Videoconference Client for Windows Mobile

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Abstract: This paper deals with a development of videoconferencing client for portable devices running the Windows Mobile operating system. The selection of suitable SIP stack (Session Initiation Protocol) for call signalling, SDP stack (Session Description Protocol) for multimedia data description, RTP stack (Real-time Transport Protocol) for multimedia data transmission, and libraries for multimedia data compression is described. Another part of this paper describes the first phase of videoconferencing client development.

Key-Words: Videoconferences, Voice over IP, Session Initiation Protocol, Real-time Transport Protocol, Session Initiation protocol, Session Description Protocol

1 Introduction
Nowadays, videoconferences are the subject of intense interest of not only international companies which use it for intercommunication between theirs branches, but also of normal people. Videoconferences have since the beginning been closely linked with the H.323 protocols family [1], which provides signalling, audio / video data compression and their transmission to other-conferences. The SIP (Session Initiation Protocol) gains ground of interest today [2], [3]. It is simpler than the H.323 standard, but it enables only call signalling. The H.323 protocol family has still a place in the videoconferencing field, mainly thanks to the H.323 desktop terminals, which have implemented this standard. The standard enables terminals not only to arrange videoconferences but also, for example, to control remote terminals, etc. The simpler SIP protocol is easy to implement in various devices. Mobile phones, smart-phones and PDAs are one group. The software for making a video-call is installed in some devices, but videoconference applications are in this area still lagging behind. Because of that we have started developing a videoconference client for mobile devices. This client is primarily designed for devices using the Windows Mobile operating system. The application development was divided into three parts. The voice SIP client will be developed in the first phase, the transmission of video signal will be added in the second part, and a connection with the RADVISION videoconference unit will be solved in the last phase. The current videoconference solution for mobile phones is solved by using a 3G gateway, which negotiates mobile phone and videoconference unit intercommunication. Our proposed solution would work thanks to data networks (WiFi, UMTS) without the 3G gateway. The proposed solution will communicate with the videoconferencing unit directly by means of the SIP protocol; in the case of wrong and unsupported audio or video codec a transcoder will be added between the mobile phone and the videoconference unit. The next chapters describe the list of available SIP, SDP and RTP stacks, the development of SIP user agent and its testing.

2 Selection of opportune libraries
An opportune selection of SIP stack for the Windows Mobile operating system is an important step in the design of an SIP user agent. A huge amount of free downloadable SIP stacks from the Internet are accessible all over the World. Most of them have implemented, in addition to the SIP stack, also the SDP stack for multimedia description. None of the SIP stacks contained the RTP stack for multimedia data transmissions. The RTP stack must be used separately during the application development. Selection was made from the following stacks:

2.1 Sofia-SIP library
Sofia-SIP is an open-source SIP user agent library, compliant with the IETF RFC3261 specification [4]. The library is written in the C language, developed primarily for the GNU/Linux (General Public License) operating system and licensed under the LGPL (Lesser General Public License). Sofia-SIP includes the SDP library, but the RTP stack is not included. This library was not used in next work because the C language is not suitable for the development of the Windows Mobile application.

2.2 GNU oSIP & eXoSIP libraries
Both libraries, oSIP [6] and eXoSIP [7], are based on the commercial antiSIP library [5]. The GNU oSIP library is licensed under LGPL, eXoSIP is licensed under GNU
Both stacks are written in the C language and do not include the RTP stack. The problem is here the same as in the previous case, namely the C language.

2.3 OpenSourceSIP
OpenSourceSIP includes two applications: OpenSBC and OSSPhone [8]. These applications provide the basic functionality of the SIP User Agent client and SIP User Agent server and can be used in further development. The applications are designed for the Windows and LINUX operating systems. Because of the great number of SIP stacks used, there can be found three license types: LGPL, GPL and MPL (Mozilla Public License). All source codes are written in the C++ language.

2.4 nSIP / nSDP libraries
nSIP/nSDP libraries were created for use in .NET 2.0 applications [9]. One of the advantages of these libraries is the development in the C# language. There exists a special class for testing the functionality of created methods in the development project. nSIP and nSDP libraries were the first candidates for the development of SIP user agent for Windows Mobile.

