

Enhancement to SIP Protocol

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Abstract— The paper is based on Enhancement to SIP Protocol. Session Initiation Protocol (SIP) is used for session establishment in IP telephony. SIP is a text based protocol. It contains messages that are exchanged between end users for the establishment of session. After the session is established the media stream is transmitted between the end users using Real Time Protocol(RTP).This papers aims at reducing the bandwidth used for transmitting the SIP Protocol by using efficient messages for Session Negotiation. The use of Options Message of SIP protocol instead of Subscribe and Notify reduces the bandwidth involved in maintaining a dialog that is created for Subscribe and Notify messages.

Keywords— Session Initiation Protocol(SIP); Real Time Protocol(RTP); Option; Subscribe; Notify

I. INTRODUCTION

The Session Initiation Protocol (SIP) is a text based protocol. It is a signaling, presence, and instant messaging protocol developed to set up, modify, and tear down multimedia sessions; request and deliver presence; send and receive instant messages; end and receive voice calls and video calls.

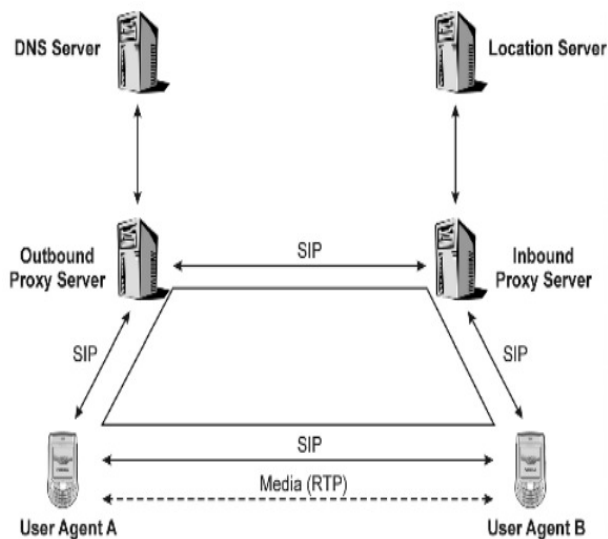


Fig. 1 SIP Architecture

As SIP is a text based protocol, it involves many messages:

A. Register

The Register method is used by a UA to notify a SIP network of its current Contact URI (IP address) and the URI

that should have requests routed to this Contact. Registration is not required to enable a user agent to use a proxy server for outgoing calls. It is necessary, however, for a user agent to register to receive incoming calls from proxies that serve that domain, unless some non-SIP mechanism is used by the location service to populate the SIP URIs and Contacts of end points. A Register request may contain a message body, although its use is not defined in the standard.^[4]

```
REGISTER sip:registrar.pqst.gr SIP/2.0
Via: SIP/2.0/UDP 201.202.203.204:5060;
branch=z9hG4bK313
Max-Forwards: 70
To: sip:abc@pqst.gr
From: <sip:secretary@academy.pqst.gr>;tag=543131
Call-ID: 48erl8132409wqer
CSeq: 1 REGISTER
Contact: sip:abc@stol.pqst.gr
Contact: abc:xyz@pq.org
Content-Length: 0
```

B. Invite

The Invite method is used to establish media sessions between user agents. An Invite usually has a message body containing the media information of the caller. The message body can also contain other session information, such as a resource list. If an Invite does not contain media information, the Ack contains the media information of the UAC.^[4]

```
INVITE sip:411@efgh.at;user=phone SIP/2.0
Via: SIP/2.0/UDP abc.edu.at:5060;
branch=z9hG4bK1d32hr4
```

Max-Forwards: 70
 To: <sip:411@efgh.at;user=phone>
 From: abc <sip:c.abc@salzburg.edu.at> ;tag=817234
 Call-ID: 12-45-A5-46-F5-43-32-F3-C2
 CSeq: 1 INVITE
 Subject: Timetables
 Allow: INVITE, ACK, CANCEL, BYE, OPTIONS,
 REFER, SUBSCRIBE, NOTIFY
 Contact: sip:c.abc@efgh.edu.at
 Content-Type: application/sdp
 Content-Length: 195

v=0
 o=abc 2890842326 2890844532 IN IP4 efgh.edu.at
 s=-
 c=IN IP4 50.61.72.83
 t=0 0
 m=audio 49172 RTP/AVP 97 98 0
 a=rtpmap:97 iLBC/8000
 a=rtpmap:98 SPEEX/8000
 a=rtpmap:0 PCMU/8000

