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

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Cisco Support Community

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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

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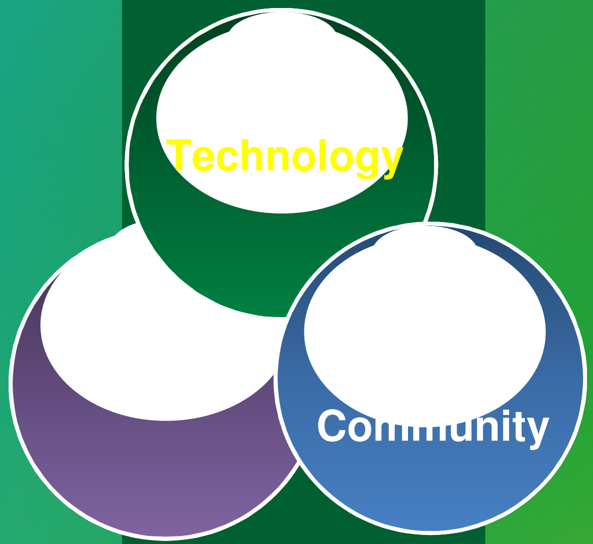


Technology – VOICE

Amit Singh

Raees 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

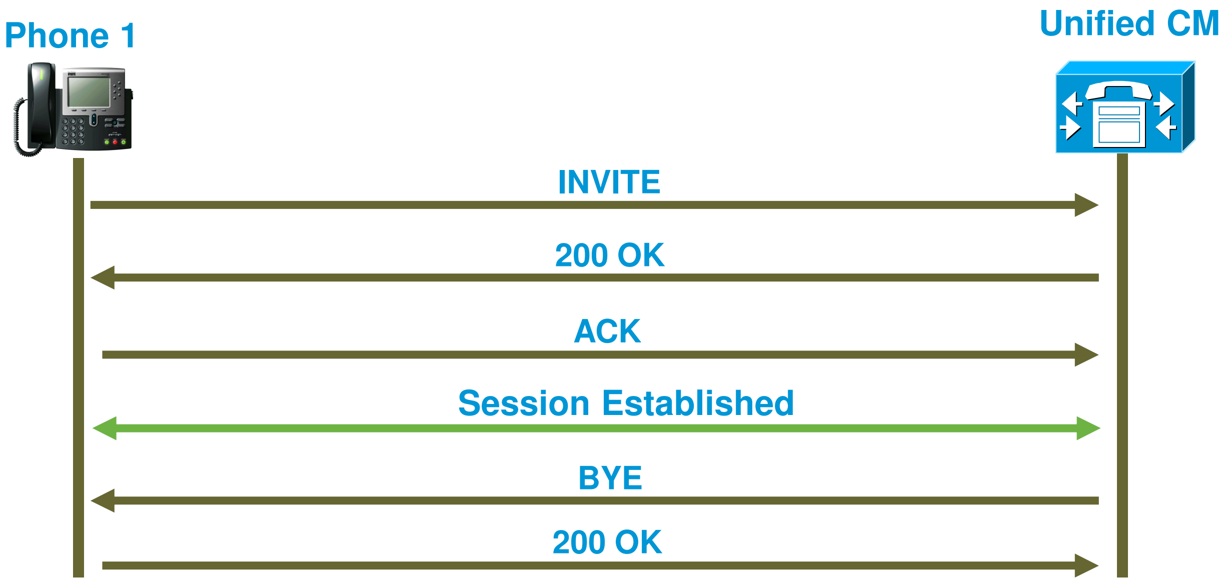
Free-text Reason

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

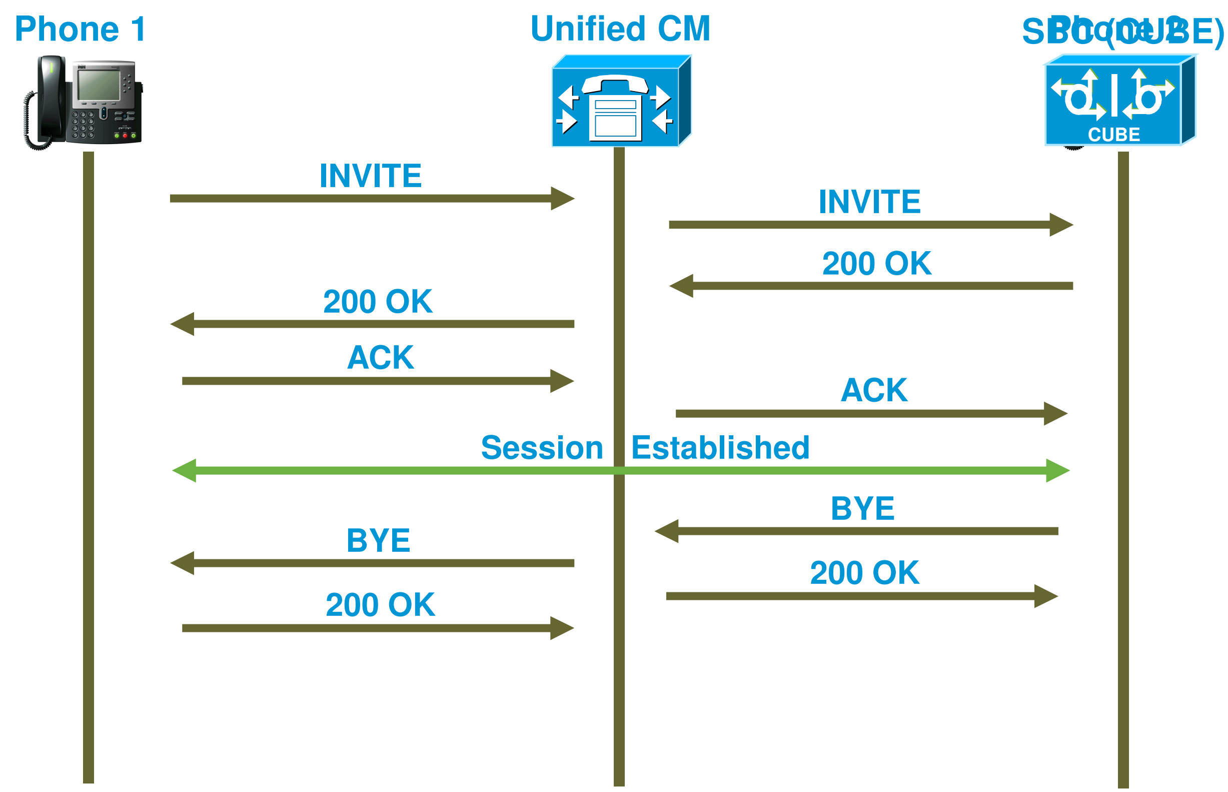
SIP Responses

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

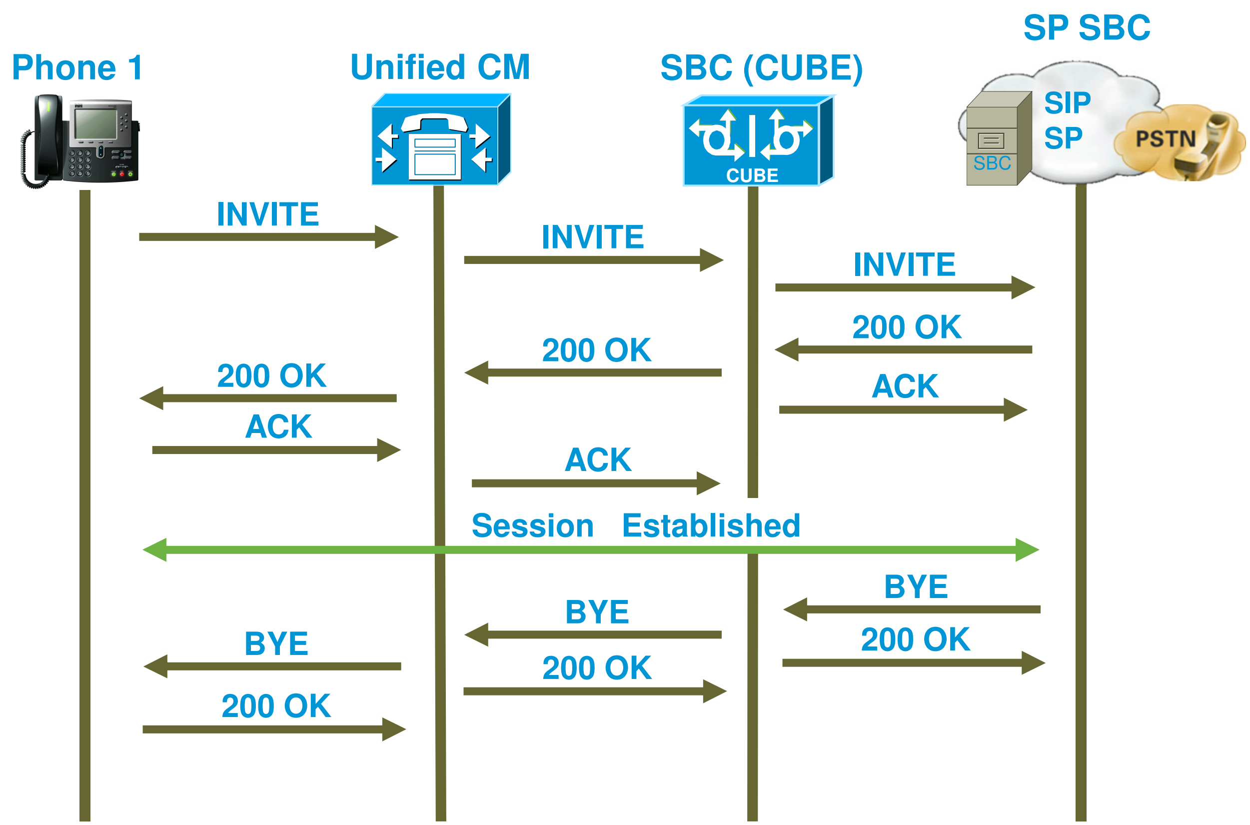
Basic SIP Call Setup



