GIPS VoiceEngine

Product Description

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VoEProdDesc08.26.2009 – REV1.1

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Chapter 1. Introduction

About this Guide

This document provides an overview of GIPS VoiceEngine family products. It describes their benefits and value proposition. It also depicts the products' architecture and provides high level technical information about their features.

Product Version Information

This product description corresponds to GIPS VoiceEngine product, starting from Release 3.3.0.

In This Guide

- Chapter 2 is an overview of GIPS products.
- Chapter 3 gives an overview of GIPS VoiceEngine.
- Chapter 4 describes VoiceEngine's technical specifications.
- Chapter 5 briefly describes the product's features.

Document Change History

Document Revision Number	Date	Change Summary
1.0	May 2009	First Release of the document
1.1	August 2009	Updated the document with the features added in release 3.3.0 - Receiver bandwidth extension - List available sound devices including current and default - Support for "Participant Placement" in client mixing mode - Support for DTMF according to RFC 4733 - Configurable comfort noise generation

Chapter 2. GIPS Products Overview

Global IP Solutions is the world's most widely deployed technology for processing real-time voice and video over IP networks. As the inventors of iLBC – an IETF industry standard codec for narrowband speech and iSAC – the defacto industry standard codec for wideband speech, Global IP Solutions' software delivers robust functionality with enough flexibility to be easily integrated in virtually any application. Our software is deployed in over 800 million end-points and resolves the full range of network impairments — like delay, jitter, packet loss, and echo. In addition, Global IP Solutions' technology offers the highest quality voice and video processing available in the marketplace as well as industry leading bandwidth optimization to ensure the most efficient use of IP networks.

GIPS' solution architecture is multi-tiered — so customers get to market faster with the right combination of features and performance.

- Tier-1 consists of quality-enhancing components (e.g., NetEQ) designed to mitigate a range of network impairments.
- Tier-2 consists of codecs to support various network environments and performance criteria.
- Tier-3 consists of voice and video engines — each designed to power voice or video communications platform while leveraging GIPS' quality-enhancing components for unrivaled performance.

This chapter gives a brief description of GIPS Engine products. For information about other GIPS products, or a more detailed description of VoiceEngine and Video Engine please refer to the GIPS web site at www.gipscorp.com.

Figure 1 GIPS Products

Client VideoEngines

VideoEngine PC is a complete voice and video processing solution optimized for softphone applications on PC platforms. It empowers applications with advanced media processing technology to handle delay and packet loss associated with IP networks, and includes both proprietary and industry standard audio and video codecs.

VideoEngine Mobile adds cutting-edge real-time voice and video capabilities to mobile applications. By implementing VideoEngine Mobile, developers can overcome the limitations of mobile IP communications, such as congested WiFi networks and background noise and echo. And because it is flexible and easy to integrate, customers can be assured that they can accelerate development and stay ahead of the rapidly evolving market for mobile voice and video.

Client VoiceEngines

Please refer to Chapter 3 - GIPS VoiceEngine Product Family Overview for an overview of GIPS Client VoiceEngines.

Server VideoEngines

Video ConferenceEngine is a powerful platform for developers to enable high-quality real-time audio and video conferencing. The platform includes all of the media processing functionalities and features of traditional hardware infrastructure-based video conferencing bridges, while offering a flexible and cost-effective software alternative that is easy to implement and maintain. Based on the award winning technology that has made Global IP Solutions the industry leader in enabling real-time communications, Video ConferenceEngine ensures a superior experience, even under adverse network conditions.

Server VoiceEngines

Voice ConferenceEngine is a powerful and complete server-based software engine that handles all aspects of audio mixing and voice related processing in a conference bridge application. Based on Global IP Solutions' award-winning and patented technology, Voice ConferenceEngine ensures minimal delay and excellent audio quality, even under adverse network conditions.

Voice MediationEngine (VME) is an advanced server based transcoding and dejittering solution that can be used in a mediation device, gateway or monitoring application. VME provides equipment manufacturers and service providers alike with a solution that solves all of the complex voice processing functions for VoIP communications, guaranteeing the best voice quality with the lowest possible delay.

Chapter 3. GIPS VoiceEngine Product Family Overview

The VoiceEngine family consists of comprehensive, packaged solutions that handle all the necessary voice processing tasks for IP networks, providing superior voice quality even under adverse network conditions. These products have been optimized for high performance on a number of platforms—from PCs to Mobile devices— allowing customers the flexibility to design almost any application or device around Global IP Solutions' cutting-edge technology.

Following is a brief description of VoiceEngine family products:

VoiceEngine PC is a full-featured voice processing framework tuned for the PC environment. It's designed to handle all voice-related tasks in real time for VoIP while incorporating quality enhancement components, including codecs, NetEQ and VQE to control delay, jitter, packet loss and echo. Because VoiceEngine PC is designed specifically for PC softphone applications, it runs efficiently on operating systems not designed for real-time communications in which artifacts such as clock drift make IP communications difficult.

VoiceEngine Mobile extends VoiceEngine functionality to mobile applications by handling complex voice processing on smartphones, dual-mode phones and PDAs. VoiceEngine Mobile enables applications to overcome limitations inherent to mobile VoIP, such as extreme packet loss on WiFi networks and excessive background noise and echo in mobile environments. Platforms include Symbian, Windows Mobile and iPhone.

