MODELLING OF INTELLIGENCE IN INTERNET TELEPHONE SYSTEM

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The paper presents a study on the possibilities for provisioning traditional telephone services on the Internet by the use of Session Initiation Protocol (SIP). Service signalling models are proposed based on the main functionality of SIP telephone system. Considered services include: Call forwarding, Abbreviate dialling, Call pick up and Virtual private network.

1. Introduction

The main aim of Intelligent network (IN) technology was centralized provisioning of value added services in a public switched telephone network. This technology facilitates to a great extend the introduction of new services in the network, as the related changes are done in a centralized node than separately in each network switching node. Session Initiation Protocol (SIP) is a signalling protocol for providing multimedia services over the Internet. SIP is seen as the future of call signalling and telephony. One reason for the rapid acceptance of SIP is that it is a powerful call control protocol. It allows intelligent end points to implement the entire suite of telephony, Private Branch Exchange (PBX), and Centrex services without a service provider, and without a controller or switch.

In provisioning of the voice over IP services the centralized nature of IN technology could not be applied easily in the Internet where the intelligence is distributed. A SIP proxy service can perform any of the IN features, such as those for numbering, routing, charging, access, and restriction, but the mechanism of interaction with the call setup process is different in comparison with those in IN.

In IN the call setup is controlled by the Service Switching Function (SSF), which delegates control to the Service Control Function (SCF) for doing the processing. After the SCF has interpreted service logic, it hands back control to the SSF. In SIP *Invite* message is routed transparently from proxy to proxy, where each proxy can do a certain part of processing.

The most important difference between IN and SIP lies in the tightness of control over the call setup signaling. The SSF in IN keeps tight control of the call state through the Basic call state mode (BCSM). The SSF can delegate control to the SCF to do service processing, but control is always handed back to the BCSM afterward. In SIP there is no central entity that keeps tight control of the call signalling. A request can be routed to a proxy for processing, but the proxy simply sends the request onward after it has done what it has to do.

The paper presents a view how to implement some of IN services on the Internet by the use of SIP. Service signalling models are proposed based on the main functionality of SIP telephone system.

2. Number translating services modeling

Number translation provides possibility for flexible routing in implementing IN service features like abbreviate dialing, call forwarding, time-dependent routing, origin-dependent routing and others. In IN service logic is hosted in service control point (SCP), and when is triggered from SSF, it requests service data point (SDP) where subscriber profiles are saved. SDP looks up in the subscriber database and returns back to SCP the number(s) the call has to be routed to.

In SIP network proxy servers, redirects servers and location servers can cooperate to provide intelligent call processing with number translation. Figure 1 shows an example how the call forwarding service can be implemented.

The calling subscriber sends an *Invite* request to the proxy server to which the user agent is configured to send all his outgoing requests. The proxy server responds with *407 Proxy authentication required*, requiring authentication. After sending acknowledgement the calling user agent resends the *Invite* containing credential. A user agent credentials are usually an encrypted username and password. If the authentication is successful, the proxy server sends *DNS query* to DNS server. The query has to identify the IP address of the proxy server that manages the called user's domain. At this time the caller's user agent receives *100 Trying response* (an indication of call progress). As a query result the proxy server receives *DNS response* with the IP address of the caller's domain.

Then the proxy server sends *Invite* to the inbound proxy server of the called user. It is possible a firewall to be installed on this proxy to protect the users. The called user agent responds with *100 Trying* to inform about the call progress. The Location server is used to locate a registration for the called party. The proxy sends *181 Call is being forwarded* informing that the called user has forwarded the incoming calls. The *302 Moved Temporarily* message contains the new URI.

Then the caller's user agent resends an *Invite* request to the user agent identified in *302 Moved Temporarily* response. Again the location server is asked for the IP address of the terminal where the called user is registered. Having the called user's IP address the proxy server sends an *Invite* request. The called user agent wants the caller to be authenticated (*401 Unauthorized* message). The caller's user agent resends a new *Invite* request with the username and password. If the authentication is successful, the caller is informed that the called user is aware (*180 Ringing*). If the called user receives the call, his user agent responds with *200 OK* that the invitation for session setup is accepted. The media session begins.

