Manual

July, 2011

Flash SIP SDK Manual

ABTO Software

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INTRODUCTION

Trends indicate that in the future all websites will include an embedded telephone. The main reason for using embedded phones is that these phones offer a much more natural method for website visitors to contact a company than regular phone calls. Furthermore webphones provide a number of advantages over traditional telephones lines. For example if you use a webphone you will have information about which product or service the website visitor is interested in, because you will know. You can know which webpage the website visitor is on (the call arrives form the given webpage). You can also find out which pages have been visited by the caller. Phone calls are free for website visitors. Website visitors can make calls to reach you but you can also make a call to reach your website visitors as soon as they open your website. The main advantage of embedded phones is that you can get in contact with more website visitors than ever and therefore you can realize more sales.

Why do you need a webphone

Introducing a webphone into your website helps you convert more website visitors into buyers. It prevents you from losing international customers who would not call you normally because international calls are expensive. A webphone boosts your call center traffic more effectively. It helps your business to keep up with technological development. Moreover it allows customers to reach your company faster. How can it be used an embedded telephone can be used in various ways to improve your communication. Find some fields of applications below:

- ✓ Website visitors can establish calls from websites for reaching the sales or support team
- ✓ The staff can make outside calls from the Intranet since a fully featured webphone can be connected to the corporate PBX as a client

- ✓ Remote employees can reach your office extensions, or can dial out through your office by logging on to a password protected webpage. This is often the only option in Hotels or when you are on the go and a VPN is not in place
- ✓ Call centers can boost traffic, by automatic outbound calls targeting website visitors, plus the number of inbound calls will also increase
- ✓ System Integrators and Value added resellers (VARs) can use it in various projects: Click-2-call, Direct dial (Based on the visited webpage a direct call can be made to an expert),
- ✓ CRM, social communities, online shops, consultancy, conferencing, voice brochure
- ✓ Internet VoIP service providers can widen the portfolio of their services.

Incoming calls from a webbrowser

A webphone enables website visitors to establish phone calls directly from a website. This phone call can be directed to the corporate IVR, to a traditional office phone, to the corporate PBX, an outside line, or it can be transferred to a mobile phone or a traditional landline phone. Based on the caller ID you can easily recognize if the incoming call comes from the web. Optionally a webphone can also be integrated with a call assistant software that will open a popup window when an incoming call is received. In this popup window important data will be presented regarding the caller party (such as call history, webbrowsing history, CRM info, etc).

Outgoing calls to reach a website visitor

A webphone also allows you to call the website visitor when she opens your webpage. Seeing that the webphone is ringing most of the website visitors will accept the call. This feature is based on the fact that Abto Webphone can notify you when a website visitor opens a webpage on your website. You can use this notification to call the visitor. The visitor will notice that the embedded soft phone is ringing and she can decide to cancel the call or pick it up. The notification includes the following information:

- ✓ Which URL has been opened by the website visitor
- ✓ Which phone number can be used to call the website visitor

To be able to use this option Abto Webphone Server needs to be configured to assign a unique phone number for each website visitor.

Why choose ABTO's web phone

- ✓ It works immediately
- ✓ The webphone can be used immediately when a visitor opens the website. No plugins or external tools need to be downloaded.
- ✓ The calls are free
- ✓ Charges won't stop website visitors from calling.
- ✓ It is compatible with your existing phone system
- ✓ You can attach Abto webphone to a PBX.
- ✓ You can forward calls to office lines, to mobile phones or to any telephone line.
- ✓ It gives a true phone call experience for website visitors
- ✓ This is important because they will feel comfortable talking on this webphone.

TECHNICAL BACKGROUND

About ABTO Software Flash Phone SDK

Flash Phone is a web softphone, that is working in web browser without installation of any specific software. It operates using SIP (Session Initiation Protocol). The Phone is compatible with any SIP server like Asterisk, FreeSWITCH and OpenSER. Everything that your client need is internet connection, speakers and microphone. The Flash Phone users can call each other, to SIP phones, and any landline or mobile phone numbers. Calling to landline and mobile phones requires paid account at third party providers.

Quick Facts

- ✓ Flash Phone can be used on any computer.
- ✓ No need to install any additional software.
- ✓ Incoming/Outgoing calls processing.
- ✓ Conferences support.
- ✓ Easily customizable.
- ✓ Calling regular phone numbers at minimal cost, choosing the most convenient IP telephony provider.
- ✓ No NAT and firewall problems.
- ✓ Calls by means of SIP protocol.
- ✓ Sending DTMF signals.
- ✓ G.711, G.729 and iLBC voice codecs.

QUICK START GUIDE

This guide explains how to integrate Flash SIP SDK quickly in your application to evaluate its functionality. You can learn how to make a VoIP call, and supported protocols, networking and media information are also provided.

