 250+ interactive video participants in a single conference room and supports video broadcast up to 1500 participants.

SPIRIT IP-MR Scalability and Compatibility



Scalable quality for different networks. SPIRT IP-MR® fluently increases voice quality due to fine internal granularity of the operational bit rate that scales in a wide range from 7.7 kbps to 34.2 kbps. 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. Both the encoder and the decoder effectively work under varying network conditions without transcoding. So there's no voice quality degradation and no need in using excessively redundant techniques that overload the network.

Ability of IP-MR® to reduce voice traffic in poor network conditions during transmission and to deliver only necessary number of coded layers to the called party guarantees decoding of meaningful voice data that complements and amplifies packet loss concealment techniques.

Layers prioritization. Using protocols with packet prioritization makes IP-MR® codec extremely robust to packet losses. Data from each layer are included in separate packets that are ranged according to the importance of contained information and then transmitted to the called party independently. High-priority packets are transmitted first of all so that key voice data are always delivered without artifacts. On the other side, packets of the lower priority are also delivered when the quality of connection allows to do that. For example, when voice is encoded with the highest quality (34,2 kbps bit rate) more than 75% of the bit stream can be lost (or discarded) but remaining meaningful voice data still can be decoded 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

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