VoiceEngine ATA/IP Phones is embedded voice processing for chips and hardware devices including IADs/ATAs, IP phones, gateways, and wireless devices. It's available on virtually any platform and runs on any OS.

Product Benefits

Some of the key benefits of GIPS VoiceEngine are listed below:

- Highest voice quality over IP networks
- Field-proven solution with over 800 million deployments
- Optimizes bandwidth to provide clear, consistent conversation and traffic flow
- Flexible high-level API for easy integration and accelerated time to market
- Optimized solution of complex speech processing components
- Robustness against packet-loss and delay

- Platform independent
- Available for mobile platforms (e.g., Windows Mobile, Symbian)
- Can be tailored to meet specific customer requirements

Value Proposition

VoiceEngine allows application developers, service providers and hardware manufacturers the ability to easily build complex voice processing technology into their solutions, without worrying about the effects of delay, jitter, packet loss, background noise and echo. This superior voice quality improves customer satisfaction and reduces churn. Additionally, GIPS believes that a reputation for high quality can increase market share and maximize revenues for its satisfied customers.

VoiceEngine is cost effective. Its multi-platform support and ease of integration reduce time to market and thus speed revenues.

VoiceEngine is a field-proven solution with over 800 million deployments worldwide. Our vast real-world experience helps minimize the operational and service issues commonly associated with new products and services.

Quality monitoring and reporting capabilities of the product allow for early detection of service issues and a proactive approach in resolving them. This approach further increases customer retention and reduces churn.

VoiceEngine provides secure communication, increasing customer satisfaction.

Architecture

Figure 2 shows how VoiceEngine fits in a VoIP soft client on a PC platform. VoiceEngine interposes between the application layer and drivers of hardware devices and processes audio streams. VoiceEngine handles all voice related tasks such as:

1. Speech processing
2. Audio settings
3. Sound card handling
4. Audio file handing
5. Audio conferencing client mixing
6. Voice quality enhancement including echo cancellation

7. RTP protocol
8. Statistics
9. Encryption
10. Miscellaneous voice handling including push-to-talk and lip synchronization

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Note:
GIPS VoiceEngine does not handle call setup.

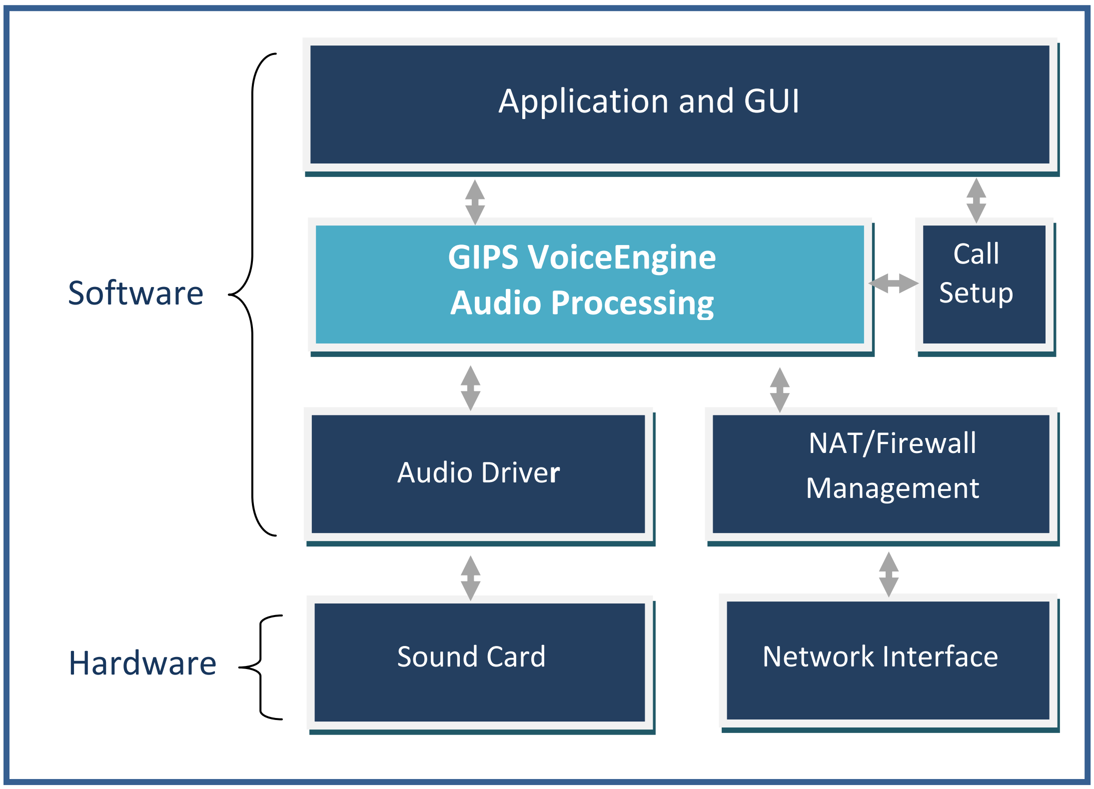

Figure 2 GIPS VoiceEngine within a Solution

In addition to the VoiceEngine products, GIPS can also provide sample integration code to further speed up the integration.

