

# SIP: Ringing Timer Support for INVITE Client Transaction

Poojan Tanna

([poojan@motorola.com](mailto:poojan@motorola.com))

Motorola India Private Limited

Outer Ring Road, Bangalore, India 560 037

**Abstract**—The time for which the Phone Call can ring should be configurable at the switch. Underlying protocol responsible for establishing the call should provide the facility to configure the Ringing Time. It has been observed that the SIP Phone Call can ring for the time indicated by Timer B. The Timer B controls transactions timeout. If the Called party does not answer the Phone before Timer B expires then the SIP call is not established. This paper suggests modification to SIP Protocol as defined in RFC 3261 with respect to its handling of Timer B and configuring the time for which SIP Phone can ring before the call is disconnected.

*Index Terms*— SIP, Timer, SIP Call No Answer.

## I. INTRODUCTION

The SIP RFC 3261 ( Section 17.1.1 ) introduces INVITE Client Transaction. It defines Timer B as INVITE Client Transaction timeout timer. In INVITE Client Transaction state machine Timer B is started in the calling state. The default value of Timer B is 32 seconds. RFC 3261 mentions explicitly the behavior if the Timer B fires in the Calling State. But it does not mention when to stop the Timer B and what to do if it fires in the Proceeding State. Here we will discuss how to solve this ambiguity in RFC 3261. The proposed solution will modify the handling of Timer B and INVITE Client Transaction state machine such that the ringing time will be independent of Timer B. The remainder of the paper is organized as follows. In Section 2 we will understand the basic SIP Call flow and role of Timer B. Section 3 describes the INVITE Client Transaction state machine and the issues with it. Section 4 presents the proposed solution and option to configure the desired ringing time. Implementation of the proposed changes on various SIP Network entities is discussed in Section 5. In Section 6 interoperability test is conducted. Section 7 comes out the conclusion.

## II. SIP CALL FLOW

### A. Overview of SIP

SIP (Session Initiation Protocol, RFC 3261)[1] is a

(conferences) such as Internet telephony calls. SIP is becoming increasing popular for IP telephony application-layer control protocol that can establish, modify, and terminate multimedia sessions applications. It provides Voice over IP (VoIP) signaling and when used in conjunction with other IETF standards like Real-time Transport Protocol (RTP) [2] for transporting real-time data and providing QoS feedback, the Real-Time streaming protocol (RTSP) [3] for controlling delivery of streaming media, and the Session Description Protocol (SDP) [4] for describing multimedia sessions it provide a complete IP telephony solution.

SIP defines four logical types of entities: User Agents, Registrars, Proxy server and Redirect servers [6]. User Agents initiate request and are usually their final destinations. Internet telephones and conferencing software are example of Users Agents. Registrars keep track of users within their assigned network domain. Proxy servers are application-layer router that forward SIP request and responses. Redirect servers receive request and then return the location of another SIP User Agent or server where the user might be found. It is quite common to find proxy, redirect, and registrar servers implemented within the same program.

### B. SIP Call Flow

Lets look at a simple call flow of a SIP based voice call and the role of Timer B. The call flow as depicted in Fig. 1 is explained as follows

1. Here Bob is calling Mary. To initiate a SIP Voice Call, Bob's phone is sending a initial INVITE request. Proxy routes the INVITE request to Mary.
2. Proxy sends 100 Trying response to Bob.
3. Mary's phone sends 180 Ringing response and it starts ringing.
4. The number of rings or the time for which Mary's phone will ring depends upon the value of Timer B

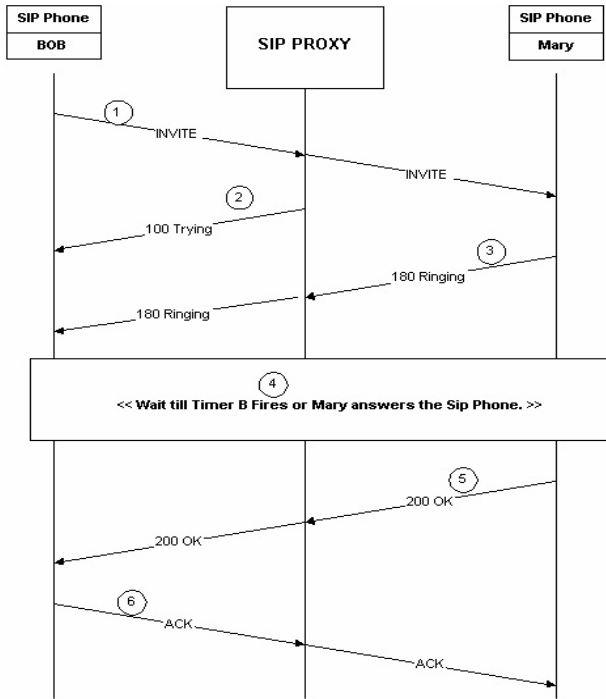


Fig. 1. SIP Call Flow

It is a protocol timer and is used for other protocol related activities. Default value of this timer is 32 seconds. Thus by default the Mary Phone will ring for 32 seconds.

