Advanced Voice over IP
Enhanced Services
Session 2105
Other Voice Sessions

- 2104 – Deploying Large Scale SP VoIP Networks
- 2005 – Understanding Voice Signaling
- 2101 – Traffic Engineering for Voice over Integrated Service Networks
- 2004 – Understanding Unified Messaging for Service Providers
- 2001 – Intro to Voice and Telephony Technology
- 2002 – Intro to Packet Voice Technologies

Agenda

- Introduction
- Technology Enablers
- Implementation Strategy
- Future Developments
Agenda

- Introduction
  - Open Packet Telephony
  - VoIP for Long Distance
- Technology Enablers
- Implementation Strategy
- Future Developments

Consider a Circuit Switch

TDM Switch

- Robust Carrier Grade Architecture
- Narrowband Access Only
- Proprietary Internal Protocols
- Limited to 64 Kb
- Line Concentration
- Digital Trunk Subsystem
- Switch-Based Features (Voice Only)
- Closed Inflexible Unmanageable
- Interface to IN Service Logic
- Costly and Inflexible
- Call Control
- Connection Control Features
- Common Channel Signaling Complex
- Administration
- Maintenance
- Billing
Open Packet Telephony Architecture

Open and Standardize the Telephony Infrastructure

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Private VoIP Toll Bypass

Enterprise Toll Bypass
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Agenda

- Introduction
- Technology Enablers
  - H.323 Overview
  - RAS Extensions
  - Empty Capability Set
  - Supplemental Services
  - Gatekeeper API/TMP
- Implementation Strategy
- Future Developments

H.323 Background

H.320 Network

H.323 Network

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H.323 Background

- ITU recommendation
- Defines multimedia applications over packet-based networks
  Voice coding is a requirement
- Leverages existing standards
- Wide market acceptance
- Facilitates interoperability between vendors
- Cisco VoIP solutions are H.323 compliant

H.323 and OSI

<table>
<thead>
<tr>
<th>Layer</th>
<th>Protocol</th>
</tr>
</thead>
<tbody>
<tr>
<td>Presentation</td>
<td>G.729(A)/G.723(.1)/G.71</td>
</tr>
<tr>
<td>Session</td>
<td>H.323</td>
</tr>
<tr>
<td>Transport</td>
<td>RTP/RTCP</td>
</tr>
<tr>
<td>Network</td>
<td>IP</td>
</tr>
<tr>
<td>Link</td>
<td>MLPPP/FR/ATM AAL1</td>
</tr>
<tr>
<td>Physical</td>
<td>---</td>
</tr>
</tbody>
</table>
H.323v2 Signaling—Fast Connect

H.323 Gateway

PSTN/Private Voice

Signaling

Bearer or Media

IP QoS Network

Setup (Fast Start, Element including Logical channels)

POTS/PSTN Call Set-up: Ringing, Answer...

Connect (Fast Start, logical channels)

RTP Stream

RTCP Stream

Q.931 Derived Call-Setup
H.225 with Optional Fast Start Elements

H.223 (TCP)

Gateway

H.323 Gateway

PSTN/Private Voice

H.323 Signaling—Direct Mode

Gateway A

GK A

GK

GK B

Gateway B

Setup

RRQ/RCF

ARQ

ACF

LRQ

LCF

Setup

Call Proceeding

Alerting/Connect

H.245 Master/Slave

H.245 Cap Exchange

H.245 OLC

Media (RTP)

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H.323 Signaling—Gatekeeper Routed

Gateway A

RRQ/RCF
Setup
ARQ
ACF
Setup
Call Proceeding
Alerting/Connect
H.245

GK A

GK

GK B

GK

Gateway B

RRQ/RCF
Setup
LRQ
LCF
Setup
Call Proceeding
Alerting/Connect
H.245

Media (RTP)

H.323 Signaling—Direct/Routed

Gateway A

RRQ/RCF
Setup
ARQ
ACF
Setup
Call Proceeding
Alerting/Connect
H.245

GK A

GK

GK B

GK

Gateway B

RRQ/RCF
Setup
LRQ
LCF
Setup
Call Proceeding
Alerting/Connect
H.245

Media (RTP)
H.323 Signaling—Direct/Routed

Gateway A

Setup

RRQ/RCF

ARQ

ACF

GK A

GK

GK B

Gateway B

Setup

Call Proceeding

Facility

Release Complete

Setup

Call Proceeding

Alerting/Connect

H.245

Media (RTP)

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Open Packet Telephony and H.323

Open Service Application Layer

Open/Standard Interface

Open Call Control Layer

Open/Standard Interface

Standards-Based Packet Infrastructure Layer

Services Plane
  Applications
    • Internet Call Waiting
    • Click to Dial
    • VPNs

