Application of the SIP Protocol in Telecommunication, Based on a Image Transmitting Application

Michal Marzec, Bartosz Sakowicz, Piotr Mazur, Andrzej Napieralski

Abstract – **This paper presents the idea of convergence of the IT and telecommunication worlds, based on an example application. The program has been developed in Microsoft Visual Web Developer 2008 environment. As one of the aims the possibilities of convergence are being described. The example application allows users to communicate by video conference using soft phone on one side and WWW browser on the other, via Internet connection.**

Keywords – **C#, SIP, convergence, IMS, video conferencing.**

I. INTRODUCTION

Based on telecommunication oriented literature review including SIP issues, the article presents possibilities that come along when implementing new technologies in comparison to those widely used for many years. Also the issues referring to new directions of converged systems development including Next Generation Networks (NGN) and IP Multimedia Subsytem (IMS) are described [1-3].

Telecommunication associated with telegraphs or even a simple telephone calls vanished long time ago. Yesterdays networks are being transformed into a platforms capable of transmitting new types of signals and data, which results in a large growth in modern solutions. It is no longer only IT oriented industry. As in many other cases the main factor for this development are the growing consumers demands. Convergence is the answer for these demands.

Having many definitions, in this article, convergence is understood as the integration of telecommunication and IT systems. One of the first concepts was the New Generation Network. Although it does not have a specified definition it can be described as a set of conceptions of how telecommunication is expected to evolve. Its structure is defined as an open one which means new components can be added at any time. What is more there is no association between the net and the communication solutions. Type of data does not depend on the type of junction any more but rather on the application implemented.

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INVITE sip:B@domain1 SIP/2.0
Via: SIP/2.0/UDP domain2
From: sip:A@ domain 2.com
To: sip:B@ domain 1.com
Call-ID: r4r3t@one. domain 2.com
Content-Type: application/sdp
Content-Length: 666

One of the most important components of modern communication networks is the Session Initiation Protocol (SIP) [4]. The SIP is a signaling, messaging protocol developed by the Internet Engineering Task Force (IETF). It is capable of setting up, modifying and tearing down multimedia sessions, requesting and sending presence and instant messages. Alike Hypertext Transfer Protocol (HTTP) SIP works in the application layer of the OSI communications model.

The architecture of SIP defines and uses few components:
User Agent Client (UAC)
- User Agent Server (UAS)
- User Agent (UA)
- Proxy server
- Redirect Server
- Location Server

They are all responsible for the most fundamental functions such as location the callee or configuration of the call parameters. SIP is a text-based protocol which makes it possible to read for human. The general schema of communication between user a and user B is shown on Fig. 1.

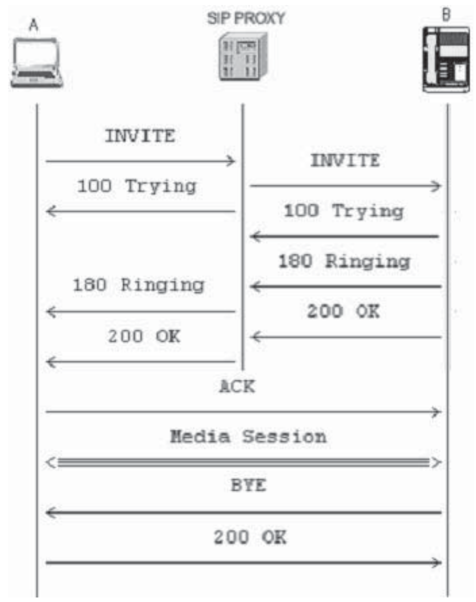

Fig. 1. The general schema of communication.

In order to connect both users must obtain their own unique Uniform Resource Identifier (URI). The URI shows the similarity to the IT world, as its formed for example: michael@domain.com. This stands for users name and the SIP domain he is currently using. The INVITE command is the most important both initiating the connection and sending information about the caller request. In this particular case the INVITE command sent from user A to user B could be formed as follows:

SIP is far faster, easier to develop and flexible standard in comparison to its precursor H.323. SIP is widely specified in IETF Request for Comments (RFC) 3261.

