SIGNALING AND DIALING: WHERE THE MAGIC HAPPENS

Nick Ciesinski

University of Wisconsin - Whitewater

“The process of establishing connections between endpoints, or between an endpoint and a gatekeeper/registrar”

SIGNALING

SIGNALING: PROTOCOLS

- **H.323**
- **SIP**
- MGCP
- SCCP (SKINNY)
- DTMF
- QSIG
- Q.931

SIGNALING: H323

- First published by the International Telegraph Union (ITU) in 1996
  - Current version approved in 2009
- Widely deployed and widely known
- Not as easy to troubleshoot as other protocols
- Common Terms
  - Terminals
  - Multipoint Control Units (MCU)
  - Gateways
  - Gatekeepers
  - Border Elements

SIGNALING: SIP

- Designed in 1996 and standardized in 1999 by IETF (RFC 2543)
  - Current version published in 2002 (RFC 3261)
- Gaining popularity in both voice and video
- Easy to troubleshoot
  - Text-based protocol
    - Uses many elements of HTTP and SMTP
- Media identification and negotiation uses Session Description Protocol (SDP)
- Common Terms
  - User Agent
  - Registrar & Proxy
  - Gateway
  - Session Border Controller & B2BUA

SIGNALING: GATEKEEPER

- Call Admission Control for H.323
  - Permit/Deny calls based on bandwidth, rules, etc.
- Translation services from E.164 to IP addresses
- Not required component of H.323
  - Generally seen in large H.323 deployments
- Does not do gateway functions but can be combined with gateway to be Session Border Controller

SIGNALING: REGISTRAR & PROXY

- Registrar: SIP endpoint (generally server) that accepts REGISTER requests
  - Puts registrations into a location service that links one or more IP addresses to the SIP URI of the user agent
- Proxy: SIP endpoint (generally server) that acts as both server and client for the purpose of making requests on behalf of other clients
- Generally registrar and proxy are the same server
- Not required in SIP deployments but highly recommended to ease issues. Some devices its required.
- Some similarities to H323's gatekeeper

SIGNALING: GATEWAYS

- Used in both H323 and SIP to interface with another network.
  - PSTN
  - Sometimes will do protocol switching
    - SIP -> H323
    - SIP -> ISDN
    - H323 -> ISDN

SIGNALING: SESSION BORDER CONTROLLERS

- Similar to a gateway sometimes confused as the same thing
- It is a device that exerts control over the signaling and possibly media
- Generally found in telecommunication networks or at network borders to link multiple customers together.
- Functions of a SBC
  - NAT traversal
  - Normalization
  - IPv4 to IPv6 interworking
  - Protocol translations
  - QoS
    - Policing
    - Call Admission Control (CAC)
    - ToS/DSCP marking
  - Media transcoding
  - Statistics and billing info

SIGNALING: B2BUA

- Back to Back User Agent (B2BUA)
  - Operates in between both ends of a call
    - Each endpoints signaling terminates on the B2BUA
    - Often also media is terminated on B2BUA
- Useful for
  - Address hiding
  - Adding value-added features available during call
  - Giving full control over the session

SIGNALING: EXAMPLE

INVITE sip:johnsmith@university.edu SIP/2.0
Via: SIP/2.0/UDP registrar.university.edu;branch=z9hG4bK776asdhds Max-Forwards: 70
To: John Smith <sip:johnsmith@university.edu>
From: Joe Brown <sip:joebrown@university.edu>;tag=1928301774
Call-ID: a84b4c76e66710@registrar. university.edu
CSeq: 314159 INVITE
Contact: <sip:johnsmith@registrar.university.edu>
Content-Type: application/sdp
Content-Length: 142

SIGNALING: SIP SDP

- Format for describing streaming media initialization
- Used in
  - Real-Time Transport Protocol (RTP)
  - Real-time Streaming Protocol (RTSP)
  - SIP
  - Standalone Multicast sessions
- Media negotiation between endpoints in SIP is done with SDP
- Like SIP also text based

SIGNALING: SDP EXAMPLE

```
v=0
o=CiscoSystemsCCM-SIP 575030 1 IN IP4 10.246.200.21
s=SIP Call
b=AS:4756
t=0 0
a=X-cisco-mux: cisco
m=audio 27964 RTP/AVP 96 101
c=IN IP4 10.242.200.2
b=TIAS:256000
a=rtpmap:96 mpeg4-generic/48000
a=fmtp:96 profile-level-id=16;streamtype=5;config=B98C00;mode=AAC-hbr;sizeLength=13;indexLength=3;indexDeltaLength=3;constantDuration=480
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=mid:1
m=video 17322 RTP/AVP 97
```

