



Welcome

Technical Services Virtual Boot Camp Session 6

Technical Services India Team



Recap – Session 5 (27th Jan)

Technology

CUCM Media Resources

- Transcoder, MTP
- OOB, In-Band DTMF & MoH
- Trace Snippets

Cisco Support Community

- Learn about CSC top contributors programs
- CSC Events
- CSC on Social Media



Q&A



Course Material

Cisco Support Community

Home | Top Contributors | Expert Corner

Home > NetPro > Online Tools and Resources > Technical Documentation Ideas

Up to Documents in Technical Documentation Ideas

Document

Created on: Dec 7, 2013 4:41 AM by Vinay Sharma - Last Modified: Jan 21, 2014 7:02 AM by Vinay Sharma

VERSION 7

Technical Services Virtual Boot Camp Series

- Session 1: LAN Switching - Technical Services Virtual Boot Camp
- Session 2: LAN Switching - Technical Services Virtual Boot Camp
- Session 3: Security - Technical Services Virtual Boot Camp
- Session 4: Security - Technical Services Virtual Boot Camp
- Session 5: Voice - Technical Services Virtual Boot Camp
- Session 6: Voice - Technical Services Virtual Boot Camp

Session 1: LAN Switching - Technical Services Virtual Boot Camp

- Session Presentation - Troubleshooting and Upgradation on Cisco LAN switches.pptx
- Video - Troubleshooting and Upgradation on Cisco LAN switches
- Q&A from Troubleshooting and Upgradation on Cisco LAN switches Session 1
- Cisco Technical Assistance Center (TAC) Support Model - Technical Services Virtual Boot Camp Series

Session 2: LAN Switching - Technical Services Virtual Boot Camp

- Session Presentation - Understanding LAN Switching Features – STP, QOS and Stacking.pptx
- Video - Understanding LAN Switching Features – STP, QOS, and Stacking



<https://supportforums.cisco.com/docs/DOC-37994...PPT>

<https://supportforums.cisco.com/videos/7517...Video>

<https://supportforums.cisco.com/docs/DOC-37851...Q&A>



Today Agenda (Session -6)

Technology

CUCM Troubleshooting

- Troubleshooting methods
- Troubleshooting SIP Call Flows
- Case studies
- Troubleshooting tools – Wireshark/Translator - X

Cisco Support Community

- How to Stay connected with experts on CSC
- How to explore & create new content, answered and unanswered discussion

Q&A





Introduction



Nirmal Sodani
Technical Support Manager



Mohit Mmangal
Manager, CSC



Amit Singh
TAC Escalation Engineer



Vinay Sharma
Lead, CSC



Raees Shaikh
TAC Escalation Engineer



Shiv Goel
Technical Support Manager



Technology – VOICE

Amit Singh

Raes Shaikh



SIP Concepts & Troubleshooting

Amit Singh and Raees Shaikh

27th-January, 2014

Agenda

- SIP Concepts
- Trace Collection
- Tools
- Initial Analysis to TAC
- Troubleshooting

Agenda

- SIP Concepts
 - User Agents
 - SIP Messages
 - Requests and Responses
 - Headers
- Media Negotiation
 - SDP
 - Early Offer vs. Delayed Offer
 - DTMF Relay
- Trace Collection
- Tools
- Initial Analysis to TAC
- Troubleshooting

User Agents

- User Agent Clients (UAC) send requests to User Agent Servers (UAS)
- User Agent Servers send responses to the requests
- Most SIP devices are both a UAC and a UAS (they both initiate and accept requests)
- Unified CM and CUBE are both Back-to-Back User Agents (B2BUA) (as opposed to Proxies)

SIP Request Methods

- **INVITE** - A user or service is being invited to participate in a multimedia session
- **ACK** - Confirms that a client has received a final response to an INVITE request
- **BYE** - Terminates an existing session; can be sent by any user agent (in a multiparty session)
- **CANCEL** - Cancels pending requests; does not terminate sessions that have been accepted
- **OPTIONS** - Queries the capabilities of servers (Also used as a keep alive) **Ex. SIP Options PING**
- **REGISTER** - Registers the user agent with the registrar server of a domain

*Reference http://en.wikipedia.org/wiki/List_of_SIP_request_methods

SIP Request Methods Cont.

- **INFO** (RFC 2976) - to send more information within an established dialog
- **PRACK** (RFC 3262) - to acknowledge a provisional response
- **SUBSCRIBE** (RFC 3265) - to tell a remote node to look for a certain event
- **NOTIFY** (RFC 3265) - to respond when that certain event occurs
- **UPDATE** (RFC 3311) - to update parameters of a session set-up
- **MESSAGE** (RFC 3428) - SIP instant messaging
- **REFER** (RFC 3515) – to “refer” one UA to communicate with another UA
- **PUBLISH** (RFC 3903) - to push UA state information to a compositor/presence server

SIP Invite Method and Headers

SIP Method
SIP URI
SIP Version

```
INVITE sip:+18775551234@172.18.159.231:5060 SIP/2.0
Via: SIP/2.0/UDP 172.18.106.59:5060;branch=z9hG4bK1515b3154665
From: "Test User 1" <sip:9195551111@172.18.106.59>;tag=97903bc0-43adcd-45510543
To: <sip:+18775551234@172.18.159.231>
Call-ID: 7c0ca800-bb01baf9-1468e-3b6a12ac@172.18.106.59
Supported: timer,resource-priority,replaces
User-Agent: Cisco-CUCM8.6
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: <sip:172.18.106.59:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 2081204224-3137452793-0000000466-0996807340
Session-Expires: 1800
P-Asserted-Identity: "Test User 1" <sip:9195551111@172.18.106.59>
Contact: <sip:9195551111@172.18.106.59:5060>;video;audio
Max-Forwards: 69
Content-Length: 864
Content-Type: application/sdp
```

