eyeP Media

Powering VoIP

eyeP Communicator v1.5 – Windows® edition

OEM communication software application

Feature list

eyeP Media

Powering VoIP

eyeP Media SA
Avenue des Baumettes 15
1020 Renens
Switzerland

Tel: +41 21.632.66.00   -   sales@eyepmedia.com - www.eyepmedia.com

eye P Media

Powering VoIP

Table of Contents

eyeP Media

Powering VoIP

1   Introduction

The purpose of this document is to provide an overview of the functions and features available on the eyeP Communicator – release 1.5.

1.1 Scope

The document is aimed at customers wanting to deploy or use the application in their network. This document is neither a user guide, or a release note or a specification or a developer's guide, which are separate documents.

eyeP Media

Powering VoIP

eyeP Media

Powering VoIP

1.2 Definitions, Acronyms and Abbreviations

PAEC   **Perceptually motivated Acoustic Echo Canceller**
Audio or video signal processing used to reduce 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 distorsion that might me 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 taht 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 accross 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 avaibility 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 provioning of softphones deployed, based on XML file of parameters.

eyeP Media

Powering VoIP

2 Product Features

This section briefly depicts the system features, at a high level, without giving any details about how the feature is implemented.

2.1 Graphical User Interface

eyeP Communicator provides a flexible interface which can be:

- ✔ reconfigured (select amongst a set of predefined communication features)
- ✔ rebranded with your own logos
- ✔ localized with various languages (support of Unicode character set)

2.1.1 Main screen

![eyeP Media logo — Powering VoIP]
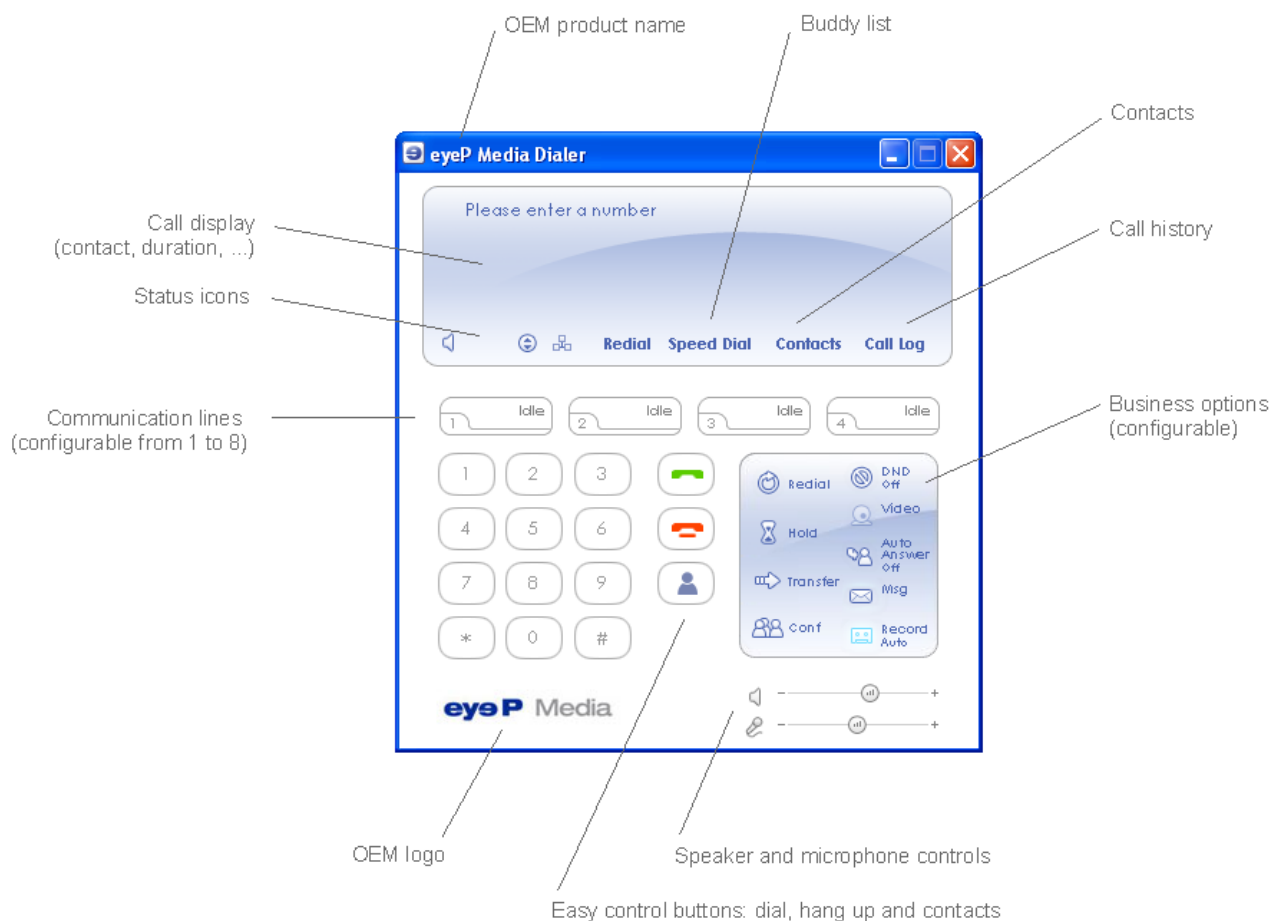
2.1.2 Contacts window