DIALING

DIALING: DESIGN & DIALPLAN

- When designing your dial plan determine who you need to call
  - Internal only or external?
  - What protocols do I have to interwork with?
  - How will external entities connect with me?
  - What is the industry doing?
  - What is easy for my users?
  - What is easy for me the administrator?
  - How can I future proof my dialing plan

DIALING: DESIGN & DIALPLAN

- Most common dialing schemes
  - URI
  - E.164
  - IP
- URI
  - username@domain.edu
  - Industry direction
  - Simple, generally the same as e-mail address
  - Not just SIP but H.323
    - H.323 Annex 0
  - Requires the use of registrar/gatekeeper if using top level @domain.edu vs @IP Address
  - Some devices do not support @ symbol on keypad

DIALING: DESIGN & DIALPLAN

- E.164
  - Plus (+) based dialing ex +15555551234
  - Easy to use we all know how to dial a phone number, right?
  - More common in voice then in video
  - ENUM (E.164 Number to URI Mapping) Database
    - A common registry/database of numbers. There are several available and are managed by different entities and some have restricted access.
      - NRENum.net (Internet2)
      - E164.org
  - Device support for + key on keypad
  - System support for + in call signaling

DIALING: DESIGN & DIALPLAN

- IP
  - Easy for administrators but confusing for end users. What's a IP?
  - More common in academia
    - Public vs Private IP's
  - Many deployments have no gatekeeper and endpoints sit outside firewall
    - Toll Fraud targets
  - Issues for SIP only endpoints
  - What happens with IPv6?
    - That's one big number to dial
  - Device move generally requires a new IP and need to give new IP to users

DIALING: DESIGN & DIALPLAN

- ENUM
  - DNS lookup using NAPTR record type
  - Some systems do not support ENUM
    - Some systems may support ENUM but a different syntax
  - Need to setup what ENUM e.164 tree you are looking at

$ORIGIN 2.4.2.4.5.5.5.5.5.5.1.e164.arpa.

IN NAPTR 100 10 "u" "E2U+sip" "!^.*$!sip:phoneme@example.net!" .

PUTTING IT TOGETHER

- Consider SIP if you have not already
  - Future
  - Easy troubleshooting
  - Easy dialing
  - Lots of registrar/proxy options available
- Make use of gateway/SBC
  - Put endpoints behind firewall with no firewall holes let the gateway anchor media
    - Easier to deal with toll fraud attempts
  - Recommendation
    - Disable SIP UDP only use TCP on outside

PUTTING IT TOGETHER

- This presentations description said something about where the magic happens, so where is the magic?
  - No real magic, just a few cheap parlor tricks

SCENARIO 1

- I have SIP devices connected to a SIP registrar/proxy and I need to make video calls to and from university A to university B. Both university A and university B only support E.164 dialing

- University A and University B
  - Can have some sort of gateway or SBC that supports ENUM
    - Calls are redirected to gateway or SBC and a DNS ENUM lookup is performed
      - Calls are sent to other universities gateway or SBC
  - Can setup a direct SIP peer between registrar/proxy servers
    - Configure call routes for other universities E.164 numbers. Calls are redirected to other universities registrar/proxy server
      - Note, some proxy/registrar servers do not anchor media!

SCENARIO 1

- University A and University B
  - Can have some sort of gateway or SBC without ENUM
    - Calls are redirected to gateway or SBC
      - Cheap Parlor Trick
        - Gateway or SBC is programmed to look for other universities E.164 numbers
          - Gateway/SBC appends @domain.edu to the dialed number
            - Call sent via standard SIP DNS SRV lookup to other university

SCENARIO 2

- I have SIP devices connected to a SIP registrar/proxy and I need to make video calls to and from university A to university B, but university B only supports direct IP calling where we support only URI dialing
- University A
  - Needs to have some sort of gateway or SBC to handle incoming H323 IP calls from university B.
    - Gateway/SBC needed to interwork H.323 and SIP calls
      - How to I convert a IP into a URI?
        - Cheap Parlor Trick:
          - Remember H323 Annex 0?
            - Can they dial by URI?
              - No, they don't have a @ key on their keypad
            - Some devices support alternate URI dialing
              - IP Address Of Gateway##URI Username
                - 10.10.10.10##joeuser

SCENARIO 2

- University A
  - Needs to have a way to call outbound IP calls to University B
    - Gateway/SBC needed to interwork H.323 and SIP calls
      - Cheap Parlor Trick:
        - SIP requires the username and domain portion in the signaling how can I fake it out?
          - Create a dialing pattern you will modify at the gateway
            - 10.20.20.20@ip.address What???
              - At gateway/SBC strip bogus domain @ip.address off incoming calling string all that is left is the IP address and then gateway sends call to IP over H.323

INTERNET2 VIDEO EXCHANGE

- Open to everyone even non-Internet2 members
  - Some services only available to members
  - Some services free others charged
- Services
  - Device registration
  - Education community dialing
  - Virtual meeting rooms (3+ participants)
  - TATA Jamvee
  - ENUM registration
- Support SIP and H.323
- E.164 and URI dialing plan

INTERNET2 VIDEO EXCHANGE

- Infrastructure
  - North America
    - Cisco Video Communications System (VCS)
    - Cisco Conductor
    - Cisco Unified Communications Manager
    - Cisco Telepresence Server
    - Cisco Unified Border Element
  - Asia (Singapore)
    - Cisco Video Communications System (VCS)
    - Cisco Conductor
    - Cisco Unified Communications Manager
    - Cisco Telepresence Server
    - Cisco Unified Border Element
- Systems running latest versions of software to take advantage of the latest features.