SIP Response and Headers

Response
code

Free-text
Reason

SIP/2.0 404 Not Found

Via: SIP/2.0/UDP 172.18.106.59:5060;branch=z9hG4bKb5291d44b969a4

From: "TEST" <sip:89915644@172.18.106.59>;tag=19210123ca7-45568313

To: <sip:+19195551212@10.81.2.30>;tag=253488-726

Date: Mon, 16 Jan 2012 04:00:22 GMT

Call-ID: e59bc600-f1319fa5-b1ea4a-3b6a12ac@172.18.106.59

CSeq: 101 INVITE

Allow-Events: telephone-event

Server: Cisco-SIPGateway/IOS-15.2.2.T

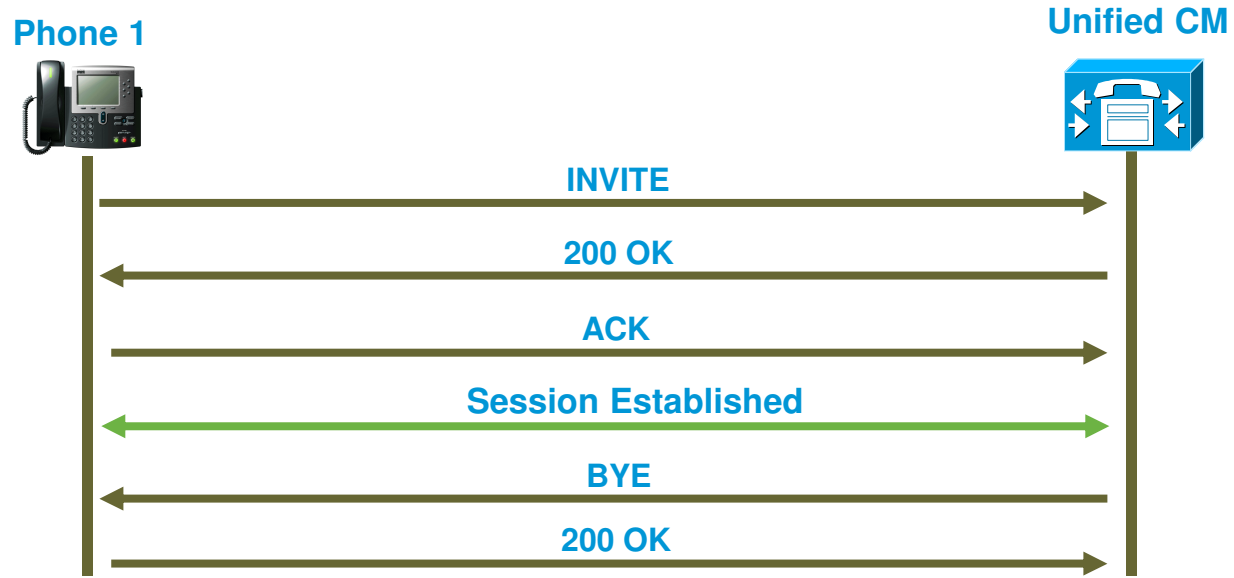
Reason: Q.850;cause=1

Content-Length: 0

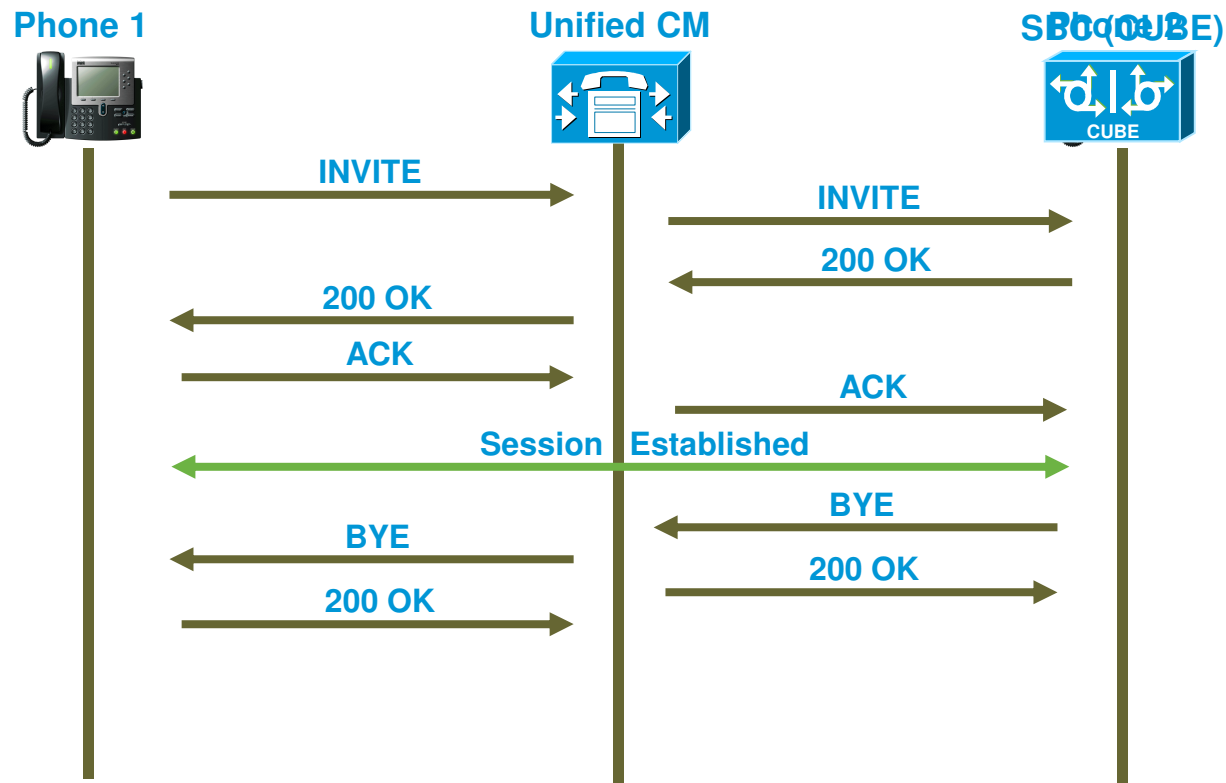
SIP Responses

Response Code	Description	Example
1xx	Informational – Request Received and Continuing to Process Request	100 Trying 180 Ringing 183 Session Progress
2xx	Success – Action was successfully received, understood, and accepted	200 OK 202 Acceptable
3xx	Redirection – Another SIP Element needs to be contacted in order to complete the request	300 Multiple Choices 301 Moved Permanently 302 Moved Temporarily
4xx	Client Error – Request contains bad syntax or cannot be fulfilled at this server	401 Unauthorized 404 Not Found 406 Not Acceptable 486 Busy Here 488 Not Acceptable Here
5xx	Server Error – Server failed to fulfill an apparently valid request	503 Service Unavailable
6xx	Global Failure – Request is invalid at any server	600 Busy Everywhere 603 Decline

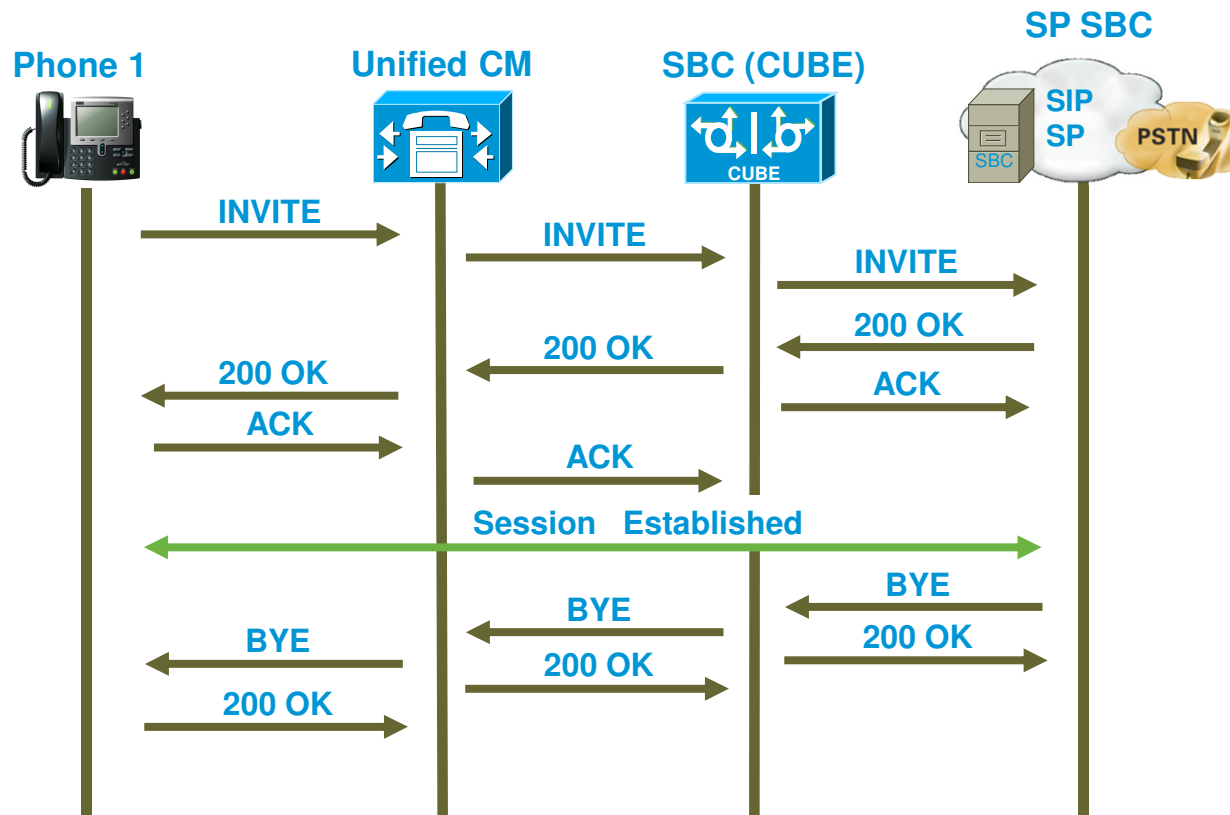
Basic SIP Call Setup



Basic SIP Call Setup with B2BUA (Unified CM)



Basic SIP Call Setup with Unified CM and CUBE



Agenda

- SIP Concepts
- Media Negotiation
 - SDP
 - Early Offer vs. Delayed Offer
 - DTMF Relay
- Trace Collection
- Tools
- Initial Analysis to TAC
- Troubleshooting

Media Negotiation

- SIP uses the offer/answer model described in **RFC 3264** to negotiate media using SDP
- One endpoint sends an offer SDP containing all the capabilities the endpoint wishes to negotiate.
- SDP contains m lines for each media stream being negotiated (i.e. audio, video, content channel, etc...)
- Receiving endpoint sends an answer SDP that contains the same or a subset of capabilities received in the offer.
- Per RFC 3264, “For each "m=" line in the offer, there **MUST** be a corresponding "m=" line in the answer. The answer **MUST** contain exactly the same number of "m=" lines as the offer.”

Session Description Protocol (SDP) - Offer

```
v=0
o=Cisco-SIPUA 26964 0 IN IP4 172.18.159.152
s=SIP Call
t=0 0
m=audio 29254 RTP/SAVP 0 8 18 102 9 116 124 101
c=IN IP4 172.18.159.152
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:102 L16/16000
a=rtpmap:9 G722/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:124 ISAC/16000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
m=video 25466 RTP/AVP 97
c=IN IP4 172.18.159.152
b=TIAS:1000000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42801E
a=recvonly
```

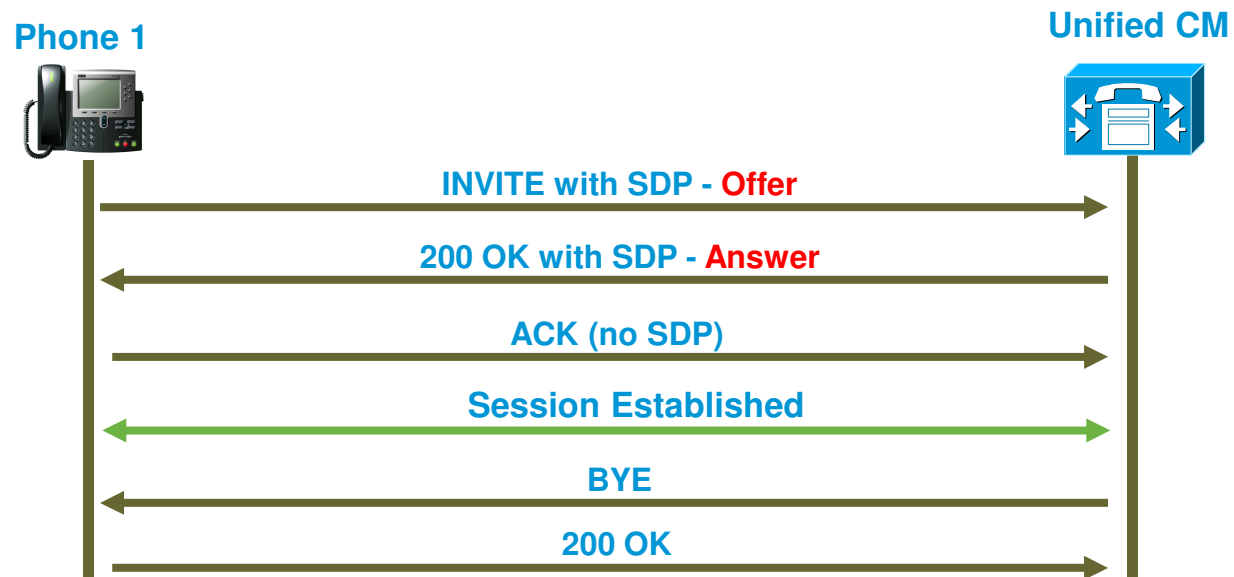
Session Description Protocol (SDP) - Answer

```
v=0  
o=CiscoSystemsCCM-SIP 2000 1 IN IP4 172.18.106.59  
s=SIP Call  
c=IN IP4 172.18.159.152  
t=0 0  
m=audio 30308 RTP/AVP 0 101  
a=rtpmap:0 PCMU/8000  
a=ptime:20  
a=rtpmap:101 telephone-event/8000  
a=fmtp:101 0-15  
m=video 0 RTP/AVP 97
```

