

eyeP Foundation v5.05 – Windows® edition

Communication over IP software development kit



Feature list

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1 Introduction

The purpose of this document is to provide an overview of the functions and features available in the eyeP Foundation product line.

1.1 Scope

The document is aimed at those wanting to know what the eyeP Foundation provides and the functions it can provide to the application hosting it. This document is neither a release note nor a specification or a developer's guide, which are separate documents.

1.2 Definitions, Acronyms and Abbreviations

PAEC	Perceptually motivated Acoustic Echo Canceller Audio or video signal processing used to suppress the echo effect.
AGC	Automatic Gain Control Audio or video signal processing used to adjust the microphone volume level so that callee does not hear the distortion that might be generated by too high microphone input.
CODEC	COder-DECoder Process of converting and compressing analog audio or video waveforms to digital information.
Conference	Ability for a user to set up an audio or video conference call with two or more parties.
DTMF	Dual Tone Multi Frequency. System that can accept touch tone and used in Interactive Voice Response (see IVR) menu systems like voicemail.
IM	Instant Messaging Technology that let users exchange text messages and files instantaneously.
IP	Internet Protocol A protocol used to communicate data across a network.
MWI	Message Waiting Indicator An indicator which signals that there is at least one voicemail message received.
Presence	A feature that allows to share information about their availability and location.
SIP	Session Initiation Protocol Standard signaling protocol used to establish a call session.
SIP account	Account that grants the user to make calls over IP.
OAMP	Operation Administration Management Provision A mechanism used by eyeP Media which allows remote management and provisioning of softphones deployed, based on XML file of parameters.

2 Product Features

This section briefly depicts the eyeP Foundation features, at a high level, without giving any details about how the features are implemented.

2.1 Overview

The eyeP Foundation is a SIP based software development kit (SDK) – bundled as an ActiveX component - used to integrate communication over IP technology into applications. In short, it embeds an audio & video streaming layer, a SIP signaling stack and the necessary call control features to establish a communication over the Internet.