Configuration

VoiceEngine with its flexible architecture can be used in any of the following configurations:

- GIPS VoiceEngine PC

- GIPS VoiceEngine Mobile (Windows Mobile, Symbian, iPhone)
- GIPS VoiceEngine IP Phone
- GIPS VoiceEngine ATA

The following figure depicts VoiceEngine PC's architecture.

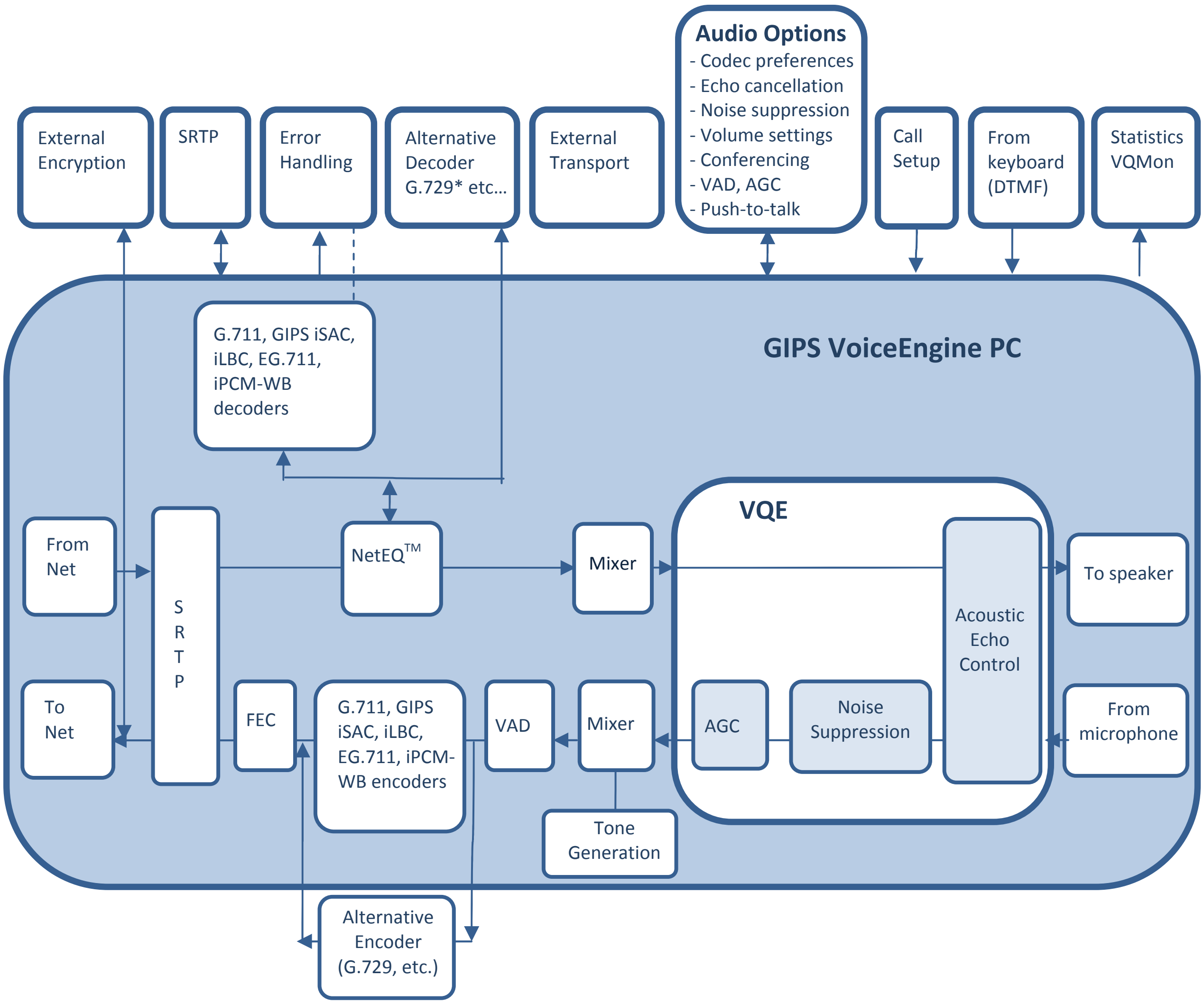

Figure 3 GIPS VoiceEngine PC Architecture & Configuration

Each configuration has a set of supported features as described in the table below:

√	Included in the product package
Optional	Available, can be added to the product package
-	Subject to special request
NA	Not Applicable

GIPS VoiceEngine Features	PC	Mobile[1] for Windows	Mobile[1] for Symbian	Mobile for iPhone	IP Phone[1]	ATA[1]
GIPS NetEQ	√	√	√	√	√	√
GIPS narrowband and wideband codecs	√	√[2]	√[3]	√[4]	√	√[5]
External narrowband and wideband codecs	Optional	Optional	Optional	Optional	Optional	Optiona l
Receiver bandwidth extension	√	NA	NA	-	NA	NA
VAD/DTX/CNG	√	√	√	√	√	√
Telephony	√	√	√	√	√	√
DTMF Detection	√	√	-	√	√	√
DTMF Generation	√	√	√	√	√	√
Call Progress Tone (CPT)	NA	NA	NA	NA	√[6]	√
Support for RFC 2833	√	√	√	√	√	√
Support for RFC 4733	√	√	NA	-	-	-
Caller ID (CID) Generation	NA	NA	NA	NA	NA	√
Audio						
Speech Level Indicator	√	-	√	-	-	√
Microphone Volume Control	√	-	-	-	√	NA
Speaker Volume Control	√	√	√	-	√	NA
Volume Scaling	√	√	√	√	√	√
Mute	√	√	√	√	√	√
Sound Card/Device						