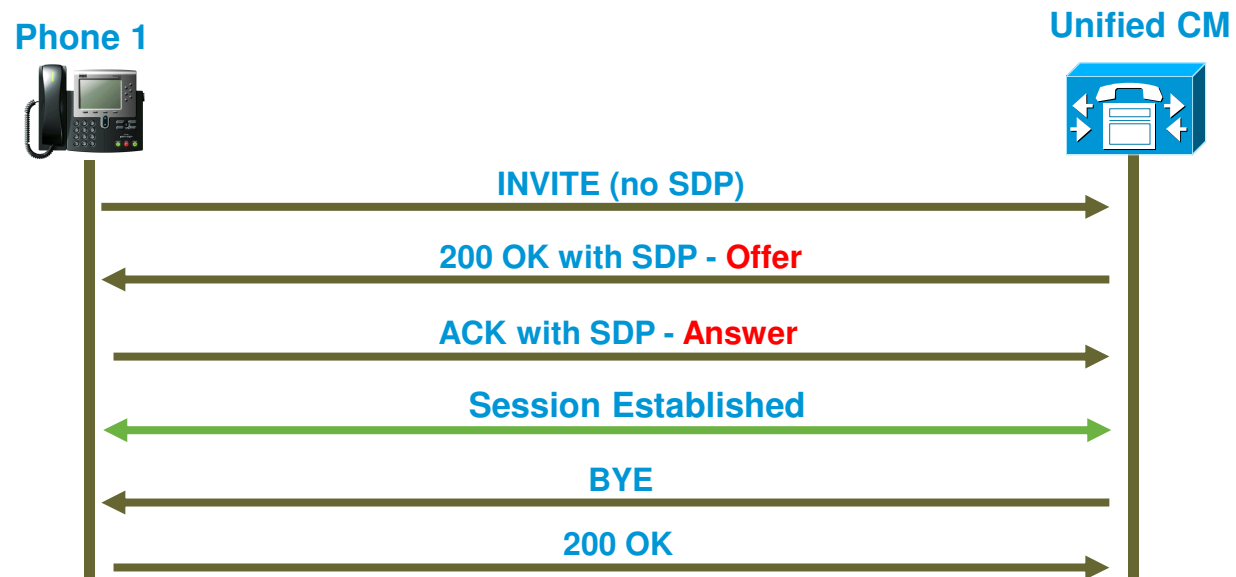
Media Negotiation – Early Offer and Delayed Offer

- Initiator of the call can send SDP offer in the INVITE – this is called an Early Offer (EO)
- Receiving endpoint can send the SDP offer in a response if the INVITE did not contain an offer – this is called a Delayed Offer (DO)
- For Early Offer, the answer is sent in a response (usually 200 OK).
- For Delayed Offer, the answer is typically sent in the ACK.

Early Offer



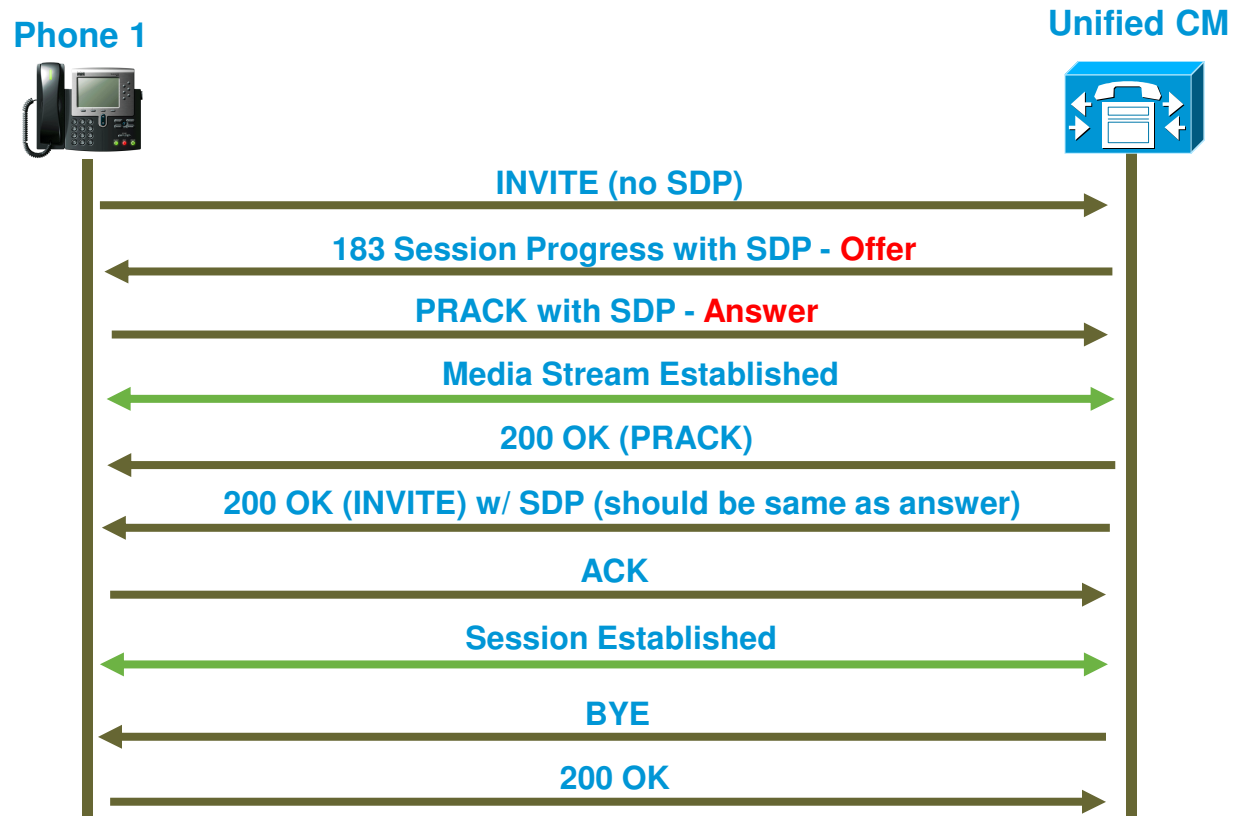
Delayed Offer



Early Media

- Delayed Offer calls do not set up media until the 200 OK (call is answered)
- If media is required prior to the call being connected, SIP has provisions for Early Media
- With Early Media on a Delayed Offer call, the offer comes from the terminating side in a provisional response (e.g. 183 Session Progress)
- Originating side sends SDP Answer in a PRACK message (defined in RFC 3262)

Early Media



Media Re-negotiation

Re-INVITE

- Either UA involved in a call can re-INVITE an existing dialog to re-negotiate parameters for the call.
- Cannot re-INVITE until any previous INVITE messages have received a final response.
- UPDATE method can also be used to re-negotiate prior to a final response.

Media Re-negotiation

Re-INVITE

```
INVITE sip:dbe40e44-0dfe-45f1-bd7f-e652098ca344@10.116.101.41:49833;transport=tls SIP/2.0
Via: SIP/2.0/TLS 172.18.106.59:5061;branch=z9hG4bK901f9c72c19221
From: "Paul" <sip:89915644@172.18.106.59>;tag=15462272~0d0d25d7-4931-4a07-83c6-b82e2c213ca7-45545776
To: <sip:89915644@172.18.106.59>;tag=0022bdd6843100702aae8e5b-4be253be
Date: Wed, 11 Jan 2012 03:08:51 GMT
Call-ID: 8c045780-f0c1fd34-8d838f-3b6a12ac@172.18.106.59
Supported: timer,resource-priority,replaces
Min-SE: 1800
User-Agent: Cisco-CUCM8.6
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 104 INVITE
Max-Forwards: 70
Expires: 180
Allow-Events: presence
Call-Info: <urn:x-cisco-remotecallinfo>; security= Authenticated; orientation= from; gci= 2-231448; call-instance= 2
Remote-Party-ID: "Paul" <sip:89915644@172.18.106.59>;party=calling;screen=yes;privacy=off
Contact: <sip:89915644@172.18.106.59:5061;transport=tls>
Content-Type: application/sdp
Content-Length: 489
```