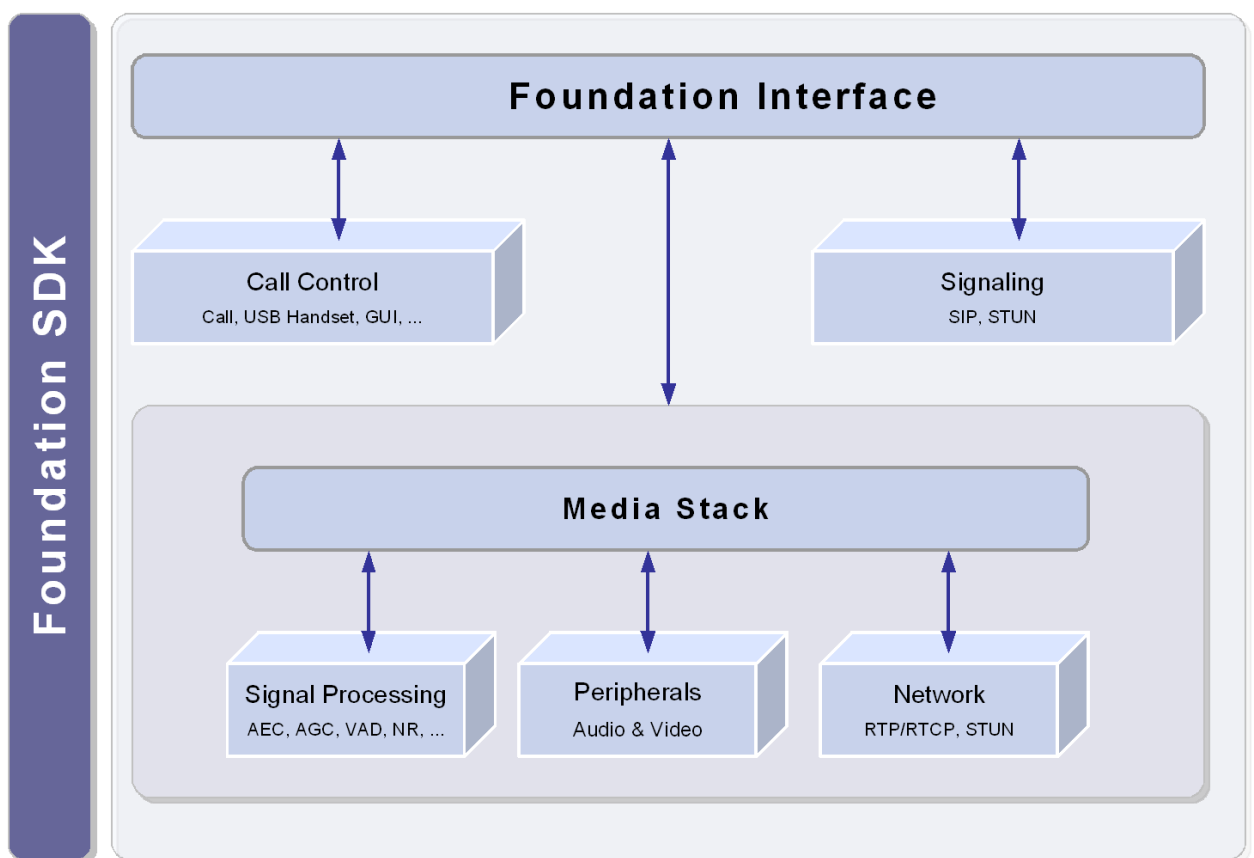


Figure 1: eyeP Foundation architecture

The goal of the eyeP Foundation is to provide a toolkit for developers who want to easily and quickly implement professional class solutions.

This is why - as shown in Figure 1 above - the toolkit proposes a simple interface (like a “Facade” design pattern) which embeds all the glue logic. This allows developers to use powerful functions like call(), drop(), answer() or transfer() without necessarily knowing the SIP signaling protocol implementation details.

This single API also increases the reliability of the solution, since this approach prevents the developer from making mistakes and keeps the communication dialog consistent.

Feature	Description
Basic call service	Allows to place or receive calls, hold/unhold calls and indicates voice mail reception.
Business call service	Allows to transfer calls, park & pickup ⁽¹⁾ calls and organize audio/video conferences.
Instant Messaging & Presence	Allows to chat and get presence status of contacts.
Audio	Allows handling of the audio peripherals (microphone and speaker) or playing an audio file, with a wide choice of audio CODECs. Audio experience can be enhanced with the help of signal processing modules like an Acoustic Echo Canceller.
Video	Allows to make video calls with webcams supported at the Operating System level, with a large choice of video CODECs.
Settings & Remote Provisioning	Allows to configure the solution, either locally or remotely, with more than 150 parameters, using XML files.
Network	Allows to finely adjust parameters related to the network (adapter, NAT/Firewall traversal, ...)

Table 1: List of major features

(1) Not available with the SIP stack yet. Please ask us for more information.

2.2 Basic Call Service

Basic call service features include standard call control functionalities:

- Place an outgoing call

The toolkit allows to place a call with either a standard phone number (calling a PSTN) or a SIP number (calling a softphone). If the option of multiple communication lines is provided, then it is possible to select a particular active line.

It is also possible for an application to request the toolkit to select or change the medium (audio or video), even when the communication is in progress.

- Receive an incoming call

The toolkit allows the reception of call(s). According to the number of communication lines provided with the toolkit, it is possible to receive multiple incoming calls on different communication lines, one call being active at any one time.

- Dropping a call

At any time, the toolkit allows to drop the communication on a line, including the rejection of an incoming call.

- Dialing DTMF

The toolkit can send DTMF digits using several methods:

- Info method (RFC 2976)
- Inband
- Outband (RFC 2833)
- Inband and outband

At reception, DTMF digits can only be received through either Info the method or the RFC 2833.

- Hold

The user may hold the communication at any time during a call session. This allows to put outside callers on hold so that user can answer or initiate another call. If video media is used, the toolkit will display the last video frame received.