5. Assuming that Mary answers the phone before Timer B expires then Mary's Phone sends 200 OK response and the call is UP.

6. Bob's Phone sends the ACK.

### III. INVITE CLIENT TRANSACTION

The INVITE Client Transaction is to initiate a SIP call. It consists of a three-way handshake. The Client Transaction sends an INVITE, the server transaction sends responses, and the client transaction sends an ACK.

#### A. SIP Call States

The INVITE Client Transaction provides its functionality through the maintenance of a state machine. The SIP RFC 3261 defines the following states for the INVITE Client Transaction state machine.

**Calling State:** This state indicates that the User Agent has initiated a new client transaction with an INVITE request. If an unreliable transport is being used, the client transaction must start Timer A. Timer A controls request retransmissions. If Timer A fires when client transaction is in this state then INVITE

request must be retransmitted. Timer B is also started in this state. If Timer B fires then the client transaction transitions to Terminated State as explained below.

**Proceeding State:** This state indicates that the client transaction has received a provisional response while in the "Calling" state. In this state, the client transaction should not retransmit the request.

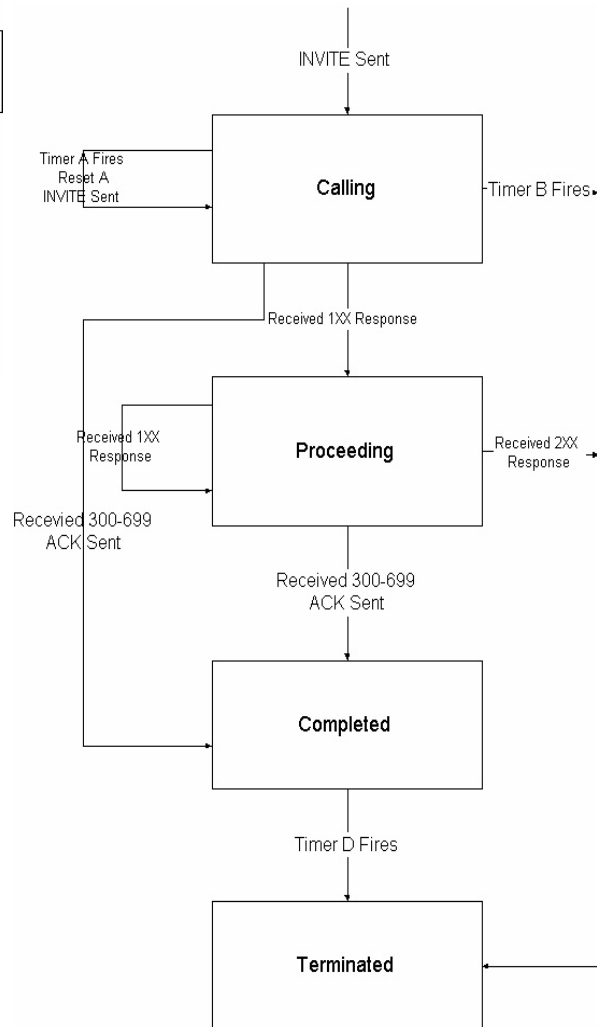


Fig. 2. INVITE Client Transaction State Machine

**Completed State:** This state indicates that the client transaction has received a 300-699 response. The client transaction should start Timer D in this state.

**Terminated State:** This state indicates that either Timer D or Timer B has fired. The client transaction can also enter this state if it receives 2xx response while in the Proceeding State. The client transaction must be destroyed the instant it enters this state. ACK is sent as a response of 2xx request and SIP Call is UP.

