

## EnduraTalk GIPS Integrated SIP SOFTPHONE ver 4.0

ENCRYPTED || VOICE QUALITY ENHANCED || PRIVATE LABELED  
EMBEDDED WITH GIPS ADVANCED VOICE ENGINE || WORKS ON ALL WINDOWS VERSIONS

EnduraTalk is a *“Private Labeled Encrypted Voice Quality Enhanced PC Windows based Softphone”* that uses SIP signaling protocol & is integrated with “GIPS Advanced Voice Engine” for exceptional voice quality experience in practically all network conditions. Embedded with GIPS Advanced Voice Engine & Encryption/Decryption technology, EnduraTalk is a unique proposition sold to VoIP Service Providers around the globe. It features an intuitive and easy-to-use GUI (Graphical User Interface) that allows callers to make VoIP calls with ease, in a secure mode & with exceptional level of voice quality.

**GIPS Integration** - The EnduraTalk PC Based Softphone is integrated with “GIPS Advanced Voice Engine” to take care of the voice quality in pc based VoIP calling. This ensures that the caller gets exceptional voice quality experience no matter what is the type of internet connection is being used. GIPS is world leader in offering such Speech Processing solution. <http://www.gipscorp.com/>

EnduraTalk has been developed keeping special focus on “Voice Quality” controlled and enhanced features with the help of GIPS Advanced voice Engine:

- ❖ Adaptive jitter buffer
- ❖ Packet loss concealment for voice
- ❖ Packets Delay Management
- ❖ Automatic Gain Control (AGC) for voice
- ❖ Voice Activity Detection (VAD)
- ❖ Acoustic Echo Cancellation (AEC) and support for speakerphone
- ❖ Codecs Supported: G.723, G.729, G.711ulaw and G.711alaw, iLBC
- ❖ VAD (Voice Activity Detection) to enable voice transmission

## Features

Basic Features		Voice Quality
<ul style="list-style-type: none"> <li>■ Real-Time Account Balance Display</li> <li>■ Real-time balance calculation and display while the call is on. The balance keeps reducing while the call is on.</li> <li>■ Time remaining display in real-time</li> <li>■ Last 20 Calls CDR history in real-time itself</li> <li>■ Connect to PSTN and mobile phones (using SIP gateways &amp; soft-switches)</li> <li>■ Microphone &amp; speaker device selector</li> <li>■ Speakerphone</li> <li>■ Microphone &amp; Speaker Volume control</li> <li>■ Audio tuning wizard</li> </ul>	<ul style="list-style-type: none"> <li>■ PC to Phone, Phone to PC &amp; PC to PC Calling</li> <li>■ Authentication - HTTP Basic, Digest Authentication</li> <li>■ Basic Telephony - Do not disturb, Auto answer, Redial</li> <li>■ SIP UDP Support</li> <li>■ Codec negotiation as per defined sequence by Service provider.</li> <li>■ Free product version upgrades</li> <li>■ Skins</li> <li>■ Address book</li> <li>■ Call Timer</li> <li>■ Credit Balance Display</li> <li>■ Make installation packet</li> <li>■ Works with all kind of Internet connections</li> </ul>	<ul style="list-style-type: none"> <li>■ Adaptive jitter buffer</li> <li>■ Packet loss concealment for voice</li> <li>■ Automatic Gain Control (AGC) for voice</li> <li>■ Voice Activity Detection (VAD)</li> <li>■ Acoustic Echo Cancellation (AEC) and support for speakerphone</li> <li>■ Codecs Supported: G.723, G.729, iLBC, G.711ulaw and G.711alaw</li> <li>■ VAD (Voice Activity Detection) to enable voice transmission.</li> </ul>
Compatibility		Technical Capability Features
<ul style="list-style-type: none"> <li>■ MS Windows(2000/XP/2003/Vista)</li> <li>■ Work with Windows Audio.</li> <li>■ Audio call - Dial and receive phone audio calls</li> <li>■ Audio codecs - G.711 uLaw/aLaw, G.723.1,G.729, iLBC</li> <li>■ Supported SIP Methods - REGISTER ,INVITE,CANCEL, INFO,BYE, ACK, REFER, SUBSCRIBE,OPTIONS, NOTIFY,MESSAGE, UPDATE</li> <li>■ RFC supported - RFC 3261, RFC 3665, RFC 2833, RFC 2327, RFC 3264, RFC 3550, RFC 3263, RFC 3891,RFC 3515, RFC 3420, RFC 3892, RFC 3265, RFC 3666, RFC 3489, RFC 3920, RFC 3921, RFC 3922, RFC 3923, RFC 4622, RFC 4854, RFC 4979, RFC 3842,</li> </ul>		<ul style="list-style-type: none"> <li>■ NAT and firewall traversal using Encryption/Decryption</li> <li>■ Microphone and speaker device selector</li> <li>■ UDP, TCP and TLS for SIP transport</li> <li>■ Proxy/WWW authentication</li> <li>■ Connect to PSTN and mobile phones (using SIP gateway/softswitch)</li> <li>■ Security offering signaling &amp; media encryption via TLS streams</li> </ul>