2.1.3 Settings window

eyeP Media

Powering VoIP

2.2 Installation
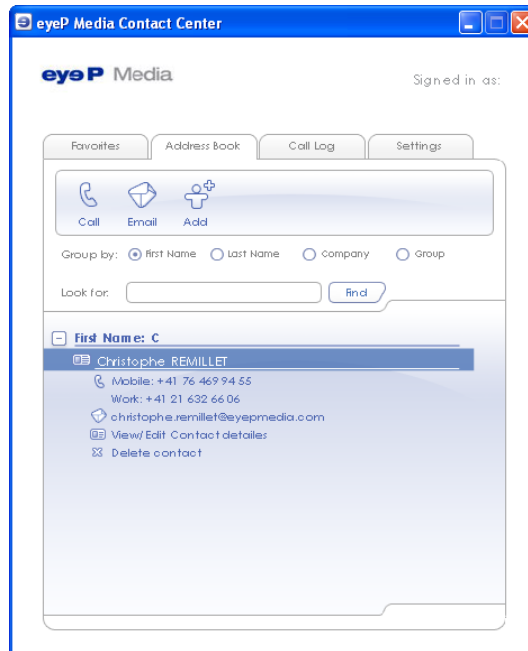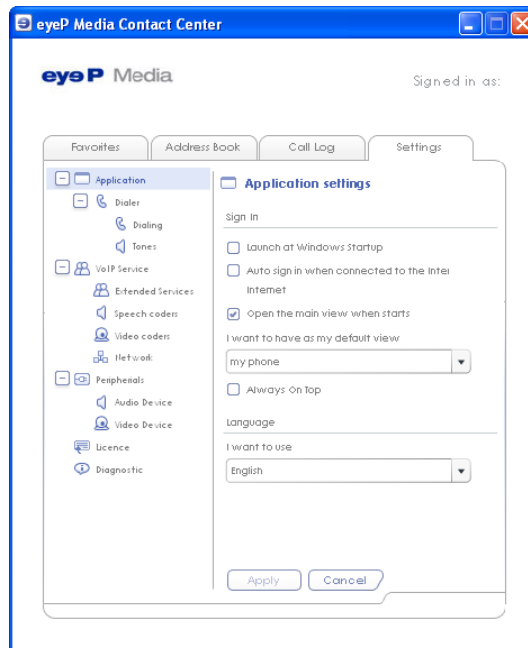
2.2.1 First launch

The very first time eyeP Communicator is launched, a tuning wizard is executed to assist the user to configure the softphone. The tuning wizard allows the:

- Selection of the preferred audio device.
  The computer machine can be equipped with more than one audio device like external speakers, a USB headset or a microphone.