INTERNET2 VIDEO EXCHANGE

- How to get more information?
  - Email: video-support@internet2.edu
- How to setup link to Intetnet2 video exchange?
  - https://questionpro.com/t/AJDgFZPdcK
- How to subscribe to services?
  - https://internet2.app.box.com/netplus-videoex-app


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SIP TROUBLESHOOTING WHAT TO DO WHEN THINGS GO WRONG

Nick Ciesinski

University of Wisconsin - Whitewater

BASIC SIP REQUEST METHODS

- **INVITE** – The invite to participate in a voice or video session
- **ACK** – Confirmation that a device has received a response to a request
- **BYE** – Terminates an existing session; can be sent by any device in a session
- CANCEL – Cancels any pending requests
- OPTIONS – Determines capabilities of systems. Can also be used for keep alive (OPTIONS PING)
- REGISTER – Registers the device (user agent) with the server for the domain.
- INFO – Send more information
- REFER – To tell one user agent to communicate with another

SIP CALL

- Call to 111@bjn.vc

SIP INVITE

INVITE sip:111@bjn.vc SIP/2.0

Via: SIP/2.0/TLS 140.146.20.8:5061;egress-zone=TraversalZone;branch=z9hG4bK3e1cc481c02192d1e814d888fd09a483366117.b02f91f5cfb9b35bb7f747d133d42b4b;proxy-call-id=7dbff6b7-4e68-4deb-ae47-d2b07495f3ac;rport

Via: SIP/2.0/TCP 140.146.20.5:5062;branch=z9hG4bK673ed65ed1b5e;received=140.146.20.5;ingress-zone=CUCM

Call-ID: e27a8500-541135db-65b66-514928c@140.146.20.5

CSeq: 101 INVITE

Remote-Party-ID: "Nick Ciesinski" <sip:ciesinsn@uww.edu;x-cisco-number=7774>;party=calling;screen=yes;privacy=off

Contact: <sip:ciesinsn@140.146.20.5:5062;transport=tcp>;video;audio;+multiple-codecs-in-ans

From: "Nick Ciesinski" <sip:ciesinsn@uww.edu>;tag=64023402~6d045f31-1dfc-45b1-b614-164f86bd8be1-44940887

To: <sip:111@bjn.vc>

Max-Forwards: 15

Record-Route: <sip:proxy-call-id=7dbff6b7-4e68-4deb-ae47-d2b07495f3ac@140.146.20.8:5061;transport=tls;lr>

Record-Route: <sip:proxy-call-id=7dbff6b7-4e68-4deb-ae47-d2b07495f3ac@140.146.20.8:5060;transport=tcp;lr>

Allow: INVITE,OPTIONS,INFO,BYE,CANCEL,ACK,PRACK,UPDATE,REFER,SUBSCRIBE,NOTIFY

User-Agent: Cisco-CUCM10.5

Expires: 180

Date: Wed, 29 Apr 2015 19:49:47 GMT

Supported: timer,resource-priority,replaces,X-cisco-srtp-fallback,X-cisco-original-called

Session-Expires: 1800

SIP INVITE

SIP/2.0 100 Trying

Via: SIP/2.0/TLS 140.146.20.8:5061;egress-zone=TraversalZone;branch=z9hG4bK3e1cc481c02192d1e814d888fd09a483366117.b02f91f5cfb9b35bb7f747d133d42b4b;proxy-call-id=7dbff6b7-4e68-4deb-ae47-d2b07495f3ac;received=140.146.20.8;rport=25026;ingress-zone=TraversalZone

Via: SIP/2.0/TCP 140.146.20.5:5062;branch=z9hG4bK673ed65ed1b5e;received=140.146.20.5;ingress-zone=CUCM

Call-ID: e27a8500-541135db-65b66-514928c@140.146.20.5

CSeq: 101 INVITE

From: "Nick Ciesinski" <sip:ciesinsn@uww.edu>;tag=64023402~6d045f31-1dfc-45b1-b614-164f86bd8be1-44940887

To: <sip:111@bjn.vc>

Server: TANDBERG/4130 (X8.5.2Alpha8)