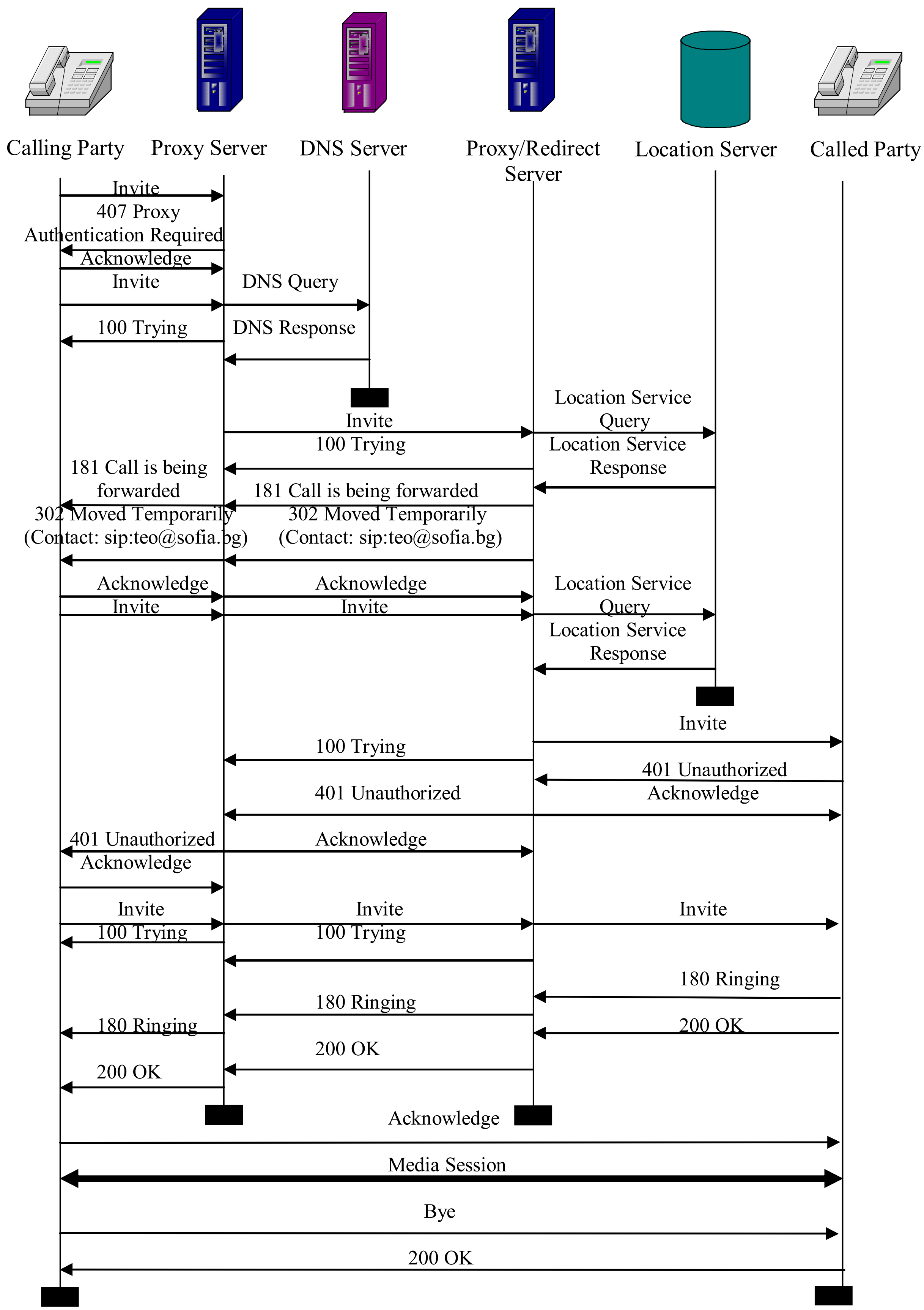

Figure 1 A Call forwarding service model implemented by SIP

The call terminates when the called party sends a *Bye*. The acknowledgment of the *Bye* with *200 OK* response causes both sides to stop sending media packets.