### B. Issues with the INVITE Client Transaction

The INVITE Client Transaction State Machine is as shown in Fig. 2. If the client transaction is in Calling State and Timer B fires then it silently disposes the client transaction. When to stop Timer B and whether it is applicable to Proceeding state is not clear in RFC 3261. If the calling party's INVITE transaction has transitioned to Terminated State, then it will be disposed. But the called party transaction will still be in Proceeding state and called party's phone will be ringing. Also, if the Called party answers the phone after the Calling Party client transaction has transitioned to Terminated State then 2xx response from the called party will not be useful to establish the SIP Call. There should be some communication from the calling party to inform the called party about the Timer B timeout so that both the parties will release the resources and gracefully dispose the transaction.

### C. Changing the value of Timer B

Timer B controls transaction timeout mechanism. The client transactions must start with  $64 * T1$ . All of the transaction timers scale with T1, and adjusting T1 changes their values. T1 is an estimate of the round-trip time (RTT), and it defaults to 500 ms.

For example, Timer A controls request retransmissions starts with T1 timer. When timer A fires, the client transaction must retransmit the request by passing it to the transport layer, and must reset the timer with a value of  $2 * T1$ .

Increasing the value of T1 to make the SIP Phone ring for more time will not be a feasible solution since it increases Timer A and the retransmission time of the INVITE request.

## IV. PROPOSED SOLUTION

The proposed solution is to introduce a new state called "Ringing State" and a new timer "Ringing Timer" in the SIP RFC 3261 INVITE Client Transaction.

### A. Ringing State

This state will indicate that client transaction has received 180 (Ringing) or 183 (Progress). It also means that Phone at the called party is Ringing. In this state Ringing Timer as explained below is also started.

### B. Ringing Timer

This Timer will be new "TIMER-L" in RFC 3261.

This new Timer will start in Ringing State. The System Administrator can configure this Timer. If this timer fires a CANCEL request is sent to the called party. On receiving 487 response, the INVITE client transaction is disposed.

### C. Ringing Timer Range

The minimum value of Ringing Timer should be equal to 32 seconds. It will ensure the compatibility with the SIP Network entities which does not have Ringing State and Ringing Timer implemented.

Maximum value of Ringing Timer should be evaluated considering these two timers. 1. ITU-T 931 recommendations timer T301. The minimum value of Timer T301 is 180 seconds. The Timer T301 is started on receiving "Alerting" Message and is stopped when "Connect" message is received. 2. Timer C is applicable to SIP Proxy. Timer C is set when it receives initial INVITE request from the client transaction. The minimum value of Timer C is 180 seconds.

The default value of the ringing timer should be 60 seconds. This will allow the Called Party's phone to ring for approximately 13 rings.

### D. Proposed State Machine

The proposed INVITE Client Transaction State Machine with the new Ringing Timer and Ringing State is shown in Fig. 3.

The operation of this new state machine is as follows:

1. The INVITE request is sent and the client transaction transitions to Calling State. Timer A and Timer B are started. When Timer A fires then INVITE request is retransmitted. Timer A is stopped if the client transaction receives provisional response. When Timer B fires then client transaction transitions to Terminated State and the client transaction is disposed.

2. When the client transaction receives 180 (Ringing) or 183 (Call Progress) response then it directly transitions to Ringing State.

3. If client transaction receives 1xx provisional response other than 180/183 then it transitions to Proceeding State. Timer B is applicable to this state. If the Timer B fires in this state then client transaction transitions to Terminated state and client transaction is disposed.

4. In the Proceeding State, if the client transaction receives 180 (Ringing) or 183 (Call Progress) then it transitions to Ringing State. Timer B is stopped. Ringing Timer is started with the configured value. Now either of these scenarios is possible a) If the

ringing timer fires in the ringing state then client transaction sends a CANCEL request. When Client transaction receives 487 response it transitions to Terminate State. b) Client transaction receives 2xx response, so it should stop the ringing timer and transitions to Terminate State. c) The client transaction received 300-699 response. It stops the ringing timer and transitions to Completed State.

5. In the completed state the Client Transaction sends ACK for 300-699 response. It also starts the Timer D. When Timer D expires, it transitions to Terminated state.

6. While in Proceeding or Ringing State if the Client Transaction receives 2xx response then it transitions to Terminate state. The SIP Call is up and the transaction is disposed. But if the Client Transaction transitions to this state since Timer B, Ringing Timer or Timer D has fired then client transaction is disposed and the SIP Call is not established.

## V. IMPLEMENTATION

The SIP Network entities acting as User Agent Client (UAC) should have the proposed INVITE Client Transaction state machine. Let us analyze some of the SIP Networks entities which need to have the proposed solution implemented.