- Mute

The toolkit allows to mute the microphone to prevent the outside caller (or parties when conferencing) from hearing the user. If video is enabled, it is possible to stop the video feed.

- Message Waiting Indication

The toolkit allows to handle message notifications which can be voice, fax, pager or multimedia messages. Typically, communication applications inform the user that messages are waiting.

2.3 Business Call Service

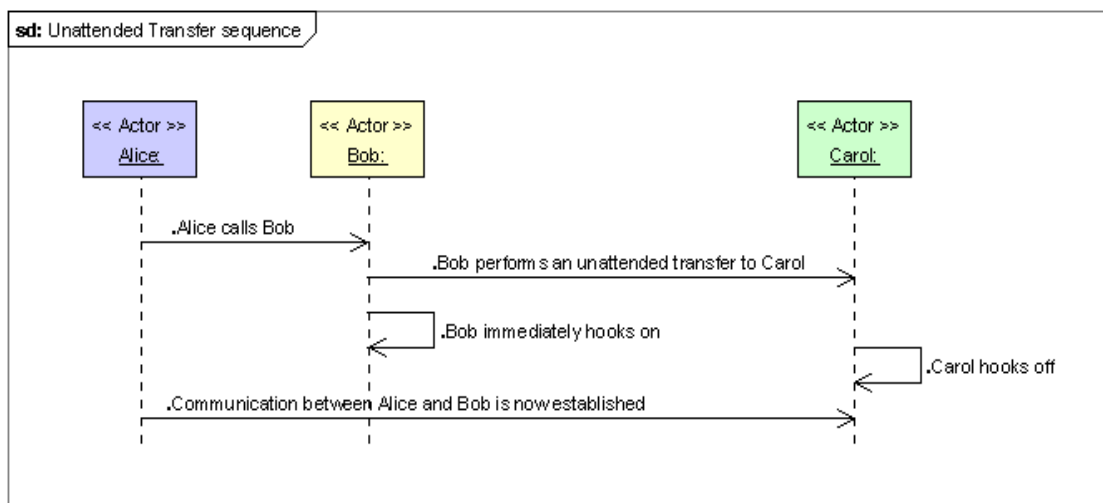
The toolkit exposes business telephony features for professional class solutions like call center applications or high-end softphones.

- Transferring a call

The user may transfer a communication at any time, either in blind or attended transfer modes.

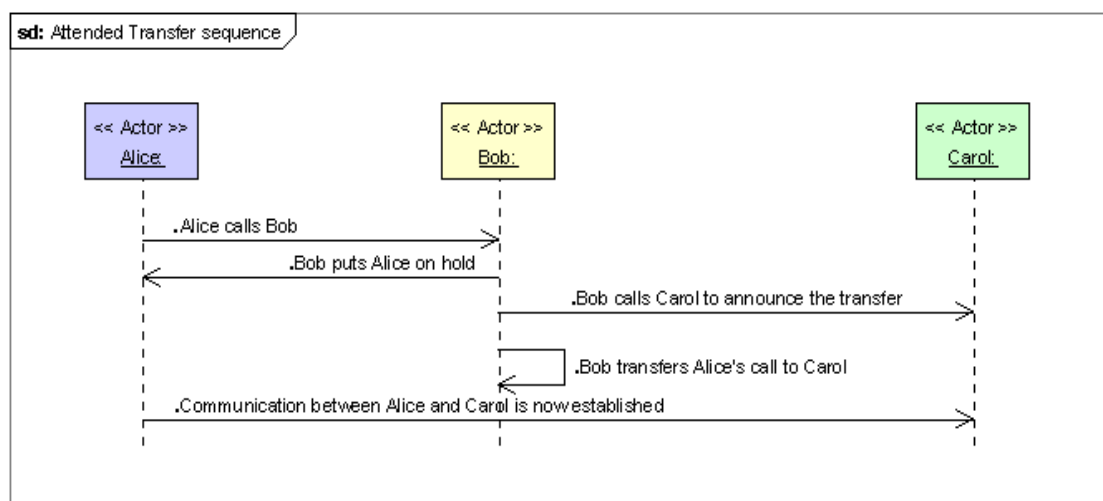
Blind (unattended) transfer

The user transfers the call to the party and goes on hook, without knowing if the call has been completed successfully.