[1] Option may be limited by hardware configuration
[2] GIPS iSAC-LC is used instead of GIPS iSAC to reduce complexity
[3] Wideband not supported due to hardware limitations
[4] Wideband only supported for handsfree due to limitations in the hardware, microphone or speaker
[5] Wideband not applicable in an ATA device
[6] Only CPT generation supported

GIPS VoiceEngine Features	PC	Mobile[1] for Windows	Mobile[1] for Symbian	Mobile for iPhone	IP Phone[1]	ATA[1]
Sound Device Selection	√	-	NA	-	√[7]	NA
Microphone Boost [8]	√	NA	NA	NA	NA	NA
Multiple Call Channel Support	√	√	√	√	√	√
Audio File Handling						
File Playback	√	√	√	√	√	√
File Recording	√	√	√	√	√	√
File Conversion Calls	√	√	√	√	√	√
Non Real-time RTP to File Conversion	√	√	√	√	√	√
External Media Processing	√	√	√	-	√	√
Push-to-Talk	√	√	√	-	√	NA
Audio Conferencing Client Mixing						
Add or Delete Mixing Participants	√	√	√	√	√	√
Number of Supported Channels	32	2	2	2	2	2
CPU Load	√	√	√	-	NA	NA
Participant Placement	√	NA	NA	NA	NA	NA
Voice Quality Enhancement						
Acoustic Echo Cancellation	√	-	-[9]	-	√	NA
Acoustic Echo Suppression	√	√	√[6]	√	√	NA
Network Echo Cancellation	NA	NA	NA	NA	NA	√
Automatic Gain Control	√	√	√	√	√	√
Noise Suppression	√	√	√	√	√	√
Noise Cancellation	√	√	√	√	√	√
High Pass Filter	√	√	√	√	√	√
VQE Instantaneous Metrics Measurement	√	-	-	-	-	NA
Video Synchronization	√	√	-	-	√	NA

[7] Available if supported by the OS
[8] Applicable only for some Windows-based sound devices
[9] On Nokia S60 3rd or 5th Ed. devices we can use the built in AEC. Please check the specification.

GIPS VoiceEngine Features	PC	Mobile[1] for Windows	Mobile[1] for Symbian	Mobile for iPhone	IP Phone[1]	ATA[1]
RTP Protocol						
Codec Payload Type Settings	√	√	√	√	√	√
FEC	√	-	-	-	√	√
RTCP	√	√	√	√	√	√
External Transport Protocol	√	√	√	√	√	√
Network QoS [10]						
TOS	√	√	√	-	√	√
Windows GQoS	√	-	-	-	-	-
Statistics						
Call Reports	√	-	-	-	-	NA
RTCP Statistics	√	√	√	√	√	√
Telchemy VQmon Support	Optional	Optional	Optional	-	Optional	Optional
Encryption						
External Encryption	√	√	√	√	√	√
Secure RTP	√	√	√	√	√	√

Table 1 **GIPS VoiceEngine Configurations and Features**

[10] Support dependent on platforms. Please check the specification.

Use Cases

In the use cases described below, all voice clients are not required to be GIPS voice-enabled endpoints, but quality is much better when GIPS technology is available on all VoIP clients due to the inclusion of GIPS advanced Voice Quality Enhancement and Packet Loss Concealment techniques. VoIP endpoints can be registered with a SIP server (or other call management architecture) which assists in the setup of the call.

Use Case One: Peer-to-Peer Voice Call

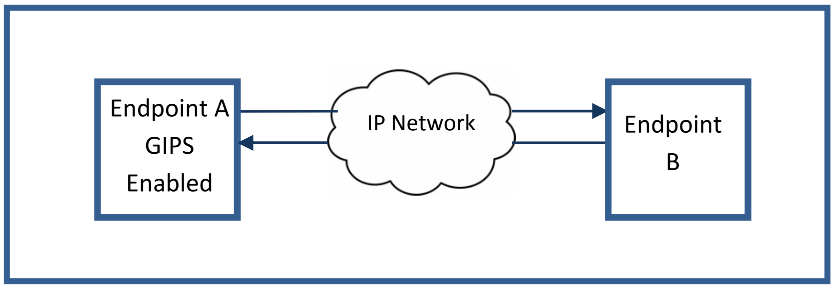

Figure 4 GIPS VoiceEngine in a Peer-to-Peer Voice Application

In this scenario, GIPS technology is integrated into an application that is installed on the endpoint A. VoiceEngine sends / receives voice streams to / from the other endpoint.

Use Case Two: Web-based VoIP Client

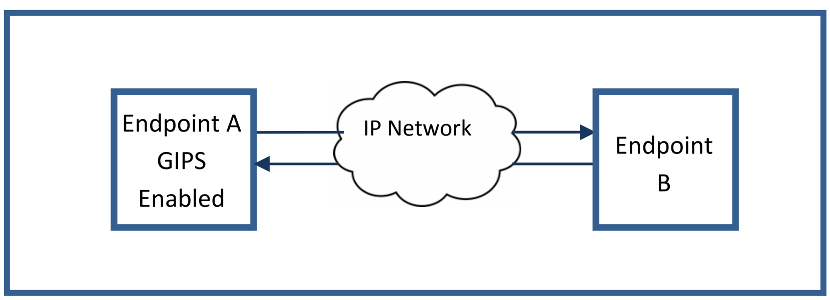

Figure 5 GIPS VoiceEngine as a Thin Video Client

VoiceEngine can be used in a web-based voice client downloaded as an ActiveX component. In the above figure, the endpoint A is a GIPS voice-enabled web client. The endpoint B can be either a web client or a client application. VoiceEngine sends / receives video streams to / from the other endpoint.