Media Re-negotiation

Re-INVITE – Stopping a Media Session

```
v=0
o=CiscoSystemsCCM-SIP 15462272 2 IN IP4 172.18.106.59
s=SIP Call
c=IN IP4 0.0.0.0
t=0 0
m=audio 19594 RTP/SAVP 9 101
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=rtpmap:9 G722/8000
a=ptime:20
a=inactive
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
m=video 19444 RTP/AVP 126
b=TIAS:1000000
a=rtpmap:126 H264/90000
a=fmtp:126 profile-level-id=42801E;packetization-mode=1;level-asymmetry-allowed=1
a=inactive
a=mid:227796888
```

DTMF Relay

- 3 Methods for passing DTMF digits over a SIP network:
 - RFC 2833
 - SIP NOTIFY
 - SIP Keypad Markup Language (KPML)

DTMF Relay

RFC 2833

- Digits are passed in the RTP stream with a unique payload type
- Capability is negotiated in SDP like any other codec

Offer

```
m=audio 30414 RTP/AVP 0 8 116 18 100 101
c=IN IP4 172.18.106.231
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:100 X-NSE/800
a=fmtp:100 192-194
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
```

Answer

```
m=audio 17236 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```


DTMF Relay

SIP NOTIFY

- Passes DTMF information in a SIP NOTIFY message telephone-event Event
- Negotiated in Call-Info header

Offer

```
INVITE sip:+19195553333@172.18.106.231:5060 SIP/2.0
Via: SIP/2.0/UDP 172.18.106.59:5060;branch=z9hG4bK9843c455840434
From: "Paul Giral" <sip:9195551234@172.18.106.59>;tag=14902469~0d0d25d7-4931-4a07-83c6
To: <sip:+19195553333@172.18.106.231>
Date: Mon, 13 May 2013 14:48:00 GMT
Call-ID: 1a189580-1901fd20-962c99-3b6a12ac@172.18.106.59
... snip ...
Call-Info: <sip:172.18.106.59:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-class=DESKTOP
... snip ...
Max-Forwards: 69
Content-Length: 0
```

DTMF Relay

- SIP NOTIFY

Answer

SIP/2.0 200 OK

Via: SIP/2.0/UDP 172.18.106.59:5060;branch=z9hG4bK9843c455840434

From: "Paul Giral" <sip:9195551234@172.18.106.59>;tag=14902469~0d0d25d7-4931-4a07-83c6

To: <sip:+19195553333@172.18.106.231>;tag=4363A830-17FC

Call-ID: 1a189580-1901fd20-962c99-3b6a12ac@172.18.106.59

... snip ...

Allow-Events: telephone-event

Call-Info: <sip:172.18.106.231:5060>;method="NOTIFY;Event=telephone-event;Duration=500"

... snip ...

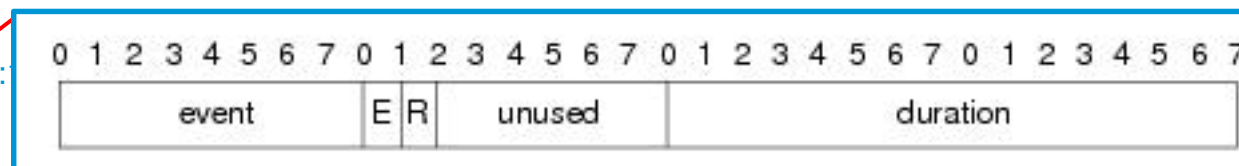
Content-Length: 601

DTMF Relay

SIP NOTIFY

- Digits passed in payload of a NOTIFY message

NOTIFY sip:172.18.106.231:5060 SIP/2.0
Via: SIP/2.0/UDP 172.18.106.59:5060;branch=z9hG4bK98443140152a0a
From: "Paul" <sip:9195551234@172.18.106.59>;tag=14902469~0d0d25d7-4931-4a07-83c6
To: <sip:+19195553333@172.18.106.231>;tag=4363A830-17FC
Call-ID: 1a189580-1901fd20-962c99-3b6a12ac@172.18.106.59
CSeq: 104 NOTIFY
Max-Forwards: 70
Date: Mon, 13 May 2013 14:48:30
User-Agent: Cisco-CUCM10.0
Event: telephone-event
Subscription-State: active
Contact: <sip:172.18.106.59:5060>
P-Asserted-Identity: "Paul" <sip:9195551234@172.18.106.59>
Content-Type: audio/telephone-event
Content-Length: 4



.d

DTMF Relay

- SIP KPML
 - Passes DTMF information in a SIP NOTIFY message kpml Event
 - Capability advertised in Allow-Events – uses SUBSCRIBE message to subscribe

Offer

```
INVITE sip:+19195554444@172.18.106.231:5060 SIP/2.0
Via: SIP/2.0/UDP 172.18.106.59:5060;branch=z9hG4bK986efd6c4e51e4
From: "Paul" <sip:9195551234@172.18.106.59>;tag=14918970~0d0d25d7-4931-4a07-83c6
To: <sip:+19195554444@172.18.106.231>
Date: Mon, 13 May 2013 15:05:24 GMT
Call-ID: 885e5780-19110134-96567f-3b6a12ac@172.18.106.59
User-Agent: Cisco-CUCM10.0
... snip ...
Allow-Events: presence, kpml
... snip ...
Session-Expires: 18000
Max-Forwards: 69
Content-Length: 0
```

DTMF Relay

- SIP KPML

Answer

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.106.59:5060;branch=z9hG4bK986efd6c4e51e4
From: "Paul" <sip:9195551234@172.18.106.59>;tag=14918970~0d0d25d7-4931-4a07-83c6
To: <sip:+19195554444@172.18.106.231>;tag=437394E8-2E1
Date: Mon, 13 May 2013 15:05:26 GMT
Call-ID: 885e5780-19110134-96567f-3b6a12ac@172.18.106.59
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO
Allow-Events: kpml, telephone-event
Remote-Party-ID: <sip:9196247285@172.18.106.231>;party=called;screen=no;privacy=off
Contact: <sip:+19196247285@172.18.106.231:5060>
Supported: replaces
Server: Cisco-SIPGateway/IOS-15.2.4.M3
Require: timer
Session-Expires: 18000;refresher=uac
Content-Type: multipart/mixed;boundary=uniqueBoundary
Mime-Version: 1.0
Content-Length: 600

DTMF Relay

- SIP KPML

Subscribe to KPML

SUBSCRIBE sip:9195554444@172.18.106.59:5060 SIP/2.0
Via: SIP/2.0/UDP 172.18.106.231:5060;branch=z9hG4bKBAE27139E
From: <sip:+19195551234@172.18.106.231>;tag=437394E8-2E1
To: "Paul" <sip:9195554444@172.18.106.59>;tag=14918970~0d0d25d7-4931-4a07-83c6
Call-ID: 885e5780-19110134-96567f-3b6a12ac@172.18.106.59
CSeq: 101 SUBSCRIBE
Max-Forwards: 70
User-Agent: Cisco-SIPGateway/IOS-15.2.4.M3
Event: kpml
Expires: 7200
Contact: <sip:172.18.106.231:5060>
Content-Type: application/kpml-request+xml
Content-Length: 327

```
<?xml version="1.0" encoding="UTF-8"?><kpml-request xmlns="urn:ietf:params:xml:ns:kpml-request"
xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-
request kpml-request.xsd" version="1.0"><pattern persist="persist"><regex
tag="dtmf">[x*#ABCD]</regex></pattern></kpml-request>
```

DTMF Relay

- SIP KPML

Send a Digit

```
NOTIFY sip:172.18.106.231:5060 SIP/2.0
Via: SIP/2.0/UDP 172.18.106.59:5060;branch=z9hG4bK986f73662cca3b
From: "Paul" <sip:9195554444@172.18.106.59>;tag=14918970~0d0d25d7-4931-4a07-83c6
To: <sip:+19195551234@172.18.106.231>;tag=437394E8-2E1
Call-ID: 885e5780-19110134-96567f-3b6a12ac@172.18.106.59
CSeq: 104 NOTIFY
Max-Forwards: 70
User-Agent: Cisco-CUCM10.0
Event: kpml
Subscription-State: active;expires=7197
Contact: <sip:9195554444@172.18.106.59:5060>
Content-Type: application/kpml-response+xml
Content-Length: 336
```

```
<?xml version="1.0" encoding="UTF-8" ?>
<kpml-response xmlns="urn:ietf:params:xml:ns:kpml-response" xmlns:xsi="http://www.w3.org/2001/XMLSchema-
instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-response kpml-response.xsd" code="200" digits="1"
forced_flush="false" suppressed="false" tag="dtmf" text="Success" version="1.0"/>
```

Agenda

- SIP Concepts
- Media Negotiation
- Trace Collection
 - RTMT
 - Logging Buffer
 - VoS CLI
- Tools
- Initial Analysis to TAC
- Troubleshooting

Unified CM Trace Configuration

Cisco Unified Serviceability
For Cisco Unified Communications Solutions

Navigation: Cisco Unified Serviceability Go

pgiralt About Logout

Alarm Trace Tools Snmp Help

Trace Co Configuration Troubleshooting Trace Settings

Related Links: SDL Configuration Go

Status
Status : Ready

Select Server, Service Group and Service

Server* 172.18.106.59 Go

Service Group* CM Services Go

Service* Cisco CallManager (Active) Go

Apply to All Nodes

Trace On

Trace Filter Settings

Select the Server

Select Service Group

Select the Service on Which Trace Needs to Be Enabled

Unified CM Trace Configuration

Trace Filter Settings

Debug Trace Level

- Cisco CallManager Trace Fields
 - Enable H245 Message Trace
 - Enable DT-24+/DE-30+ Trace
 - Enable PRI Trace
 - Enable ISDN Translation Trace
 - Enable H225 & Gatekeeper Trace
 - Enable Miscellaneous Trace
 - Enable Conference Bridge Trace
 - Enable Music On Hold Trace
 - Enable CM Real-Time Information Server Trace
 - Enable SIP Stack Trace
 - Enable Annunciator Trace
 - Enable SoftKey Trace
- Enable CDR Trace
- Enable Analog Trunk Trace
- Enable All Phone Device Trace
- Enable MTP Trace
- Enable All GateWay Trace
- Enable Forward & Miscellaneous Trace
- Enable MGCP Trace
- Enable Media Resource Manager Trace
- Enable SIP Call Processing Trace
- Enable SCCP Keep Alive Trace
- Enable SpeedDial Trace
- Enable SIP Keep Alive (REGISTER Refresh) Trace

Enable SIP Stack Trace is NOT needed to see SIP Messages.
Do not enable SIP Stack Trace prior to 9.0 unless directed by TAC

Gathering a Packet Capture from Unified CM

- Use the Platform CLI command 'utils network capture'