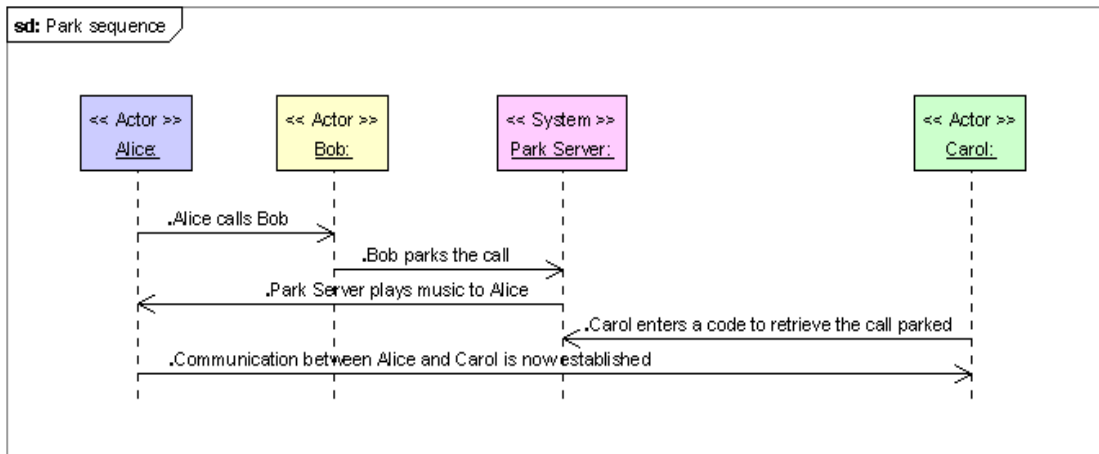
Attended transfer

The user transfers the call to somebody and waits to establish the communication before transferring the call.



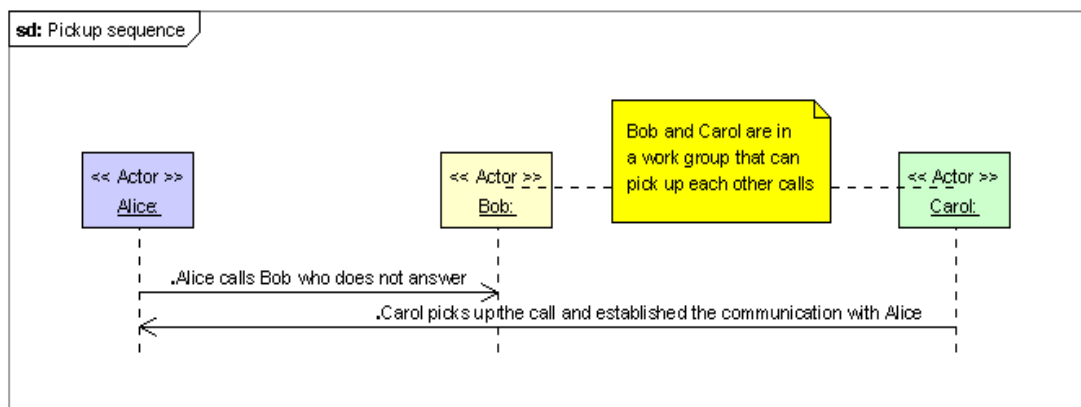
- Park (1)

The user parks a call with the help of a park server playing music and somebody retrieves the call.



- Pickup (1)

The caller attempts to call a person who does not answer and who is part of a work group in the company. Then, another party in the work group picks up the call.



(1) Currently, this feature is available with Sylanro's platform only.

- Conference

Audio conferencing can be organized in two ways:

Local conferencing

The communication application is in charge of organizing and managing the conference with other parties. This means that media streams from other participants must be handled locally by the conferencing machine. Therefore, the maximum number of participants depends on the available bandwidth and CPU characteristics of the hosting machine.