If the Abbreviate dialling service is implemented on the IN platform the service logic asks a database and finds out the correspondence between the abbreviated number and the destination number. If the service is implemented on the Internet through SIP, then the caller's proxy server with the embedded functionality of redirect server asks a supported database and translates the abbreviated number into the destination number.

3. Call pick up service modeling

Call pick up service allows the called subscriber to receive telephone calls from any terminal after dialling a personal authorization code or a password. When the service is activated, the dialled authorization code is checked by the service logic. If the service is offered on the Internet, it is also accessible through a special code that is authenticated. Figure 2 shows how the call pick up service can be implemented by the use of SIP.

As the call processing signaling flow follows that in the previous example, just the specific signaling messages for the call pick up service will be discussed.

After successful authentication, the caller receives *100 Trying* and *180 Ringing* responses indicating that the *Invite* request is accepted and the called user is aware about the incoming call. At this moment the called user hears that there is an incoming call for him and inserts an authorization code from the closest terminal. If the called user authentication is successful, the called user's agent receives the *Refer* request with *Replaces* header field. The called user's agent sends an *Invite* with *Replaces* header field to the caller's user agent. At receiving the *Invite (Replaces)* the caller's user agent responds with *200 OK*. The caller's user agent must terminate the pending dialog with *Cancel* and to transfer all resources and state from the existing dialog to the newly established dialog. The *200 OK* response acknowledges that the cancel request is accepted by the user agent of the originally dialed terminal. The called user agent sends an *Acknowledgment* and the user agent of the originally dialed terminal sends *487 Request terminated* response to the calling user. The media session begins.

An advanced use of *Refer* is in implementation of a common PSTN or PBX feature known as attended transfer. In this feature, the transferor is assumed to be in a dialog (in a session) with the transferee. The transferor places the transferee on hold then sends an *Invite* to another party, called the transfer target. After the session is established between the transferor and the transfer target, the transferor then puts the target on hold. Now the transferor has two on hold sessions. The transferor then sends a *Refer* to the transferee which causes the transferee to generate a new *Invite* to the target. The successful *Invite* replaces the existing session between the transferor and the transfer target. When the transferee receives notification that the transfer was successful, the session between the transferor and the transferee is terminated with *Bye*.