II. APPLICATION COMPONENTS AND TOOLS USED

The choice of programming technology has been foregone by thoroughly conducted examination of available professional literature concerning SIP applications. As a supplementary work a softphone was written in order to ensure the compatibility of video codecs between the application, the server and the softphone. The thesis resulted with a program written in C# on the Microsoft Visual Web Developer platform, which in this case, allows to transfer image to the web browser. The choice between C# and Java was conditioned by the speed of working application [5]. The task required smooth transfer between caller and callee as the communication had to be live. Another advantage of C# was the simplicity of creating

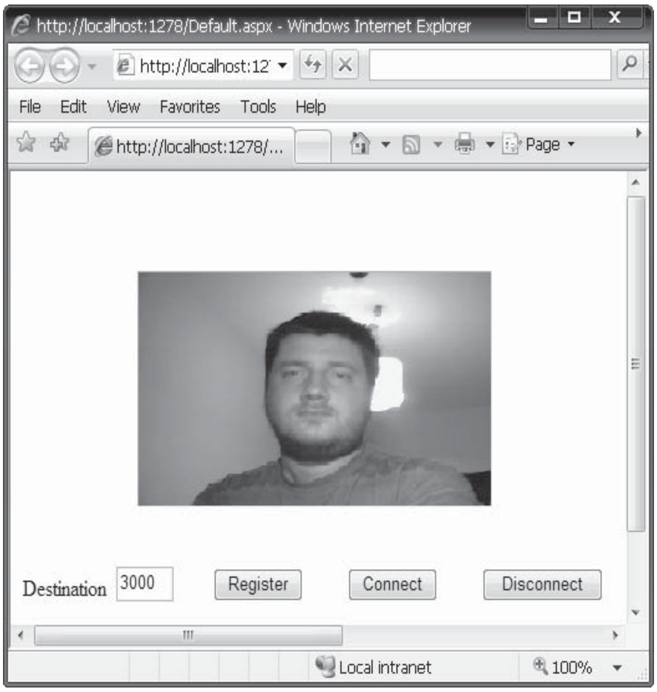

Fig. 2 The user interface

interface which has tremendous importance in developing web applications. The program was written in a way reducing to the minimum necessity to configure after installation. The interface in a web browser is as intuitive as possible, allowing user to execute the basic functions. The interface presented on Fig. 2, includes buttons "Register", "Connect", "Disconnect"; text field "Destination" and of course the video displaying field. The application requires a telecommunication server operating on the SIP protocol. This condition is fulfilled among others by the 3CX Phone System v7.1. It is an easy to use phone system integrating different kinds of communication networks such as GSM or those operating on IP addresses. For proper performance of the application installing Internet Information Services (IIS) was required.

III. Application description

The general idea of how and what the application does is presented on Fig. 3.

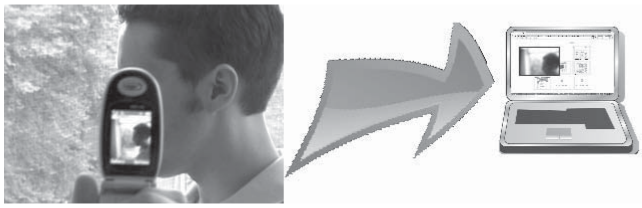

Fig. 3 General idea of the application

While starting the program sends to the browser code responsible for generating user interface. When pressing the "register" button it is registering on a already running server by exchanging the "REGISTER" and "200 OK" commands, as a standard SIP soft phone. After entering desired contact and pressing the "connect" button the application sends request "INVITE" to the server, from where it is transferred to the callee. The caller receives statement "100 TRYING". The connection is rejected unless being approved by the callee in certain time. Whereas the callee accepts the call his phone sends "200 OK" statement, which is then transfered to the application. As an answer "ACK" is sent and the connection is set. Video data is being sent from the soft phone to the application. The image is being continually saved to a file from which it is presented in the WWW browser. When the connection is turned down by any user statement "BYE" is sent and as an answer to it "200 OK". The schema of communication in this particular case is presented on Fig. 4.

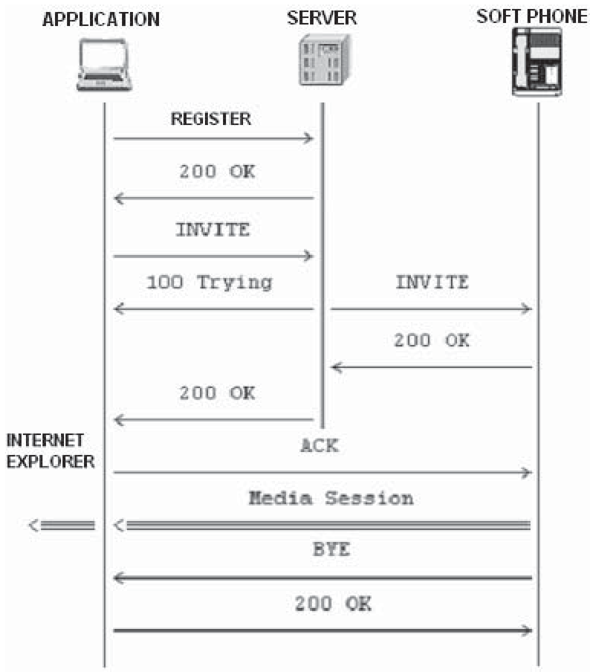

Fig. 4. The schema of communication.

