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A Review of Formal Descriptions of Session Initiation Protocol

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Abstract. The Session Initiation Protocol is used for establishing peer-to-peer sessions with one or more participants. Internet telephone calls, multimedia distribution, conference calls, and multiplayer online games are examples of these sessions. Since formal description is the first step of modeling and analysis of a system, this paper reviews formal descriptions of this protocol and suggests ideas to build more realistic model.

Keywords: Session Initiation Protocol, Multimedia distribution, Colored Petri net, conference calls, online games.

1 Introduction

The Session Initiation Protocol (SIP) is a signaling protocol for establishing a simple two-way voice-over-IP calls or a multi-point collaborative multimedia conference session [1]. The authors of [1] presented a formalized and executable Colored Petri Net (CPN) model of SIP. The authors of [2] carried out the verification of the SIP Invite scenario with CPN.

Colored Petri Nets (CPN) extend the vocabulary of ordinary Petri Nets and add features that make them suitable for modeling large systems [1, 3]. A hierarchical CPN model consists of several modules called pages. A transition can be substituted by a page [4, 5].

Formal description is the first step of modeling and analyzing of a system. This paper reviews formal descriptions of SIP.

2 SIP Overview [1]

SIP is a text-based peer-to-peer signaling protocol based on the request/response transaction model. A logical entity that crates a new request is a User Agent Client (UAC) whereas a logical entity that responds to a SIP request is a User Agent Server (UAS). A SIP User Agent (UA) can be thought of as a software component or SIP entity that interacts with the user. SIP UAs can be lightweight clients suitable for embedding in mobile handsets or PDAs or soft-phones as desktop applications. A UA

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itself has a client element, the User Agent Client (UAC) and a server element, the User Agent Server (UAS). SIP specifies several server elements: Proxy Server, Registrar, Redirect Server, Presence Server as shown in Figure 1 [1].

A proxy server is an intermediary entity acts as both a server and a client for the purpose of making requests on behalf of other clients. A proxy server primarily plays the role of routing. A registrar is a server that accepts register requests and places the information it receives in those requests into the location service for the domain it handles. A redirect server is a USA that generates 3xx responses to requests it receives, directing the client to contact an alternate set of addresses. A presence server manages subscription and notifications [1].

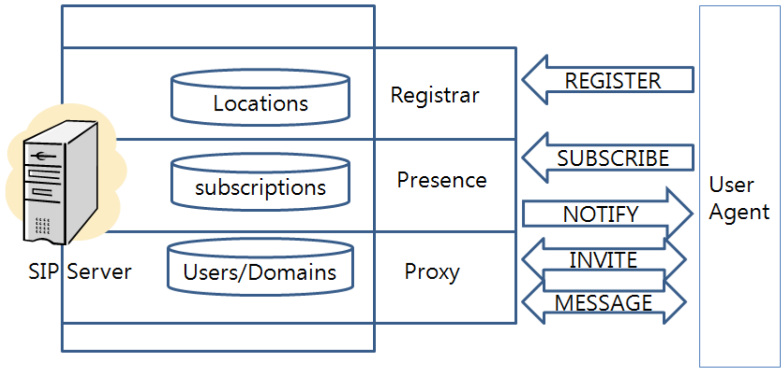

Fig. 1. Main SIP components and their relationships [1]

Interactions between these components take place in a series of message exchanges. A transaction consists of a single request and one or more final responses with zero or more provisional responses. The process of a SIP transaction is described in four state machines: two for invite transactions and two for non-invite transactions. Each set consists of a client transaction machine and a server transaction machine. Figure 2 depicts the non-invite client transaction state machine [1, 6].

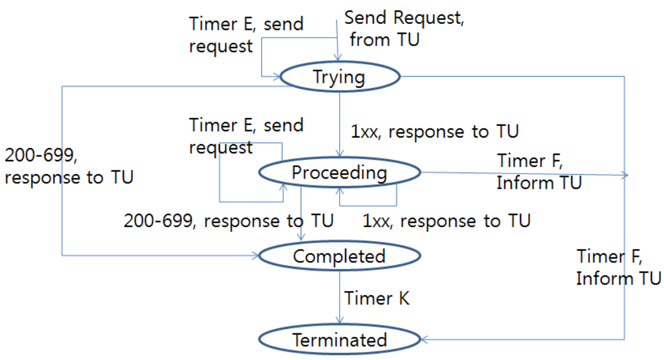

Fig. 2. SIP non-invite client transaction state machine

To send a request, a client transaction user (TU) hands the request to this underlying machine. The machine enters the Trying state, passes the request to the transport layer and starts the re-transmission timer E to a specified value. It moves to the Proceeding state if a provisional response is received and moves to the Completed state if a final response is received. For unreliable transport, it remains in the Completed state to absorb a retransmission of responses and enters the Terminated state when timer K expires [1].

SIP invite transaction was investigated in [2]. The caller sends the INVITE message to initiate the session. The message is delivered to the proxy server, then to the callee. After the message finally reached the callee and the callee agreed to establish the session, the callee might send the response back to the caller via a proxy.

3 Colored Petri Net Model of SIP

A hierarchical CPN model of the overall flow of information in SIP is shown in Fig. 3. UAC/UAS in this figure corresponds to the User Agent/SIP Server in Fig. 1. A UAC consists of Transaction User (TU) and a Client Transaction Machine depicted in Fig. 2. Requests are generated by the TU and handed to the underlying Client Transaction Machine for transmission to the server. The flow in the reverse direction is similar and responses received by the Client Transaction machine are passed on to the TU.

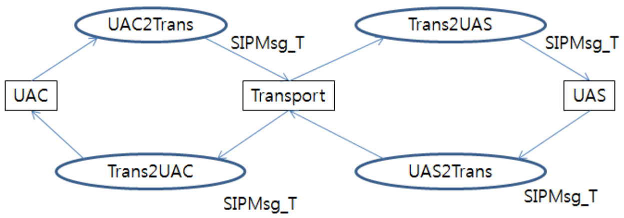

Fig. 3. The topmost level CPN model of SIP [1]

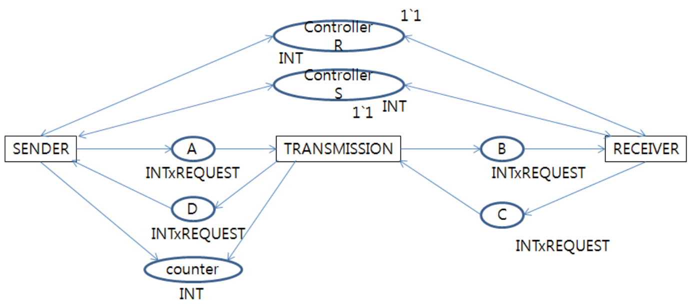

Fig. 4. A CPN model of the SIP invite scenario [2]

The response message comes back through the From Client Transaction and enables the Receive Response transition. The return code contained in the message is compared to see if it is the final response code. If it is the final response code and the

CID matches a CID in the Wait for Response, the transaction is completed and the user is ready to send the next request. Otherwise it continues to wait for a final response [1].

A CPN model of the high level abstraction overview for the SIP Invite scenario is shown in Fig. 4 [2].

4 Conclusions

The authors of [1] built a CPN model of SIP non-invite scenario. They used crisp numbers to represent delay times. The authors of [2] built a CPN model of the SIP invite scenario under the assumption that the message will not be lost as messages are carried over the reliable communication medium. We are planning to build a comprehensive and more realistic CPN model of SIP. In our model, delay time would not be crisp and the communication medium would not be 100% reliable.

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