

SIP Interoperability Troubleshooting with the ClearSight Analyzer

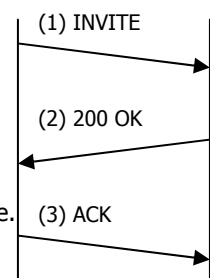
Chizuru Yajima – July 2005

IP telephony services have been progressively introduced by telecom operators and adopted by companies to take advantage of the lower call costs, management costs, and other savings they provide. SIP (Session Initiation Protocol) is the most commonly used call signaling protocol for making IP telephone calls. Although the SIP protocol has been specified in the RFC3261 as a standard, problems remain when interconnecting between devices from different vendors as some detailed specifications are not defined. These different implementations of the standard are referred to as "dialects" and the different dialects used by each vendor are a cause of interconnection problems between SIP servers, telephones, and other terminals. To solve these problems, the various parties involved in providing the telephone service must conduct SIP interconnection tests, and a LAN analyzer is an essential tool for collecting, analyzing, and troubleshooting data from these tests. This case study describes an example of SIP protocol analysis using the ClearSight Analyzer from ClearSight Networks of the USA. www.clearsightnet.com

Figure 1 SIP Call Setup Procedure

The following three steps are necessary for SIP call setup:

- (1) The originating telephone sends the INVITE message to request a connection.
- (2) The terminating telephone returns a 200 OK response when the call is answered.
- (3) The originating telephone sends an ACK method to confirm reception of the response.



Problem

When an IP telephone is connected to a SIP server made by a different manufacturer, speech can only be heard in one direction.

Analysis

The ClearSight Analyzer is used to capture data (can also be used for Real Time analysis), display it in a ladder view, and check the three packets shown in Figure 1. This shows that the SIP call is connected successfully. However, the data flow also shows that the RTP (Realtime Transport Protocol) voice transmission protocol is only operating in one direction. When the data is analyzed in more detail it becomes apparent that the 200 OK response is being sent repeatedly. The fact that the response is being resent even though the ACK method is reaching the terminating telephone suggests that the terminating telephone is not recognizing the ACK message.

Proceeding next to investigate "why the terminating telephone is not recognizing the ACK message", it can be seen that the ACK message is being sent directly from the originating telephone to the terminating telephone without going via the SIP server. (See Figure 2 , step (3).)

This suggests that the terminating telephone can only recognize ACK messages that come via the SIP server. In order to make the ACK message go through the SIP server, it is necessary to include a Record-Route header in the INVITE method containing the address information for the SIP server (SIP URI).

When the INVITE message from the captured data is viewed in ClearSight Analyzer, it can be seen that the SIP server is transferring the INVITE without adding a Record-Route header. (If no Record-Route header is used, the ACK method is sent to the address (SIP URI) indicated by the Contact header in the 200 OK response. Viewing the frame containing the 200 OK response shows that the Contact header contains the redirected address (SIP URI) of the terminating telephone.)