### A. Media Gateway

Media Gateway interfaces the PSTN to IP world. For the SIP Call initiated from the PSTN side, Media Gateway will act as UAC of the SIP Call hence it should have the proposed INVITE Client Transaction state machine. These proposed changes are being implemented at commercial Media Gateways. The INVITE Client Transaction state machine of these devices is Ringing State if the called party phone is ringing. If the Called Party Media Gateway acting as User Agent Server (UAS) for this INVITE client transaction does not have proposed solution implemented then it will remain in the Proceeding State. If the called party does not answer the call within the configured ringing time and the ringing timer fires, then calling party Media Gateway which has proposed changes implemented will send CANCEL request and gracefully bring down the SIP call.

### B. SIP Proxy

The proposed solution is not mandatory for the SIP Proxy. If the SIP Proxy does not have proposed solution implemented then it will remain in the

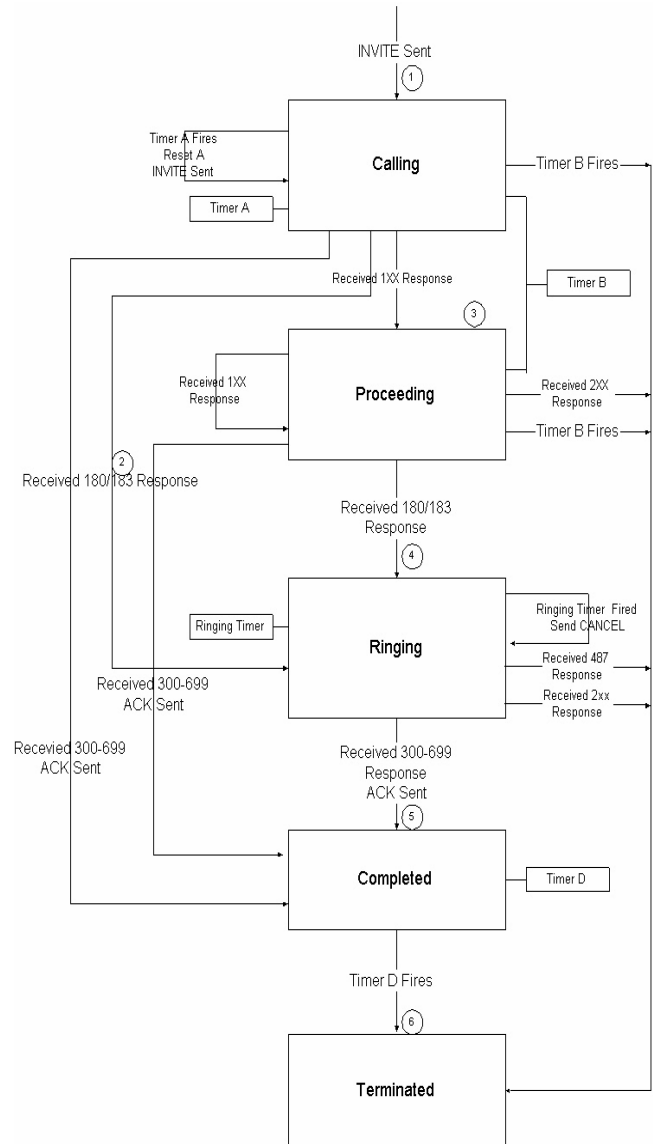


Fig. 3. Proposed INVITE Client Transaction State Machine

Proceeding State even though User Agent's INVITE Client Transaction will be in Ringing State. Both the parties will move to Terminated State together.

### C. SIP Phone

The calling party phone should have the proposed solution implemented since it acts as UAC for INVITE transaction. So if the SIP Phone which has the proposed solution implemented initiates a SIP Call and the called party does not answer this call within the ringing time then ringing timer fires. Then as per proposed INVITE Client Transaction SIP Phone will send CANCEL request and will gracefully bring down the call.

### D. SIP Registrar

SIP Registrar keeps track of users in the network domain. It is not involved in the INVITE Client Transaction state machine. So the proposed solution is not applicable to SIP registrar.

## VI. INTEROPERABILITY

To test the interoperability commercial SIP Proxy was used to setup a SIP call. This SIP Proxy does not have the proposed solution implemented.

The setup details are as shown in Fig. 4.

Here Bob is the calling party and Mary is the called party. Since Bob and Mary are using ISDN Phones hence the Media Gateways MG1 and MG2 will be SIP Endpoints respectively.

