THE PROPOSAL OF AN IMS CLIENT

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ABSTRACT

In this paper, the issue of monitoring user availability and services for the voice transmission in SDS Ericsson 4.1 are addressed. The possibility of employing a Java application for Symbian OS emulator and the Windows XP using platform ICP (IMS Client Platform) are discussed. Also voice transmission parameters are measured for assessing the quality of service (jitter, delay and packet loss).

1. INTRODUCTION

IP Multimedia Subsystem is a generic architecture used as the standard of support for advanced telecommunications services to provide fixed and mobile networks convergence. Providing services to users is solved by IMS AS (Application Server). AS communicates with IMS (CSCF, Call Session Control Function) through SIP (Session Initiation Protocol) protocol, which is also the main signaling protocol in the IMS. Thanks to mutual cooperation between AS servers can IMS offer to users different types of services for access networks and end devices. The platform is formed by many entities ensuring the full functionality of the system. It is possible to divide the IMS entities to six sections: session and routing management, databases, application servers, interworking functions, support functions and charging entities [1].

The system for providing services in the IMS architecture consists of JavaSE (client modules tested for OS Symbian and OS Windows XP) and JavaEE/SIP applications in SDS Ericsson using integrated ICP platform. An example of application creation for Symbian is shown in [2].

The project solution (Figure 1:) is divided into three parts:

- Database of users (*information storage*),
- Module for database access and management (*processing client requirements and communication with database server*),
- Client module (*user interface*).

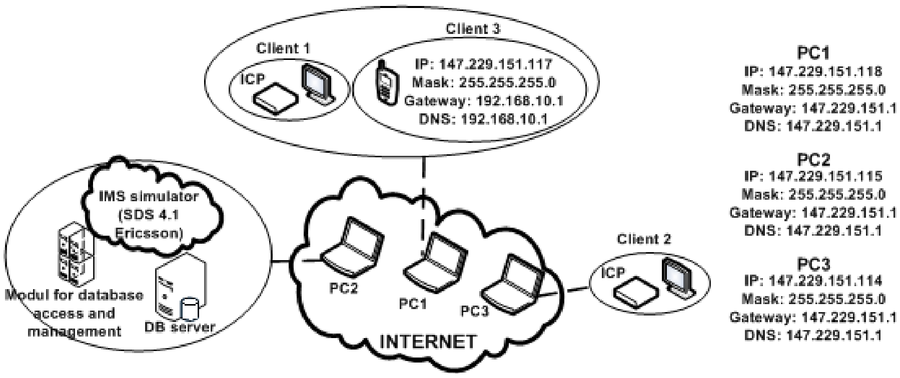

Figure 1: Graphic representation of the proposed system

The applications are tested on free PCs. PC1 (IP: 147.229.151.118) and PC3 (147.229.151.114) stand for the client part and PC2 (IP: 147.229.151.115) stands for testing of:

- IMS simulator,
- Module for database access and management,
- Designed database.

Due to the fact that it is not possible to run ICP platform on Windows more times, it is necessary to test the client module on the PC3. The client application is simulated as a Java application for PC (Windows XP) and executed as an emulator (Symbian) as shown in Figure 2:. The Ericsson Service Development Studio was installed to PC1 and PC3 due to appropriate users' registration.

