

GIPS VoiceEngine™ Embedded for ATAs

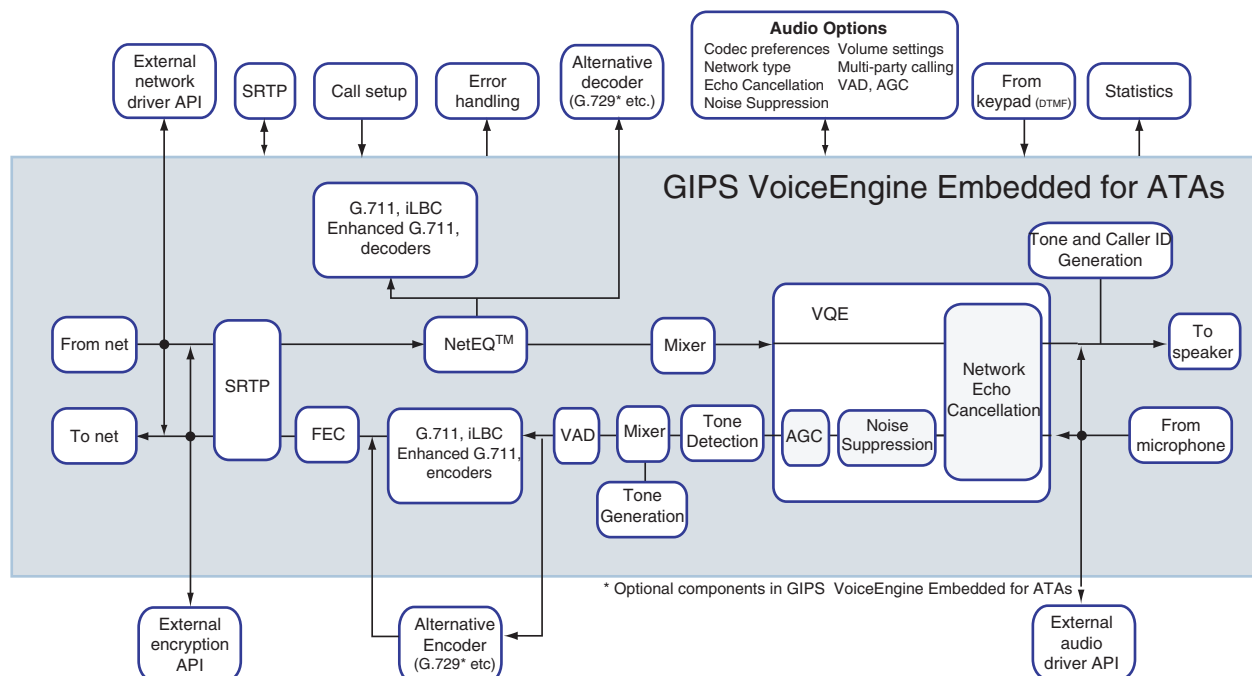
Consistent and unmatched voice quality across platforms

GIPS VoiceEngine Embedded for ATAs is an extension to the world-renowned GIPS VoiceEngine product family. Building on the success of GIPS VoiceEngine, Global IP Solutions has created an embedded solution that meets the highest voice processing requirements of today's hardware devices. This member of the VoiceEngine family is specifically designed to deliver the highest quality and ease of integration in CPU based residential gateways - often referred to as TAs (Terminal Adaptors) or IADs (Integrated Access Devices).

The comprehensive GIPS VoiceEngine Embedded package provides:

- Accelerated time-to-market and reduced development costs
- Simplified integration of sound-processing software with high-level API
- Field-proven, high quality solution
- Sophisticated speech enhancements including network echo cancellation
- Flexible and rich feature set

GIPS VoiceEngine Embedded for ATAs provides manufacturers with a solution that solves all of the complex voice processing functions in one integral piece of software that guarantees the best voice quality with the lowest possible delay. GIPS VoiceEngine Embedded for ATAs takes full advantage of Global IP Sound's patented technology and expertise to bring to market industry-leading VoIP enabled devices.





GLOBAL IP SOLUTIONS

GIPS VoiceEngine™ Embedded for ATAs

| Feature | VoiceEngine Embedded for ATAs |
|---|-------------------------------|
| Same high level API on all platforms | ✓ |
| Low latency and high packet loss robustness with GIPS NetEQ and GIPS codecs | ✓ |
| GIPS narrowband and wideband codecs | ✓ ¹ |
| Standard narrowband and wideband codecs | Optional |
| Forward Error Correction (FEC), RFC 2198 | ✓ |
| Re-samplings filters | ✓ |
| VAD/DTX/CNG | ✓ |
| Send DTMF | ✓ |
| DTMF Detection | ✓ |
| FAX support | Optional |
| Call progress tone generation | ✓* |
| Call progress tone detection | ✓* |
| Caller ID generation | ✓ |
| Compatible with any call setup protocol (e.g. SIP, H.323) | ✓ |
| RTP Handling or support for customer specific media transport protocol | ✓ |
| RTCP and statistics | ✓ |
| SRTP | ✓ |
| Support for addition of external encryption technology | ✓ |
| Voice Quality Enhancement | ✓ |
| - Network echo cancellation | ✓ |
| - Noise Suppression | ✓ |
| - Automatic gain control | ✓ |
| Audio mixing/Multi-party calling between 3 and 5 participants | ✓ |
| Multiple channel support (call waiting etc.) | ✓ |
| Support for multiple hardware ² and OS platforms ³ | ✓* |

1 = GIPS iSAC not included

2 = Currently available for ARM, MIPS, XScale and ColdFire CPUs

3 = Currently available for Linux, eCOS and VxWorks operating systems

* As GIPS is constantly innovating, please contact a sales representative for the most recent list of supported countries and platforms