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:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
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
```

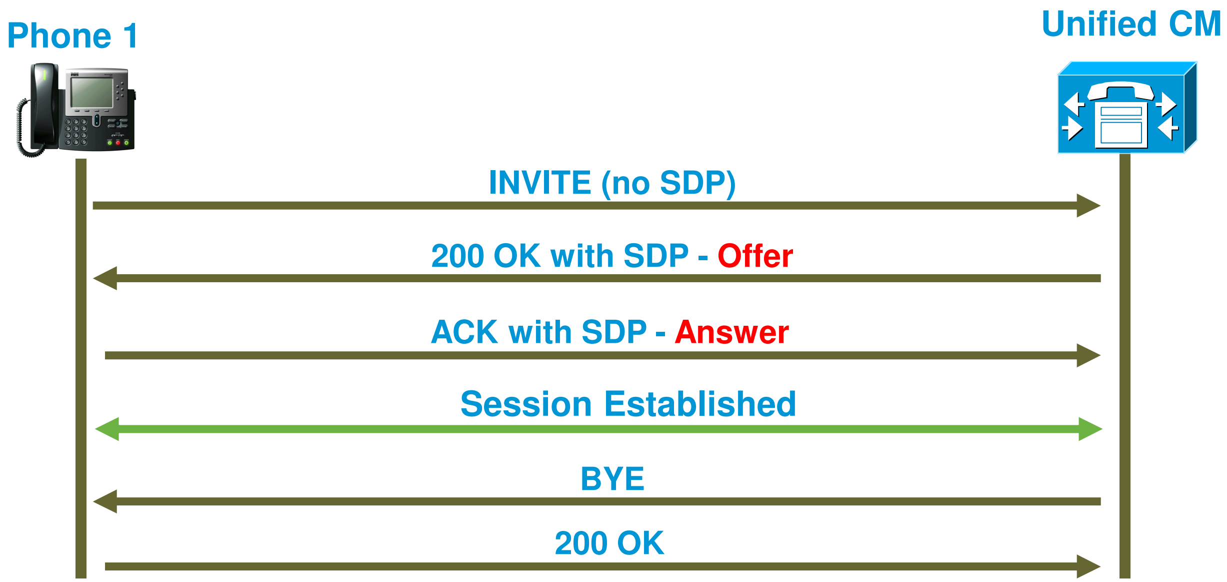
Media Negotiation – Early Offer and Delayed Offer

- 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


Phone 1
Unified CM
INVITE with SDP - Offer
200 OK with SDP - Answer
ACK (no SDP)
Session Established
BYE
200 OK

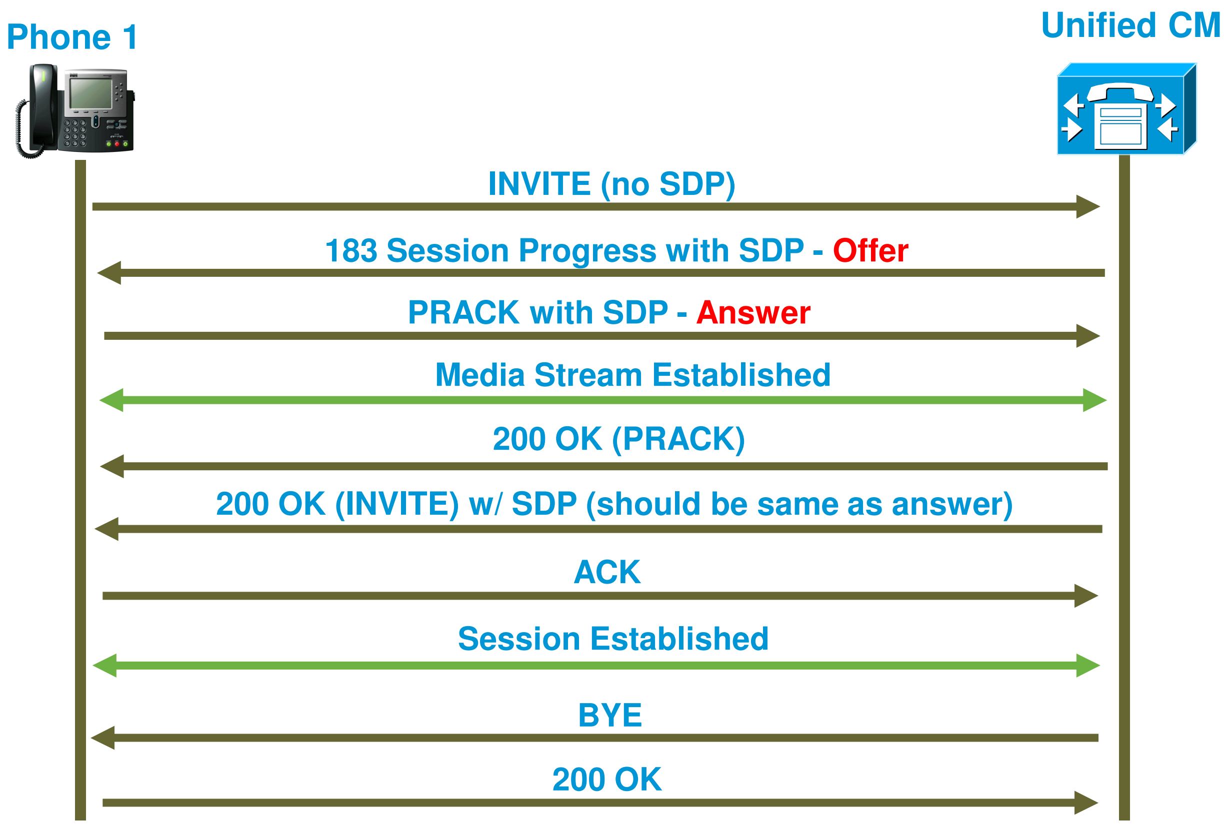
Delayed Offer



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



Phone 1
Unified CM
INVITE (no SDP)
183 Session Progress with SDP - Offer
PRACK with SDP - Answer
Media Stream Established
200 OK (PRACK)
200 OK (INVITE) w/ SDP (should be same as answer)
ACK
Session Established
BYE
200 OK

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-remotecc:callinfo>; 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:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
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 Giralt" <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

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DTMF Relay

- SIP NOTIFY

Answer

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.106.59:5060;branch=z9hG4bK9843c455840434
From: "Paul Giralt" <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

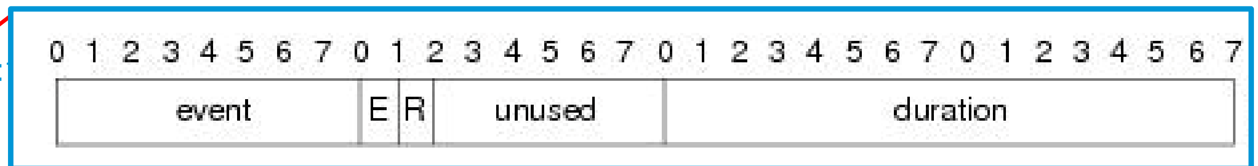
DTMF Relay

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 14:48:1
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
```