Content-Length: 0

SIP INVITE

SIP/2.0 180 Ringing

Via: SIP/2.0/TLS 140.146.20.8:5061;rport=25026;received=140.146.20.8;branch=z9hG4bK3e1cc481c02192d1e814d888fd09a483366117.b02f91f5cfb9b35bb7f747d133d42b4b;egress-zone=TraversalZone;proxy-call-id=7dbff6b7-4e68-4deb-ae47-d2b07495f3ac;ingress-zone=TraversalZone

Via: SIP/2.0/TCP 140.146.20.5:5062;received=140.146.20.5;branch=z9hG4bK673ed65ed1b5e;ingress-zone=CUCM

Call-ID: e27a8500-541135db-65b66-514928c@140.146.20.5

CSeq: 101 INVITE

Contact: "BlueJeans" <sip:111@bjn.vc:5061;transport=tls>

From: "Nick Ciesinski" <sip:ciesinsn@uww.edu>;tag=64023402~6d045f31-1dfc-45b1-b614-164f86bd8be1-44940887

To: <sip:111@bjn.vc>;tag=0b9aefa1-82cb-4ec0-bc40-d905ca989b06

Record-Route: <sip:proxy-call-id=039ccdf1-5955-4e67-98c8-333d7086ac19@140.146.22.2:5061;transport=tls;lr>

Record-Route: <sip:proxy-call-id=039ccdf1-5955-4e67-98c8-333d7086ac19@140.146.22.2:7001;transport=tls;lr>

Record-Route: <sip:proxy-call-id=7dbff6b7-4e68-4deb-ae47-d2b07495f3ac@140.146.20.8:5061;transport=tls;lr>

Record-Route: <sip:proxy-call-id=7dbff6b7-4e68-4deb-ae47-d2b07495f3ac@140.146.20.8:5060;transport=tcp;lr>

Allow: PRACK,INVITE,ACK,BYE,CANCEL,UPDATE,SUBSCRIBE,NOTIFY,INFO,OPTIONS

Content-Length: 0

SIP INVITE

SIP/2.0 200 OK

Via: SIP/2.0/TLS 140.146.22.2:5061;rport=27229;received=140.146.22.2;branch=z9hG4bKe4ca822581768356c98e2f055606f490164599.51a33a259a017cb8400d654eb9ef193d;egress-zone=DNSZone;proxy-call-id=039ccdf1-5955-4e67-98c8-333d7086ac19

Via: SIP/2.0/TLS 140.146.20.8:5061;rport=25026;received=140.146.20.8;branch=z9hG4bK3e1cc481c02192d1e814d888fd09a483366117.b02f91f5cfb9b35bb7f747d133d42b4b;egress-zone=TraversalZone;proxy-call-id=7dbff6b7-4e68-4deb-ae47-d2b07495f3ac;ingress-zone=TraversalZone

Via: SIP/2.0/TCP 140.146.20.5:5062;received=140.146.20.5;branch=z9hG4bK673ed65ed1b5e;ingress-zone=CUCM

Call-ID: e27a8500-541135db-65b66-514928c@140.146.20.5

CSeq: 101 INVITE

Contact: "BlueJeans" <sip:111@bjn.vc:5061;transport=tls>

From: "Nick Ciesinski" <sip:ciesinsn@uww.edu>;tag=64023402~6d045f31-1dfc-45b1-b614-164f86bd8be1-44940887

To: <sip:111@bjn.vc>;tag=0b9aefa1-82cb-4ec0-bc40-d905ca989b06

Record-Route: <sip:proxy-call-id=039ccdf1-5955-4e67-98c8-333d7086ac19@140.146.22.2:5061;transport=tls;lr>

Record-Route: <sip:proxy-call-id=039ccdf1-5955-4e67-98c8-333d7086ac19@140.146.22.2:7001;transport=tls;lr>

Allow: PRACK,INVITE,ACK,BYE,CANCEL,UPDATE,SUBSCRIBE,NOTIFY,INFO,OPTIONS

Supported: 100rel

Content-Type: application/sdp

Content-Length: 1074

SIP INVITE

ACK sip:111@bjn.vc:5061;transport=tls SIP/2.0

Via: SIP/2.0/TLS 140.146.20.8:5061;egress-zone=TraversalZone;branch=z9hG4bK7dd945b06c26fb981b62ec5067df9e7a366118.b02f91f5cfb9b35bb7f747d133d42b4b;proxy-call-id=7dbff6b7-4e68-4deb-ae47-d2b07495f3ac;rport

Via: SIP/2.0/TCP 140.146.20.5:5062;branch=z9hG4bK673ef1b208f6;received=140.146.20.5;ingress-zone=CUCM

Call-ID: e27a8500-541135db-65b66-514928c@140.146.20.5

CSeq: 101 ACK

From: "Nick Ciesinski" <sip:ciesinsn@uww.edu>;tag=64023402~6d045f31-1dfc-45b1-b614-164f86bd8be1-44940887