Use Case Three: Multi-party Voice Calls

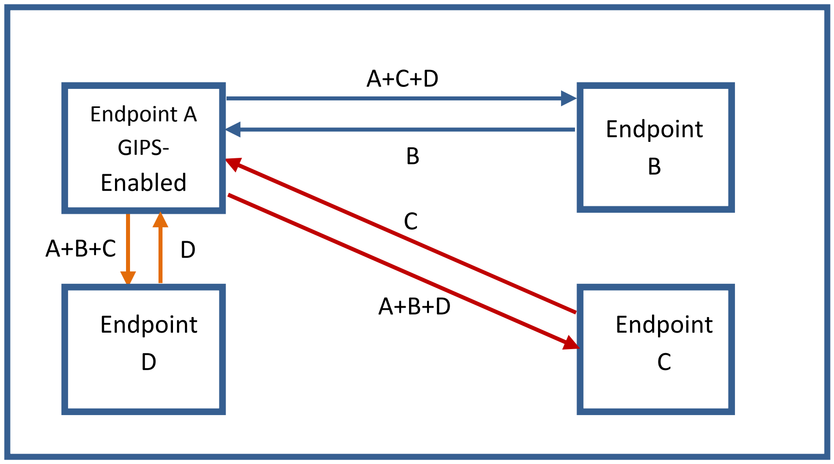

Figure 6 GIPS VoiceEngine in a Multi-party Voice Call

In this scenario, GIPS technology is integrated into an application that is installed on the endpoint A. Endpoint A initiates a multi-party voice call and performs all of the audio mixing. Other endpoints send media streams to endpoint A. In order to prevent endpoints B, C, and D from hearing their own voice in the mixed stream, VoiceEngine creates a unique mixed stream for each endpoint that contains only the audio streams from all other endpoints and returns the unique mixed stream to each client. VoiceEngine can support up to 32 channels in a multi-party voice call, depending on the processing power of the PC or device.

Chapter 4. Technical Specifications

Supported Platforms

The platforms supported today are listed below. For further information about the OS version numbers for the supported platforms, please refer to **GIPS VoiceEngine - Release Notes**.

GIPS VoiceEngine PC

Supported OS: Windows 2000, XP, Vista, Mac OS X, Linux

Processor: Minimum required is Intel Pentium III 1.0 GHz or equivalent

Note:
If you load your CPU up to ~100%, voice processing and capabilities to handle the IP packets will be affected, leading to degraded voice quality.

GIPS VoiceEngine Mobile for Windows

VoiceEngine Mobile for Windows is currently supported on the following platforms:

- Windows Mobile
- Windows CE

Note:
For the list of Windows Mobile limitations and devices tested by GIPS, please refer to **GIPS VoiceEngine Mobile for Windows - Integration Note** and **Tested Windows Mobile Hardware Devices.**

GIPS VoiceEngine Mobile for Symbian

VoiceEngine Mobile for Symbian is supported on Nokia S60 3rd and 5th Editions.

Note:
- For the list of Symbian limitations and devices tested by GIPS, please refer to **GIPS VoiceEngine Mobile for Symbian - Integration Note**.
- VoiceEngine Mobile for Symbian does only support full duplex voice for a defined list of mobile devices. Please contact your GIPS sale representative to receive the list of supported full duplex devices.

GIPS VoiceEngine Mobile for iPhone

VoiceEngine Mobile for iPhone is currently supported on iPhone OS. It supports iPhone and iPod Touch 2nd generation devices.

> **Note:**
> For the list of iPhone limitations and devices tested by GIPS, please refer to **GIPS VoiceEngine Mobile for iPhone - Integration Note**.

GIPS VoiceEngine IP Phone

VoiceEngine IP Phone can be embedded into a number of platforms, including but not limited to ARM, TI, and ADI chipsets. For the most updated list of supported platforms and their requirements, please contact your GIPS sales representative.

GIPS VoiceEngine ATA

VoiceEngine ATA can be embedded into a number of hardware platforms. For the most updated list of supported platforms and their requirements, please contact your GIPS sales representative.

GIPS Audio Codecs

The four GIPS codecs that are included in GIPS VoiceEngine are designed specifically to maintain high quality in packet networks.

iLBC

GIPS iLBC delivers basic speech quality better than G.729A and similar to G.729E, while offering substantially better quality over congested networks with packet loss.

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.

GIPS iSAC

iSAC is an adaptive 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.

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.

Additional Audio Codecs

G.711

G.711 is the international standard for encoding telephone audio on 64 kbps channel and is included in all VoiceEngine products.

GIPS VoiceEngine can handle any external codec. Examples of codecs in commercial use are:

- G.729
- G.729.1
- G.723.1
- G.722
- G.722.1
- GSM Fullrate
- GSM AMR

Note that external codecs are optional components in VoiceEngine.