- Test the speaker (audio out),
  by playing a sample audio message. The user can adjust the speaker volume level.

- Test the microphone (audio in),
  by recording the user voice during 5 seconds and replaying the sound recorded. The user can adjust the sensitivity of the microphone.

- Select and test the preferred video device,
  User is asked to select a preferred video device and can preview the webcam. If no video device is connected, then user may skip this step.

- Select the network adapter.
  Because computers might be connected to the Internet through various network adapters (TCP/IP, WiFi, …), User can select its favorite network adapter here.

- Select the connection type.
  The system has no idea of the real performance of the network connection. Thus user should specify the network connection type (DialUp, xDSL 128 kb/s, xDSL 256 kb/s, LAN).

- Define the SIP account.
  User specifies its SIP account which has been provided by its service provider. SIP account should be made of :
  - URI address (like "myaccount@myprovider.com")
  - outbound proxy
  - register name
  - register password
  - account name (the name displayed)

Once completed, the dialer window is displayed by default. If parameters have been set correctly, user should see that a network connection is available and its SIP account has been registered. Then user is ready to make calls or receive calls on Internet.

2.2.2 Login

The system provides an 'application level' authentication mechanism based on an account and password, which is a separate concept from SIP authentication.

Coupled to an OAMP server, the system can be programmed to automatically request a sign-in. To help the user, a combo box with the last three account names is displayed so that frequent users only have to enter the password field.

Once signed-in, the OAMP server sends the user SIP account properties to the system. It can send personal user information too, like its personal address book for example.

eyeP Media

Powering VoIP

2.3 Telephony features

2.3.1 Dialing

User has multiple options to dial a number; he may:

- Compose the phone number,
  by clicking either on the dialpad keys or by entering the digits directly from the keyboard. By setting the parameter to permit SIP number entry, the user can call using the SIP number - alphanumerical string of characters.

  To ease the phone number composition, the system provides an auto-completion feature. When user starts to dial the number, the system attempts to identify any matching phone numbers defined in the personal address book.

  User may correct the phone number by using the backspace key. Copy/paste feature can be used and will replace the whole phone number, which is particularly interesting when viewing a document containing contacts.

  Alternatively, the user can take the line by pressing on the call button first, then dial the phone number and hear the corresponding DTMF codes. The user may also select a communication line first, for example line 2, and then dial the phone number.

- Press the Redial button.
  The last number dialed is composed automatically.

- Select a contact or a buddy.
  User selects the appropriate phone number (work, home, mobile or SIP) of the contact or buddy. If many audio lines are available, only the first audio line will be used (e.g. selecting line 2 will not allow accessing the list of contacts).

- Select a call log.
  User can select a call which has been log amongst the history. If many audio lines are available, only the first communication line will be used (e.g. selecting line 2 will disable access to the list of contacts).

2.3.2 Incoming call

When receiving an incoming call, the system plays the ringing sound on the default speaker of the computer. This allows user - who do not necessarily use a headset all the day long - to hear the ringing tone and know that an incoming call is pending. In parallel, the dialer displays a ringing message on the screen.

To answer, the user can either click on the call button or click on the message box for accepting the call. If user rejects the call, the ringing sound is switched off and a message is returned to the network indicating that system is not available.

In case of missing the incoming call, the system reports missed calls into the call logs. A message box remains on the dialer screen, indicating that a missed call occurred and proposes to the user to call back immediately.

2.3.3   Establishing the communication

Once user has dialed the phone number, he presses on the call button. The system then displays the name of the contact, if it corresponds to a contact name, and informs the user that phone is ringing on the called side.

2.3.4   During the communication

When the call is established, the system displays the time elapsed (each second) and the callee phone number.

During the communication, the user may adjust, in real time, the speaker volume and the microphone sensibility, including changing the audio device whilst in conversation (for example enabling the speaker) or muting the audio.

Telephony options can be activated, again in real time, like recording the conversation, transferring the call, organizing a conference, holding the call or enabling the video (the party should have the video enabled too).