To: <sip:111@bjn.vc>;tag=0b9aefa1-82cb-4ec0-bc40-d905ca989b06

Max-Forwards: 69

Route: <sip:proxy-call-id=039ccdf1-5955-4e67-98c8-333d7086ac19@140.146.22.2:7001;transport=tls;lr>,<sip:proxy-call-id=039ccdf1-5955-4e67-98c8-333d7086ac19@140.146.22.2:5061;transport=tls;lr>

User-Agent: Cisco-CUCM10.5

Date: Wed, 29 Apr 2015 19:49:47 GMT

Allow-Events: presence

X-TAATag: 824826cf-561c-40a3-8de8-fc18000372c8

Content-Length: 0

SIP ACK

ACK sip:111@bjn.vc:5061;transport=tls SIP/2.0

Via: SIP/2.0/TLS 140.146.20.8:5061;egress-zone=TraversalZone;branch=z9hG4bK7dd945b06c26fb981b62ec5067df9e7a366118.b02f91f5cfb9b35bb7f747d133d42b4b;proxy-call-id=7dbff6b7-4e68-4deb-ae47-d2b07495f3ac;rport

Via: SIP/2.0/TCP 140.146.20.5:5062;branch=z9hG4bK673ef1b208f6;received=140.146.20.5;ingress-zone=CUCM

Call-ID: e27a8500-541135db-65b66-514928c@140.146.20.5

CSeq: 101 ACK

From: "Nick Ciesinski" <sip:ciesinsn@uww.edu>;tag=64023402~6d045f31-1dfc-45b1-b614-164f86bd8be1-44940887

To: <sip:111@bjn.vc>;tag=0b9aefa1-82cb-4ec0-bc40-d905ca989b06

Max-Forwards: 69

Route: <sip:proxy-call-id=039ccdf1-5955-4e67-98c8-333d7086ac19@140.146.22.2:7001;transport=tls;lr>,<sip:proxy-call-id=039ccdf1-5955-4e67-98c8-333d7086ac19@140.146.22.2:5061;transport=tls;lr>

User-Agent: Cisco-CUCM10.5

Date: Wed, 29 Apr 2015 19:49:47 GMT

Allow-Events: presence

X-TAATag: 824826cf-561c-40a3-8de8-fc18000372c8

Content-Length: 0

SIP BYE

BYE sip:111@bjn.vc:5061;transport=tls SIP/2.0

Via: SIP/2.0/TLS 140.146.20.8:5061;egress-zone=TraversalZone;branch=z9hG4bK6e6375cd10419701e6bbeaeaeb0808e0366119.b02f91f5cfb9b35bb7f747d133d42b4b;proxy-call-id=7dbff6b7-4e68-4deb-ae47-d2b07495f3ac;rport

Via: SIP/2.0/TCP 140.146.20.5:5062;branch=z9hG4bK673f11b68b9c;received=140.146.20.5;ingress-zone=CUCM

Call-ID: e27a8500-541135db-65b66-514928c@140.146.20.5

CSeq: 102 BYE

From: "Nick Ciesinski" <sip:ciesinsn@uww.edu>;tag=64023402~6d045f31-1dfc-45b1-b614-164f86bd8be1-44940887

To: <sip:111@bjn.vc>;tag=0b9aefa1-82cb-4ec0-bc40-d905ca989b06

Max-Forwards: 69

Route: <sip:proxy-call-id=039ccdf1-5955-4e67-98c8-333d7086ac19@140.146.22.2:7001;transport=tls;lr>,<sip:proxy-call-id=039ccdf1-5955-4e67-98c8-333d7086ac19@140.146.22.2:5061;transport=tls;lr>