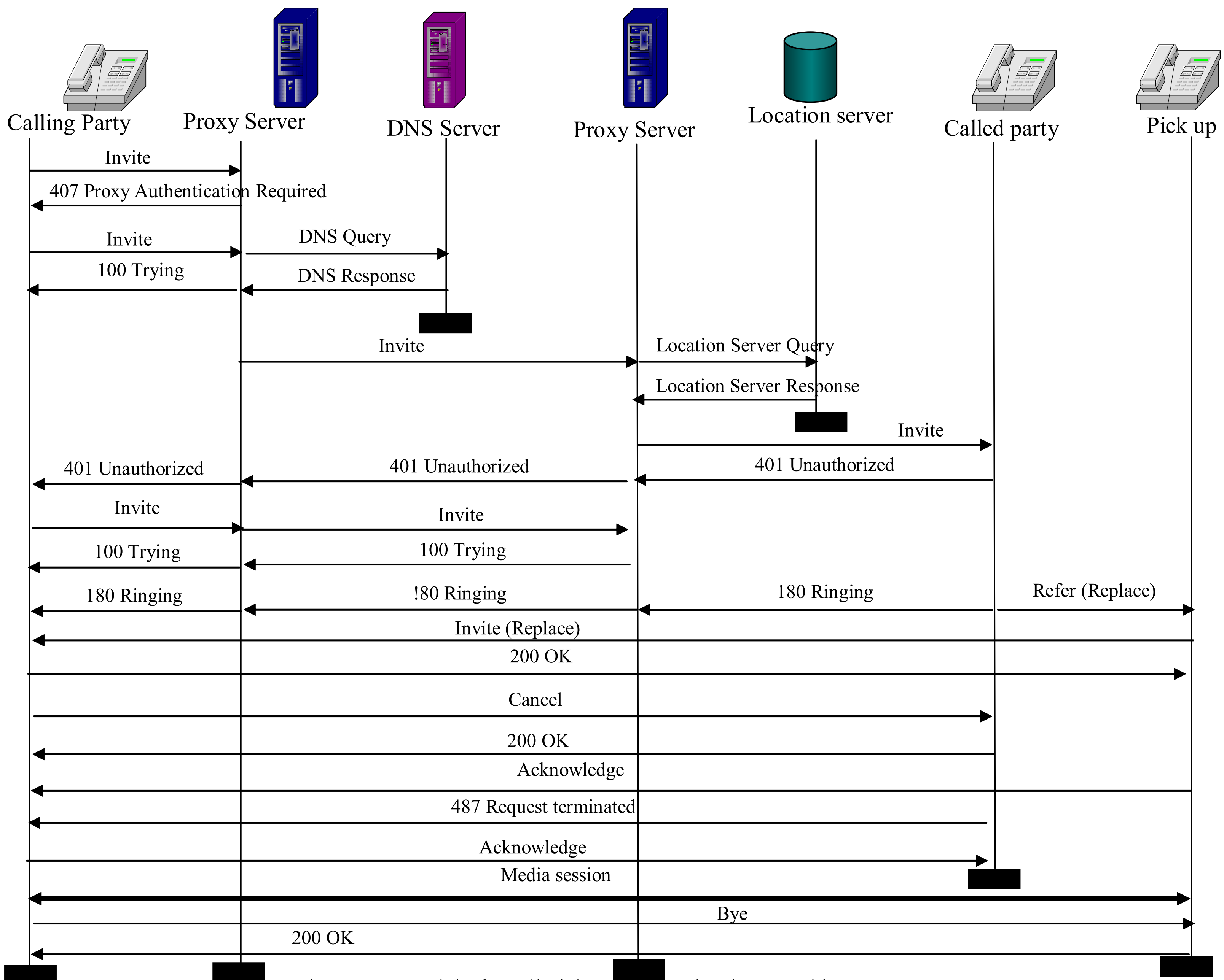

Figure 2 A model of a call pick up service implemented by SIP

4. A model of VPN service implemented by SIP

Virtual private network (VPN) service allows the users to use public network resources to simulate the functionality of a private network without using dedicated resources. The VPN may include private network capabilities, such as dialling restrictions, private numbering plan, hold, call transfer, and so on. VPN users can, for instance, dial an extension number to make an internal call, just as if they were connected to a PBX. For external calls, they use a prefix to reach the public network, just as any call from a private to a public network would.