Flash SIP SDK is an excellent software development kit that allows you to establish VoIP calls easily and quickly. You can utilize this SDK to establish VoIP calls from your application and to ensure SIP and RTP communication. The great thing is that you do not need any previous knowledge about SIP and RTP protocols but it is enough to use this SDK! Also you can easily integrate Flash SIP SDK into your application and it is ready to make VoIP calls immediately!

Prerequisites

- ✓ Web browser
- ✓ Flash SIP SDK
- ✓ SIP Server or SIP account from your service provider

Getting started

With Flash SIP SDK you can easily establish VoIP calls by a SIP account. The SDK includes a softphone that demonstrates all the available options and functionalities with Flash SIP SDK.

How to make a call - Main steps

1. Open Flash SIP SDK
2. Register with a SIP account
3. Handle calls
4. Leave the page

Supported protocols - Transports

- ✓ UDP
- ✓ TCP

Networking

Flash SIP SDK is fully compatible with all PBX and ATA device that support SIP protocol.

Media

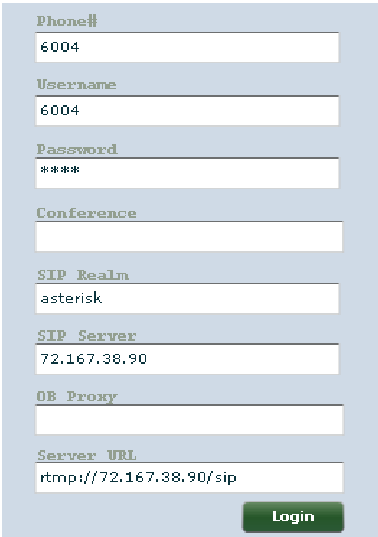
Flash SIP SDK supports the following codecs:

- ✓ G711 A-Law (PCMU)
- ✓ G711 u-Law (PCMA)
- ✓ G729
- ✓ Speex

Flash SIP SDK also enables you to send and receive DTMF tones and to play WAV files and record conversations in WAV format.

Basic configuration steps

In order to use our flash sip client you have to enter sip account details:



After clicking on login button you will be able to launch the following application interface:

In conclusion

To introduce a really effective VoIP solution into your application you only need to follow some very simple configuration steps. Without any previous special knowledge for SIP you can instantly and easily experience the benefits of VoIP technology with Flash SIP SDK!

FEATURES OF FLASH SIP SDK

Empower your system with the following powerful webphone related features and just discover how they can make your webpages much better!

CAPABLE OF DELIVERING CRYSTAL CLEAR SOUND

When you integrate Flash SIP SDK your application will be capable of delivering crystal clear sound for all browsers.

SUPERIOR VOICE QUALITY

Superior voice quality is also guaranteed due to the digital voice processing features.

EXTENDED CODEC SUPPORT

To achieve superior voice quality extended codec support has been included into Flash SIP SDK. Flash SIP SDK supports for both narrowband and wideband, codecs that's why it works with all type of Internet connections. The following codecs are supported to improve voice quality:

- ✓ G711 Alaw
- ✓ G711 Ulaw
- ✓ G729 Alaw
- ✓ iLBC

SIP PROXY AUTHENTICATION

Flash SIP SDK allows registering with the SIP proxy server by providing Login ID and Login password.

DIAL/RECEIVE PHONE CALLS

You can dial and receive phone calls through any SIP based server, gateway or Internet Telephony Service Provider (ITSP).

DTMF TONES GENERATION

VoIP SIP allows webpages to generate Dual Tone Multi Frequency (DTMF) tones.

MICROPHONE & SPEAKERS VOLUME

User can control Microphone and Speakers volume directly.

UDP AND TCP SUPPORT

User Datagram Protocol (UDP) and Transmission Control Protocol (TCP) are supported effectively.

COMPREHENSIVE CONFIGURATION SUPPORT

- ✓ Select media input/output devices (on-the-fly as well during a conversation/conference)
- ✓ Configurable ports (RTP, SIP UDP, SIP TCP, STUN, TURN, ICE)
- ✓ SIP proxy

CONNECTIVITY

It connects to a supported VoIP PBX or to a VoIP service provider over the Internet. Supports firewall pass-through (STUN/TURN).

SOURCE CODE

Full source code can be purchased.

RFC SUPPORTED

- ✓ RFC 2327
- ✓ RFC 2833
- ✓ RFC 3261
- ✓ RFC 3264
- ✓ RFC 3550

SUPPORTED SIP METHODS

- ✓ ACK
- ✓ BYE
- ✓ CANCEL
- ✓ INFO
- ✓ INVITE
- ✓ OPTION
- ✓ REGISTER
- ✓ SUBSCRIBE

BASIC TELEPHONY AND TELEPHONE FUNCTIONS

- ✓ Auto answer
- ✓ Call history
- ✓ Do Not Disturb(DND)
- ✓ DTMF
- ✓ Hold
- ✓ Multiple SIP lines
- ✓ Redial
- ✓ Transfer
- ✓ Voice call recording
- ✓ Voice conferencing