2.5 PTLib and OPAL libraries
PTLib (Portable Tools Library) and OPAL (Open Phone Abstraction Library) [10] were created by the RSDevs Company. Both libraries are designed exclusively for use with the Windows Mobile operating system and are developed in the C++ language. The included RTP [11] stack is one of the big advantages of these libraries. The OPAL library supports the G.711 audio codec and the H.261 video codec, which is another big advantage of theirs. Both libraries run under the MPL license and were selected for the implementation of videoconferencing user agent.

3 Videoconferencing user agent
The developed videoconferencing user agent is based on the SIP protocol. The development can be divided into three phases. A standard software VoIP user agent (Voice Over IP) must be created in the first phase. This user agent should allow user registration with the SIP REGISTRAR server, and point-to-point calling through the PROXY or REDIRECT server. In the second phase, the developed user agent should allow the conference calls with three or more users. The video codecs in addition to the audio codecs should be implemented in the developed application in the last phase.

The first phase is described in this paper. Figure 1 shows the basic block diagram of the developed application. The graphical user interface is initiated after the user agent is started; its state is checked at the same time. If the application is started for the first time, the registration with the REGISTRAR server must be created. If it is repeated, the log-in window is started. The application is in the normal state after the user is signed in. In case of bad registration or enrolment, the whole process is repeated.

![Block diagram of first operations after the application is started](image)

The user agent waits for the process that happens after the user has signed in. This depends on the decision whether the application will be on the caller or the called side.

In the case the user agent is on the caller side, the SIP message INVITE is sent to the PROXY server after the user dials the number. The time interval of waiting for the response to the INVITE message is set on the caller side. The SIP request ACK is sent after the 200 OK response is received and then the call is established. For the multimedia transmission, the RTP protocol is used. In the second case, the user agent is on the called side and waits for the INVITE message. After this message is received, the 180 ringing and the 200 OK messages are sent to the caller side and the user agent waits for the final ACK message. The call is then established and the multimedia data transmission through the RTP protocol is started.

The call can be terminated before the call is established or after the call is established. In the first case the CANCEL message is sent, in the second case the BYE message is sent. The message which is expected as the
response to the request CANCEL or BYE is 200 OK. The call is terminated after this response is received. The application returns back to waiting for the process which happens.

Only audio calling is enabled in the first phase. The G.711 codec was used in this phase because of certain support on other devices. The OPAL library was used for audio compression, and for generating the SIP request messages, SIP response messages, SDP messages, and RTP messages. This library co-operates with the PTLib library. The PTLib is used to access hardware tools such as network components, microphone, speakers, etc. The graphical user interface of the final application is shown in Figure 4.

4 Application structure
The application implemented in the C++ language is based on the graphical user interface created by means of MFC (Microsoft Foundation Class Library), and on the OPAL library including the SIP and RTP stacks and on the PTLib library (Fig. 3).

5 Application test
The user agents used for testing the developed application are: software phones – SJPhone and X-Lite; hardware phones: Linksys SPA 942 and Grandstream GXV-3000. The Wireshark sniffer software was used to record all communication between two devices. Various possibilities and scenarios were tested. To connect all the devices, a local area network was used. REGISTRAR and the PROXY server were also connected to this network. The Windows Mobile 6.5 operating system with the developed application was installed on the HTC Touch Diamond 2 phone. The REGISTER, INVITE, BYE, CANCEL requests and the common 1xx, 2xx, 3xx, 4xx, 5xx, and 6xx responses were tested with the developed application. The user agent executed all of these common situations without any problem.

6 Future work
The development of a videoconferencing application for the Windows Mobile operating system, as it was presented, is in the first phase. When audio-conferencing with 3 or more users is enabled, the possibility of sending video signals will be dealt with in the last phase. All of the available video codecs suitable for videoconference connection, such as the H.261, H.263 and H.264 video codecs, will be tested for video signal
compression. The quality of the compressed video and the processing time will be tested. The application will be connected to the RADVISION videoconferencing unit and will be tested together with this unit.

Fig. 5: Scheme of the future work

Conclusion
This paper discussed the development of the videoconferencing client for mobile devices running on the Windows Mobile operating system. A new application, which implements the SIP, SDP and RTP protocols, and enables the basic SIP communication, was created. All the program functions were tested and no complication was discovered.

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