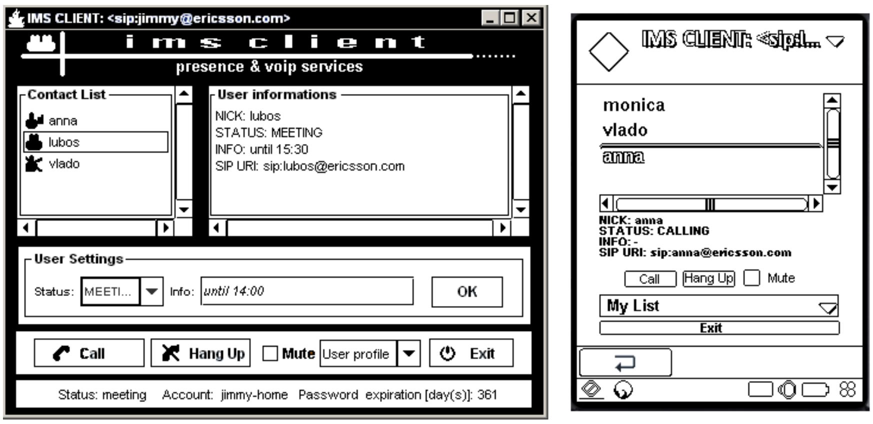

Figure 2: Created applications for Windows XP and Symbian

2. THE ANALYSIS OF COMMUNICATION

2.1. REGISTRATION PROCESS IN THE IMS

The process of client registration (Figure 3:) is performed with the help of ICP platform, which is implemented by the providers of development tool. The ICP platform can be used in the java desktop application for the Windows XP and also in the application for the OS Symbian. The process of registration in the IMS consists of an initial SIP REGISTER request. The request is sent from the user (IP: 147.229.151.114) to the core simulator of IMS (IP: 147.229.151.115). In the case of correct data transmission to the simulator the positive respond for the request is SIP OK.

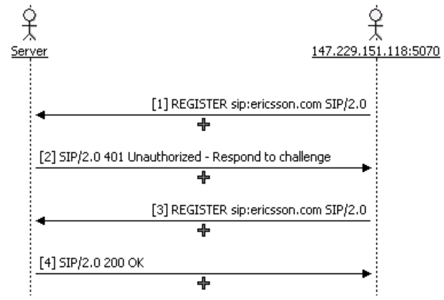

Figure 3: The process of user registration captured by Visual Traffic Flow in SDS

2.2. THE PRESENCE SERVICES

The signaling of presence services is based on SIP messaging; especially PUBLISH, SUBSCRIBE and NOTIFY.

The SIP PUBLISH method is used for the user's presence information updating and restored every three minutes. The message body carries the presence information (e.g. the user's presence status) and the user's information (username and password) to login into the database server in the format of XML (eXtensible Markup Language). If the login information (username and password) sent by the application are correct, the update process passes off. It is also important to define headers of the PUBLISH request (the *Event, Expires, Content-Type* headers) [3]. The SIP PUBLISH method is automatically sent during switching off the client application. Then the value of user's status is offline.

Messages SUBSCRIBE and NOTIFY are used for determining the clients availability located in the buddy list. The response of the servlet application with required information is the SIP NOTIFY. It is necessary to define the header values for the SUBSRIBE method: *Supported*, *Event*, *Expires* and the *Accept* [3]. After receiving the requirement, the servlet application acknowledges the message by SIP OK and generates NOTIFY response. Except the main headers must the NOTIFY method define headers for the presence services (*Event, Require, Expires, Subscription-state, Content-Type* and *Require*) [3]. The body of SIP NOTIFY message carries the information in the format of MIME (Multipurpose Internet Mail Extensions). The information is divided into the various sequences with the help of the boundary value. The start value of this parameter is declared in the *Content-Type* header. The settings of headers (SIP PUBLISH, SUBSCRIBE and NOTIFY) are defined in [3].

It is necessary to define an automatic update user's presence information for data updating. The principle of the automatic update is illustrated in Figure 4:.

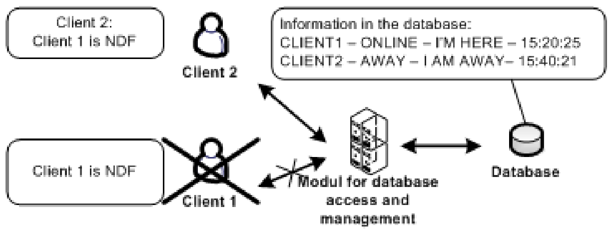

Figure 4: The principle of an automatic update of user's presence information

Client 1 does not send the presence information from unknown reason and Client 2 still has Client 1 in the buddy list. When the module for database access and management (servlet application) receives the SIP SUBSCRIBE request, in which the Client 2 wants to know the presence information of Client 1, then the servlet application checks the expire time information in the database. The value of the time information in the database server is changed after updating the presence information (after receiving the SIP PUBLIS message). If the time information is older than ten minutes, the servlet application sends the SIP message to Client 1. In this message is the actual presence information required. If Client 1 still does not response, the servlet application sends to Client 2 the response (SIP NOTIFY) with status “NDF”. The “NDF” indicates the absence of Client 1.