User-Agent: Cisco-CUCM10.5

Date: Wed, 29 Apr 2015 19:49:47 GMT

P-Asserted-Identity: "Nick Ciesinski" <sip:ciesinsn@uww.edu>

X-TAATag: 824826cf-561c-40a3-8de8-fc18000372c8

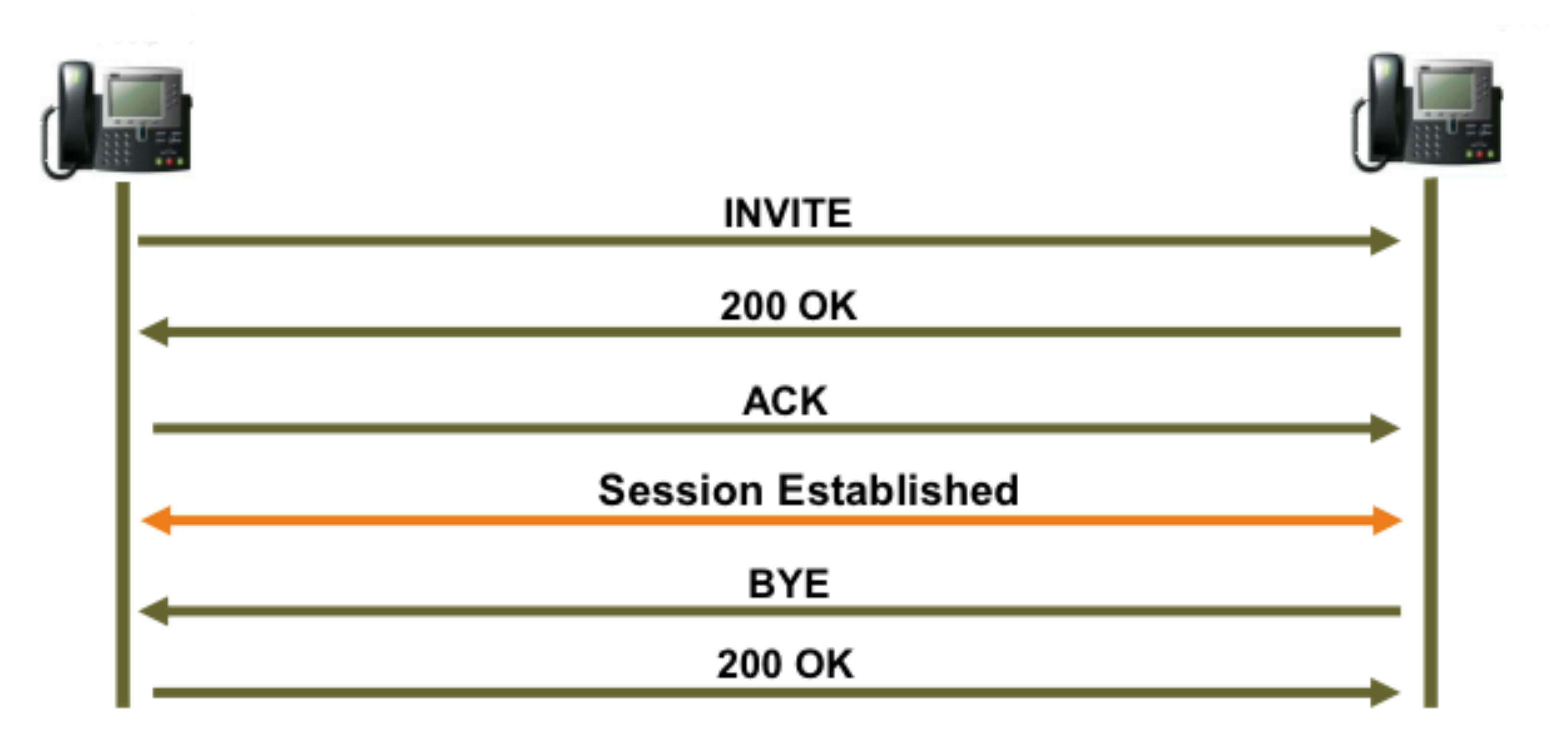
Reason: Q.850 ;cause=16

Content-Length: 0

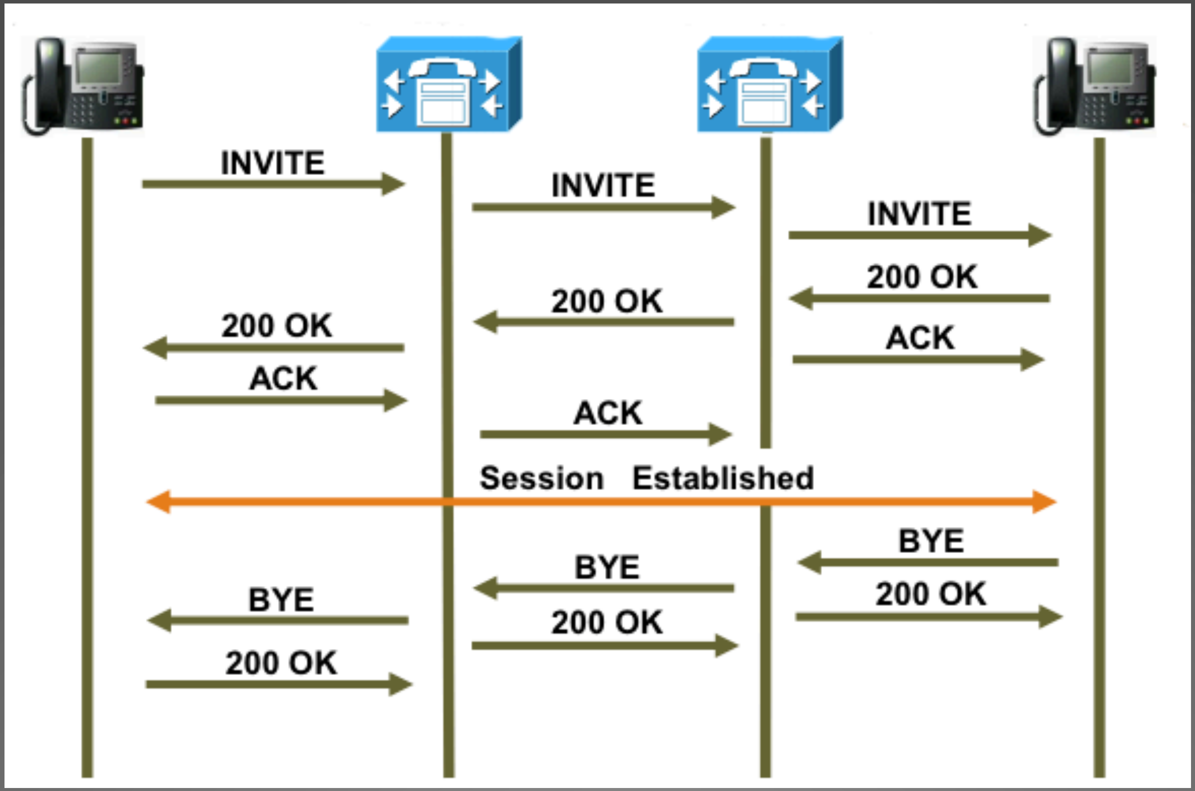
SIP RESPONSES

- 1XX – Informational
- 2XX – Success
  - 200 OK
- 3XX – Redirect
  - 301 Moved Permanently
  - 302 Moved Temporarily
- 4XX – Client Error
  - 404 Not Found
  - 486 Busy Here
- 5XX – Server Error
  - 503 Service Unavailable

BASIC CALL SETUP



COMMON CALL SETUP



SDP

FIRST DEVICE SENDS ITS CODECS

```
m=audio 51050 RTP/AVP 107 108 109 110 9 104 105 0 8 15 18 101
b=TIAS:128000
a=rtpmap:107 MP4A-LATM/90000
a=fmtp:107 bitrate=128000;profile-level-id=25;object=23
a=rtpmap:108 MP4A-LATM/90000
a=fmtp:108 bitrate=64000;profile-level-id=24;object=23
a=rtpmap:109 MP4A-LATM/90000
a=fmtp:109 bitrate=56000;profile-level-id=24;object=23
a=rtpmap:110 MP4A-LATM/90000
a=fmtp:110 bitrate=48000;profile-level-id=24;object=23
a=rtpmap:9 G722/8000
a=rtpmap:104 G7221/16000
a=fmtp:104 bitrate=32000
a=rtpmap:105 G7221/16000
