GLOBAL IP SOLUTIONS

GIPS NetEQ™

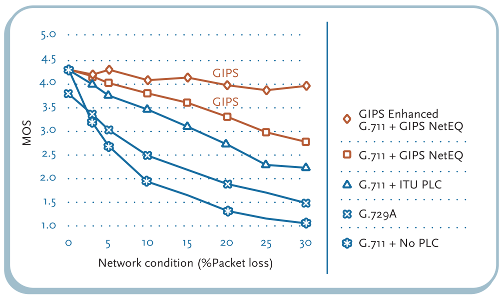
Maximize sound quality, minimize delay

GIPS NetEQ™ is an advanced speech-processing solution for IP telephony systems that delivers dramatic improvements in sound quality while minimizing buffering latency. Designed for use on individual network edge devices, the embedded software is available in either a narrowband or wideband system implementation. The performance of the narrowband (8 kHz sample rate) mode of the software is capable of delivering PSTN or better voice quality for IP telephony. NetEQ-wb advances IP communication to a new level by supporting 16 kHz and 32 kHz sampling to achieve higher fidelity than conventional phones. In all implementations, NetEQ maintains very high voice quality even under high packet loss conditions.

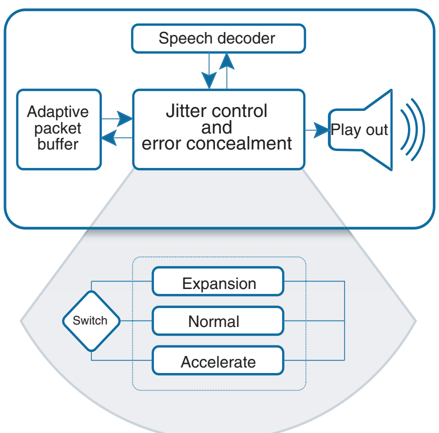
Both versions of NetEQ offer the flexibility of a single-ended solution, meaning hardware designers and service providers need only to deploy it on the receiving end to improve end user voice quality. The software can quickly adapt to changing network conditions and offers much higher resolution than other available solutions.

Key benefits

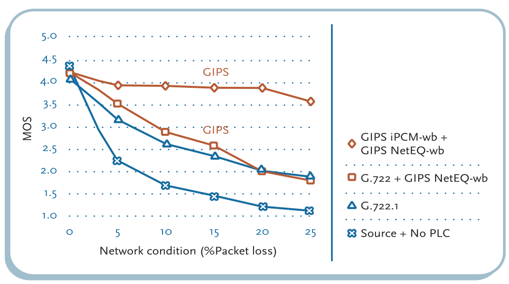
- Dramatically improves sound quality
- Minimizes jitter buffer delay, with reductions of 30 – 80 ms compared to the best alternative adaptive jitter buffers
- Deployed only at the receiving end
- Reduces need for network over-provisioning
- Compatible with all standard codecs



Tests of NetEQ (above) and NetEQ-wb (below) show voice quality performance over a range of network packet loss conditions. These tests were performed by Lockheed Martin Global Telecommunications (formerly Comsat), an independent test laboratory.



The combination of an advanced, adaptive jitter buffer control and an error concealment system in a single unit is the key to NetEQ performance and the excellent quality of service it delivers.



Source: Lockheed Martin Global Telecommunication. Score system range: 1 = bad, 2 = poor, 3 = fair, 4 = good, 5 = excellent

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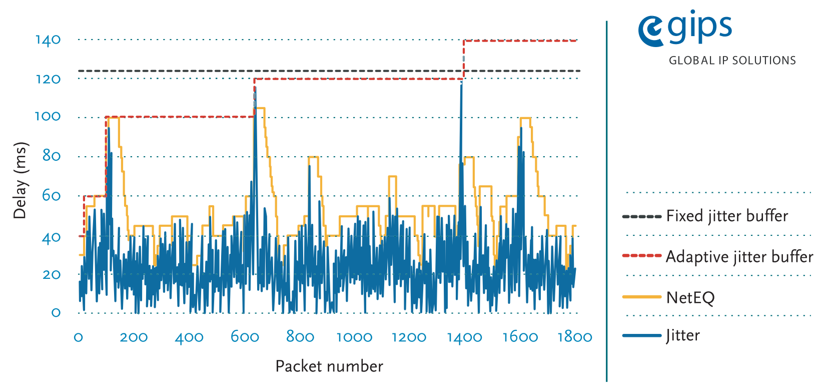

The unique non-sequential process implemented in NetEQ combines jitter buffer and error concealment in one unit. The unit detects delay variations and makes corrections before play-out and therefore produces a speech signal with highly improved sound quality. The high-resolution adaptive jitter buffer implemented in NetEQ is able to adjust quickly to the incoming jitter and operate very efficiently.

SPECIFICATION

	NetEQ	NetEQ-wb
CODECS	Compatible with any codec through the generic codec interface. Currently implemented for GIPS Enhanced G.711, G.711, iLBC, G.729, G.729A, G.723.1, 16-bit PCM, DVI, GSM FR, GSM, EFR, and AMR	Compatible with any codec through the generic codec interface. Currently implemented for iPCM™-wb, iSAC™, G.722, G.722.1, G.722.2 (AMR-wb), Siren, and 16-bit PCM
PACKET SIZE	Supports 10-60 ms speech frame sizes	Supports 10-60 ms speech frame sizes
IMPLEMENTATION	Fixed point ANSI C code and DSP (TI C54x, C55x, C62x, C64x, Starcore 810x, ADI, ZSP and MIPS) off-the-shelf. Other DSP platform opt. on request	Fixed point ANSI C code and DSP (TI C54x, C55x, C62x, C64x, Starcore 810x, and ADI) off-the-shelf. Other DSP platform optimizations on request
COMPATIBILITY	Needs to be implemented only on the receiving side	Needs to be implemented only on the receiving side
COMPLEXITY	3.6 MIPS/channel for a TI C54x processor	4.8 MIPS/channel for a TI C54x processor
MEMORY	Static: 1.0 + 2.5*N kWord16 where N is number of channels Dynamic: 1.0 kWord16 Program: 17 kWord16	Static: 1.0 + 5.8*N kWord16 where N is number of channels Dynamic: 3.1 kWord16 Program: 17.1 kWord16
DELAY	Significantly lower than for stand-alone jitter buffers	Significantly lower than for stand-alone jitter buffers
ADAPTIVE BUFFER SIZE	Dynamic, minimizes delay at millisecond resolution	Dynamic, minimizes delay at millisecond resolution
PACKET LOSS CONCEALMENT	Integrated, superior quality compared to other algorithms	Integrated, superior quality compared to other algorithms
OTHER INFORMATION	Support for RTP, RTCP Support for RFC 2198 (FEC), RFC 2833 (DTMF) Support for Comfort Noise Generation according to RFC 3389	Support for RTP, RTCP Support for RFC 2198 (FEC), RFC 2833 (DTMF) Support for Comfort Noise Generation according to RFC 3389