Chapter 5. Product Features

Please note that some features may not be available on all platforms. Refer to **Table 1 GIPS VoiceEngine Configurations and Features** on page 17 for the list of features available for each configuration/ platform.

Speech Processing

NetEQ

GIPS NetEQ is an advanced jitter buffer and packet loss concealment unit that delivers dramatic improvements in sound quality while minimizing latency for IP telephony systems. This one-sided, embedded solution enables high quality voice over networks, even with poor infrastructure. NetEQ reduces jitter buffer delay 30-80 ms compared to the best alternative solutions and automatically provides a solution to the clock drift problem present in VoIP hardware. NetEQ is codec independent and available in the following modes:

Default: This is the standard mode for VoIP calls. The trade-off between low delay and jitter robustness is optimized for high-quality two-way communication. NetEQ's packet loss concealment and signal processing capabilities are fully employed.

Streaming: In the case of one-way communication such as a passive conference participant, a webinar, or a streaming application, this mode can be used to improve the jitter robustness at the cost of increased delay.

FAX: When this mode is selected, NetEQ will do as few delay changes as possible, trying to maintain a high and constant delay. Meanwhile, the packet loss concealment efforts are reduced.

VAD/DTX/CNG

GIPS' Voice Activity Detection (VAD), Discontinuous Transmission (DTX) and Comfort Noise Generation (CNG) software components are used to reduce the average data rate during silent periods of speech. It is possible to enable and disable silence suppression and comfort noise generation functionality.

Voice Activity Detection

GIPS VAD reliably detects the presence or absence of speech.

Comfort Noise Generation

Comfort Noise Generation (CNG) analyzes background noise to generate a comfort noise signal.

Comfort Noise Generation in Absence of Audio Packets

VoiceEngine allows configuring the behavior of comfort noise generation in cases where no audio packet is received for reasons other than a silent period. Push-to-talk, severe and bursty packet loss where the PLC cannot recover, and communication interruption due to network anomalies are some examples of such cases.

Bandwidth Extension

VoiceEngine supports bandwidth extension on the receiving side, upsampling the received wideband codec (16 kHz) to super wideband (48 kHz) for richer user experience.

Telephony

DTMF Detection

VoiceEngine can detect in-band or out-of-band DTMF tones as an option. Out-of-band DTMF tones are first detected by GIPS NetEQ and then processed as in-band tones. When a DTMF tone is detected, the application is notified via an event from VoiceEngine. The following DTMF tones are detected:

Detected Event/Digit
0-9
*****
#
A-D

DTMF Generation

VoiceEngine can generate and send DTMF tones/events in-band or out-of-band, according to RFC 2833 and RFC 4733. It is also possible to set VoiceEngine so that when a DTMF tone is sent, the same DTMF tone is played out on the speaker.

Call Progress Tone (CPT)

VoiceEngine supports Call Progress Tone generation and detection. Call Progress Tones in telephony are audible tones that are sent from the PSTN or a PBX to the calling parties to indicate the status of the phone call. For additional information on CPT, please contact your GIPS sales representative.

Caller ID (CID) Generation

GIPS VoiceEngine ATA supports CID generation according to Telcordia/ Bellcore, ETSI, and NTT DTMF standards. For additional information on CID generation, please contact your GIPS sales representative.

Audio Level Control

Audio can be configured according to the users’ preferences or by using the default settings as described below.

Speech Level Indicator

The speech level indicator provides a filtered speech input or output level that is suited for display with a progress bar or similar indicator. The progress bar can then be divided into different color fields such as .

Volume Scaling

Volume Scaling enables the change of the output signal level if there is a problem with the signal level.

Microphone and Speaker Volume Control

VoiceEngine allows users to set microphone and speaker levels.

Mute

It is possible to mute the microphone completely without affecting the sound device volume. Note that data (zeros) is still being transmitted.

Sound Card/Device

VoiceEngine handles recording and playout on the sound card and allows the user to set microphone and speaker volume using a simple interface.

Both recording and playout are implemented so that the encoding/decoding complexity and the delay are minimized. The playout method is adaptive to give the best possible performance independent of the sound card, computer and operating system.

Sound cards and operating systems usually have built in re-sampling filters; however, the quality of these varies. To ensure high quality audio, VoiceEngine is equipped with re-sampling filters designed by GIPS to have high quality.

Sound Device Selection

VoiceEngine can return the list of current and default sound devices. It can automatically select the default sound device in Windows/Linux or Mac. It is also possible to select devices other than the preferred devices, such as an external USB device.

Microphone Boost

It is possible to disable the microphone boost on most Windows based sound devices.

Multiple Call Channel Support

VoiceEngine supports several audio channels. For example, it can send, receive, and play out several audio streams. This enables functionality such as:

- Call Waiting
- Call Hold

Audio File Handling

VoiceEngine supports several audio file handling functions, including the common recording of a voice call, as well as playback of stored speech or music files.

File Playback

It is possible to play background music locally, stream music or mix music with the microphone.

File Recording

Recording can be done from three different sources:

- The microphone, regardless if a call is started or not
- The playout of a specific audio channel. If the channel is currently not playing, nothing will be recorded

- The entire call. At least one channel must be sending to be able to record the entire call. For a multi-party call, all sources are mixed with the microphone signal

The recorded data can be output to either a file or to an OutStream object. Wav files can only be output to files directly, since the header needs to be updated after the recording is finished.