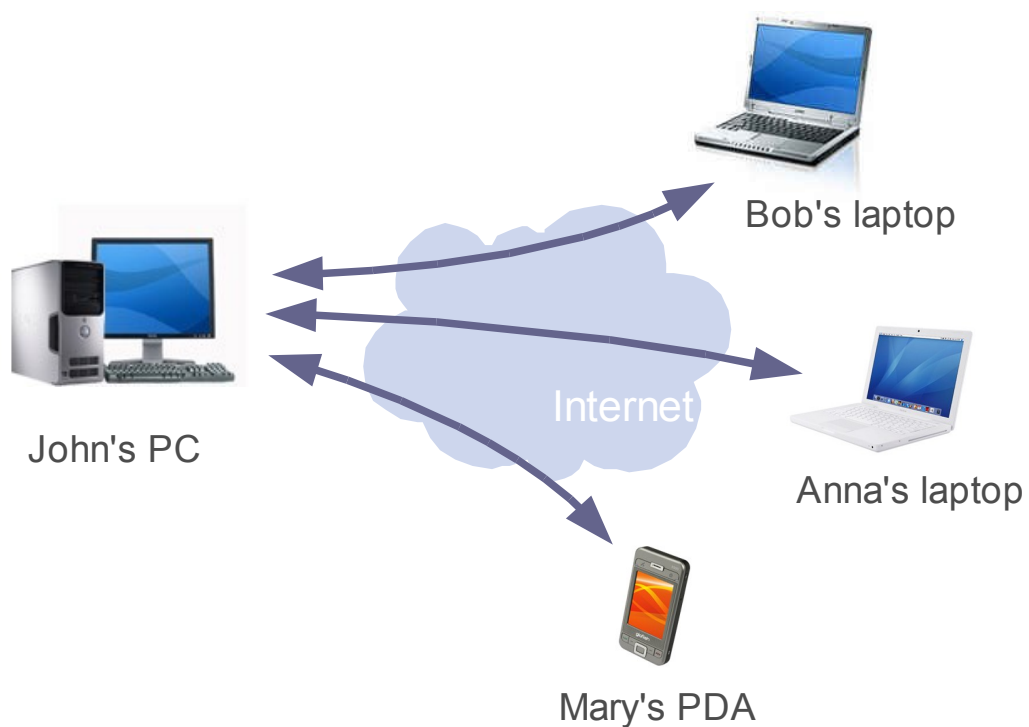


Figure 2: Local conferencing (here hosted by John's computer)

Network conferencing

In such case, a conferencing server is provided on the network and the communication application running on each user device should have been configured with the server address. Such configuration allows much more participants than local conferencing, due to better performance of the conferencing server and no bandwidth bottleneck.

