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GENERALIZED NET MODEL OF INTERNET TELEPHONY

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Abstract: This paper presents the construction of a generalized net model of the process of Internet Telephony using VoIP protocol. It is realized by the Freeswich platform which is installed on a virtual machine with Ubuntu. The model can help to observe and analyze the process.
Keywords: Generalized net, Voice over Internet Protocol, Session Initiate Protocol, Real-time Transport Protocol, Real-Time Control Protocol, Internet telephony.

1. Introduction

The paper presents a model of VoIP communication between two users.

Internet telephony or IP telephony, or VoIP *(Voice over Internet Protocol)* is a technology which allows the voice transfer (telephony) owing to the infrastructure of the Internet. The term may refer to a connection between two computers, two telephones or a computer and a telephone, as long as the signal is carried in a portion of its way through IP packets. Upon receipt of analogue voice (standard voice) by phone, Voice Gateway first digitalizes the signal and compresses the new digital signal into standard blocks of data known as IP packets. They are sent through Internet to an entrance of Voice Gateway, where the process is reversed. With this technology it is possible to make three different types of calls: computer to computer, computer to phone, and phone to phone.

Internet telephony process is initiated by the calling party – the system: a computer, a microphone and headphones, that wants to connect to the called party – a phone from the public telephone network. The Internet provider of the calling party, having the necessary software, connects to the called party and provides a telephone number to the Internet provider which provides a service (VoIP). Using a microphone, the calling party then speaks in it and the voice signal is transferred to the Voice Gateway, where it is digitalized (if the signal is not digital). From here the IP packets are transferred through Internet on a route that is determined by the Voice Gateway of the provider until they reach the remote Voice Gateway. Then, on its part it converts the IP packets into voice signal and transfers the voice of the local Public Switched Telephone Network to the called party. From here, the phone of the called party will signal incoming calls. Both sides can conduct fully bidirectional (duplex) conversation. The same example can easily describe the links: computer to phone and computer to computer [9, 10].

VoIP has the advantage of reducing the price of long distance calls. Its disadvantages are loss and delay of packets, signal interference and echo, congestions on the network. The best way to overcome these problems is to use network equipment, supporting prioritization of packets along the whole lenght of the connection.

For modeling the VoIP communication we use the theory of Generalized Nets (for GNs see [1, 2, 3]).

2. Generalized net model

The Generalized Net model is presented in Fig. 1.

The call is realized by several protocols: Session Initiate Protocol (SIP) [4, 5], Real-time Transport Protocol (RTP) [6], Real-Time Control Protocol (RTCP) [7].

The algorithm for realization of the VoIP consists of six steps [4, 5]:

1. User A dials user B (A sends a request SIP Invite);
2. User B picks up (B returns Status: 183 Session Progress/Status: 200 OK). There is already established a session/channel between phones;
3. User A returns SIP Ack (Acknowledge) (the conversation can start);
4. The platform creates RTP/RTCP channel between users (voices are transferred here as IP packets);
5. B hangs up to A (B sends a request SIP: Bye);
6. A returns Status: 200 OK and connection ends.

The generalized net consists of three transitions which represent respectively:

Z_1 – The activity of user *i*;

Z_2 – The activity of Server;

Z_3 – The activity of user *j*.

The transitions of the GN are the following.

Initially, there are α_i^1-tokens that are located in place L_5 with initial characteristic "User *i*" (i =1, 2, …, n). In the next time-moments these token can splits into two or more. One of them, let it be the original α_i^1-token, will continue to stay in place L_5, while the other α-tokens will move to transition Z_3, passing via transition Z_2.

In the first time-moment, there is one β-token that is located in place L_{14} with initial characteristic "Server". In the next time-moments this token is split into two or more. One of them, let it be the original β-token, will continue to stay in place L_{14}, while the other β-tokens will move to transitions Z_1 and Z_2.

Also initially, there are α_j^2-tokens that are located in place L_{18} with initial characteristic "User *j*" (j =1, 2, …, m). In the next time-moments these token can split into two or more. One of them, let it be the original α_j^2-token, will continue to stay in place L_{18}, while the other α-tokens will move to transition Z_1, passing via transition Z_2.