The following compression options can be applied to the file recording:

- No compression, 16 kHz PCM samples
- ILBC compression, as specified in RFC 3952
- AMR compression, as specified in RFC 3267
  Note that AMR must be included in VoiceEngine for this mode to be supported
- WAV files
  - Uncompressed with sampling frequency 8000 or 16000
  - G.711 u-law
  - G.711 A-law

File Conversion Calls

VoiceEngine can save the audio file in a compressed format before storing or sending the file via email.

This feature can be used to convert a .wav file of arbitrary PCM format to a file with PCM 16 kHz 16bits/sample.

The currently supported formats are either iLBC or AMR codec. The specification of the file formats can be found in RFC 3267 for AMR and in RFC 3952 for the iLBC codec. Note that the codec must be enabled in VoiceEngine to be used for file compression.

Non Real-time RTP to File Conversion

GIPS VoiceEngine enables users to very quickly convert stored RTP packets to a file. Please note that a timestamp for each packet is required to reproduce the call.

External Media Processing

It is possible to add custom voice processing effects to the audio in VoiceEngine, such as a “voice changer” or “voice disguise”.

Push-to-Talk

Push-to-Talk (PTT) makes bandwidth-efficient communication possible over IP networks, providing direct one-to-one, and one-to-many, voice communication service to soft VoIP clients.

The principle of PTT is simple; VoIP calls are started with a push of a button and connected directly to an active talk group. PTT calls are one-way communications, while one person speaks, the other(s) listen. With just one press of a button, speaker(s) take turns responding to each other on a first-come-first-served basis.

PTT services are also referred to as walkie-talkie services.

Audio Conferencing Client Mixing

VoiceEngine supports several channels. It can send, receive, mix and play out several audio streams at the same time. This feature enables functionality such as multiple-party conferencing.

Add or Delete Conferencing Participants

It is possible to add and delete conferencing participants during a call.

Number of Supported Channels

An application can display to an end user the maximum number of voice channels supported by VoiceEngine for their configuration.

CPU Load

VoiceEngine can report used capacity in percent. This metric can be used to get an understanding of how many additional channels can be added to the current application. If this value is higher than 70%, it is not recommended to add any more voice channels to a conference. This function is only supported on Windows.

Participants Placement

VoiceEngine provides support for Participant Placement. This feature can be used to separate participants in a conferencing scenario by panning them differently from left to right speaker. This effect is mostly noticed when using headphones. Note that the panning is only done on the receiving side and therefore in a conference scenario relay mode is needed.

Voice Quality Enhancement

Voice Quality Enhancement (VQE) module improves the quality of VoIP conversations by removing echo and noise, and by adjusting speech levels to achieve a consistent and comfortable listening experience. GIPS VQE is the first product designed specifically for VoIP that combines all of the speech processing components necessary to meet tough quality demands in one comprehensive module.

Acoustic Echo Cancellation

Acoustic Echo Cancellation (AEC) is optimized to eliminate echo caused by acoustic feedback in real-time communications. This highly efficient speech processing software cancels acoustic feedback without audible echo or clipping.

Acoustic Echo Suppression

Acoustic Echo Suppression (AES) is low complexity software that eliminates echo caused by acoustic feedback in devices in which complexity is an issue, such as Mobile Phones, PDAs and low cost devices. It is possible to set the level of echo suppression.

High Pass Filter

The near-end microphone signal may contain a DC bias and/or (very) low frequency noise that may degrade the performance. This feature enables or disables a high-pass filter to remove unnatural disturbances.

Automatic Gain Control

Automatic gain control (AGC) automatically adjusts the level of an audio signal to achieve consistent and comfortable audio levels. In a VoIP call, the audio levels depend on user audio settings, equipment type and environment. As a result, the AGC feature is essential to avoid signal clipping or too low signal level.

The AGC can be set in 4 different modes; Off, Analog, Digital, Analog and Digital. Note that analog control requires the user to provide the coupling between the mixer and the VQE.

Noise Suppression

Noise Suppression (NS) suppresses background noise without distorting voice quality. The Noise control can be set in 4 different modes; Off, Mild, Medium, and Aggressive (Noise Cancellation).

Noise Cancellation

Noise Cancellation (NC) removes background noise without affecting voice volume. The NC is active when the noise control mode is set to “Aggressive”.

VQE Metrics Measurements

VoiceEngine can be set to measure the instantaneous speech level, noise level, and echo for all active channels of the received and transmitted signals.

Speech and noise level measurements are performed according to ITU-T P.56. The internal measurement interval is fixed and set to 1.5 seconds. All measurements and calculations take place in the VQE module (refer to Figure 3 GIPS VoiceEngine PC Architecture & Configuration on page 14).

VoiceEngine measures:

- Echo Return Loss (ERL)
- Echo Return Loss Enhancement (ERLE)
- Remained far-end echo after echo cancellation (RERL)
- Amount of echo cancelled within the echo canceller at the point just before its NLP (A-NLP)

Network Echo Cancellation for Integrated Access Device (NEC-IAD)

GIPS NEC-IAD is an efficient speech processing echo cancellation software that provides excellent sound quality without audible echo or clipping. It is designed for use in ATA/IAD gateways at the edge of a packet network to cancel 4-2 wire echo conditions.