```
admin:utils network capture ?
Syntax:
utils network capture [options]
options optional page, numeric, file fname, count num, size bytes, src addr, dest
addr, port num, host protocol addr
```

```
admin:utils network capture file capturefile count 100000 size ALL host ip 10.1.1.1
Executing command with options:
  size=ALL          count=100000          interface=eth0
  src=              dest=              port=
  ip=10.1.1.1
```

```
admin:file list activelog platform/cli
capturefile.cap
dir count = 0, file count = 1
```

```
admin:file get activelog platform/cli/capturefile.cap
Please wait while the system is gathering files info ...done.
Sub-directories were not traversed.
Number of files affected: 1
Total size in Bytes: 24
Total size in Kbytes: 0.0234375
Would you like to proceed [y/n]? y
```

CUBE Debugging

- When debugging in IOS, configure logging buffered to a fairly large value (based on available memory)
- Disable logging to the console with command 'no logging console'
- Enable timestamps for debugs
- Make sure router has NTP enabled

```
service timestamps debug datetime msec localtime  
service timestamps log datetime msec localtime
```

```
logging buffered 10000000  
no logging console
```

```
clock timezone IST -5 0  
clock summer-time EDT recurring
```

```
ntp server 10.14.1.1
```

Agenda

- SIP Concepts
- Media Negotiation
- Trace Collection
- Tools
 - Real Time Monitoring Tool
 - TranslatorX
 - Wireshark
- Initial Analysis to TAC
- Troubleshooting

Troubleshooting Tools

- Real Time Monitoring Tool (RTMT)
- TranslatorX
- Wireshark

RTMT Session Trace Tool

Session Trace Features

- Allows you to search for a call based on calling or called number
- Does not depend on Call Detail Records
- Session trace only traces SIP sessions in detail
- Can display raw SIP messages
- Uses correlation tags to include all call legs related to the call selected
- On versions 8.5 and 8.6, can only be used on calls for which traces still exist on the server. Unified CM 9.0 allows viewing traces that have been archived off-server.

TranslatorX Tool

- Features
- Parses through Unified CM CCM/SDI Trace Files (SDL in 9.0+)
- Drag-and-Drop support for .txt as well as .gz files.
- Latest version supports IOS CUBE ccsip debugs, event-trace, and VCS diagnostic log files
- Decodes SIP, SCCP, H.323, MGCP, Q.Sig, and ISDN Q.931 messages
- Call List based on CDR information in the Traces
- Can generate multi-protocol ladder diagrams
- Sophisticated filtering capabilities
- Download for Windows, Mac OS X, and Linux from:
<http://translatorx.cisco.com/>
- NOTE: Do not call TAC for support on TranslatorX (although many TAC engineers use it so feel free to mention you're using it)

TranslatorX Tool

Cisco Unified Communications Trace Translator

Filters Enabled New Filter Filters... Clear Filters 0 Filters Configured Call List... Search Clear

Timestamp	Node/Interface	Device IP	Direction	Protocol	Message Name	TCP Handle/From Tag	Call Ref / ID
07/19/2012 10:19:36.652	172.18.106.60	10.150.45.172	In	SIP	200 OK	642328532	c3555180-811778-2c...
07/19/2012 10:19:36.832	172.18.106.60	172.18.106.41	In	SIP	INVITE	EE315E10-26C	9AA21AD0-DOE311E1...
07/19/2012 10:19:36.833	172.18.106.60	172.18.106.41	Out	SIP	100 Trying	EE315E10-26C	9AA21AD0-DOE311E1...
07/19/2012 10:19:36.836	172.18.106.60	172.18.106.225	Out	SIP	INVITE	9079691-0d0d25d7-...	c3555180-811778-2c...
07/19/2012 10:19:36.842	172.18.106.60	172.18.106.225	In	SIP	100 Trying	9079691-0d0d25d7-...	c3555180-811778-2c...
07/19/2012 10:19:36.860	172.18.106.60	172.18.106.225	In	SIP	404 Not Found	9079691-0d0d25d7-...	c3555180-811778-2c...
07/19/2012 10:19:36.860	172.18.106.60	172.18.106.225	Out	SIP	ACK	9079691-0d0d25d7-...	c3555180-811778-2c...
07/19/2012 10:19:36.863	172.18.106.60	172.18.106.41	Out	SIP	404 Not Found	EE315E10-26C	9AA21AD0-DOE311E1...
07/19/2012 10:19:36.867	172.18.106.60	172.18.106.41	In	SIP	ACK	EE315E10-26C	9AA21AD0-DOE311E1...
07/19/2012 10:19:44.207	172.18.106.60	10.80.75.34	In	SIP	REFER	6400f1151765182f6c...	6400f115-17650ad0-...
07/19/2012 10:19:44.209	172.18.106.60	10.80.75.34	Out	SIP	202 Accepted	6400f1151765182f6c...	6400f115-17650ad0-...
07/19/2012 10:19:44.254	172.18.106.60	10.80.75.34	Out	SIP	REFER	683927583	c81a0580-811780-2c...
07/19/2012 10:19:44.356	172.18.106.60	10.80.75.34	In	SIP	REFER	6400f115176518317...	6400f115-17650ad1-...
07/19/2012 10:19:44.358	172.18.106.60	10.80.75.34	Out	SIP	202 Accepted	6400f115176518317...	6400f115-17650ad1-...
07/19/2012 10:19:44.443	172.18.106.60	10.80.75.34	In	SIP	200 OK	683927583	c81a0580-811780-2c...
07/19/2012 10:19:47.735	172.18.106.59	10.82.3.177	In	SCCP	TimeDateReq.	(0204405)	
07/19/2012 10:19:47.735	172.18.106.59	10.82.3.177	Out	SCCP	DefineTimeDate	(0204405)	
07/19/2012 10:19:47.735	172.18.106.59	10.82.3.177	Out	SCCP	DisplayPromptStatus	(0204405)	0
07/19/2012 10:19:48.193	172.18.106.59	10.116.33.244	Out	SIP	REFER	293950898	c3555180-811778-40...

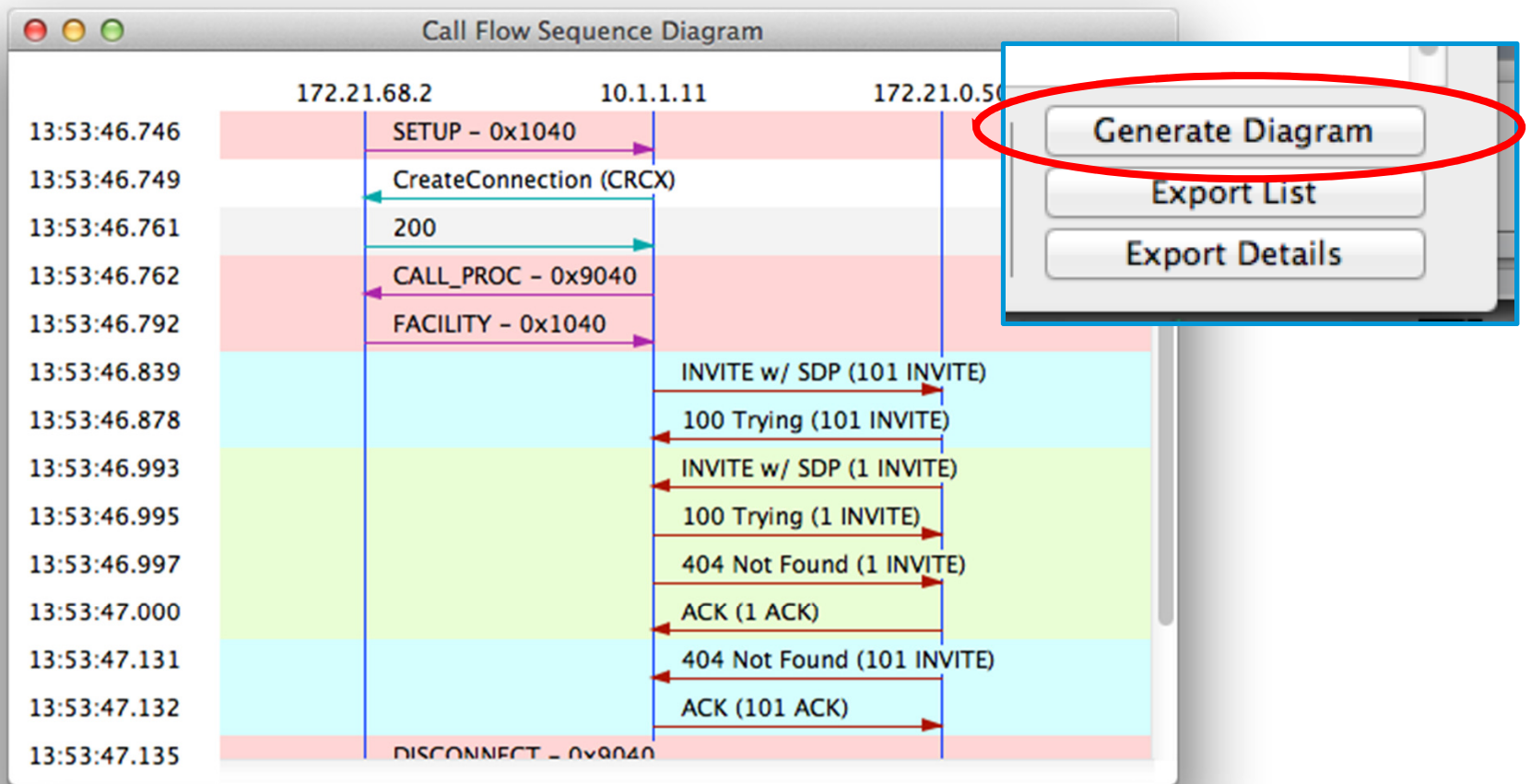
41677692.001 | 10:19:36.836 | AppInfo | SIPtcp - wait_sd1SPISignal: Outgoing SIP TCP message to 172.18.106.225 on port 5060 index 237195 [23872826,NET]