C. Options

The Options method is used to query a user agent or server about its capabilities and discover its current availability. The response to the request lists the capabilities. A proxy never generates an Options request.^[7] A user agent or server responds to the request as it would to an Invite (i.e., if it is not accepting calls, it would respond with a 4xx or 6xx response). A success class (2xx) response can contain Allow, Accept, Accept-Encoding, Accept-Language, and Supported headers indicating its capabilities. Feature tags (such as audio, video, and is focus) should be included with the Contact header field. An Options request may not contain a message body. A proxy determines if an OPTIONS request is for itself by examining the Request-URI. If the Request-URI contains the address or host name of the proxy, the request is for the proxy. Otherwise, the Options is for another proxy or user agent and the request is forwarded.^[5]

OPTIONS sip:user@carrier.com SIP/2.0
 Via: SIP/2.0/UDP abc.efgh.xyz.edu.uk
 ;branch=z9hG4bK1834
 Max-Forwards:70
 To: <sip:ghi@cam.ac.uk>
 From: J.C. Maxwell <sip:cd.max@abc.efgh.edu.uk>
 ;tag=34
 Call-ID: 747469e729acd305
 CSeq: 29 OPTIONS
 Content-Length: 0

D. 200 OK

The 200 OK response has two uses in SIP.^[4] When used to accept a session invitation, it will contain a message body containing the media properties of the UAS(called party). When used in response to other requests, it indicates a successful completion or receipt of the request.

SIP/2.0 200 OK
 Via: SIP/2.0/UDP abc.efgh.xyz.edu.uk;tag=512A6
 ;branch=z9hG4bK0834 ;received=192.0.0.2
 To: <sip:tolls@cam.ac.uk>;tag=432
 From: J.C. Max <sip:cd.max@ abc.efgh.edu.uk > ;tag=34
 Call-ID: 747469e729acd305
 CSeq: 29 OPTIONS
 Contact: <sip:will@tuto.cam.ac.uk>;audio;video
 Allow: INVITE, OPTIONS, ACK, BYE, CANCEL,
 REFER Supported: replaces, join
 Accept-Language: en, de, fr
 Content-Type: application/sdp Content-Length: 170
 v=0
 o=jc 2590845378 2590945578 IN IP4 tuto.cam.ac.uk
 s=-
 c=IN IP4 tutors.cam.ac.uk
 t=0 0
 m=audio 32852 RTP/AVP 96 0
 a=rtpmap:96 SPEEX/8000
 a=rtpmap:0 PCMU/8000
 m=video 82852 RTP/AVP 34
 a=rtpmap:34 H263/90000

E. Subscribe

The SUBSCRIBE method is used by a UA to establish a subscription for the purpose of receiving notifications (via the NOTIFY method) about a particular event. A successful subscription establishes a dialog between the UAC and the UAS. The subscription request contains an Expires header field, which indicates the desired duration of the existence of the subscription.^[6] After this time period passes, the subscription is automatically terminated. The subscription can be refreshed by sending another SUBSCRIBE within the dialog before the expiration time. A server accepting a subscription returns a 200 OK response also containing an Expires header field. The expiration timer can be the same as the request, or the server may shorten the interval, but it may not lengthen the interval. There is no UNSUBSCRIBE method used in SIP—instead a SUBSCRIBE with Expires:0 requests the termination of a subscription and hence the dialog.