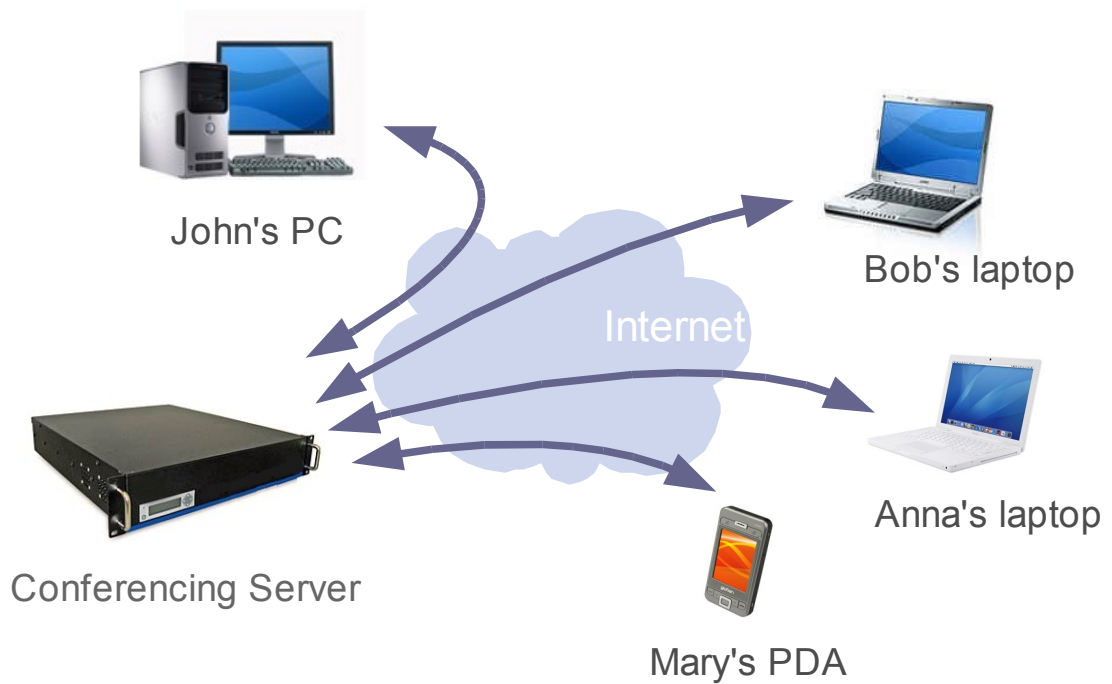


Figure 3 : Network conferencing

2.4 Instant Messaging & Presence

The toolkit goes beyond simple phone features and offers extra communication functionalities:

- Instant Messaging

The toolkit proposes to send or receive instantaneously a message to/from another party, which must use a SIP account.

- Presence

The toolkit proposes a simple SIP presence monitoring. Typically, application may display a presence status of the contacts based on their SIP registration states.

2.5 Audio

The toolkit accesses the multimedia peripherals (sound, video and microphone) through encapsulated OS specific separated modules. For example, when running under Microsoft OS, the toolkit uses modules based on the Wave API for audio.

- Play & record audio file

The toolkit proposes to record or play WAV audio file. Hence, application may record phone calls for example.

- Monitor the microphone and speaker

In order to improve the user experience, the toolkit can monitor any change in the microphone or speaker regarding sensitivity or volume. This allows to monitor in real time any change made at the application level, like reducing the speaker volume.

- Audio processing

Sound quality is enhanced with the help of advanced signal processing features like:

- Acoustic Echo Cancellation (AEC) which cancels any echo signal forwarded to the other party
- Noise Reduction (NR) which reduces the “background” noise
- Automatic Gain Control (AGC) which corrects any fluctuation of your voice.

- Sound adapter

The toolkit allows to get the list of all audio adapters and to select the active one. This is particularly useful when more than one sound card is detected, the user deciding which sound adapters should be used by default in order to get the best audio experience.

2.6 Video

Like the audio features, the toolkit allows to handle video, providing also an abstract interface to pilot the Operating System video peripherals. For example, when running under Microsoft OS, the toolkit uses modules based on the DirectX API for video.

- Video Adapters

The toolkit allows to get the list of all video adapters and to select the active one. For example, several webcams can be connected to the computer.

- Settings

Main parameters like the frame rate, the format, the video bandwidth and the CODEC are provided to fully control video quality .

2.7 Settings and Remote Provisioning

To configure the toolkit, two methods are proposed :

1. Setting the parameters manually

The parameters can be directly set by filling in the corresponding field values. It should be noticed that a default configuration window can be displayed if necessary (with the default Windows look & feel), avoiding developers to (re) write a specific configuration window.

2. Loading the parameters from an XML file

An XML file containing the eyeP Media's structure of parameters (OAMP parameters) can be loaded, either locally or remotely. For remote provisioning & management, the application is responsible to handle the remote requests and responses (HTTP/HTTPS requests for example).

It should be noted that setting parameters are applied when the toolkit is initialized. This means that parameters are kept in memory for successive call sessions, avoiding the application to reload the parameters after each call session.

The toolkit exposes more than one hundred and fifty (150) parameters to let developers finely adjust the communication application. A summarized list of main topics is shown here:

OAMP parameter topics
SIP account (user, password, ...)
Audio
Video
RTP
Network
System (URL, upgrade, ...)
Application settings

2.8 Network support

The toolkit provides methods, properties and parameters to handle network connections and security issues.