```

INVITE sip:899176938172.18.106.225:5060 SIP/2.0
Via: SIP/2.0/TCP 172.18.106.60:5060;branch=z9hG4bK2d6be2612E03c2
From: "pending" <sip:86695363748172.18.106.60>;tag=9079691-0d0d25d7-4931-4a07-83c6-b82e2c213ca7-60184383
To: <sip:899176938172.18.106.225>
Date: Thu, 19 Jul 2012 14:19:36 GMT
Call-ID: c3555180-811778-2c3d0b-3c6a12ac@172.18.106.60
Supported: 100rel,timer,resource-priority,replaces
Min-SE: 1800
User-Agent: Cisco-CUCM9.0
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, FRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback,X-cisco-original-called
Call-Info: <sip:172.18.106.60:5060>;method=NOTIFY;Event=telephone-event;Duration=500"
    
```

Lines Processed: 495869 SCCP H.245 Exclude SCCP and MGCP Keepalives Generate Diagram
 Msgs Processed: 16729 SIP MGCP Exclude SIP REGISTER Export List
 Msgs Displayed: 1907 Q.931 / H.225 MGCP Backhaul Exclude SIP OPTIONS Export Details

TranslatorX Tool



Wireshark

- Open Source network packet capture and analysis tool
- Available at <http://www.wireshark.org>
- Available for Windows, Mac OS X, and UNIX/Linux
- Provides VoIP Call and SIP analysis

Wireshark

The screenshot displays the Wireshark interface for a file named '3mxxp_info_problem.pcap'. The main pane shows a list of captured packets with columns for No., Time, Source, Destination, Protocol, and Info. Packet 14 is selected, showing a SIP INVITE request over TCP. The packet details pane is expanded to show the Session Initiation Protocol (SIP) structure, including the Request-Line, Method (INVITE), Request-URI, and Message Header (Via). The raw packet bytes pane at the bottom shows the hexadecimal and ASCII representation of the selected packet.

No.	Time	Source	Destination	Protocol	Info
10	9.699290	172.18.159.142	172.18.107.81	SIP	Continuation
11	9.739101	172.18.107.81	172.18.159.142	TCP	sip > 49152 [ACK] Seq=587 Ack=587 Win=431 Len=0 TSV=46822474
12	13.853885	172.18.107.81	172.18.159.198	TCP	[TCP segment of a reassembled PDU]
13	13.853888	172.18.107.81	172.18.159.198	TCP	[TCP segment of a reassembled PDU]
14	13.853890	172.18.107.81	172.18.159.198	SIP	Request: INVITE sip:89917403@172.18.159.198:5060;transport=tcp
15	13.854187	172.18.159.198	172.18.107.81	TCP	sip > 25255 [ACK] Seq=1 Ack=1049 Win=15720 Len=0 TSV=306368 TS
16	13.862181	172.18.159.198	172.18.107.81	TCP	sip > 25255 [ACK] Seq=1 Ack=1267 Win=16768 Len=0 TSV=306369 TS
17	13.892493	172.18.159.198	172.18.107.81	TCP	[TCP segment of a reassembled PDU]
18	13.892494	172.18.159.198	172.18.107.81	SIP	Status: 180 Ringing
19	13.893626	172.18.107.81	172.18.159.198	TCP	25255 > sip [ACK] Seq=1267 Ack=760 Win=431 Len=0 TSV=468228628

Packet 14 details:

- Frame 14: 284 bytes on wire (2272 bits), 284 bytes captured (2272 bits)
- Ethernet II, Src: Cisco_14:0f:c1 (00:26:98:14:0f:c1), Dst: Tandberg_81:8e:09 (00:50:60:81:8e:09)
- Internet Protocol, Src: 172.18.107.81 (172.18.107.81), Dst: 172.18.159.198 (172.18.159.198)
- Transmission Control Protocol, Src Port: 25255 (25255), Dst Port: sip (5060), Seq: 1049, Ack: 1, Len: 218
- [Reassembled TCP Segments (1266 bytes): #12(524), #13(524), #14(218)]
- Session Initiation Protocol
 - Request-Line: INVITE sip:89917403@172.18.159.198:5060;transport=tcp SIP/2.0
 - Method: INVITE
 - Request-URI: sip:89917403@172.18.159.198:5060;transport=tcp [Resent Packet: False]
 - Message Header
 - Via: SIP/2.0/TCP 172.18.107.81:5060;egress-zone=Intranet;branch=z9hG4bKcf88821899620c769f205f97df8ccb383327440.813d8dea66ea1a816737f69e10427ca

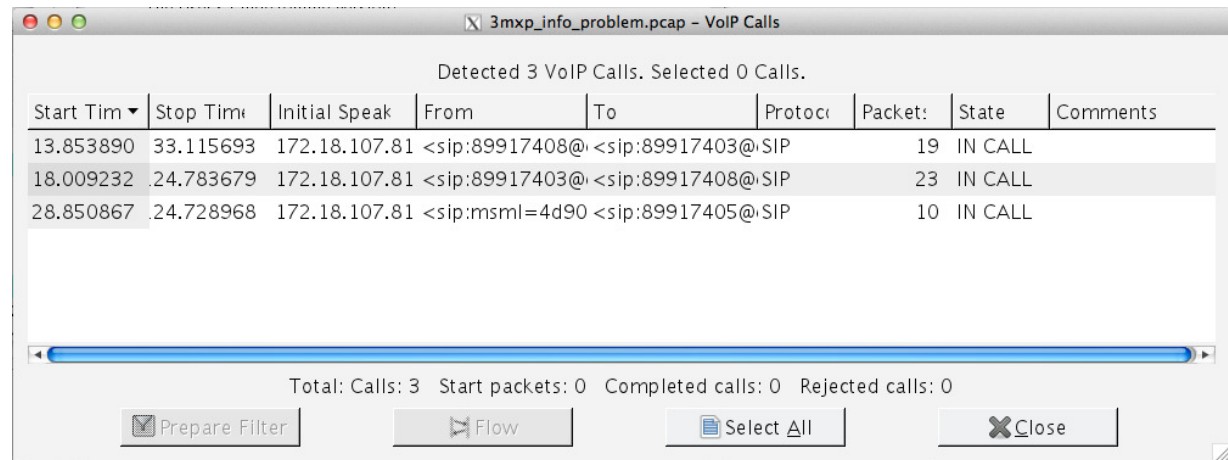
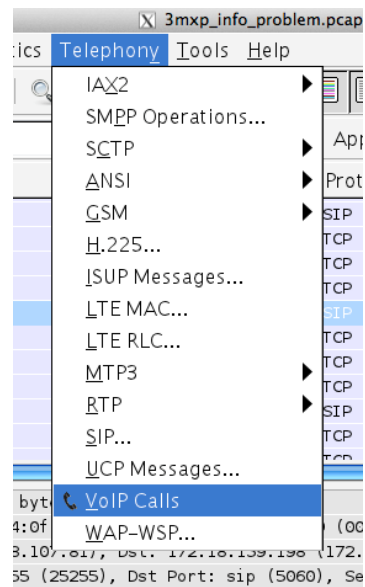
Raw packet bytes:

```

0000 00 50 60 81 8e 09 00 26 98 14 0f c1 08 00 45 00  .P....&.....E.
0010 01 0e 2e 91 40 00 3e 06 aa 1c ac 12 6b 51 ac 12  ...@.>...kq..
0020 9f c6 62 a7 13 c4 a5 4e 33 7a 60 10 e1 a9 80 18  ..b...N 3z'....
0030 01 9f ca 0c 00 00 01 01 08 0a 1b e8 99 ec 00 04  .....
0040 1c fe 4f 50 54 49 4f 4e 53 2c 50 52 41 43 4b 2c  ..OPTION S,PRACK,
    
