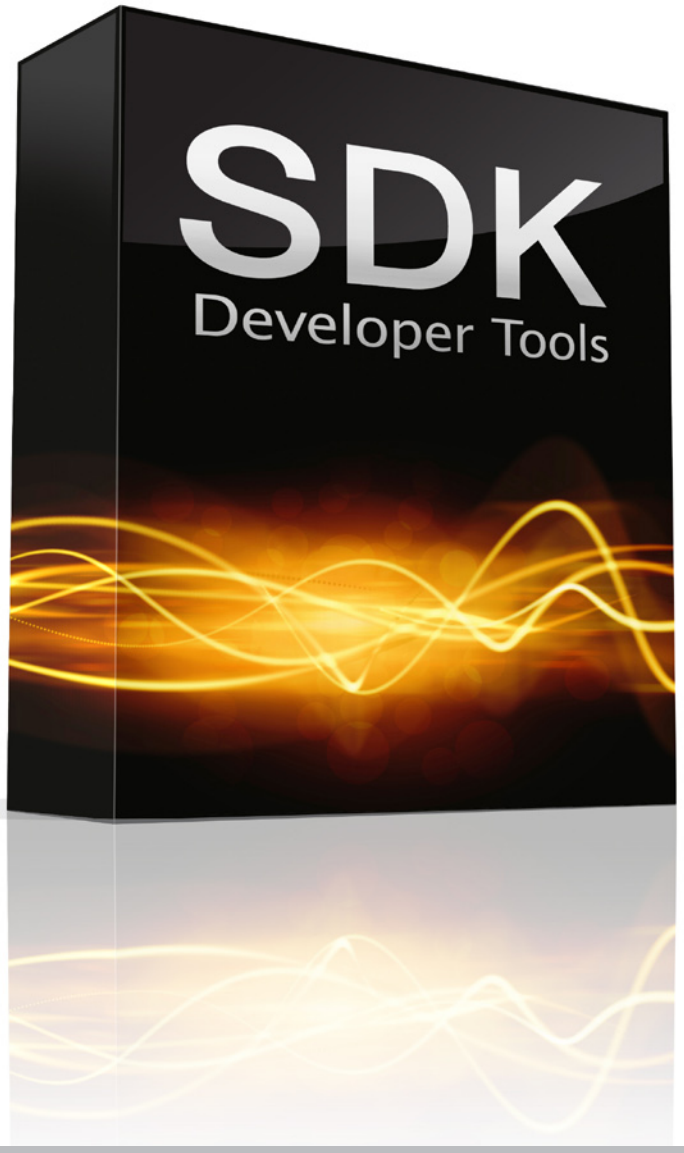
COUNTERPATH



Developer Tools

Solution Brief

CounterPath SDKs

Robust development tools for superior SIP-based VoIP applications

CounterPath offers three Software Development Kits (SDKs) to enable Service Provider, OEM and Enterprise developers to create customized SIP-based VoIP applications. Our SDKs enable flexible, rapid application development options for creating desktop softphone clients, Rich Internet Applications (RIA) and telephony extensions into existing enterprise and web applications.

Each of the SDKs is built upon our core libraries which have been field tested and deployed to hundreds of customers worldwide.

C++ Edition SDK

The CounterPath C++ Edition SDK offers access to our core softphone libraries, enabling engineers with solid VoIP experience to develop complex voice, video, IM and Presence applications. The basis for CounterPath's softphones, the C++ SDK allows the developer to build his own graphical user interface and applications, and implement different calling behaviors for features such as call answer, call transfer, conferencing and more.

Applications: Building of stand-alone softphone applications.
Skill level: Advanced VoIP knowledge required.

COM Edition SDK

The CounterPath COM Edition SDK is an easy-to-learn SDK that enables developers to quickly and easily build a stand-alone softphone with standard telephony features or integrate standard telephony features into existing Windows applications. Using high level programming languages such as C#, Visual Basic or Delphi and the SDK's simple Application Programming Interface (API), developers with little VoIP experience can develop rich voice, video, IM and Presence applications.

Applications: Integration of softphone functionality into existing Windows applications.
Skill level: Basic VoIP knowledge required.

Active X - Web Edition SDK

The CounterPath ActiveX Edition SDK enables the developer to build standard telephony features into their website, adding real – time communications capabilities within the web browser. The SDK offers an easy-to-use API for high level development programming environments such as Javascript, Adobe Flash and OpenLaszlo and is ideal for developers with little or no prior knowledge of telecommunications or VoIP protocols.

Applications: Voice-enabling of web pages.
Skill level: Basic VoIP knowledge required.

Key Features

- Carrier-grade audio and video codecs
- Security and encryption features (TLS, SRTP by request)
- Enhanced Quality of service (QoS)
- Conferencing
- Bandwidth scaling
- Contact Management
- IM and Presence (SIMPLE and XMPP)
- Open standards-based
- Interoperable with all of the major equipment vendors

Benefits

- Rapid market deployment
- Simplified integration with Microsoft applications
- Simplified integration with SIP and other VoIP applications
- Easy integration with third party web applications

What's Included

- Detailed Developer Guide
- Sample code
- Developer support options

SDK trial versions are available by request. Please contact us for more information.