a=fmtp:105 bitrate=24000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:15 G728/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=trafficclass:conversational.audio.immersive.aq:admitted
```

```
m=video 51052 RTP/AVP 97 126 96 34 31
b=TIAS:5952000
a=label:11
a=answer:full
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=420016;packetization-mode=0;max-mbps=245000;max-fs=9000;max-cpb=200;max-br=5000;max-rcmd-nalu-size=3456000;max-smbps=245000;;max-fps=6000
a=rtpmap:126 H264/90000
a=fmtp:126 profile-level-id=428016;packetization-mode=1;max-mbps=245000;max-fs=9000;max-cpb=200;max-br=5000;max-rcmd-nalu-size=3456000;max-smbps=245000;;max-fps=6000
a=rtpmap:96 H263-1998/90000
a=fmtp:96 QCIF=1;CIF=1;CIF4=1;CUSTOM=352,240,1
a=rtpmap:34 H263/90000
a=fmtp:34 QCIF=1;CIF=1;CIF4=1
a=rtpmap:31 H261/90000
a=fmtp:31 CIF=1;QCIF=1
a=content:main
a=rtcp-fb:* nack pli
a=trafficclass:conversational.video.immersive.aq:admitted
m=application 51054 UDP/BFCP *
a=userid:182
```

SDP

SECOND DEVICE RESPONDS WITH WHAT WILL BE USED

m=audio 5046 RTP/AVP 9 101

a=rtcp:5047

a=rtpmap:9 G722/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-15

a=sendrecv

m=video 5048 RTP/AVP 126

b=TIAS:1472000

a=rtcp:5049

a=rtpmap:126 H264/90000

a=fmtp:126 profile-level-id=42801f;max-mbps=108500;max-fs=3600;packetization-mode=1

