GLOBAL IP SOLUTIONS

# GIPS iSAC

## Delivers wideband communication over low and high bit rate connections

iSAC is an adaptive VoIP codec, part of the GIPS MediaWare™ voice processing software suite, specially 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. Because of this high quality user experience, iSAC has become the codec of choice for wideband VoIP communications, and is deployed in hundreds of millions of endpoints.

The codec automatically adjusts transmission rates from a low of 10 kbps to a maximum of 32 kbps. This flexibility makes iSAC well suited for VoIP calls with wideband quality, real-time multimedia, conferencing, distance learning and multi-user gaming using IP network connections. It also handles non-speech audio, such as music and background noise,  exceptionally well.

iSAC performs well under packet loss, especially when used in conjunction with GIPS NetEQ™. This combination can deliver high quality voice, even in unmanaged, high-jitter and severe-packet-loss environments, delivering a greatly improved end-user experience while minimizing delay.

It is also available in a low-complexity mode, which is ideal for resource constrained devices, like mobile phones and PDAs. The low-complexity mode is also ideal for conferencing applications. This version facilitates the use of iSAC in almost any scenario, enabling high quality communication across multiple platforms.

**iSAC is ideally suited for**

- Applications and devices with constrained bandwidth requirements
- Applications and devices that benefit from wideband speech
- Providers and users who work with "real-life" networks and require reliability

**Features**

- Excellent trade-off between bit-rate and audio quality for low rate connections such as modems
- Automatically adjusts the bit rate for best quality or use a bit rate set to a fixed value
- Available in a low-complexity mode for resource intensive hardware devices
- Packet-loss robustness that exceeds current standards
- Uses the full 8 kHz available audio bandwidth for 16 kHz sampling
- Interoperable with hundreds of millions of existing VoIP endpoints

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| SPECIFICATIONS | |
|---|---|
| PACKET SIZE | Adaptive, 30 - 60 ms |
| BIT RATE | Adaptive and variable, range 10 - 32 kbps |
| SAMPLING RATE | 16 kHz |
| AUDIO BANDWIDTH | 8 kHz |
| IMPLEMENTATION | Floating point ANSI C<br>Fixed point version available for most common DSP platforms[1] |
| COMPLEXITY | Comparable to G.722.2 |
| ALGORITHMIC DELAY | Frame size plus 3 ms |
| QUALITY | Comparable to G.722.2 |

| SPECIAL LOW-COMPLEXITY FEATURES | |
|---|---|
| BIT RATE | 40 kbps average |
| COMPLEXITY | 6-10 MIPS[2] |
| QUALITY | Comparable to G.722.2 |

1. For a complete list of supported platforms, please contact a GIPS sales representative
2. Subject to change and dependent on platform