Call Control Plane
  H.323 Gatekeeper
    • Translation/Routing
    • Registration
    • H.323 Zones

Connection Plane
  H.323 Terminals
  H.323 Gateways
  IP Routers and Switches

Legend:
1. Incoming call forwarded to gateway
2. Incoming call presented to Application Server
3. Incoming call presented to user
4. Transfer to PC requested by user
5. H.323 Call setup between telephone and PC
6. Speech path established

Example: Internet Call Waiting

Internet Call Waiting

GK

Packet Network Infrastructure

ICW Server

Legend:
1. Incoming call forwarded to gateway
2. Incoming call presented to Application Server
3. Incoming call presented to user
4. Transfer to PC requested by user
5. H.323 Call setup between telephone and PC
6. Speech path established
Directory Gatekeeper—Scaling

Small Network—Gateways only

Small Network—Simplified with a Gatekeeper

Medium Network—Multiple Gatekeepers

Medium-Large Network—Multiple Gatekeepers and a Directory Gatekeeper

Gateway, Gatekeeper, Directory Gatekeeper

gkk gkk gkk

gkk gkk gkk

LRQ Forwarding

Dir GK maintains no state
Accomplished by adding TTL field
Supported in 12.03(T)

ACF

Setup
LRQ Forwarding

- Use of LRQ forwarding to find applications

GK A -> LRQ -> Dir GK -> LRQ -> GK B

Gateway A

Phone A

GK

ACF

ARQ

Setup

GK

IP Phone

RAS Extensions

- Enhance RAS Messages to include more information fields from PRI interfaces
- Support “CanMapAlias” field in ARQ and LRQ for call rerouting
- Support pass-thru RIP message to endpoint by gatekeeper
ARQ/LRQ Extensions

ARQ
- redirectIEInfo
- callingOctet3a
- displayInformationElement
- interfaceSpecificBillingID
- interfaceDescription

LRQ
- redirectIEInfo
- callingOctet3a
- gatewaySrcInfo
- displayInformationElement

RAS Extensions

PSTN  PRI  Setup  CP

ARQ  RIP  ACF  ARQ  ACF

LRQ  RIP  LCF

GK  B

APP  Server

User Reg.

Remapping info

Tunneled RDN, Display
canMapAlias = True

Remapping info

Tunneled RDN, Display
gatewaySrcInfo
canMapAlias = True

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ICW—Route Call To PSTN

ICW—Accept Call

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Moving Connections

MTP = Media Termination Point  
SSP = Skinny Station Protocol  
AVVID = Architecture for Voice Video and Integrated Data

Empty Capability Set
ECS with Unified Messaging

- RTP Stream
- RTCP Stream
- Unified Messaging Server
- LDAP
- PSTN/Private Voice
- H.323 Gateway
- IP Network
- ECS
- Master-Slave Determination
- Capability Exchange
- OLC (EP-D)
- RTP Stream
- RTP Stream
- RTP Stream
- RTCP Stream

Supplementary Services

- H.450 is an optional component of H.323v2
- H.450 defines supplementary features for H.323v2
  - H.450.1 defines signaling between endpoints
  - H.450.2 defines call transfer
  - H.450.3 defines call diversion (call forward on busy, no answer, or no reply)
- Unlike ECS, H.450 requires a new call setup
- H.450 PDUs are sent in H.225 signaling messages between endpoints
Call Transfer without Consultation

Call Diversion with H.450
Gatekeeper Transaction Messaging Protocol and API

- GKTMP provides a transaction-oriented application protocol that allows an external application to modify gatekeeper behavior by processing specified RAS messages.

GK TMP Messages Supported

- ACF—Admission Confirm
- ARJ—Admission Reject
- ARQ—Admission Request
- LCF—Location Confirm
- LRJ—Location Reject
- LRQ—Location Request
- RCF—Registration Confirm
- RRQ—Registration Request
- RRJ—Registration Reject
- URQ—Unregistration Request
GK TMP Message Interpretation

- For RRQ, URQ, the application server performs gatekeeper authorization, storing endpoint RAS gatekeeper IP addresses, and maintaining gatekeeper resource control.
- For ARQ, LRQ, the application server performs authorization and digit translation functions and returns terminating IP addresses or a new E.164 address to the gatekeeper for re-origination by the originating gateway.
- For LCF, LRJ the application server intercepts location responses from a distant gatekeeper and modifies the message fields before responding to the originating gateway.