- NAT/Firewall traversal

The toolkit provides STUN support to address NAT & Firewall traversal. Note that TURN server is not supported (TURN is still in a draft state).

It should be noticed that eyeP Media recommends use of Session Border Controllers (SBC) to address NAT/Firewall issues.

- Network adapter

Like for audio and video, the toolkit can get the list of network adapters available (network card, WiFi, ...). It provides an event mechanism which can be used to handle any network connection change.

Notice that the network connection itself is supplied by the Operating System, hence this is not the role of the toolkit to establish an Internet connection.

3 Other Product Requirements

3.1 Supported standards

3.1.1 RFC standards

- SIP Working Group
 - RFC 2976 – INFO Method
 - RFC 3261 – Session Initiation Protocol
 - RFC 3263 – Locating SIP Servers
 - RFC 3265 – Specific Event Notification
 - RFC 3420 – Internet Media Type message/sipfrag
 - RFC 3428 – Extension for Instant Messaging
 - RFC 3515 – Refer Method
 - RFC 3581 – Extension for Symmetric Response Routing
 - RFC 3891 – Replaces Header
 - RFC 3892 – Referred By Mechanism

- SIPPING Working Group
 - Draft - Session Initiation Protocol Call Control - Transfer
[draft-ietf-sipping-cc-transfer-04.txt]
 - Draft - Session Initiation Protocol Call Control - Conferencing For User Agents
[draft-ietf-sipping-cc-conferencing-05.txt]
 - RFC 3842 – Message Summary and Message Waiting Indication Event Package

- SIMPLE Working Group
 - RFC 3856 – Presence Event Package

- MMUSIC Working Group
 - RFC 2327 – Session Description Protocol
 - RFC 3264 – Offer/Answer Model
 - RFC 3605 – RTCP Attribute

- AVT Working Group
 - Draft - RTP payload format for the Speex Codec
 - RFC 2032 – RTP payload format for H.261
 - RFC 2190 – RTP payload format for H.263
 - RFC 2429 – RTP payload format for the 1998 version of ITU-T Rec. H.263
 - RFC 2833 – RTP payload for DTMF digits, Telephony tones and Telephony signals
 - RFC 3550 – RTP: A transport protocol for real-time applications
 - RFC 3551 – RTP profile for audio and video conferences with minimal control
 - RFC 3555 – Mime type registration of RTP payload formats
 - RFC 3952 – RTP payload format for iLBC Speech
 - RFC 3984 – RTP payload format for H.264

- Other RFCs
 - Draft – XML based data format for conveying Presence Information [draft-rosenberg-imp-pidf-00.txt] (used instead of RFC 3863)
 - RFC 1035 – Domain Names – Implementation and specification
 - RFC 2782 – A DNS RR for specifying the location of services (DNS SRV)
 - RFC 2915 – The Naming Authority pointer (NAPTR) DNS Resource Record
 - RFC 3489 – Simple Traversal of UDP through NATs (STUN)

3.1.2 CODECS

- Audio
 - G.711 u-Law and A-Law
 - G.722,
 - G.726
 - G.729
 - Speex (8 and 16 kHz)
 - GSM 6.10
 - iLBC
- Video
 - H.261
 - H.263
 - H.264

3.2 System Requirements

- Processor:
 - * audio only: Intel Pentium II 400 Mhz or equivalent (minimum)
 - * full features: Intel Pentium III 833 Mhz or equivalent (minimum)
- RAM:
 - * audio only: 128 Mbytes (minimum)
 - * full features: 256 Mbytes (minimum), 512 Mbytes (recommended)
- Full duplex sound card , speaker or headset and microphone
- IP network connection (LAN, Wireless, broadband)
- Bandwidth:
 - audio only : 56 kbps
 - video : 256 kbps or higher
- Operating systems:
 - Windows 98 Second Edition
 - Windows 2000
 - Windows Millenium Edition
 - Windows XP
 - Windows Vista (ask us)
- Download size : 10,1 Mbytes
- Hard disk space : 18 Mbytes