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DTMF Relay

- SIP KPML

<u>Answer</u>

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

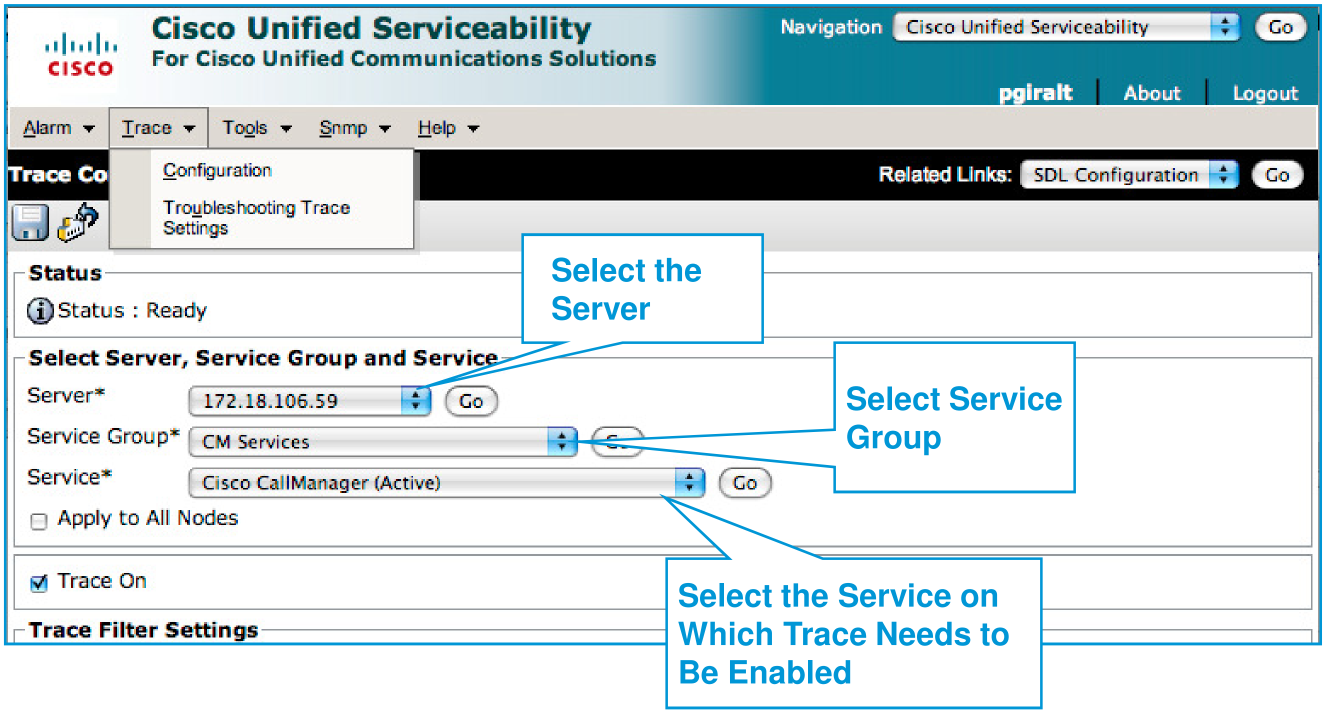
```
NOTIFY sip:172.18.106.231:5060 SIP/2.0
Via: SIP/2.0/UDP 172.18.106.59:5060;branch=z9hG4bK986f73662cca3b
From: "Paul" <sip:9195554444@172.18.106.59>;tag=14918970~0d0d25d7-4931-4a07-83c6
To: <sip:+19195551234@172.18.106.231>;tag=437394E8-2E1
Call-ID: 885e5780-19110134-96567f-3b6a12ac@172.18.106.59
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"/>
```