2.3. THE VoIP SERVICES

The main difference between the signaling of the presence service and VoIP service is the absence of servlet application in the case of VoIP communication (end-to-end). The SIP endpoint needs to send an INVITE message to establish a call. The SIP Decline message is used for the invitation session refusing. The session is finished by the SIP BYE message. The VoIP service uses the real-time RTP protocol (Real-time Transport Protocol) with the AMR (Adaptive Multi-Rate) codec.

In the Figure 5: is shown the measured characteristic of a jitter parameter. There are illustrated the values of a jitter parameter depending on time (on the left is characteristic for direction from PC3 to PC1, on the right is characteristic for reverse direction). It can be seen that the jitter is rising in the first second and then is almost invariable.

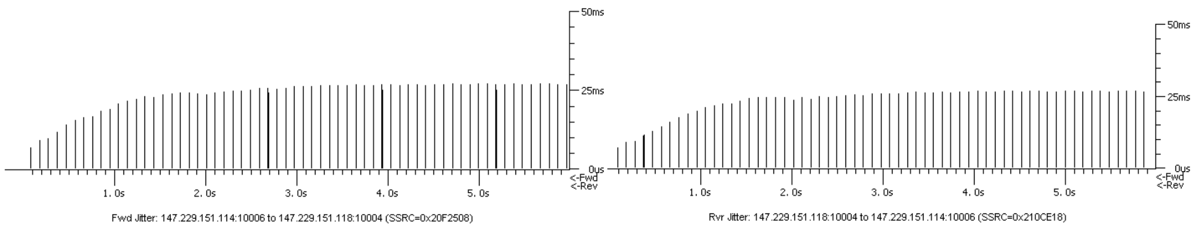

Figure 5: Jitter characteristics measured with the help of WireShark (on the left: PC3 >> PC1, on the right: PC1 >> PC3)

Table 1: indicates measured values of the maximum and the mean jitter, the value of maximum delay and the packet loss.

	PC3 >> PC1	PC1 >> PC3
Max. jitter	26.54 ms	26.52 ms
Mean jitter	25.17 ms	25.18 ms
Max. delay	95.26 ms	95.20 ms
Packet loss	0 %	0%

Table 1: Measured parameters with the help of WireShark

The comparisons of the measured parameters for assessing the quality of services (Table 1:) with the values of parameters from [4] are shown in Table 2:. The maximum value of jitter for the good quality of services is 20 ms [4].

Parameter	**Quality**
jitter	suitable

delay	good
packet loss	good

Table 2: The application quality measurement

3. CONCLUSION

This article describes client application proposal for the IMS platform created in Ericsson Service Development Studio. Designed example provides services to determine the client availability and voice transmission. The client application was simulated as a Java application for PC (Windows XP) and executed as an emulator (Symbian, *P1i a M600*). These services can be provided to users as stand-alone services or in special packages. It will be also possible to expand on IPTV services and videoconferencing.

The application is based on SIP signaling; especially PUBLISH messaging (for the information recovery describing the availability status of the client), SUBSCRIBE messaging (a request for sending data describing the status of users located in the client list) and NOTIFY messaging (information describing the attendance client status). For the voice transmission is used RTP protocol with AMR codec. The client application was performed with the help of ICP platform, which is not fully working in the Symbian emulator. Therefore, the service for voice transmission works only as a Java desktop application.

For quality of services assessing were measured next parameters: jitter, delay and packet loss. According to ITU (International Telecommunication Union), if the delay is kept below 150 ms, then the application would not be significantly affected. The values of maximum delay were 95.26 ms and 95.20 ms. The jitter values slightly got over 20 ms because there are various kinds of factors reflected to the quality. ITU also recommends that the packet loss kept bellow 0.5% is good which is sufficiently implemented.

ACKNOWLEDGEMENTS

This paper has been supported by projects FRVŠ 2954/2010/F1a and FRVŠ 2986/2010/G1.

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