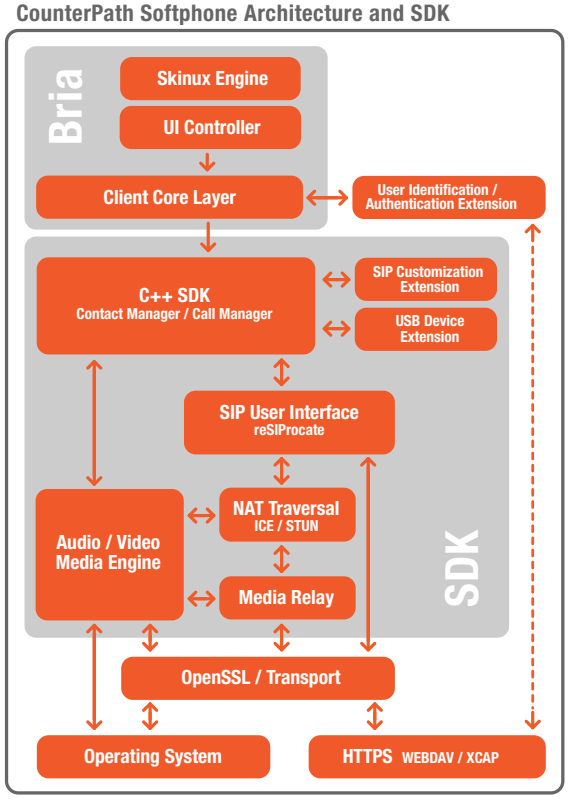
Architectural Overview

CounterPath's SDKs are built on open standards, leveraging on SIP (Session Initiation Protocol) to create, run and terminate multi-media sessions. They are built on ITU codec standards for voice and video (for example, G.711, G.722.2, G.723, G.726, G.729, H.263, and H.264), and the SIP SIMPLE and XMPP/Jabber standards for Instant Messaging and Presence.

NAT Traversal

CounterPath softphone clients are STUN, ICE and TURN enabled. STUN is a protocol that allows an endpoint to determine its IP address with respect to the STUN server. It also allows the endpoint to characterize the type of NAT or firewall it is located behind. ICE is a methodology for establishing connections between two endpoints by collecting and advertising a list of possible IP addresses that the client may be reached at. Some of these addresses are gathered using STUN. A client will attempt to establish contact with another client using the addresses advertised in SDP using the ICE candidates to determine the optimal path. In the event of a restrictive NAT/ firewall, a relay solution such as TURN is required to establish a call. TURN allows a client to acquire a public address which it can then advertise to another client. CounterPath is actively involved in the IETF standardization process for STUN, TURN and ICE.

CounterPath Softphone Architecture and SDK



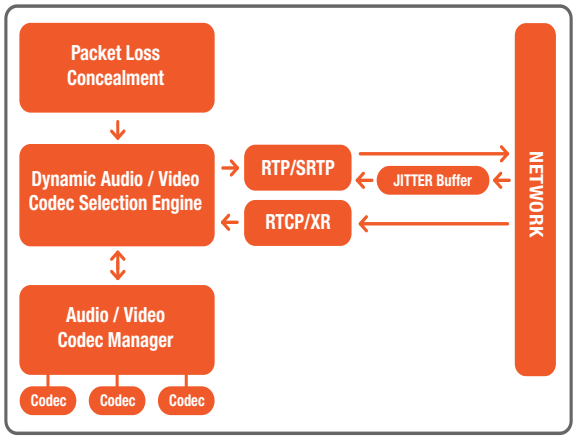
Audio and Video Engines

CounterPath's Audio and Video Engines select voice and video codecs to optimize the sound and video quality to the user's bandwidth. CounterPath softphones include a dynamic jitter buffer to re-order out-of-order packets, adjust to changing network delay, and compensate for existing network congestions and overloads. CounterPath clients support AGC (Automatic Gain Control) and AEC (Acoustic Echo Cancellation), implement several packet-loss-concealment algorithms, and noise-reduction mechanisms.

CounterPath client's audio/video media engine provides the ability to monitor network conditions and make adjustments to improve call quality to optimum levels if the link degrades. This means Voice and video are adjusted or calibrated to the user's or transport bandwidth using the following mechanisms: When the user manually configures the connection type (i.e. dialup, LAN, etc.), or when the automatic bandwidth detection feature is enabled, the codec suite is evaluated and modified based on the users bandwidth settings.

For example, under low bandwidth conditions G711u might be disabled, while G729 would be enabled. Codecs are also adjusted on the fly based on feedback from the remote party (RTCP reports).

CounterPath's Audio/Video Media Engine



Developer Tools - SDK Features

Telephony Features

- Multiple SIP lines
- Hold
- Do not disturb
- Call Ignore
- Call history
- Call transfer
- Voice/video call recording
- Voice/video conferencing

Enhanced Features

- SIP compliance to 3261 SIP standard
- Address book – remote storage with WebDav and XCAP
- Security via signaling encryption (TLS) and media encryption (SRTP)
- STUN and ICE NAT traversal and XTunnels for firewall traversal
- Enhanced Quality of Service for voice and video calls
- Instant Messaging and Presence using the SIMPLE and XMPP protocols
- Zero-touch configuration of audio and video devices
- Acoustic echo cancellation, automatic gain control, voice activity detection
- Support for a long list of narrow, wideband carrier-grade codecs including G.729
- Support for the H.263 and H.264 video codecs
- Automatic selection of the best codec based on the other party's capability, available bandwidth and network condition

SIP Extensions

- Send/Receive Out-Of-Dialog REFERS (with/without modification) & other SIP messages
- Subscription to SIP event packages during a specific call
- SIP preprocessor functionality-inspection/repair of SIP messages
- Adornment of SIP messages:
  - INVITEs
  - REGISTERs
  - NOTIFYs
  - REFERs
  - REINVITEs
  - MWI Subscriptions
  - Out-of-Dialog REFER sends
  - Out-of-Dialog SIP Message sends
  - Outgoing IM Messages

Supported Languages / Localizations

- CounterPath SDK C++ Edition is available in English (US and UK).
  Note: all documentation is available in English only

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