a=rtcp-fb:* nack pli

a=rtcp-fb:126 nack

a=rtcp-fb:* ccm fir

a=rtcp-fb:* nack sli

a=rtcp-fb:* ack rpsi

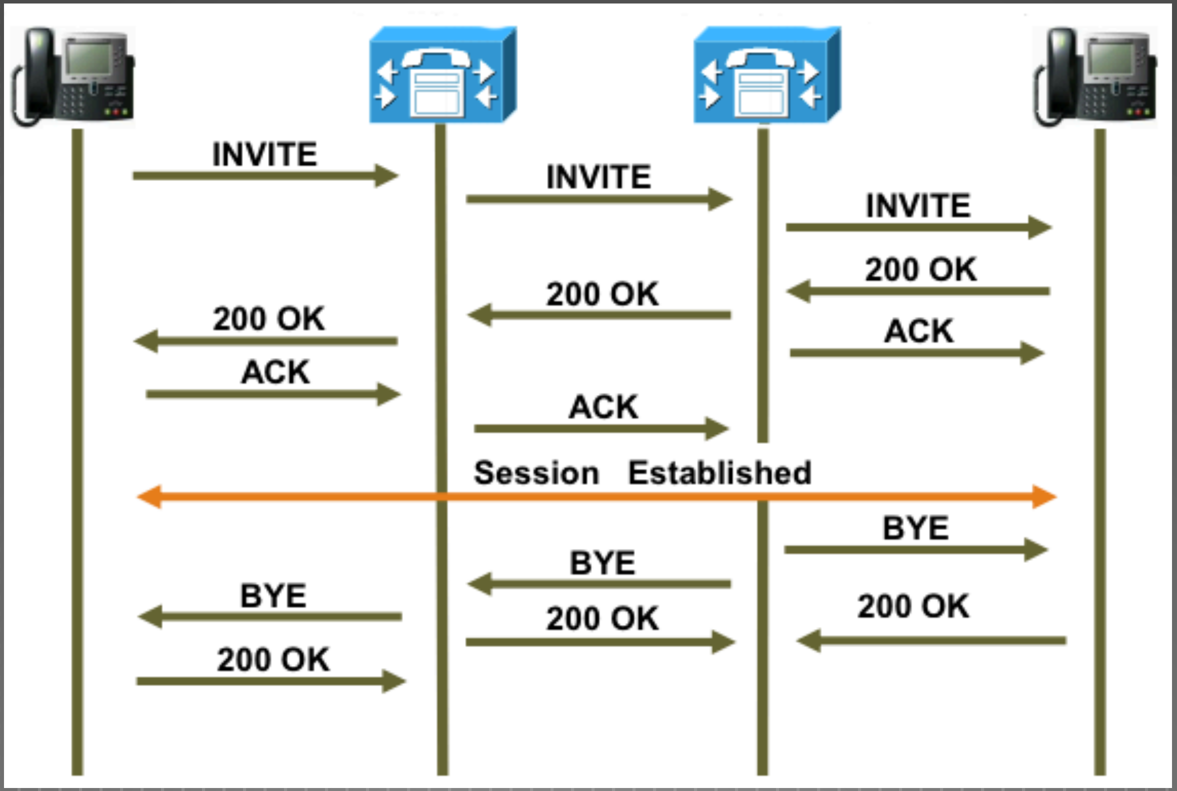
a=rtcp-fb:* ccm tmmbr

a=content:main

a=label:11

a=sendrecv

COMMON SEEN ISSUE



WHERE TO START

- Find the device that sent the BYE
  - SIP messages may not give all the details to why a call failed on all hops in the call path
    - Especially in B2BUA sessions
  - Turn debugging on (if not already) and do another call and capture traces from device sending the BYE
    - All devices have their own set of debug settings
      - Cisco CUBE
        - Debug ccsip messages (SIP messages)
        - Debug voip ccapi inout (Device messages)
      - Cisco/Tandberg VCS/Expressway
        - Maintenance -> Diagnostics -> Diagnostic Logging

COMMON ISSUES

- 404 Errors
  - Wrong number dialed
  - Incorrect translations taking place
- Media Negotiation Failure
  - One side set to delayed offer other side expecting early offer
    - Delayed offer
      - SDP offered by called device in 200 OK
        - Return SDP offered in ACK
    - Early offer
      - SDP offered by calling device in INVITE
        - Return SDP offered in 200 OK

COMMON ISSUES

- Media Negotiation Failure
  - No SDP media codecs in common
    - Verify settings and if devices support a common codec
    - Bandwidth restrictions set on server limit the use of certain codecs
  - Codecs in common but no audio or video or one way
    - Verify in SDP that the IP listed in C= lines are actually accessible outside firewall
      - In NAT situations sometimes you must enable fixups to re-write the IP on the firewall/NAT device
    - Media does not have to be anchored by the signaling device
      - Verify media is flowing through, not around device and being caught by a firewall restriction

BIGGEST TIPS

- Look at things one hop at a time!
- Verify code versions of endpoints and registrars/proxy
  - Sometimes features are added that one side may not understand
    - iX Application Media


Q.
A.