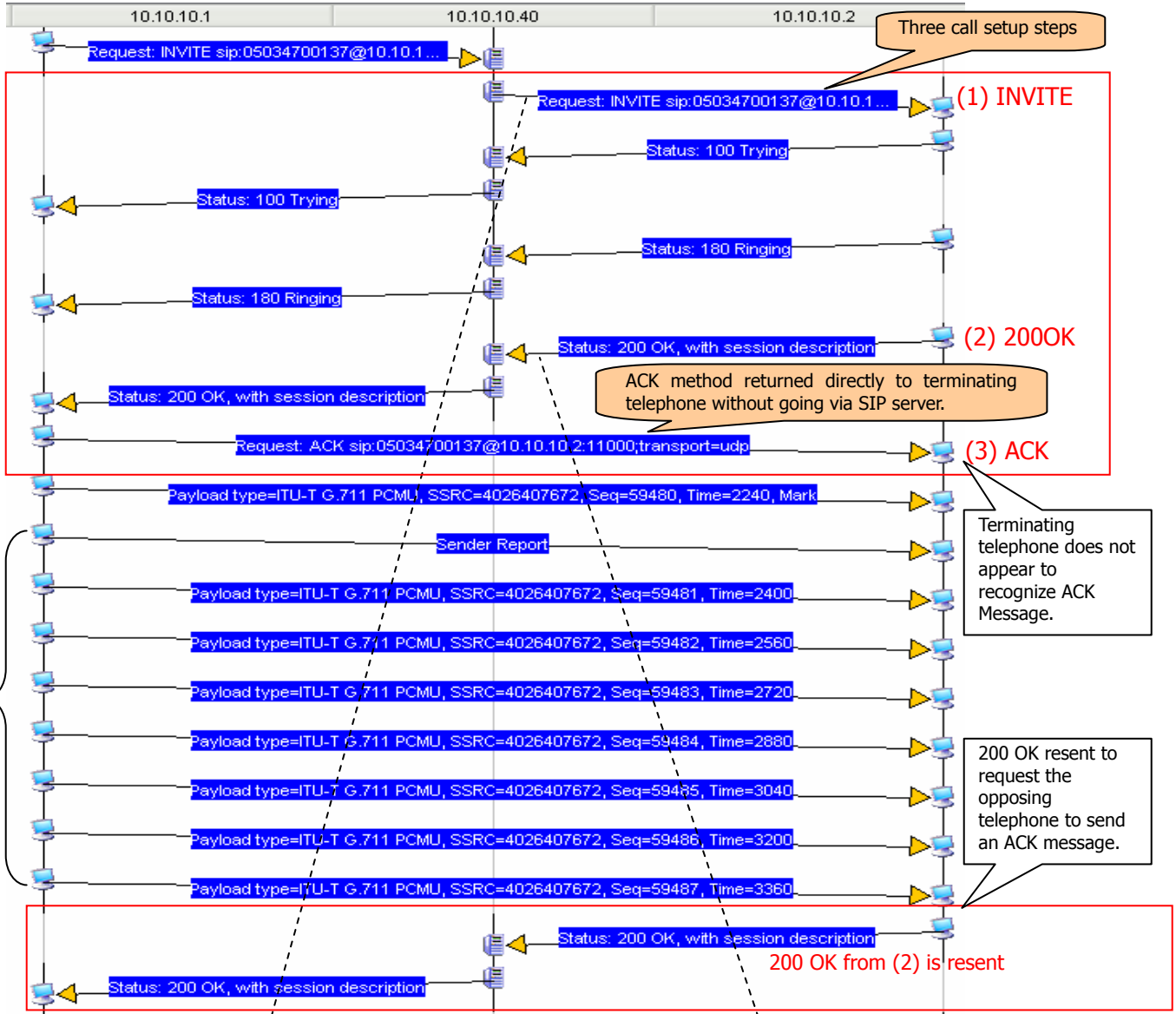
Solution

The SIP server used in the system has a setting that controls use of the Record-Route header. Speech between the telephones started working correctly when this setting was turned ON.

This example identified a problem caused by the protocol implementation in the terminating telephone not accepting ACK methods that do not come from the SIP server. The problem was solved by changing the SIP server setting to match the operation of the terminating telephone.

The case study demonstrates the importance of testing the operation of SIP servers with IP telephones, soft phones, and other terminals before installation, and shows how a LAN Analyzer like the ClearSight Analyzer that can translate the flow of data and display it in ladder format is an essential tool for conducting these tests.

Figure 2 ClearSight Analyzer's Ladder View



RTP
One side only

Session Initiation Protocol

- Request-Line: INVITE sip:05034700137@10.10.10.2:11000;transport=udp SIP/2.0
- Method: INVITE
- Resent Packet: False
- Message Header
- Via: SIP/2.0/UDP 10.10.10.40:5060;branch=fcf3e9c380801d4e639961dde26a7c84.0
- Via: SIP/2.0/UDP 10.10.10.1:11000;branch=z9hG4bK1173376624-6505354347
- To: sip:05034700137@10.10.10.40
- SIP to address: sip:05034700137@10.10.10.40
- From: sip:05034701111@10.10.10.40;tag=7852117532-935342251
- SIP from address: sip:05034701111@10.10.10.40
- SIP tag: 7852117532-935342251
- Max-Forwards: 69
- CSeq: 1 INVITE
- Call-ID: -41367-7306887360-90005@10.10.10.1
- Contact: <sip:05034701111@10.10.10.1:11000;transport=udp>
- Min-SE: 180
- Session-Expires: 180
- Supported: timer
- Content-Length: 492
- Content-Type: application/sdp

**INVITE message sent via SIP server:
No Record-Route header.**

Session Initiation Protocol

- Status-Line: SIP/2.0 200 OK
- Status-Code: 200
- Resent Packet: False
- Message Header
- Via: SIP/2.0/UDP 10.10.10.1:11000;branch=z9hG4bK1173376624-6505354347
- To: sip:05034700137@10.10.10.40;tag=-99032779429569456770
- SIP to address: sip:05034700137@10.10.10.40
- SIP tag: -99032779429569456770
- From: sip:05034701111@10.10.10.40;tag=7852117532-935342251
- SIP from address: sip:05034701111@10.10.10.40
- SIP tag: 7852117532-935342251
- CSeq: 1 INVITE
- Call-ID: 41367-7306887360-90005@10.10.10.1
- Contact: <sip:05034700137@10.10.10.2:11000;transport=udp>
- Require: timer
- Session-Expires: 180;refresher=uas
- Supported: timer
- Content-Length: 199
- Content-Type: application/sdp

**200 OK:
Terminating address set in Contact header.**