## Functionality

- **PC to Phone** – Caller can make outgoing call to any telephone/Mobile
- **Phone to PC** – Incoming calls may be received on EnduraTalk. These incoming calls maybe from another instance of EnduraTalk on some other computer or from any telephone (like incoming DID service).

- **PC to PC** – Callers may indulge in voice calling among themselves. Service providers may offer free voice calling among it's users to further popularize and penetrate the service.
- **EnduraTalk** is meant for “**Service Providers**” who intend to procure a high quality softphone to cater to their customers. Such “Service Providers” might be “VoIP Service Providers”, Hosted/Managed Platform Providers, ITSPs, Broadband Operators, PC to Phone Service Providers and any other.
- Any service that requires a Desktop/Laptop based Softphone can utilize the power of EnduraTalk in their service offering.
- **Security**
  - The Softphone is also equipped with Encryption/Decryption technique whereby each packet (both signaling & RTP) are encrypted at the client side and then sent over IP to the server whereby they are decrypted. This ensures completely secure voice conversation. This also make sure that the softphone functions behind all types of firewalls & NAT. Along with this, it also works behind all blocked port situations

## Technology

- EnduraTalk is developed using C++ as it's language and has “GIPS (Global IP Sound) Advanced Voice Engine” integrated for it's Voice Quality handling.
  - GIPS Advanced Voice Engine takes care of the Media (RTP) part by managing the parameters:
    - Adaptive jitter buffer
    - Packet loss concealment for voice
    - Automatic Gain Control (AGC) for voice
    - Voice Activity Detection (VAD)
    - Acoustic Echo Cancellation (AEC) and support for speakerphone
    - Codecs Supported: G.723, G.729, iLBC, G.711ulaw and G.711alaw
    - VAD (Voice Activity Detection) to enable voice transmission.
- EnduraTalk is developed with C++, Visual Basic as development languages and can be installed on Windows platforms.

## Benefits

- **Voice Quality Handled** – Ascent is the only company in the world that has developed the capability to have GIPS integrated in our SIP softphone and that also has a model to sell the same to the service providers. That means, as a service provider, you have a unique edge over others by having GIPS voice quality advantage.
- **Security Through Encryption/Decryption** – Encryption/Decryption mechanism makes completely secure voice communication through EnduraTalk softphone, makes it work behind all kinds of firewalls & NAT and makes it work behind all kinds of blocked ports.
- **Customization / Support** – The application is open to customization and any required feature / functionality may be incorporated in short time. This would ensure that you can create a differentiation with varied unique feature set and also can change with the changing technologies and customer requirements.

- **Private Labeling** – EnduraTalk is very flexible on the GUI side. We can incorporate various skins (GUI) and that would help you in creating different brands in the market to tap more market segments.
- **Wide Interoperability** – EnduraTalk has been widely tested for interoperability with wide range of class 5 category softswitches like NexGE, Digitalk, Voicemaster / Sysmaster, Nextpoint, Asterisk, SER, PortaOne. EnduraTalk is interoperable with any standard SBC.