SUBSCRIBE sip:ptolemy@rosettastone.org SIP/2.0
 Via: SIP/2.0/UDP proxy.ele.co.uk:5060
 ;branch=z9hG4bK348471123

Via SIP/2.0/UDP parl.ele.co.uk:5060
 ;branch=z9hG4bKADA ;received=192.0.3.4
 Max-Forwards: 69
 To: sip:ptol@stone.org
 From: abc <sip:abc@ele.co.uk>;tag=1814
 Call-ID: 452k59252058dkfj34924lk34
 CSeq: 3412 SUBSCRIBE
 Allow-Events: dialog
 Contact: <sip:ele@parl.ele.co.uk>
 Event: dialog
 Content-Length: 0

F. Notify

The Notify method is used by a user agent to convey information about the occurrence of a particular event.^[3] A Notify is always sent within a dialog when a subscription exists between the subscriber and the notifier. However, it is possible for a subscription to be established using non-SIP means (no Subscribe is sent) and may also be implicit in another SIP request type (for example, a Refer establishes an implicit subscription). Since it is sent within a dialog, the NOTIFY will contain a To tag, From tag, and existing Call-ID.

Notify requests contain an Event header field indicating the package and a Subscription-State header field indicating the current state of the subscription.^[5] The Event header field will contain the package name used in the subscription. Currently defined packages are listed in Table 4.8. The Subscription-State header field will either be active, pending, or terminated. A Notify is always sent at the start of a subscription and at the termination of a subscription. If a Notify contains incremental (delta) state information, the message body will contain a state version number that will be incremented by 1 for each Notify sent. This way, the receiver of the Notify can tell if information is missing or received out of sequence.

NOTIFY sip:ty@parl.ele.co.uk SIP/2.0
 Via: SIP/2.0/UDP cart.rosettastone.org:5060
 ;branch=z9hG4bK3841323
 Max-Forwards: 70
 To: Thomas Young
 <sip:tyoung@elasticity.co.uk>;tag=1814 From: <sip:ptol@rosettastone.org>;tag=5363956k
 Call-ID: 452k59252058dkfj34924lk34
 CSeq: 3 NOTIFY Contact:
 <sip:ptolemy@cart.rosettastone.org>
 Event: dialog
 Subscription-State: active; expires=180
 Allow-Events: dialog
 Content-Type: application/xml+dialog

Content-Length: ...

II. LITERATURE SURVEY

SIP Protocol establishes session between participating User agents using messages.^[1] The Subscribe and Notify Messages establish a dialog for propagating the messages, which is a loss of bandwidth since Notify message is sent only at the beginning and end of the dialog.^[9] This results in loss of resources allocated for the dialog. This creates performance degradation impact on the network, since there is no transmission of data there is loss of dedicated bandwidth for retrieving the capability of remote user using the Subscribe and Notify messages.^[2]

III. EXISTING METHODS

The Subscribe and Notify Messages establish a dialog for propagating the messages, which is a loss of bandwidth since Notify message is sent only at the beginning and end of the dialog. This results in loss of resources allocated for the dialog.

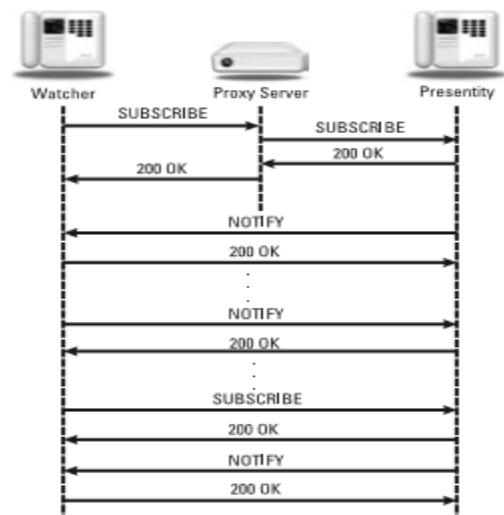


Fig. 2 Subscribe and Notify Call Flow

IV. PROPOSED METHODOLOGY

Options Message can be used as a substitute for Subscribe and Notify SIP messages. Option message requests the capabilities of the User Agent and the User Agent sends its response in 200 OK message to the requesting User Agent.^[8] This reduces the bandwidth since dialog need not be maintained between participating user agents. The User Agents can exchange information using Offer/Answer Model by exchanging Options and 200 OK messages. It also eliminates the overhead of resubscribing in the dialog the increase the life time of the dialog.^[4]