Whilst in a call session, the system may receive incoming calls on the remaining communication lines. In such case, the user may choose to reject the call, transfer the call or join the conference. The first communication line will automatically be placed into hold state in such case.

2.3.5   Ending communication

The user or the network (the party) may decide at any time to stop the communication. The call session is then recorded into the call history and the system is ready to proceed with a new call session.

2.3.6   Hold the communication

The user may hold the communication at any time during a call session. If video media is used, the system displays the last video frame received. Note that call duration continues anyway.

2.3.7   Transferring a call

The user may transfer a communication at any time by clicking the transfer button. Then user enters a phone number or selects a contact phone number (the phone destination) and then clicks call or transfer button to complete the operation.

2.3.8 Conference

Conferencing can be organized in two ways:

- Local conferencing.

  The system is then in charge of organizing and managing the conference with other parties. The maximum number of participants is subject to change according to the available bandwidth, since all audio streams will converge on the user computer.
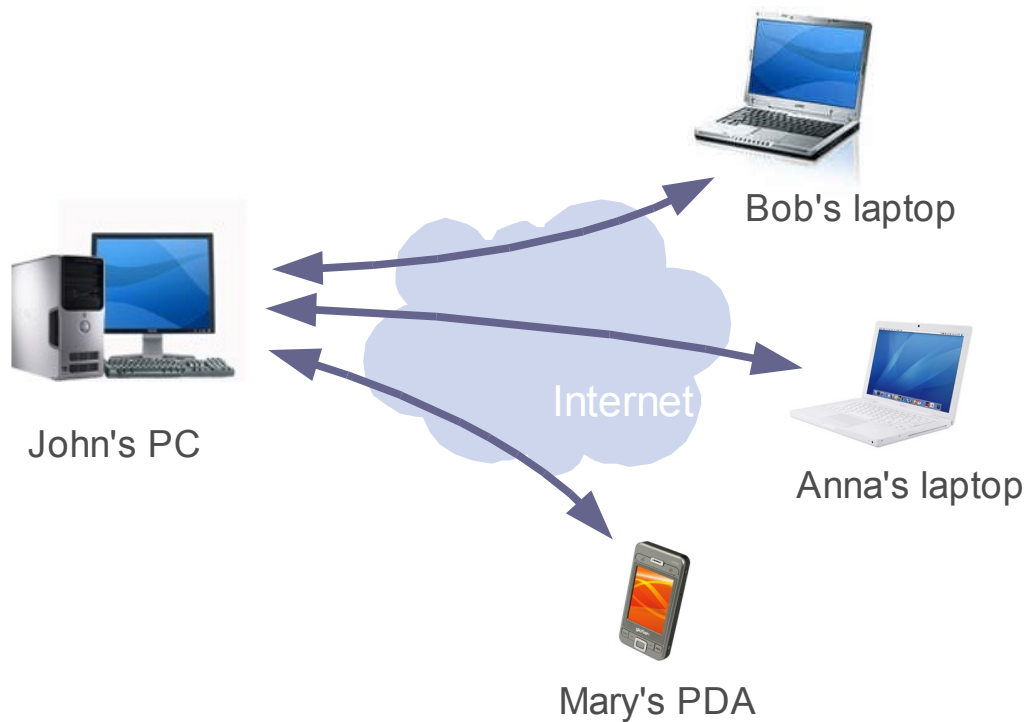
- Network conferencing

  In such case, a conferencing server is provided on the network and the system should have been configured with the relevant server address.