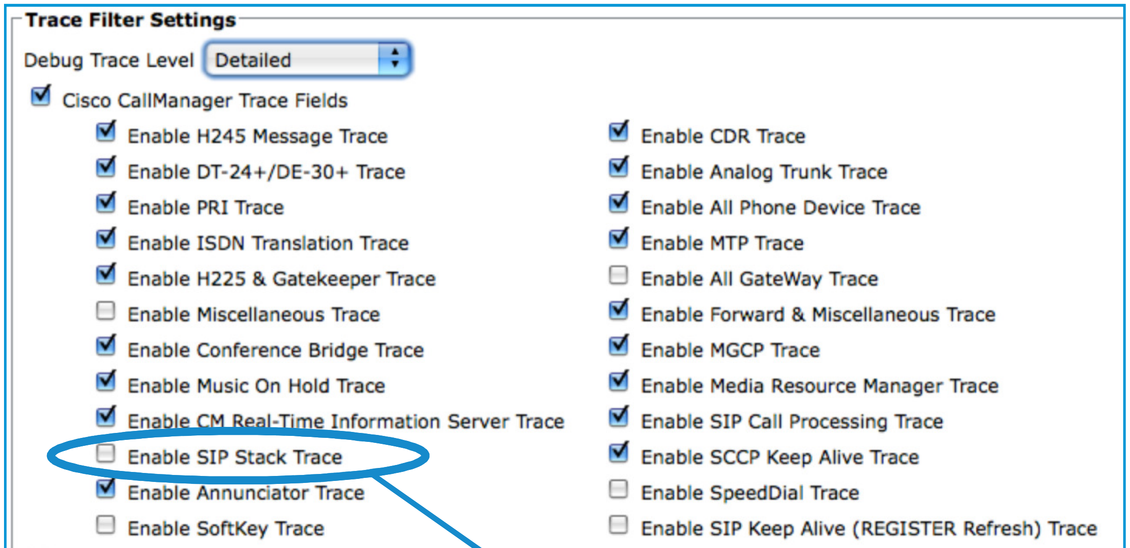
Agenda

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

Unified CM Trace Configuration



Unified CM Trace Configuration



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

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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
```

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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.

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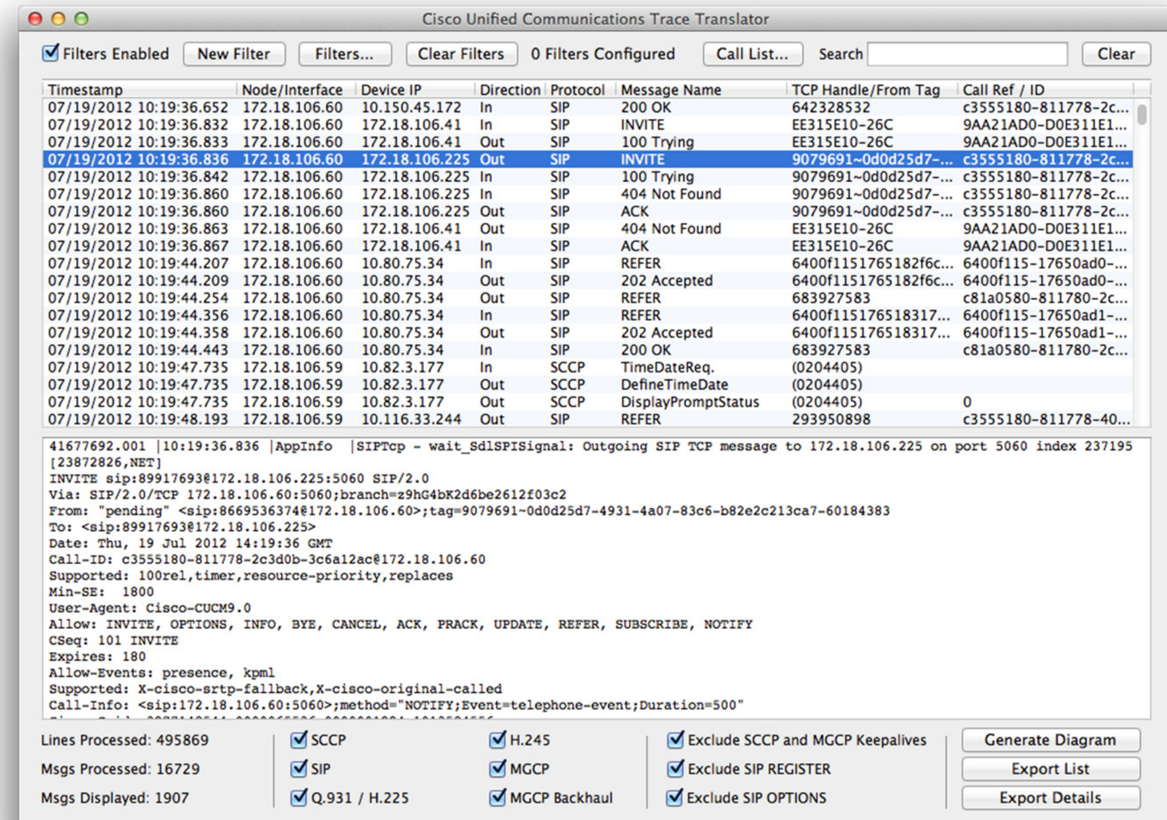