```

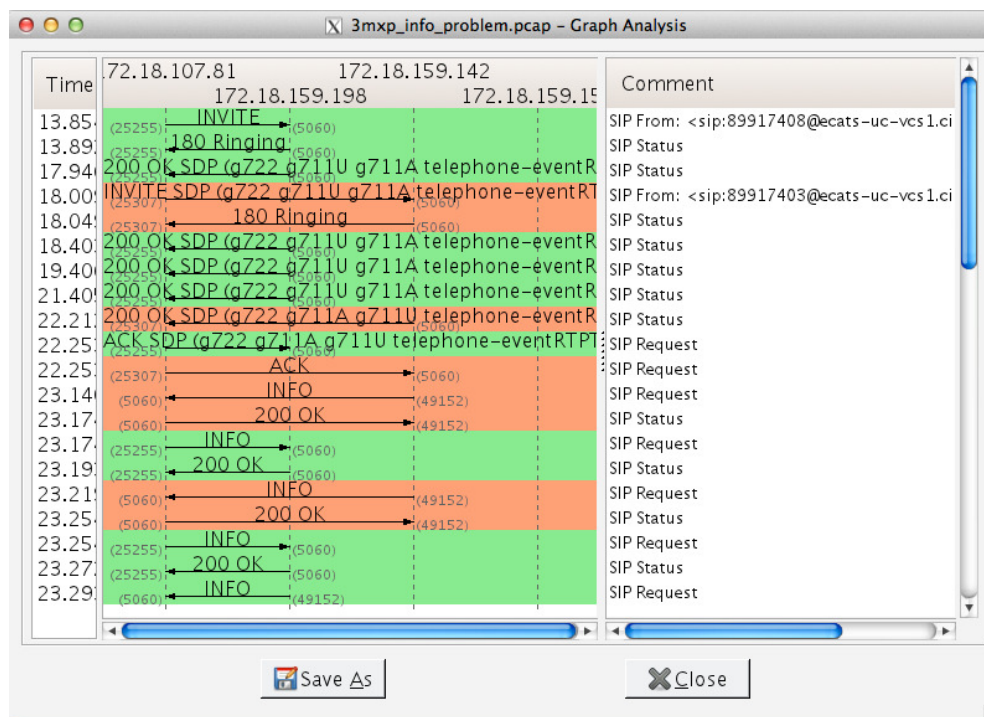
Wireshark

- VoIP Call Analysis



Wireshark

- VoIP Call Ladder Diagram



Agenda

- SIP Concepts
- Media Negotiation
- Trace Collection
- Tools
- **Initial Analysis to TAC**
- Troubleshooting

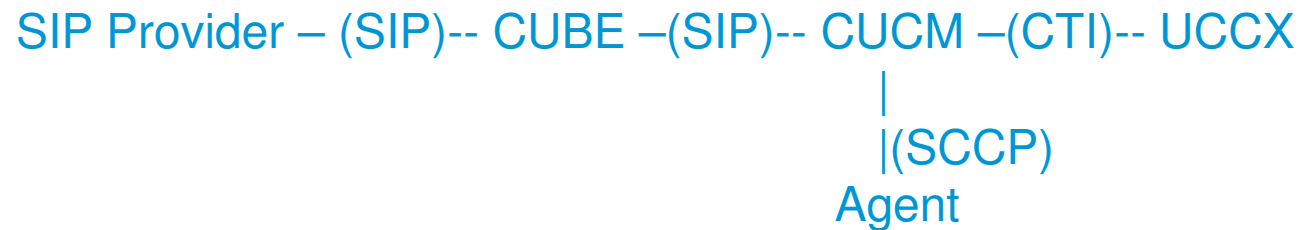
Information to Include

- Crisp Problem Description
- Business Impact
- Network Topology or LLD/HLD
- Complete Product Version (Ex. CUCM, UCCX, IOS, Phone F/w....)
- Call Related Data (Called/Calling Party, Time-Stamp, Components Involved)

Agenda

- SIP Concepts
- Media Negotiation
- Trace Collection
- Tools
- Initial Analysis to TAC
- **Troubleshooting**

Troubleshooting Call Flow



Approach

- Problem Identification
- Log Collection
- Identify Solution
- Apply/Test Fix

Questions

SIP Phones A, B and C are registered to Call Manager.
Phone A calls Phone B and Phone B starts ringing.

a.) How many Call ID's would be created for this Call?

1. One
2. Two
3. Three
4. None of the Above

b.)Phone A goes on-hook before Phone B answers the call.
Which of the below SIP messages would indicate this ?

1. INVITE
2. PRACK
3. BYE
4. CANCEL
5. 100 Trying

c.)Phone A goes on-hook after Phone B answers the call.
Which of the below SIP messages would indicate ?

1. 100 Trying
2. PRACK
3. BYE
4. CANCEL
5. INVITE

d.)Phone B answers and transfers the CALL to Phone C
Which of the below SIP messages would indicate ?

1. INVITE
2. 180 Ringing
3. REFER
4. 302 Moved Temporarily
5. 200 OK

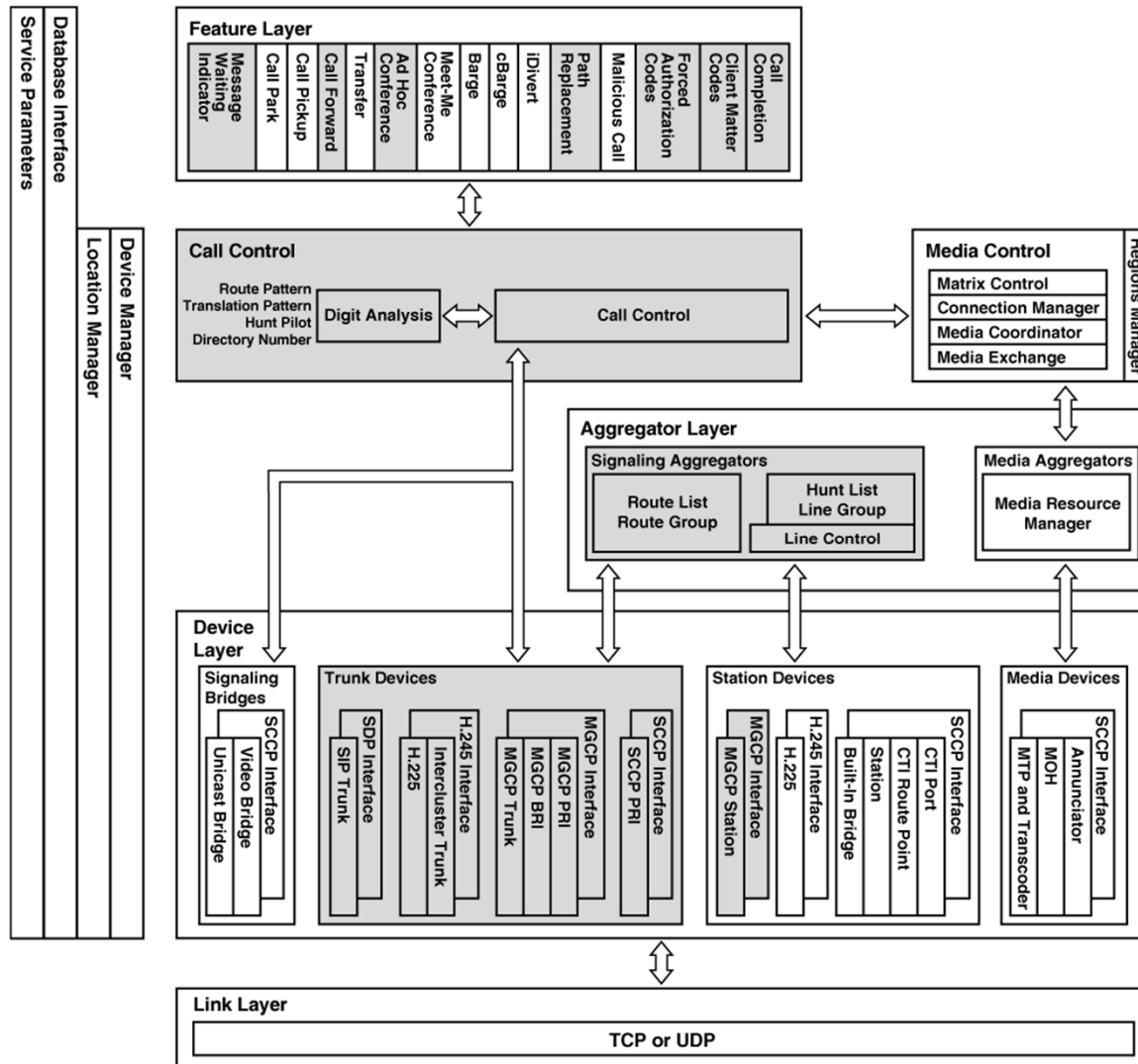
e.)Which of the following is always sent by a UAC to a UAS.
Which of the below SIP messages would indicate ?

1. INVITE
2. 100 Trying
3. 180 Ringing
4. All of the above
5. None of the above

Thank you.



CUCM Architecture





Cisco Support Community

Mohit Mmangal

Vinay Sharma

What is Cisco Support Community (CSC)?

- Its a **dynamic knowledge** base where you can collaborate.
- Create & **access the latest technical support content** (e.g. white papers, best practices, troubleshooting guides, etc.).
- You can **solve issues regarding your Cisco products** and software on a highly secure enterprise platform.
- The **open content model** enables everyone to view content and submit feedback.
- As a registered user, you can view, edit and create content.



Last Session Recap...

- How to create a Blog or Video – Editing, rating, subscription, sharing on social media.
- How to personalize your view in CSC



Today's Agenda...

- Learn about CSC top contributors programs
- CSC Events
- CSC on Social Media



CSC Live Demo...

CISCO Cisco Support Community

Home Top Contributors Expert Corner

Search the Support Community

Home

All Content Your View

ANNOUNCEMENT: [Upcoming Cisco Support Community Webcast Schedule](#) Show Details

Navigate to a Community Topic and Post

Network Infrastructure

- WAN, Routing and Switching
- LAN, Switching and Routing
- Network Management
- Remote Access
- Optical Networking
- Getting Started with LANs
- IPv6 Integration and Transition
- Design and Architecture
- EEM Scripting
- Other Subjects

Security

- VPN
- Security Management
- Firewalling
- Intrusion Prevention Systems/IDS
- AAA, Identity and NAC
- Physical Security
- MARS
- Email Security
- Web Security
- Other Subjects

Service Providers

- Metro
- MDI

Collaboration, Voice and Video

- IP Telephony
- Video Over IP
- Jabber Clients
- Unified Communications Applications
- TelePresence
- Digital Media System
- Contact Center
- Other Subjects

Wireless - Mobility

- Security and Network Management
- Wireless IP Voice and Video
- Getting Started with Wireless
- Other Subjects

Services, Solutions and Architectures

- Partner Support Service
- Smart Call Home
- Smart Care

Online Tools and Resources

- Cisco Bug Discussions
- Technical Documentation Ideas
- Cisco Technical Support Mobile Apps
- Support Community Help

Data Center

- Application Networking
- Server Networking
- Storage Networking
- Unified Computing
- Wide Area Application Services (WAAS)
- Other Subjects

Small Business

- Network Storage
- OnPlus Service
- Routers
- Security
- Surveillance
- Switches
- Voice and Conferencing
- Wireless

Cisco Social

- Behind the Scenes
- Cisco Cafe
- Community Ideas
- Southern California Cisco User Group (SCCUG)

Welcome to the Cisco Support Community

Engage, collaborate, co-create and share with your fellow experts on any Cisco technology or solutions in technical support forums in six different languages. Participate in live expert events and join the ongoing technical support forum in our communities.

Cisco Support Community Named 2013 Gold Stevie® Award Winner for Innovation in Customer Service



[Learn More](#)

Cisco Support Community Tech-Talk

Understanding Cisco Unified



Top Contributors

Overview (customize) All Content (edit) Documents (add) Blog (edit) Set as default tab

New Account

- Blog Post
- Discussion
- Document
- Poll
- Task
- Update
- Video
- Project
- Space

Navigation

Top Contributors

- Cisco Designated VIPs
- Community Spotlight Awards
- Events Top Contributor
- Leaderboards
- Hall of Fame
- Meet the Experts

Expert Corner

- Ask the Experts
- Documents
- Facebook Forums
- Videos
- Webcasts

Meet The Experts



Marvin Rhoads

A network engineer with Presidio, Marvin Rhoads is a top contributor in the Network Management, Optical, and Security Management forums. When asked why, he says, "My main motivation is continuous learning, which is one of the reasons I continue to do hands-on engineering... that the best way to learn something is to teach someone." [Read More >](#)

Meet The Experts



Recognition Programs

Learn about Cisco Support Community top contributors programs, and how you can participate and earn recognition for technical expertise in the forums.

Hall of Fame

As the premier technical support community for Cisco customers and IT professionals, we recognize the importance of user participation. The **CSC Hall of Fame** showcases an elite group of long-time contributors.