John's PC

Bob's laptop

Internet

Conferencing Server

Anna's laptop

Mary's PDA

eyeP Media
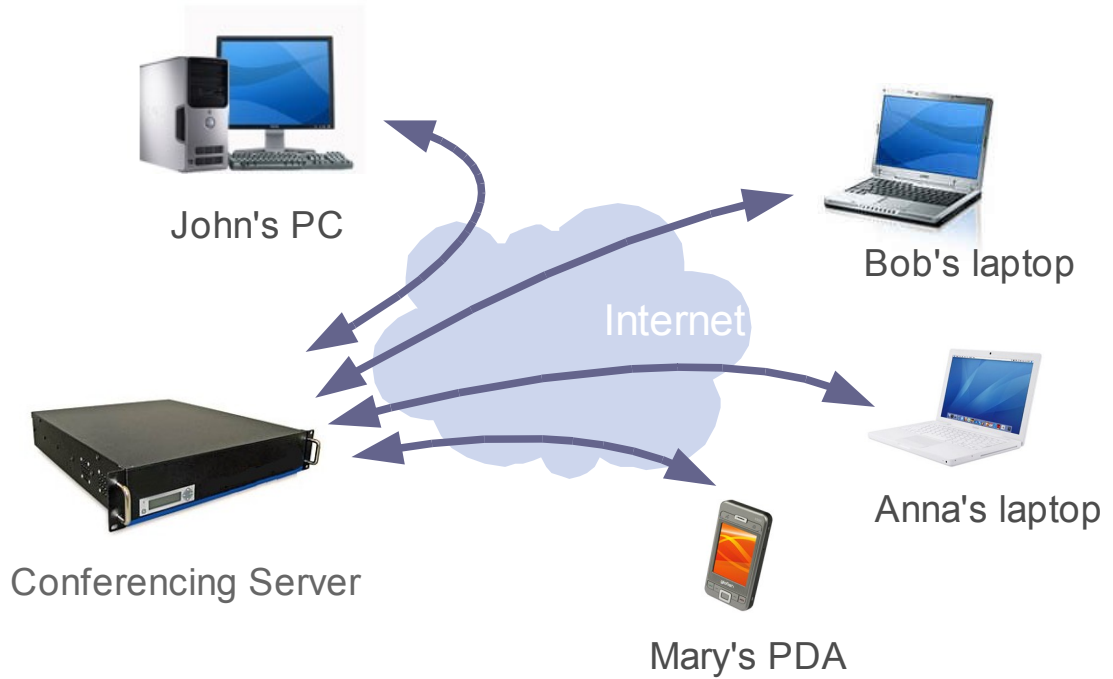
2.4 Other Communication features

2.4.1 Voice mail

The system can interface with voice mail services. To do so, the user simply clicks on the message button.

Such service could be either a web page (the user accesses the web site of its voice mail service provider) or can be consulted through the voice media only. In any case the network address should be defined to get such service.

eye P Media

Powering VoIP

2.5 Remote provisioning

2.5.1   OAMP mechanism

The system proposes an Operation Administration Management and Provisioning (OAMP) mechanism with a simple XML based interface. It allows:

- provisioning and management of subscribers and their respective user profiles

- authentication of subscribers and any associated license management

- remote upgrade of the soft phone software

- display of an operator defined information banner on the system
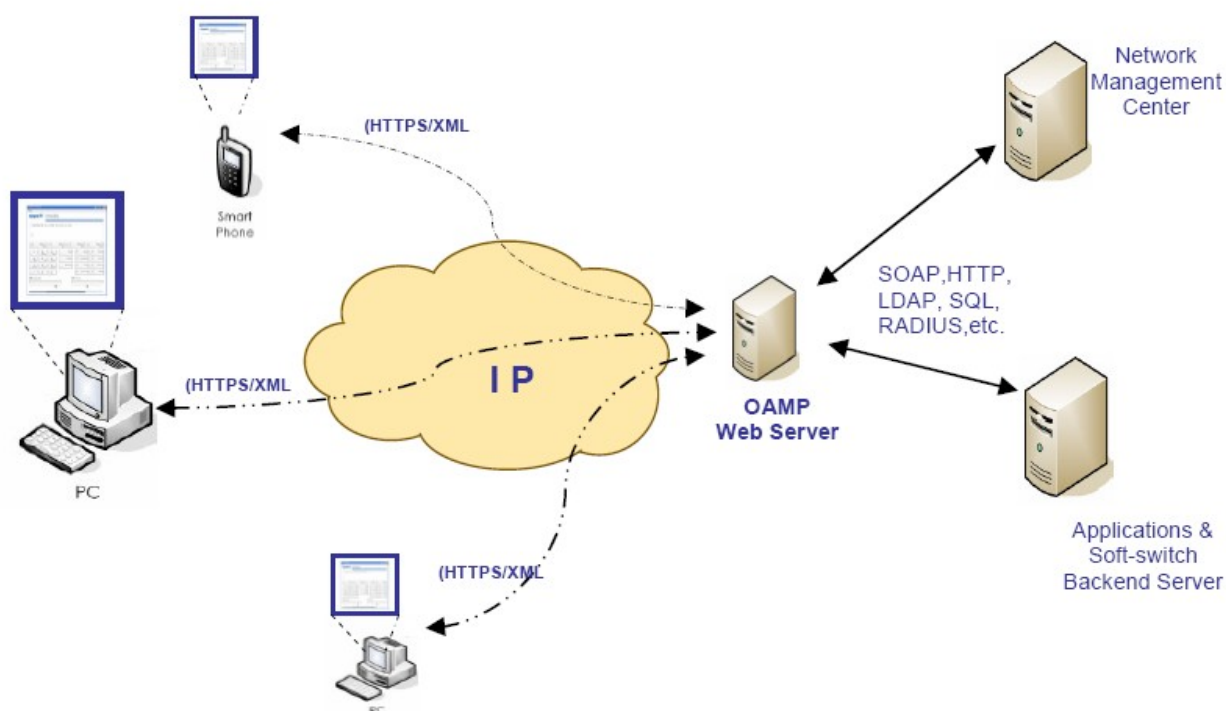
2.6 Application level features

2.6.1 Settings

The system provides three settings methodology:

- <u>Settings the parameters locally.</u>
  In such case, the system proposes access to the settings panel and parameters are stored as XML files into the installation directory of the system. A settings panel is provided to allow the user changing the most obvious parameters like SIP account properties, the network connection or the audio/video settings.

  Note that according to the administration rights of the computer, the user may access these XML files directly.

- <u>Setup file.</u>
  The service provider may provision the parameters of the system to be distributed as a setup file.

- <u>Remote provisioning</u>.
  The service provider can operate a TFTP or HTTP(S) server to remotely provision the system (see OAMPML concept).

  At login, the system sends a provisioning request to the server which answers by sending an XML file containing the user parameters. Note the system is able to upgrade its version.

  In order to satisfy the high level of requirements of telecommunication operators and service providers, the system exposes most of its technical parameters for fine tuning. In brief, the following global parameters might be changed:

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

eye P Media

Powering VoIP

2.6.2 Contacts

The User can add manually any contact, person or company, into the softphone. The contact information contains:

- Name (including first name and last for a person, name for a company)
- Phone numbers: work, mobile, home
- Email address
- SIP address
- Category (employees, friends, partners, …)

Categories can be dynamically defined and User may change at any time the category of the contact. Note that a contact belongs to only one category at a time.

The User can directly call or email the contact from the list. He can also filter the contacts by first name, last name, company or group and make search, according to the criteria defined.

List of contacts can be provisioned by the server if necessary.

2.6.3 Buddy list

The user can add any contact, from the personal address book into the buddy list. Presence information is displayed in real time.

List of buddies can be provisioned by the server if necessary.

2.6.4 Call logs

The system traces the history of calls, recording the type of call (incoming, outgoing or missed calls), the contact name or the phone number, the date and the call duration.

To increase the user efficiency, information could be filtered by date, duration, contact name, missed call and type of call. If recording feature has been enabled, then User can play again the audio file of the call session recorded.

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

eyeP Media

Powering VoIP

- Other RFCs

  Draft – XML based data format for conveying Presence Information
      [draft-rosenberg-impp-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.722.2
  - G.726
  - G.729
  - Speex (8 and 16 kHz)
  - GSM 6.10
  - iLBC

- Video
  - H.261
  - H.263
  - H.264

eyeP Media

Powering VoIP

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 : 10,2 Mbytes

- Hard disk space : 20 Mbytes