TranslatorX Tool

- 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



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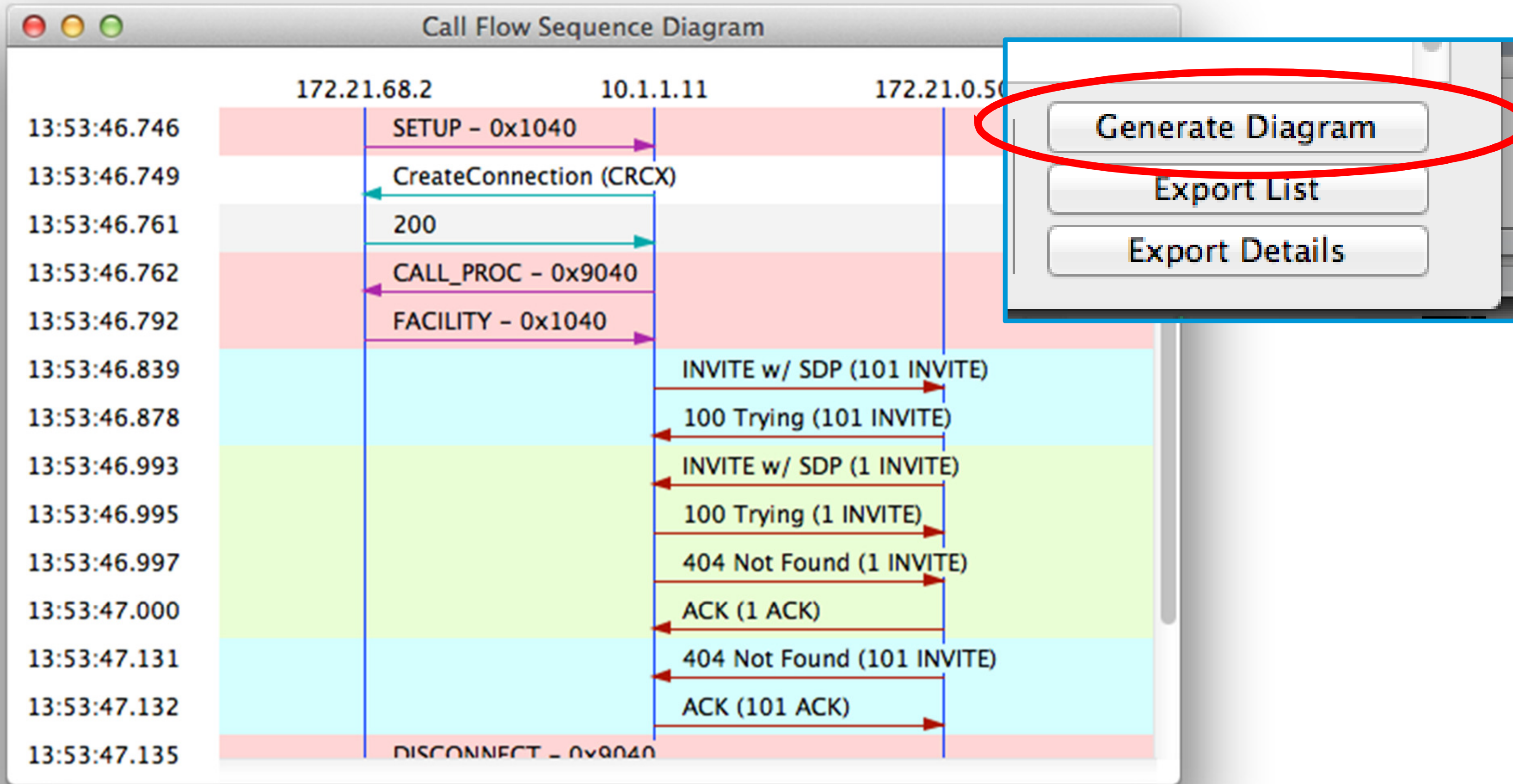
TranslatorX Tool



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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

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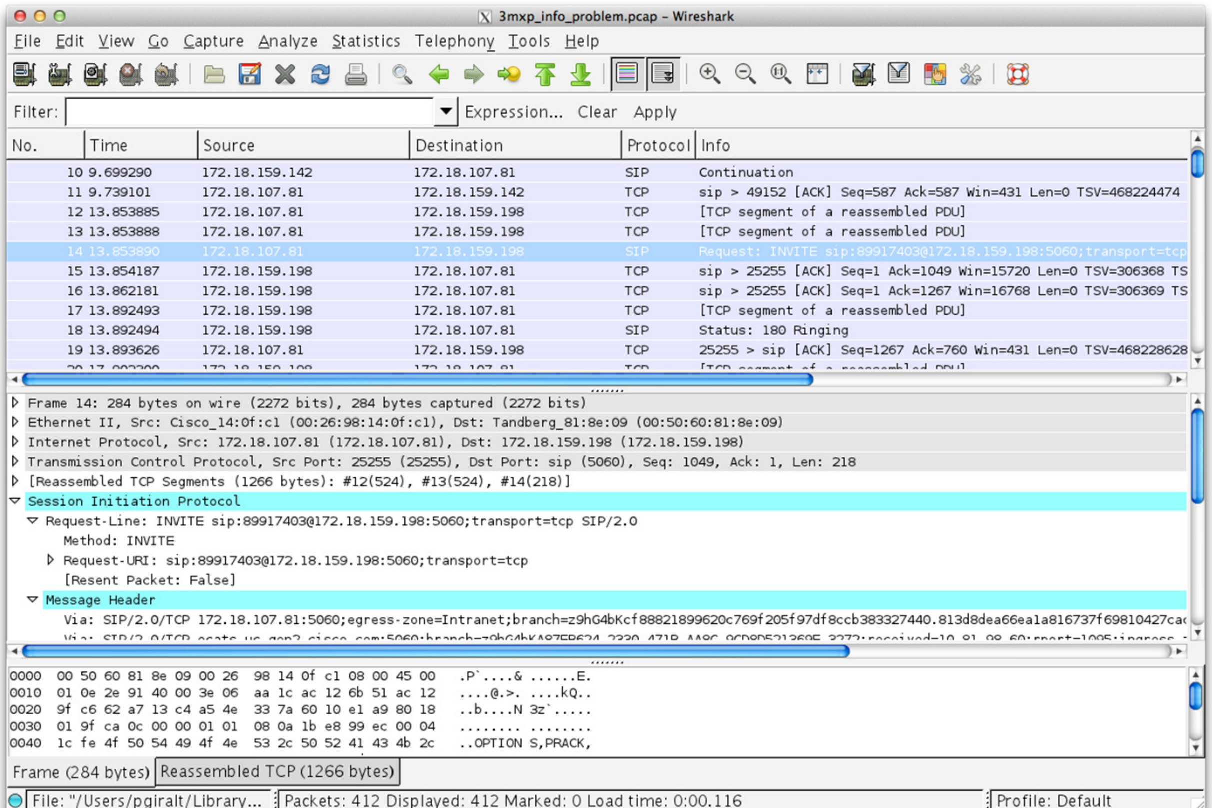
Wireshark



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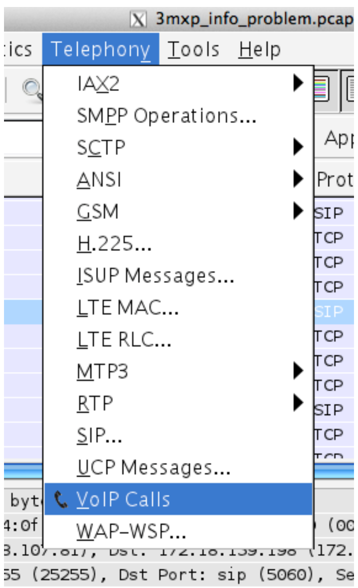