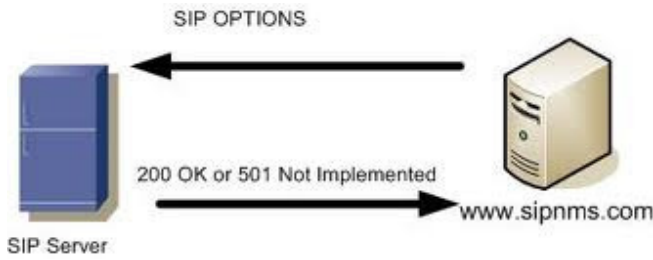


Fig. 3 Options Message Architecture

V. INPUT AND COMPUTATION

A. Input

User Agents and Proxy Server for Session Establishment.



Fig. 4 SIP Soft Phone



Fig. 5 SIP Server

B. Computation

The request for capabilities of a user agent can be made by sending Options message.

```

OPTIONS sip:user@carrier.com SIP/2.0
Via: SIP/2.0/UDP abc.kin.camb.edu.uk
    ;branch=z9hG4bK1834
Max-Forwards:70
To: sip:will@cam.ac.uk
From: J.C. Max <sip:jam.max@kin.camb.edu.uk> ;tag=34
Call-ID: 747469e729acd305
CSeq: 29 OPTIONS
Content-Length: 0
    
```

The reply to the Options request is sent using 200 OK message. The capabilities of the user agent are passed in the body of the 200 OK message.^[10]

```

SIP/2.0 200 OK
Via: SIP/2.0/UDP100.101.102.103:5060
    ;branch=z9hG4bK17a
To: Heis<sip:we.heis@munich.de>;tag=314159
From: E. Schro <sip:schroe5244@wave.org>;tag=42
Call-ID: 4827311-391-32934
CSeq: 1 INVITE
Contact: <sip:we.heis@200.201.202.203>
Content-Type: application/sdp
Content-Length: 159
v=0
o=heis 2890844526 2890844526 IN IP4 200.201.202.203
s=phone call
c=IN IP4 200.201.202.203
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/800
    
```

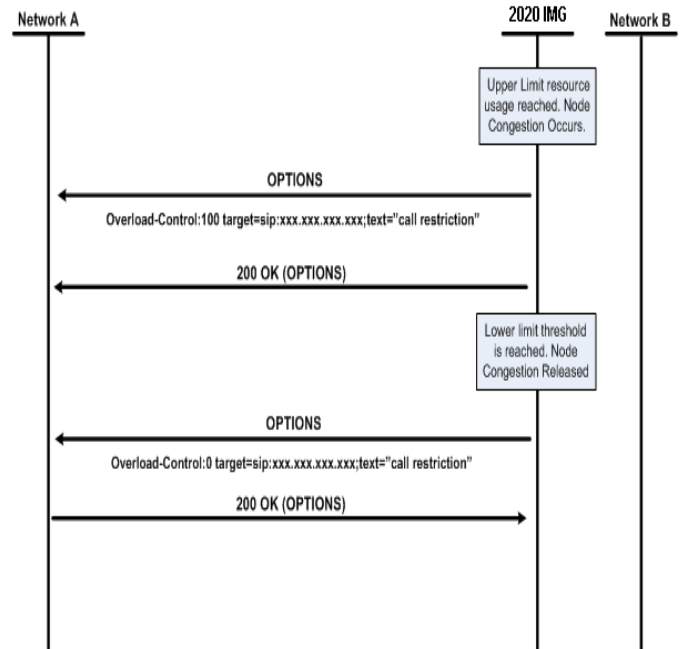


Fig. 6 Options Message Flow

VI. ALGORITHM

Input: User Agent requests Capability of another User Agent

Output: The capabilities of the User Agent is sent in the 200 OK message

- [1] Register the User Agent to the SIP registrar.
- [2] User Agent sends Subscribe request to the User Agent whose capability is to be known.
- [3] The Receiving User Agent sends its capabilities in the Notify message.
- [4] Analyse bandwidth utilization using wireshark for Subscribe and Notify session.
- [5] User Agent sends Options request to the User Agent whose capability is to be known.
- [6] The Receiving User Agent sends its capabilities in the 200 Ok message.
- [7] Analyse bandwidth utilization using wireshark for Options and 200 Ok messages.