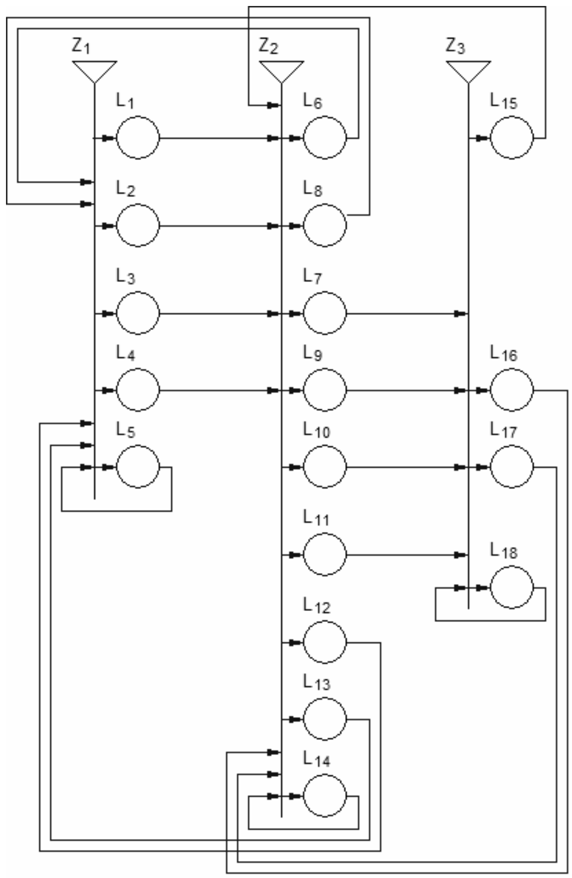

Figure 1. GN-model of the Voice over Internet Protocol.

Everywhere below i is the calling user, j is the corresponding user, and $i = 1, 2, \ldots, n$, $j = 1, 2, \ldots, m$.

$$Z_1 = \langle \{L_5, L_6, L_8, L_{11}, L_{12}\}, \{L_1, L_2, L_3, L_4, L_5\}, r_1 \rangle,$$

where:

$$r_1 = \begin{array}{c|ccccc}
 & L_1 & L_2 & L_3 & L_4 & L_5 \\
\hline
L_5 & w_{5,1} & w_{5,2} & w_{5,3} & w_{5,4} & True \\
L_6 & False & False & False & False & True \\
L_8 & False & False & False & False & True \\
L_{11} & False & False & False & False & True \\
L_{12} & False & False & False & False & True
\end{array},$$

where:

- $w_{5,1}$ = "There is a *SIP Invite query* from user *i* to user *j*",
- $w_{5,2}$ = "There is a *SIP Acknowledge query* from user *i* to user *j*",
- $w_{5,3}$ = "There is a *RTP/RTOP stream* from user *i* to user *j*",
- $w_{5,4}$ = "There is a *SIP Bye query* from user *i* to user *j*".

The α-tokens that enter places L_1, L_2, L_3 and L_4 obtain the characteristic respectively:

- "User_number *i*, User_number *j*, header, *SIP Invite query*" in place L_1,
- "User_number *i*, User_number *j*, header, *SIP Acknowledge query*" in place L_2 and
- "User_number *i*, User_number *j*, header, *RTP/RTOP stream* (*IP packets)*" in place L_3,
- "User_number *i*, User_number *j*, header, *SIP Bye query*" in place L_4.