Visit the Hall of Fame

How do I become a Hall of Fame member?

Cisco Designated VIPs

The Cisco Designated VIP program recognizes the top external individual contributors in Cisco's online communities, including the Cisco Support Community (CSC), Cisco Learning Network (CLN) and the Cisco Developers Network (CDN).

Class of 2013! See the list of current Designated VIPs

How do I become a Designated VIP?

Community Spotlight Awards are designed to recognize and thank individuals who help make our communities the premier online destination for Cisco enthusiasts. The Cisco Support Community (CSC), Cisco Learning Network (CLN) and the Cisco Developers Network (CDN) will each have a variety of awards.

View the Awardees >

How do I learn a Community Spotlight Award?

Cisco Subject Matter Experts are dedicated professionals in their areas who devote time to the Cisco Support Community by answering technical questions and collaborating with fellow community members to create documents that can be used by all Cisco enthusiasts.

[View the Cisco Employee Leaderboards >](#)

Events Top Contributor

This program recognizes Cisco experts in the Cisco Support Community (CSC) that host technical events (Webcasts, Ask the Experts, Tech Talks, and Facebook Forums.) With this program, Cisco recognizes the positive, valuable influence that our top Cisco experts exert on the communities.

Expert Corner

Overview (customize) All Content (edit) Blog (add) Set as default tab

Navigation

Top Contributors

- Cisco Designated VIPs
- Community Spotlight Awards
- Events Top Contributor
- Leaderboards
- Hall of Fame
- Meet the Experts

Expert Corner

- Ask the Experts
- Documents
- Facebook Forums
- Videos
- Webcasts

Meet The Experts



Julio Carvajal

Meet Julio Carvajal, Star of Cisco Support Community's Most Popular Spanish Language Webcast.

[Read More >](#)

Facebook Forums

Facebook Forum is an hour long live chat with a Cisco expert on a certain topic that takes place every month.

[Configuring and Troubleshooting MPLS Traffic Engineering](#)

Dec 18, 2012 In Service Providers

Engage, Collaborate, Co-Create and Share with Experts!

Expert Corner is a collection of special contributions to the community made by our experts. It includes programs such as Webcasts, Ask the Expert, Expert Videos and Documents.

Ask the Experts Documents Videos Webcasts Tech-Talks

Cisco subject matter experts engage in discussions with you on specific networking issues. Each Q&A event runs for a two-week period.

Mobile Wireless: How Your Cellular Phone Surfs the Internet

with **Deepak Michael**

Learn and ask questions about mobile wireless and get an overview of long term evolution (LTE) and a detailed explanation of the subscriber call flow.

Ends March 15, 2013

[Join the Discussion](#)

High Availability on Wireless Lan Controller (WLC)

with **Madhuri C.**

Learn and ask questions about configuring and troubleshooting High Availability (HA) on Wireless Lan Controller (WLC).

Ends March 22, 2013

[Join the Discussion](#)

Catalyst 3K Series Update

with **Dan Schour**

Get an update on the Cisco Catalyst 3K series.

Ends March 22, 2013

[Join the Discussion](#)

[Click here for previous "Ask the Expert" Q&As >](#)

Recent Blog Posts

[Experience Cisco Live London with Cisco](#)

Cisco Support Community Hall of Fame

Paolo Bevilacqua

Paolo is a former Cisco Technical Marketing Engineer for the 7200 router and voice. Before that I had jobs at Cisco as...

[Read More >](#)

Rick Burts

I've been in data communications for years, with experience in technologies from old SNA to current...

[Read More >](#)

Joe Marcus Clarke

Joe Marcus Clarke has been with Cisco since 1998, working on the network management TAC team in North...

[Read More >](#)

Rob Huffman

I've been working in the Telecom industry for many years most of which have been spent quite happily...

[Read More >](#)

Giuseppe Larosa

I was born in 1967. I had an Engineering degree in 1993 I University...

[Read More >](#)

Jon Marshall

Jon Marshall, a network connoisseur appreciates the NetPro culture that amazed that people...

[Read More >](#)

Edison Ortiz

Edison Ortiz is a Network Connoisseur with the Advanced Services concentrates...

[Read More >](#)

Jaime Valencia

I've been involved in voice as the AVVID team back in 2001 been with the PDU UC team in Network Consulting Engineer

[Read More >](#)

Leaderboards (customize)

Navigation

Top Contributors

- Cisco Designated VIPs
- Community Spotlight Awards
- Events Top Contributor
- Leaderboards
- Hall of Fame
- Meet the Experts

Expert Corner

- Ask the Experts
- Documents
- Facebook Forums
- Videos
- Webcasts

Mobile Leaderboard

User Name	Points
★ VIP Scott Fella	2,856
★ ROGER KALLBERG	397
★ Tarik Admani	257
★ George Stefanek	256
★ Karsten Jansen	220

Related Links

[How do I rate in the Community? Learn more about ratings](#)

Point Lists:

External: Current Month | All Time Internal: Current Month | All Time

External: Current Month

User Name	Points	Average Rating	Questions Answered
★ VIP Scott Fella	393	4.8	41
★ Chris Deren	240	4.6	19
★ Jonni Fors	186	4.9	28
★ Peter Paluch	115	5	12
★ ROB HUFFMAN	112	4.9	8
★ Ayodeji Olatunwo (Olatunwo)	104	5	5
★ Tom Watts	103	4.3	10
★ Giuseppe Larosa	80	4.7	3
★ Leo Laahoo	79	5.3	8
★ Terry Cheema	75	5	5
★ Blau grana	58	4.8	4
★ Gregory Snipes	50	5	8



On Cisco Support Community



Upcoming Sessions.....

January “Month of Voice Technology”

- Session 5 – 27th Jan 2014
- Session 6 – 30th Jan 2014

February “Month of Unified Computing Technology”

- Session 7 – 11th Feb 2014
- Session 8 – 18th Feb 2014

And many more.....Months and Technologies





Unified Computing Technology

Technology

UCS C Series Install/Upgrade.

UCS B & C Series Firmware Install/Upgrade.

UCS H/W and S/W Interop

Logs for Troubleshooting.

Part Replacement/ RMA



Q&A

Thank you.

