ENUM in Cisco Products

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Current Cisco ENUM support

- Cisco CallManager
- Unified Communication Manager
- Cisco Voice Gateways, SRST, Cisco CallManager Express
- Cisco ASA/PIX Firewall
- IP phones
- Cisco Softswitch
- BTS-10200 & PGW2200
- Cisco SIP Proxy Server
Cisco IOS ENUM support

Voice Gateway

PBX

ISDN PRI/BRI
analog

IP-IP Gateway

CallManager
Express

LAN

CallManager
cluster

SoftSwitch

IP

ENUM-enabled
(Internet)

IP-to-IP
ENUM Challenges – Incoming Call

- Caller has to respect my (DNS-propagated) signaling protocol choice
- Interoperability:
  - DTMF Relay (in-band/out-of band)
  - Media Setup (early/delayed)
  - Codec Negotiation (iLBC, G.711, G.729,...)
ENUM Challenges – Outgoing Call

- I have to respect signaling protocol choice (H.323, SIP), and I don’t know in advance what it will be.

- Interoperability:
  - DTMF Relay (in-band/out-of band)
  - Media Setup (SIP early/delayed, H.323 slow-start/fast-start)
  - Codec Negotiation (iLBC, G.711, G.729,...)
ENUM Challenges – Security

- Do I want to...
  
  open my IP PBX to any anonymous incoming call?  
  **TOLL FRAUD / MALICIOUS CALL**

  open my IP PBX to unlimited number of incoming calls?  
  **DENIAL OF SERVICE**

  expose my IP PBX directly to the internet?  
  If firewall (with NAT) is used does it support all possible signaling protocols and its nuances?  
  **ENDPOINT SECURITY**
ENUM Challenges - Summary

- **Interoperability**
  - Protocol support (SIP, H.323)
  - DTMF relay (RFC2833, SIP INFO, SIP NOTIFY, H.245-alpha, ...)
  - Media setup (SIP early/delayed media, H.323 fast/slow start)
  - Codec negotiation (G.711, G.729, iLBC, G.723, G.726, G.722, ...)

- **Security**
  - Denial of service
  - Endpoint/IP PBX protection

- **Call Quality**
  - Call Admission Control – controlling maximum number of calls

- **Accounting/Traffic Monitoring**
  - CDR

- How to transparently provide ENUM service to a VoIP system that does not support it?
Cisco IP-to-IP Gateway as ENUM Proxy

- Network/Topology Hiding for Voice and Video Calls (Media Proxy)
- Protocol Support - H.323 and SIP
- Voice Codecs – G.711, G.729, G.726, G.723, G.728, Transparent
- Video Codecs – H.261, H.263 and H.264
- Codec Filtering
- Media - Media Flow Through and Media Flow Around
- DTMF Interworking – H.245 Alphanumeric, Signal, RFC2833, SIP NOTIFY
- Fax/Modem – T.38, Passthrough, Cisco Fax Relay, Modem Passthrough
- Security – TLS, IPSec with SRTP
- ENUM
- Signaling Interworking
- Supplementary Services
- Transcoding – G.711, G.729, iLBC
- Transport Mode - TCP, UDP
- Number Translation
- Quality of Service
- Call Admission Control
- Call Detail Records
- TCL/VXML Support
- Rotary Support (dial-peer selection)
ENUM-enabling the existing solution

CallManager

Generic IP PBX

H.323
SIP

PSTN

PRI

Terminated on TDM

Cisco IOS Voice Gateway

PSTN

PRI

ENUM → H.323, SIP

Cisco IOS Voice Gateway upgraded to IP-IP

CallManager

Generic IP PBX

H.323
SIP

SBC

ENUM-enabling the existing solution
ENUM with IP-IP Gateway Practical Example

- ENUM implementation in IOS
  [Link](http://www.cisco.com/en/US/products/sw/iosswrel/ps1839/products_feature_guide09186a00800b5dbf.html#wp1061771)
is bound to dial-peers, i.e. does not allow dynamic protocol choice based on ENUM response

- TCL script in IP-IP gateway and external ENUM-query server can solve it
Summary

- ENUM supported in IOS (TDM gateways, CallManager Express, IP-IP gateway)
- Add-on to existing rich VoIP features in IOS
- IP-IP as proxy for non-ENUM clients/IP PBXs
Voice Call Support

H.323-H.323

<table>
<thead>
<tr>
<th>In Leg</th>
<th>Out Leg</th>
<th>Support</th>
</tr>
</thead>
<tbody>
<tr>
<td>Fast Start</td>
<td>Fast Start</td>
<td>Bidirectional</td>
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<tr>
<td>Slow Start</td>
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H.323-SIP

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<tbody>
<tr>
<td>Fast Start</td>
<td>Early Offer</td>
<td>Bidirectional</td>
</tr>
<tr>
<td>Slow Start</td>
<td>Delayed Offer</td>
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SIP-SIP

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😊 Delayed Offer to Slow Start Support in Future
Transcoding

- Supports conversion from one codec type to another for the voice call (e.g. from G.729 to G.711 or vice versa)

- Packetizations supported:
  - G.711: 10ms, 20ms and 30ms
  - G.729: 10ms, 20ms, 30ms, 40ms, 50ms and 60ms

- Same Codec different packetizations is not supported

- Transcoding requires a dedicated DSP

<table>
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<tr>
<td>G.711 a-law/µ-law</td>
<td>G.729, G.729A, G.729B, G.729AB</td>
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Support for iLBC < > G.711 in 12.4(15)T
# H.323—SIP DTMF Interworking

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Call Admission Control Mechanisms

1. Total calls
2. CPU
3. Memory
4. RSVP
5. IP call capacity
6. Max-connections