GIPS NEC - IAD is ITU-T G.168-compliant and cancels voice signal echo in an ATA/IAD. This flexible package offers adaptive comfort noise insertion and supports linear, A-law and µ-law inputs.

Video Synchronization

This feature can be used to synchronize video with the voice stream. You can set the maximum playout delay in order to synchronize the voice with video.

RTP protocol

VoiceEngine contains an RTP stack with the main features:

- Send and receive RTP packets according to RFC 3550
- Send and receive RTCP sender and receiver reports according to RFC 3550

Codec Payload Type Settings

VoiceEngine provides a list of supported codecs and their recommended settings, such as payload type number. For codecs using a dynamic payload type, it may happen that another SIP-based client has chosen a different dynamic payload type corresponding to the same codec. GIPS VoiceEngine default payload type can be changed to the same dynamic payload type number as was stated in the SIP INVITE SDP-message.

FEC

VoiceEngine supports forward error correction according to RFC 2198. Note that it's always preferred to use one of GIPS robust codecs instead of FEC, since FEC adds end-to-end delay. When a low bit-rate standard codec is used (e.g.: G.729, G.723.1 etc), FEC might be beneficial. The redundant payload in the FEC system must use the same codec as the primary payload. Once FEC is enabled, RTP data is sent with the RED payload instead of the codec payload. For more details, and SDP information, refer to RFC 2198.

RTCP

It is possible to enable and disable RTCP support in VoiceEngine. If enabled, RTCP statistics are calculated and Sender Reports will be issued. VoiceEngine can also generate RTCP Receiver Reports if used as a passive participant who does not send any RTP packets.

External Transport Protocol

The standard configuration of VoiceEngine uses RTP to transmit data over the network, and supports usage of an external transport protocol.

Sending Raw Data

It is possible to send packets with raw data over the channel, without using the standard RTP or RTCP header. For example, this feature can be used to simplify the implementation of STUN.

Network QoS

The following fields can be set to prioritize VoIP if implemented by the network:

TOS

It is possible to set the TOS (Type Of Service) field in the IP header in the outgoing stream.

Windows GQoS

Since TOS setting is not supported in Windows 2000 and Windows XP, it is possible to use the Windows GQoS API to modify the Diffserv codepoint and also the 802.1p marker bits, when supported. This is done by setting a GQoS service level that the Windows operating system will map to a Diffserv codepoint and to an 802.1p setting. The default values, copied from http://msdn.microsoft.com, are listed in the table below.

Service Type	DSCP	802.1p
Network Control	30 (6)	7
Guaranteed Service	28 (5)	5
Controlled Load	18 (3)	3
Qualitative	0 (0)	0
All other traffic	0 (0)	0

Table 2 Default values for GCOS service levels

Note:
This setting is not supported for the Mac and Linux platforms

Statistics

The statistics that VoiceEngine provides can be used:

- By an application programmer to make decisions about different actions depending on the network quality
- To inform an end-user of the network quality
- By the Enterprise or Service Provider to make decisions on the network configuration

VQE Instantaneous Metrics

VoiceEngine can provide real-time reports on:

- Speech level for the transmitted and received speech signals
- Noise level for the transmitted and received signals
- Echo level for the near-end and far-end signals

DeadOrAlive Metrics

VoiceEngine provides the status of each channel by assigning a Boolean value to the metric DeadOrAlive. The decision is based on a mix of variables for each channel such as time since last valid RTP packet was received, comfort noise state, and some additional internal factors.

Call Report

VoiceEngine can generate call reports in text files, containing a summary of all the metrics listed below:

- Long-term speech and noise/silence level metrics (min, max, average) for receiving and transmit sides according to ITU-T P.56
- Long-term echo metric (minimum, maximum and average) of the following in dB: ERL, ERLE, RERL (ERL + ERLE) and A_NLP
- Long-term minimum, maximum and average levels for Round Trip Time (RTT)
- Total amount of dead and alive connection detections during a VoIP session

RTCP Statistics

The RTCP statistics are computed according to RFC 3550 with parameters, such as jitter, delay and packet loss.

Telchemy VQmon Support

The Telchemy VQmon®-EP (End-Point) can be integrated and supported by GIPS VoiceEngine. The generated VoIP metrics are used by Telchemy VQmon to produce MOS and R factor call quality scores and diagnostic data. VoiceEngine has also implemented support for both RFC 3611 (RTP Control Protocol Extended Reports) and VQMon alert handler that trigger a specified function to be called upon a certain quality condition. Please see Telchemy VQMon documentation for more details around VQmon-EP.

Please note that VQMon software needs to be licensed separately from Telchemy and is an optional feature in VoiceEngine.

Encryption

VoiceEngine can support a variety of encryption schemes, depending on customer need.

External Encryption

VoiceEngine provides an interface that enables inclusion of an external encryption scheme to the RTP stream.

Secure RTP

VoiceEngine can be delivered with a reference implementation of Secure RTP (SRTP) using an open source implementation available at http://srtp.sourceforge.net/. This implementation is specified in RFC 3711.

Error Handling

VoiceEngine provides a full set of error handling codes to facilitate correct integration, as well as handling of run-time failures in the PC. VoiceEngine can also generate a trace file to facilitate debugging and trouble-shooting.

GIPS Sample Code

In addition to the VoiceEngine software libraries, GIPS also provides sample integration code to further speed integration.