GLOBAL IP SOLUTIONS

GIPS Codecs

Unmatched VoIP Clarity

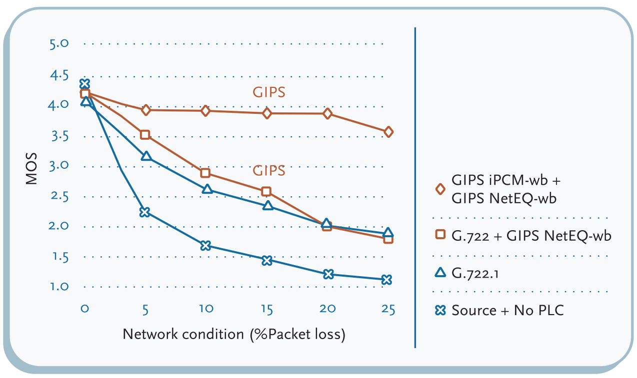
GIPS Family of Codecs

Global IP solutions offers a full set of codecs that are tailored to meet your VoIP communications needs. The codec suite includes wideband and narrowband codecs that produce a robust, clear sound, even under heavy packet loss. When this sound clarity is combined with GIPS NetEQ™, delay and jitter are managed to produce the highest quality VoIP coversation available.

	Narrowband	Wideband
Dial-up	iLBC 13.3 / 15.2 kb/s	iSAC 10 - 32 kb/s
Broadband	Enhanced G.711 64 kb/s	iPCM-wb 80 kb/s

GIPS iPCM-wb™

iPCM-wb is a high-quality, low-complexity wideband codec that provides excellent resiliency against packet loss, resulting in significantly higher sound quality than PSTN. When deployed in end-to-end IP communication, iPCM-wb ensures excellent speech quality for high-end telephony and for special user requirements (such as conference calls) where excellent sound is required.



Key benefits

- Higher basic speech quality than current standards, such as G.722.1 and G.722.2
- Packet-loss robustness that significantly exceeds current standards
- Lower delay than standard solutions when combined with NetEQ
- Compatible with GIPS Enhanced G.711
- Allows more efficient network provisioning

These tests were performed by Lockheed Martin Global Telecommunications (formerly Comsat), an independent test laboratory. Score system range: 1 = bad, 2 = poor, 3 = fair, 4 = good, 5 = excellent

GIPS iSAC™

iSAC is an adaptive VoIP codec that is specifically designed to deliver wideband sound quality in both low- and high-bit rate applications. Even at dial-up modem data rates, iSAC delivers better than PSTN sound quality by adjusting transmission rates to give the best possible listening experience for the existing connection speed.

Features

- Excellent trade-off between bit-rate and audio quality for low rate connections such as modems
- Automatically adjust the bit rate for best quality or use a bit rate set to a fixed value
- Efficient speech codec; good quality for any audio signal
- Packet-loss robustness that exceeds current standards
- Uses the full 8 kHz available audio bandwidth for 16 kHz sampling

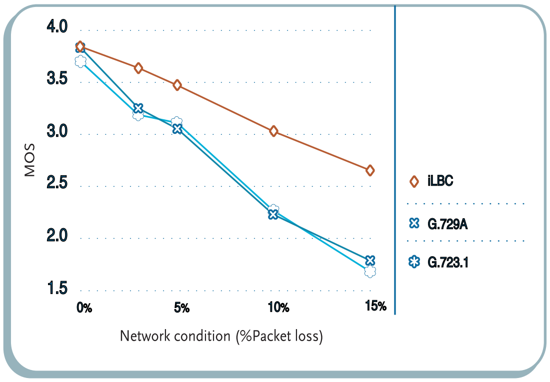
GLOBAL IP SOLUTIONS

GIPS Codecs

GIPS iLBC

GIPS iLBC from Global IP Solutions is a royalty-free codec that delivers basic speech quality better than G.729A and similar to G.729E, while offering substantially better quality over congested networks with packet loss.

iLBC is the first codec ever to be standardized by the IETF (RFC 3951 and RFC 3952) and is designated by CableLabs as a mandatory component of PacketCable voice-over-cable telephony systems.



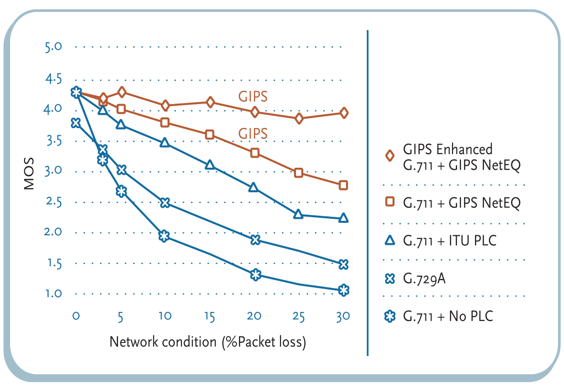
Tests were performed by Dynastat, Inc., an independent test laboratory.

Key benefits

- Royalty-free codec that provides better speech quality than G.729A and G.723.1.
- Better packet loss robustness compared to other low-bit rate codecs, including G.729A, G.729E, G.723.1 and G.728
- Supports multiple frame sizes giving increased flexibility
- Works with GIPS NetEQ to provide low delay and high packet loss robustness for low-bit rate codecs

GIPS Enhanced G.711

GIPS Enhanced G.711 is the GIPS improved version of the standard G.711 codec and provides excellent packet-loss robustness. This modified and enhanced codec maintains high speech quality over heavily loaded networks.



These tests were performed by Lockheed Martin Global Telecommunications (formerly Comsat), an independent test laboratory.

Key benefits

- High basic speech quality equal to PSTN/G.711
- Superior packet-loss robustness compared to G.711
- Lower delay than standard solutions when combined with NetEQ
- Allows more efficient network provisioning