$$Z_2 = \langle\{L_1, L_2, L_3, L_4, L_{14}, L_{15}, L_{16}, L_{17}\}, \{L_6, L_7, L_8, L_9, L_{10}, L_{11}, L_{12}, L_{13}, L_{14}\}, r_2\rangle,$$

where:

$$r_2 = \begin{array}{c|ccccccccc} & L_6 & L_7 & L_8 & L_9 & L_{10} & L_{11} & L_{12} & L_{13} & L_{14} \\ \hline L_1 & False & False & False & False & False & False & False & False & True \\ L_2 & False & False & False & False & False & False & False & False & True \\ L_3 & False & False & False & False & False & False & False & False & True \\ L_4 & False & False & False & False & False & False & False & False & True \\ L_{14} & w_{14,6} & w_{14,7} & w_{14,8} & w_{14,9} & w_{14,10} & w_{14,11} & w_{14,12} & w_{14,13} & True \\ L_{15} & False & False & False & False & False & False & False & False & True \\ L_{16} & False & False & False & False & False & False & False & False & True \\ L_{17} & False & False & False & False & False & False & False & False & True \end{array},$$

where:

- $w_{14,6}$ = "The user *i* is trying to connect with user *j*: *Status Trying*"
- $w_{14,7}$ = "There is a *SIP/STD Invite query* from user *i* to user *j*",
- $w_{14,8}$ = "The session from user *i* to user is in *Progress&Status is OK* "
- $w_{14,9}$ = "There is a *SIP Acknowledge query* from user *i* to user *j*",
- $w_{14,10}$ = $w_{14,12}$ = "There is a *RTP/RTOP stream* between user *i* to user *j*",
- $w_{14,11}$ = $w_{14,13}$ = "There is a *SIP Bye query* from user *j* to user *i*".

The β-tokens that enter places L_6, L_7, L_8, L_9, L_{10}, L_{11}, L_{12} and L_{13} obtain the characteristics respectively:

- "User_number *i*, User_number *j*, header, *Status Trying*" in place L_6,
- "User_number *i*, User_number *j*, header, *SIP Invite query*" in place L_7,
- "User_number *i*, User_number *j*, header, *Session Progress&Status OK*" in place L_8,
- "User_number *i*, User_number *j*, header, *SIP Acknowledge query*" in place L_9,
- "User_number *i*, User_number *j*, header, *RTP/RTOP stream* (*IP packets)*" in places L_{10} and L_{12},
- "User_number *i*, User_number *j*, header, *SIP Bye query*" in places L_{11} and L_{13}.

$$Z_3 = \langle \{L_7, L_9, L_{10}, L_{13}, L_{18}\}, \{L_{15}, L_{16}, L_{17}, L_{18}\}, r_3 \rangle,$$

where:

$$r_3 = \begin{array}{c|cccc} & L_{15} & L_{16} & L_{17} & L_{18} \\ \hline L_7 & False & False & False & True \\ L_9 & False & False & False & True \\ L_{10} & False & False & False & True \\ L_{13} & False & False & False & True \\ L_{18} & w_{18,15} & w_{18,16} & w_{18,17} & True \end{array},$$

where:

- $w_{18,15}$ = "The session from user *i* to user is in *Progress&Status is OK* ",
- $w_{18,16}$ = "There is a *RTP/RTOP stream* from user *i* to user *j*",
- $w_{18,17}$ = "There is a *SIP Bye query* from user *i* to user *j*".

The α-tokens that enter places L_6, L_7, L_8, L_9, L_{10}, L_{11}, L_{12} and L_{13} obtain the characteristic respectively:

- "User_number *i*, User_number *j*, header, *Session Progress&Status OK*" in place L_{15},
- "User_number *i*, User_number *j*, header, *RTP/RTOP stream* (*IP packets)*" in place L_{16} and
- "User_number *i*, User_number *j*, header, *SIP Bye query*" in place L_{17}.

The presented model of the VoIP is realized by platform Freeswich that is installed on virtual machine with Ubuntu. The communication between users is tested in Prof. Asen Zlatarov University, Burgas [8, 9]. The clients that we used were SIP-phones and laptops.

3. Conclusion

In the paper was constructed a GN-model of IP telephony, based on VoIP protocol. By using VoIP, the decisions are based and realized by IP protocol, which gives users practically unlimited mobility.

The presented model gives possibility to simulate the processes. Usage of hierarchical operators will make the model more specific. The set of characteristics taken in real processes, will give possibility for the use of the presented model for a real communication.

The authors, together with some colleagues, have been preparing an extensive further research on this theme.

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