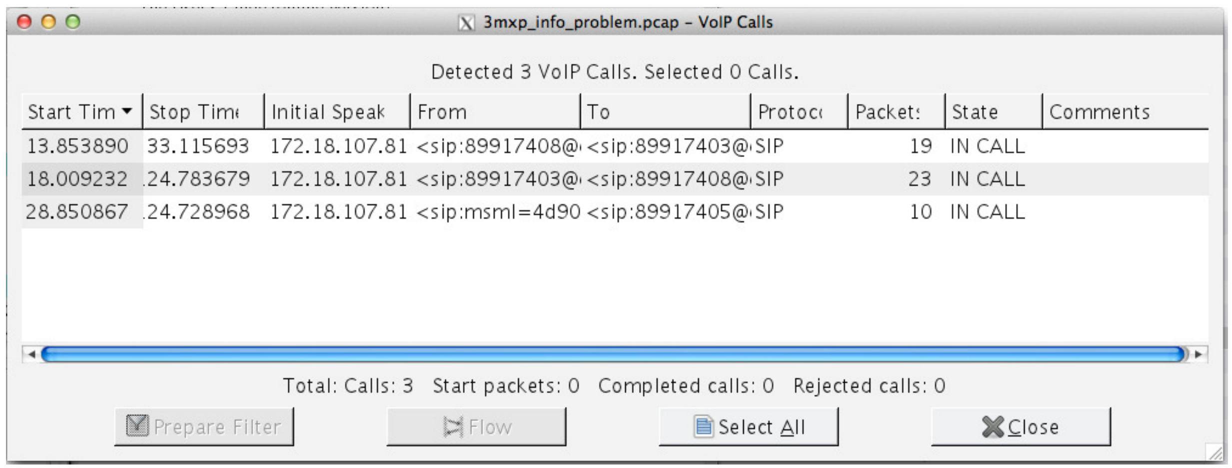
Wireshark

- VoIP Call Analysis





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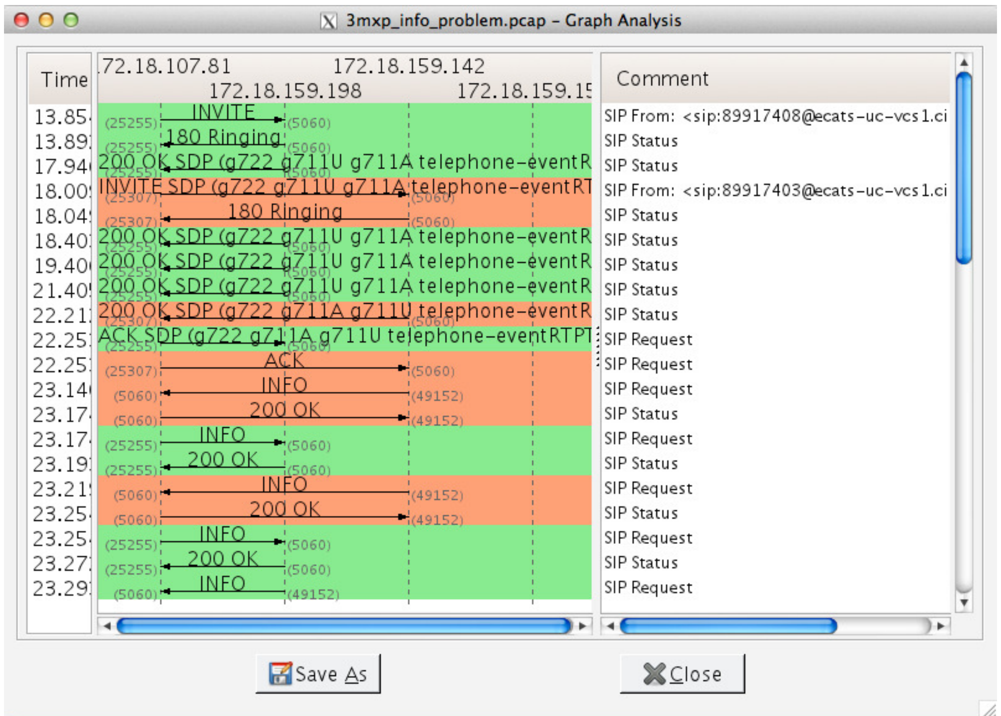
Wireshark

- VoIP Call Ladder Diagram



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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

SIP Provider – (SIP)-- CUBE –(SIP)-- CUCM –(CTI)-- UCCX
|
|(SCCP)
Agent

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

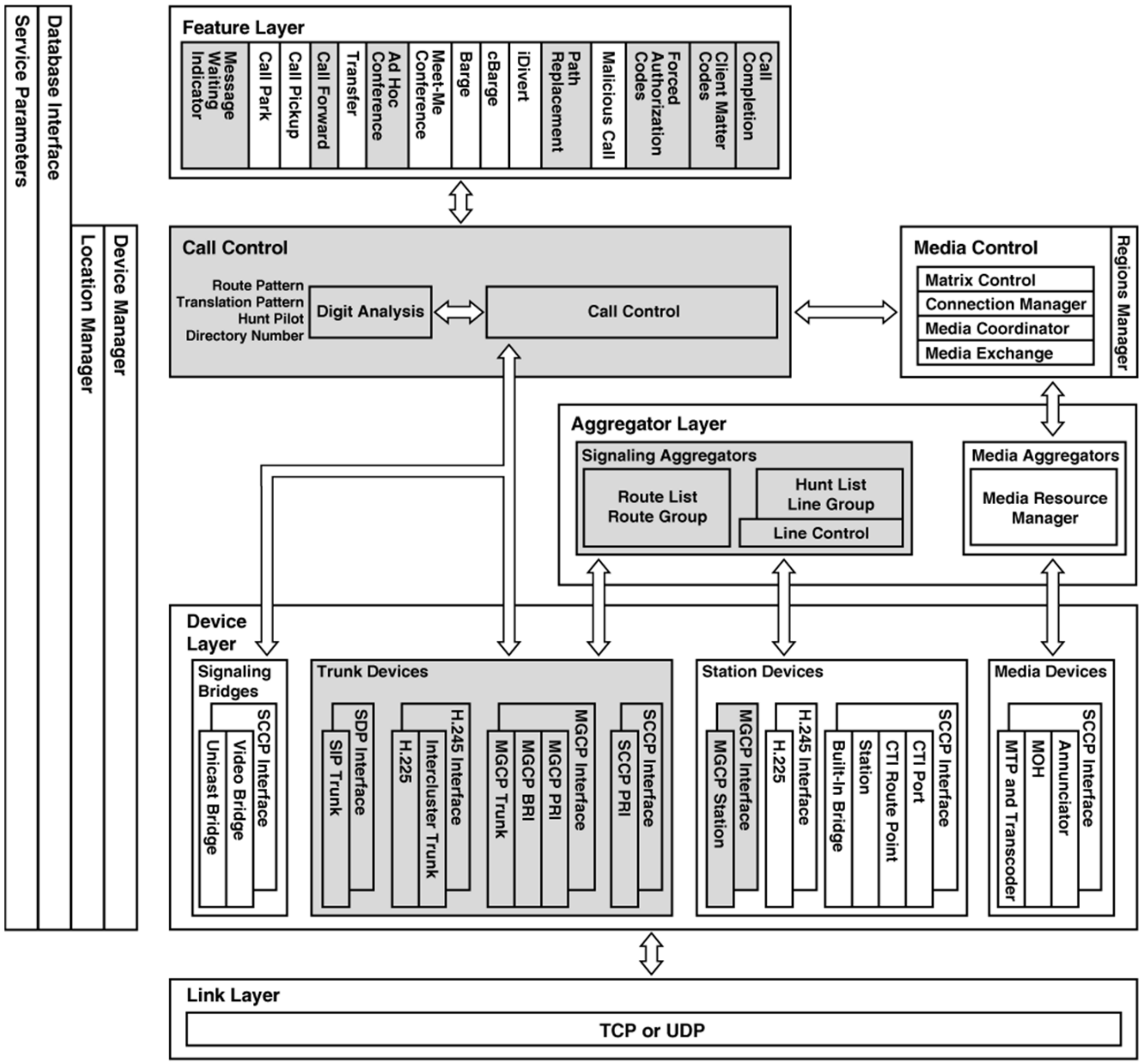