GK TMP Example
**Hookflash Relay**

- Supported with H.245 DTMF relay (alpha or numeric, but not with Cisco-RTP DTMF relay; Also works with Fast Connect

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**Agenda**

- **Introduction**
- **Technology Enablers**
- **Implementation Strategy**
  - Infrastructure
  - Application Zones
  - Billing
  - Security
- **Future Developments**
Building a Network

Application Zone/Ecosystem Partner
- Services Plane
  - Internet Call Waiting
  - Click to Talk
  - Virtual Second Line
  - Unified Messaging
- Call Control Plane
  - H.323 Gatekeeper
- Services Plane
  - IP-PBX
  - IP-ACD
  - IP-Conference
- Connection Plane
  - Soft-Phone Resources:
    - DSP, MCU
    - Database
  - Internet Call Waiting
  - Click to Talk
  - Virtual Second Line
  - Unified Messaging
  - Unified Messaging
  - IP-PBX
  - IP-ACD
  - IP-Conference
  - Soft-Phone
  - IP-Phone
  - IP-Conference
  - IP-ACD
  - IP-PBX

Infrastructure Zone
- Services Plane
  - Prepaid
  - Wholesale
  - Network Security
  - Network QoS
  - Accounting
- Call Control Plane
  - H.323 Gatekeeper
- Services Plane
  - H.323 Gateways
  - Network Security
  - Network QoS
  - Accounting
- Connection Plane
  - H.323 Gateways

Zone Interfaces

Release 1: 12.0(4)XH1
- GK-GK Protocol (LRQ)
- App Server
- GK
- Packet Network
- Packet Network
- PSTN

Release 2/3: 12.0(7)T, 12.1(1)T
- GK API
- User-User Protocol (Empty Capability Set, H.450)
- App Server
- GK
- Packet Network
- Packet Network
- PSTN
Each of the Call Legs can generate Start and Stop records.
Each call leg reports the NTP time for when the SETUP was issued, the Call was CONNECTED and the DISCONNECT was received.
The Stop records have the required information for billing.
IOS accounting for Voice uses standard RADIUS attributes where possible.
Other attributes are “packed” into the Acct-Session-Id field (attribute 44).
max length 256 characters—defined to contain 10/-separated fields.
The various call leg records for a single call can be organized by the connection ID, (one of the fields in the Acct-Session-Id).

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**Standard Supported RADIUS Attributes (RFCs 2138 and 2139)**

<table>
<thead>
<tr>
<th>Attribute</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>NAS- IP-Address</td>
<td>4</td>
<td>IP address in 4 hex octets (ASCII string)</td>
</tr>
<tr>
<td>NAS-Port-Type</td>
<td>61</td>
<td>4 octets (used for MLPPP)</td>
</tr>
<tr>
<td>User-Name</td>
<td>1</td>
<td>ASCII string field up to 63 octets</td>
</tr>
<tr>
<td>Called-station-Id</td>
<td>30</td>
<td>1 or more octets (DNIS phone number, ASCII string)</td>
</tr>
<tr>
<td>Calling-Station-Id</td>
<td>31</td>
<td>1 or more octets (ANI phone number, ASCII string)</td>
</tr>
<tr>
<td>Acct-Status-Type</td>
<td>40</td>
<td>4 octets (hex ASCII number)</td>
</tr>
<tr>
<td>Service-Type</td>
<td>6</td>
<td>4 octets (hex ASCII number)</td>
</tr>
<tr>
<td>Acct-Session-Id</td>
<td>44</td>
<td>&quot;overloaded&quot; for CDR/ASCII string up to 256 bytes</td>
</tr>
<tr>
<td>Acct-Input-Octets</td>
<td>42</td>
<td>4 octets stop records only (hex number)</td>
</tr>
<tr>
<td>Acct-Output-Octets</td>
<td>43</td>
<td>4 octets stop records only (hex number)</td>
</tr>
<tr>
<td>Acct-Input-Packets</td>
<td>47</td>
<td>4 octets stop records only (hex number)</td>
</tr>
<tr>
<td>Acct-Output-Packets</td>
<td>48</td>
<td>4 octets stop records only (hex number)</td>
</tr>
<tr>
<td>Acct-Session-Time</td>
<td>46</td>
<td>4 octets (hex number rep. Seconds) stop records only</td>
</tr>
<tr>
<td>Acct-Delay-Time</td>
<td>41</td>
<td>4 octets (hex number in seconds)</td>
</tr>
</tbody>
</table>
Overloaded *Acct-Session-Id* Field Descriptions:

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>session id</td>
<td>The standard (RFC 2139) RADIUS <em>account-session-id</em></td>
</tr>
<tr>
<td>call leg setup time</td>
<td>The Q.931 setup time for this connection in NTP format.</td>
</tr>
<tr>
<td>gateway id</td>
<td>The name of the underlying gateway. Name string is of form “gateway.domain_name”</td>
</tr>
<tr>
<td>connection id</td>
<td>A unique global identifier used to correlate call legs that belong to the same end-to-end call. The field consists of 4 long words (128 bits). Each long word is displayed in hexadecimal value and separated by a space character.</td>
</tr>
<tr>
<td>call origin</td>
<td>Indicates origin of the call relative to the gateway. Possible values are “originate” and “answer”.</td>
</tr>
<tr>
<td>call type</td>
<td>Indicates call leg type. Possible values are: “Telephony” and “VoIP.”</td>
</tr>
<tr>
<td>connect time</td>
<td>The Q.931 connect time for this call leg in NTP format. (stop only)</td>
</tr>
<tr>
<td>disconnect time</td>
<td>The Q.931 disconnect time for this call leg in NTP format. (stop only)</td>
</tr>
<tr>
<td>disconnect cause</td>
<td>Documented in Q.931 specification. Can be in the range of 1-160. (stop only)</td>
</tr>
<tr>
<td>remote IP address</td>
<td>IP address of the remote gateway used in this connection (stop only)</td>
</tr>
</tbody>
</table>

**Security**

- **Subset of H.235**
  - Cisco GWs to Gks
  - Cisco Access Token, using H.235 ClearTokens
  - Subscription-based, with hashing (H.235, section 10.3.3)
  - RADIUS used authentication

- **Registration security**
  - GW authentication
  - Authentication on registration (RRQ), as well as per call (ARQ)
  - Included on full registrations only (not lightweight)
  - Uses GW configuration (passwd and H.323 alias)

- **Admission security**
  - Use authentication
  - IVR: User ID and PIN
H.235 Security—Registration

- RRQ
- RCF/RRJ
- ARQ
- ACF
- Access Request
- Access Accept/Reject
- SETUP
- ARQ
- Access Request
- Access Accept
- ACF
- ALERTING and CONNECT

H.235 Security—Admission

- IVR
- ARQ (User)
- ACF
- Access Req (User)
- Access Accept
- SETUP (GW)
- ARQ (GW)
- Access Req (GW)
- Access Accept
- ACF
- ALERTING and CONNECT
Implementation Strategy

- Start with infrastructure example—VPN with overlapping dial plan

Adding Unified Communications
Internet Call Waiting

Agenda

- Introduction
- Technology Enablers
- Implementation Strategy
- Future Developments
  - H.323 Version 3
  - Session Initiation Protocol
  - Media Gateway Control Protocol
H.323 Version 3

- Approved 9/99
- Connection maintain and re-use
- Annex E—multiplexing
- Annex G—interdomain operation
- Supplementary features
  - H.450.4 (call hold), H.450.5 (park,pickup),
  - H.450.6 (call waiting), H.450.7 (message waiting)

SIP

- SIP defined by IETF MMUSIC working group as RFC 2543
- Defines transactions between clients and servers for setting up multimedia connections between two or more parties
- Uses URL style addresses and syntax
- MIME definition for multimedia (SDP)
- Simple extensible protocol
  - Methods—define transaction
  - Headers—describe transaction
  - Body—SDP
SIP Call Flow with Proxy

SIP Proxy

Register
OK (200)
Invite
Trying (100)
Ringing (180)
OK (200)
ACK

Register
OK (200)
Invite
Ringing (180)
OK (200)
ACK

Session Established

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SIP Call Flow with Redirect

SIP Redirect Server

Invite
Moved (302)
ACK
Invite
Ringing (180)
OK (200)
ACK

Session Established

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SIP Example: Call Forward Busy

SIP Proxy Server

EP-A

Invite

Ringing (180)

OK (200)

Trying (100)

EP-B

Register

Invite

Busy (486)

ACK

Invite

Ringing (180)

OK (200)

ACK

Session Established

EP-C

SIP Example: Call Transfer

EP-A

Bye (also C)

OK (200)

Bye (also C)

OK (200)

Invite (req A)

Invite (req A)

Trying (100)

Ringing (180)

Ringing (180)

OK (200)

OK (200)

ACK

ACK

Session Established
Media Gateway Control

- Allows remote control of various devices
- Create, modify, and delete connections; Generates and detect events (tones); Tracks resource states
- Fits in well with multimedia call signaling (i.e. H.323 and SIP)
- Strong support for existing telephone networks (SS7)

---

GC Protocol Architecture

- SS7 Network
  - SCTP
  - Call Agent: MGCP, H.248
  - H.323, SIP, ISUP
- PSTN Network
  - Media
  - Call Agent: MGCP, H.248
  - SCTP
- SS7 Network
Summary

- Market for VoIP Enhanced Services is developing rapidly
- H.323 has large installed base and is maturing
- SIP/MGCP are very promising and will require interworking

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