The proposed solution is implemented only on MG1. On MG1, the ringing timer is configured to the value of 60 seconds. SIP call is made via commercial SIP Proxy. Mary did not attend the phone for 60 seconds. The SIP Call flow between the Endpoints MG1 and MG2 is as shown in Fig. 5.

## VII. CONCLUSION

This paper has proposed an amendment in RFC 3261. Proposed solution introduces Ringing State and Ringing Timer in the INVITE Client Transaction state machine. These proposed changes will remove the ambiguity in the SIP RFC 3261 regarding the behavior if the Called Party does not answer the phone before Timer B expires. It also gives flexibility to the Service Provider to have desired ringing time.

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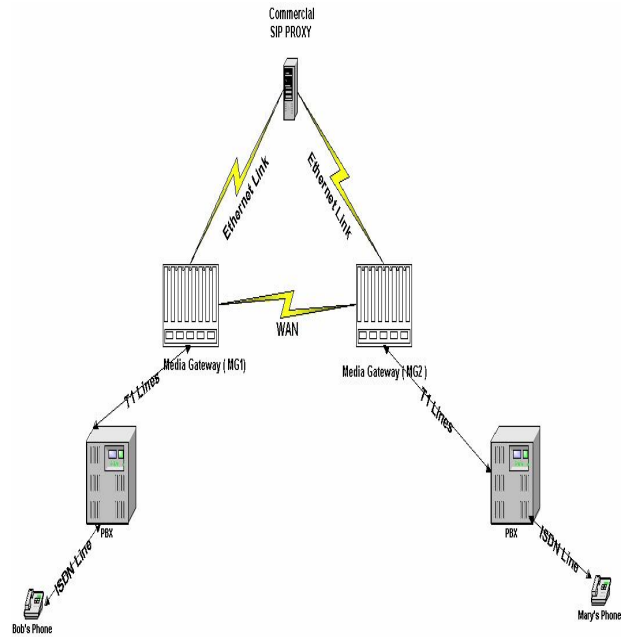


Fig. 4. Interoperability Setup

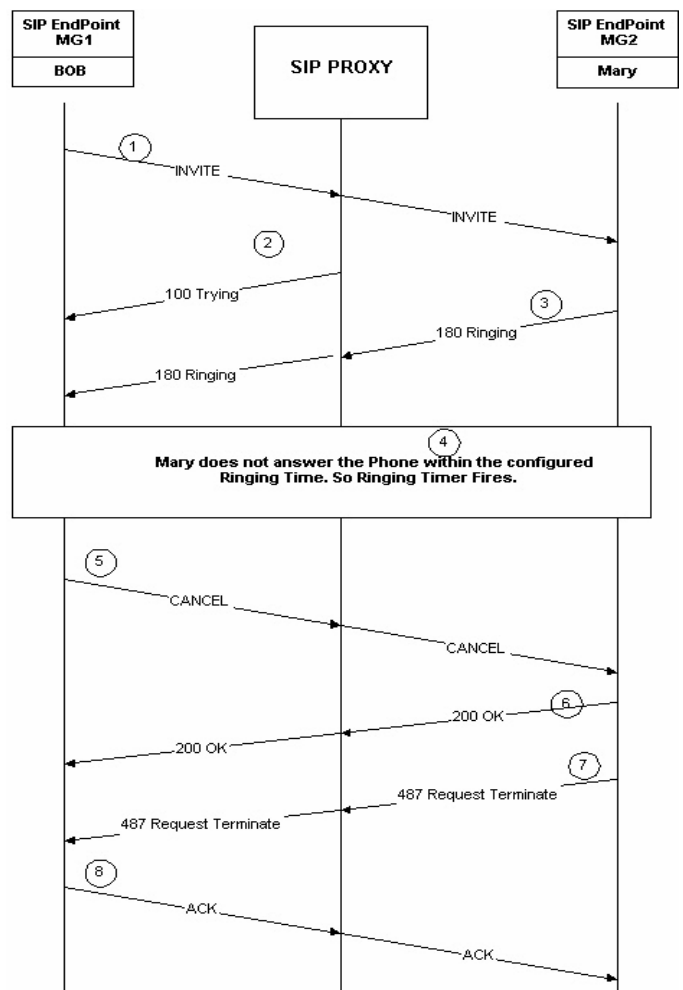


Fig. 5. Proposed SIP Call Flow

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