VII. EXPERIMENTATION AND CRITICAL ANALYSIS

SIP protocol is used for session establishment, hence usage of more bandwidth by SIP protocol message creates overhead. The use of Subscribe and Notify messages for knowing capability information creates a dialog during a session which causes wastage of bandwidth since Notify message is sent only during the initiation and termination of the dialog. Options message request capability and the requested User Agent sends capability in 200 OK message. This process does not waste bandwidth by creating an unused dialog.

User Agents are registered to the SIP server. One user agent send Subscribe request to the other user agent in order to gain the other user's capability. The other user sends Notify response to the requesting agent. This involves session creation. The performance of bandwidth utilization is measured using wireshark software.

Similarly Options message is sent from one user agent to another whose capability is to be known. The response for the request is sent by the receiving agent using the 200 Ok message. The bandwidth utilization is measured using wireshark software.

Comparing the bandwidth utilization of Subscribe and Notify messages to the Options and 200 Ok messages it is found that Options and 200 Ok messages utilize less bandwidth compared to Subscribe and Notify messages.

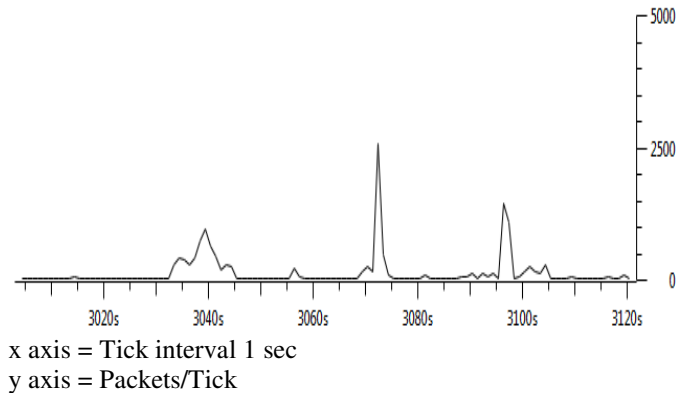


Fig. 7 Graph for Subscribe and Notify Message flow

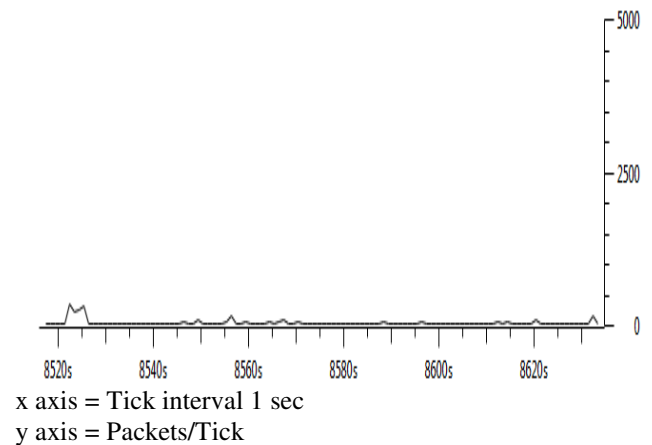


Fig. 8 Graph for Options and 200 Ok Message flow

VIII. CONCLUSION

In this work it was possible for minimizing and efficiently using the bandwidth. From this work it is possible to reduce the overhead on bandwidth. Since Session Initiation Protocol is a session establishment protocol, the more efficiently the SIP messages are communicated increase better bandwidth utilization and increases network performance. Hence the usage of Options and 200 Ok messages to investigate the capability of a User agent is more beneficial compared to the Subscribe and Notify messages for addressing capability.

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