e.)Which of the following is always sent by a UAC to a UAS.
Which of the below SIP messages would indicate ?

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

Thank you.

CISCO

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…

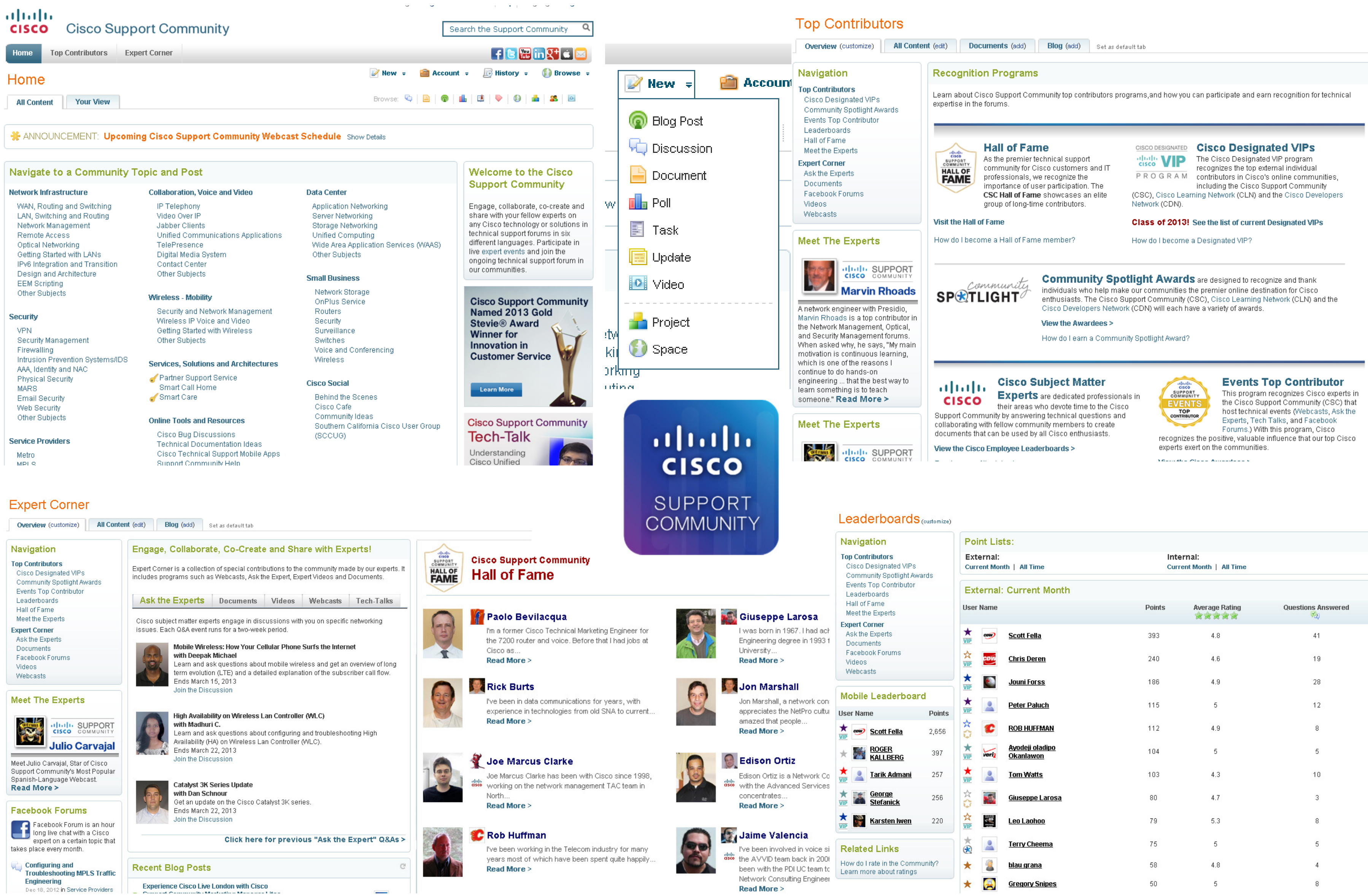
- How to create a Blog or Video – Editing, rating, subscription, sharing on social media.

- 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…



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.