III. APPLICATION SECTIONS

Three main stages of how the application works can be distinguished:

registering on the server,

connection initiation,

data transmission.

Registering the user is possible through defining the user and the "REGISTER" command. Part of the code responsible for this operation is presented below.

```
client=new
SipClient("127.0.0.1","3001","");
client.Register("sip:127.0.0.1","sip:3001
@127.0.0.1", "sip:3001@127.0.0.1");
```

First line defines the user number – 3001 whose IP address is 127.0.01. The second line describes also the IP address of the server no which the user ought to be registered, also the number and login if the user are included.

"CreateConnecion()" is the command responsible for initiating the connection, it includes parameters required for the setting of communication, as following:

```
Owner owner = new Owner();
owner.Username = "3001";
owner.SessionID = 16264;
owner.Address = "127.0.0.1";
session.Owner = owner;
session.Name = "SIP call";
connection.Address = "127.0.0.1";
```

Name and the IP address of the user are defined, as well as the session identification. The caller is declared as the one initiating the connection, which also has to be named in this case the name is "SIP call". At the end the address of the server handling the connection has to be defined. Afterwards the type of communication must be declared, in this case connection is meant to transfer video data. Code defining type of the transfer is presented below.

```
Media media1 = new Media();
media1.Type = "video";
media1.Port = rtpPort;
media1.TransportProtocol = "RTP/AVP";
media1.MediaFormats.Add("31");
media1.Attributes.Add("rtpmap",
"31 H261/90000");
media1.Attributes.Add("fmtp",
"31 QCIF=1;CIF=1");
```

The type of transfered data is defined as video, protocol used in connection as RTP/AVP and codecs required for sending the video data.

Connection is terminated by the method `client.Disconnect();`

In order to ensure expected application performance libraries dedicated to the SIP technology had to be added, as well as those enabling to process data from the network, create interface and display live image in the browser. The most difficult to find were the libraries operating on the SIP protocol, lastly found on the www.independentsoft.de web page, declared as follows:

```
using Independentsoft.Sip;
using Independentsoft.Sip.Sdp;
using Independentsoft.Sip.Methods;
```

IV. SUMMARY AND CONCLUSIONS

The application confirms assumption of versatility of programs based on the SIP programming model. The process of creating revealed the problem with formulating correct connection, setting commands and a divergence in codecs of softphones, especially with video conferencing. Developing applications depending on the SIP protocol is still evolving and having only the possibility to work on freeware platforms and environments causes many difficulties. What is more the range of professional literature concerning SIP issues is very poor. Developing in JAVA turned out to be very difficult in case of SIP operating applications, which caused the transition to C#, where wider documentation was available. The process required acquiring strong knowledge in SIP technology issues and similar solutions already widely applied in telecommunication. The problem with matching video codecs was solved by developing a soft phone which cooperates with the application and the server.

After stronger recognition of the developing techniques and the possibilities of the SIP protocol, the range of web applications possible to create is almost unlimited. The communication schema of SIP is similar to the one of HTTP. The difference between them is the fact that SIP is asynchronous, which means that one request can collect many responses. The variety of terminals possible to apply in many different connections from simple SIP phone to personal computers, shows the versatility of that solution. The transformation of telecommunication networks into more comprehensive data transmitting solutions takes place for many years now. The specification H.323 on how to use other telecommunication protocols gradually gives way to the Session Initiation Protocol. This trend allows further development in converged telecommunication solutions [6]. Also the SIP protocol was the answer for both costumers demands and the competition in the communication market. Thanks to the SIP simple conversation requires far less bandwidth which leaves more space for new types of connections using the very same connection. Another important advantage of SIP is full independence on the platform on which application can be installed. As a part of IP Multimedia Subsystem the SIP protocol is constantly evolving, showing new possibilities of telecommunication.

Acknowledgements

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