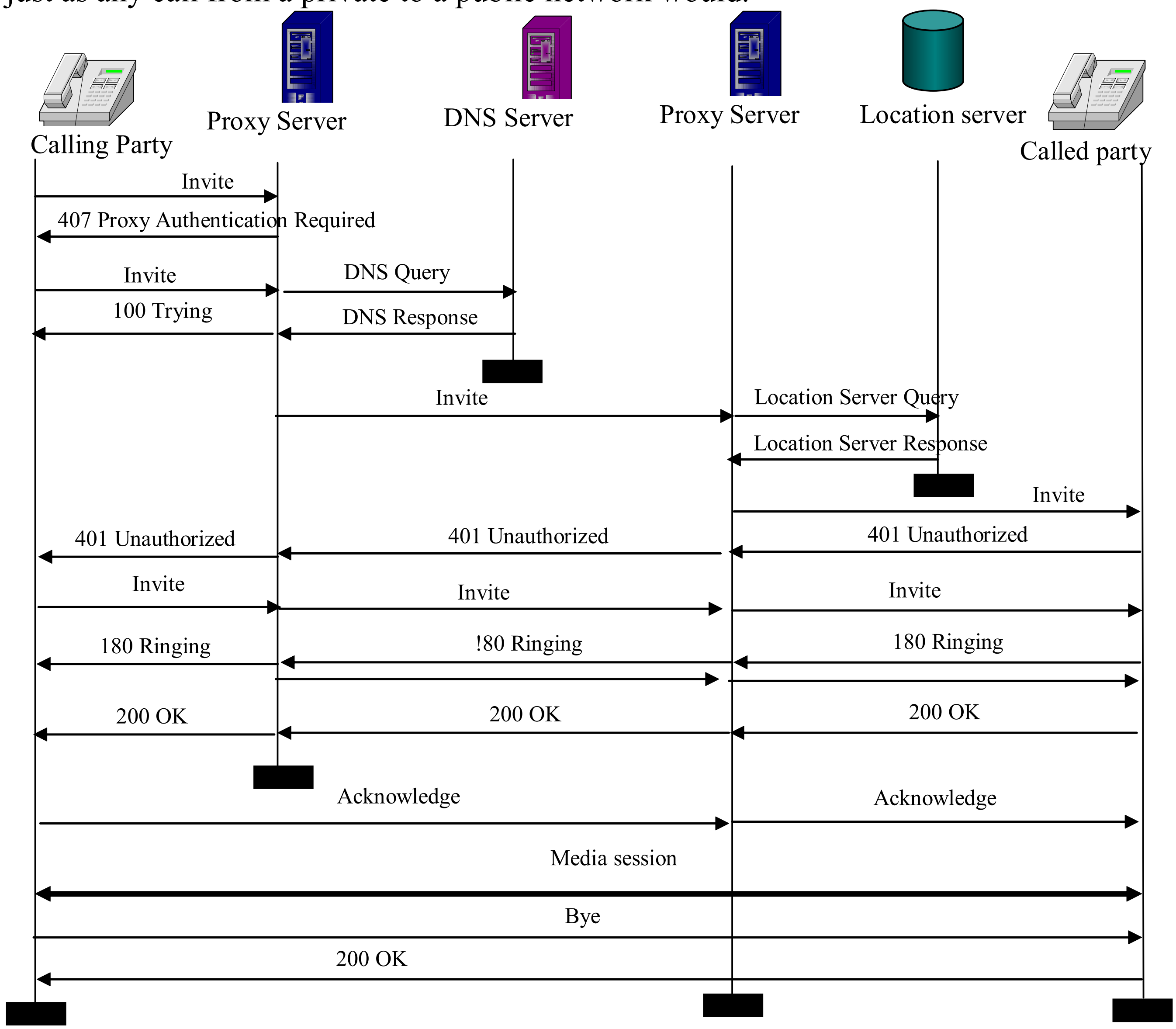

Figure 3 A model of an external VPN call implemented by SIP

If the VPN service is implemented on Internet and is used for telephony, it may consist of two parts: VPN on IP level and internal Internet telephony system running over it. So the offered service has to provide features like quality of service support, encrypting at network level and dynamic allocation of bandwidth on demand. On this intranet various private numbering plans can be maintained and within it different service features can be implemented much as they would be on the public Internet or a physically contiguous intranet.

For the VPN service the model of an external VPN call is considered. Figure 3 presents a model of an external call of VPN user in an Internet telephony system.

A firewall is installed on the proxy server to protect VPN resources and screen outgoing calls. The proxy server checks the user profile for restrictions in a business group. If the caller is authorized for making outgoing call, his *Invite* request is received, else the proxy server rejects the request with *503 Service unavailable* response.

5. Conclusion

The paper presents a study on how the SIP can be used to perform the services of traditional IN protocols. A deep analysis shows that Internet model transforms the location at which many services are performed. In general, end-user terminals are assumed to be much more intelligent than in traditional telephone model; thus many services which traditionally had to reside within the network can be moved out to the edges, without requiring any explicit support for them within the network. For example, the abbreviated dialing in Internet telephony would typically do this work; either by storing an internal table of locally defined short address for the actual address it would send, or (for setups more analogues to VPNs) by having user terminal configured to consult a local database server for address translation queries. This translation could also be performed by a local proxy server through which the caller sends its outgoing call requests.

Other services can be performed by widely separated specialized servers which result in call setup information traversing paths which might be extremely indirect when compared with the physical network's actual topology.

However the service is performed it is evident that all the services we are accustomed to use in circuit switched networks may be available in future packet switched networks. Many traditional telephone services can be implemented in Internet by the use of SIP protocol and its extensions.

References

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