GLOBAL IP SOLUTIONS

GIPS ConferenceEngine

Ensuring a high quality VoIP conferencing bridge

GIPS ConferenceEngine is a powerful and complete server-based software plug-in that handles all aspects of audio mixing and other voice related tasks in a conference bridge application. The ConferenceEngine solution provides high quality mixing of several audio streams. Based on award-winning solutions and patented technologies developed by Global IP Solutions, it also ensures minimal delay and excellent audio quality even under adverse network conditions.

GIPS ConferenceEngine equips manufacturers with an open, flexible and high quality solution that can be rapidly and easily integrated into a conference server application.

Key benefits

- Ideal for enterprise conferencing applications
- Ease of integration
- Guarantees best voice quality for clear communications during conference calls
- Ensures lowest possible delay for real time communications
- Unsurpassed robustness against network impairments for confident deployments
- In-house resources can be used to focus on features to differentiate product rather than voice processing
- Call capacity limited only by server capacity
- Portable to platform of choice

Features

- Handles all audio related tasks in a conference bridge
- Large set of control functionality
- Wideband audio, with seamless mixing between wideband and narrowband audio
- Automatic gain control (AGC), VAD/DTX/CNG
- Background noise detection
- Mitigates effects of different time references in each client
- Supports all GIPS codecs and standard codecs
- Support for addition of external encryption technology
- Support for DTMF
- SIP reference

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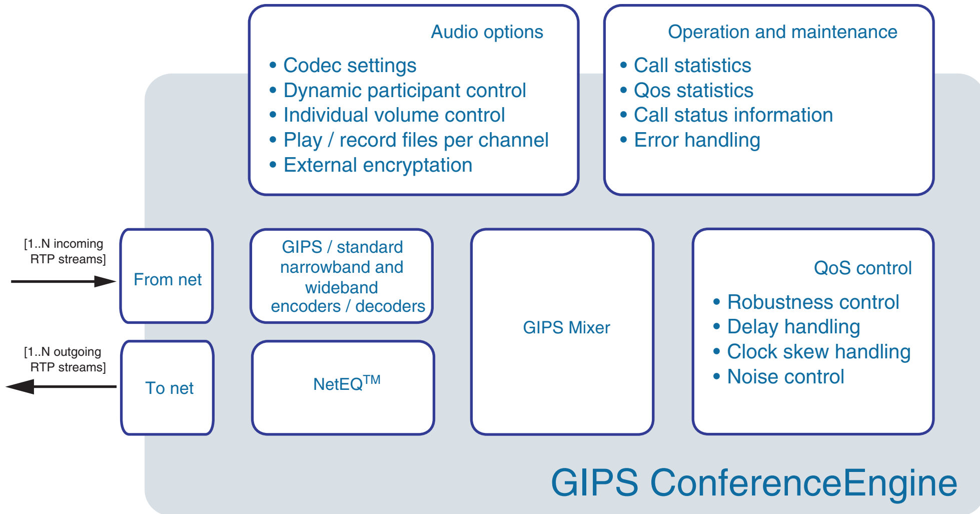

SPECIFICATION

FUNCTIONALITY	Listens for incoming RTP streams Generates outgoing RTP streams with mixed audio
CODECS	G.711, GIPS Enhanced G.711, iPCM-wb, iLBC, iSAC, other standards such as G.729, G.723.1 on demand
QUALITY OF SERVICE	GIPS codecs together with NetEQ provide extreme robustness to packet loss and jitter
AUDIO PLAYOUT / RECORDING	Play sound files to all channels or individual channels Record sound files from all channels or individual channels
AUDIO CONTROL	Individual codec settings for every channel Seamless transcoding to ensure maximum quality for every channel Set volume manually for individual channels Mute individual channels
AUDIO ENHANCEMENTS	True mixing to allow multiple speakers Wideband audio Seamless mixing between wideband and narrowband audio Automatic gain control DTMF detection Background noise detection Clock drift compensation to ensure minimal delay
SECURITY	Support for external encryption technology
CAPACITY	Capacity limited only by server capacity Reentrant implementation ensures multiple conferences per process
SUPPORTED PLATFORMS	Windows 2000/XP, Linux (Pentium CPU) Planned for Solaris (SPARC CPU)