
NETWORKERS
Cisco Systems


Advanced Voice over IP
Enhanced Services
Session 2105
Cisco Systems

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

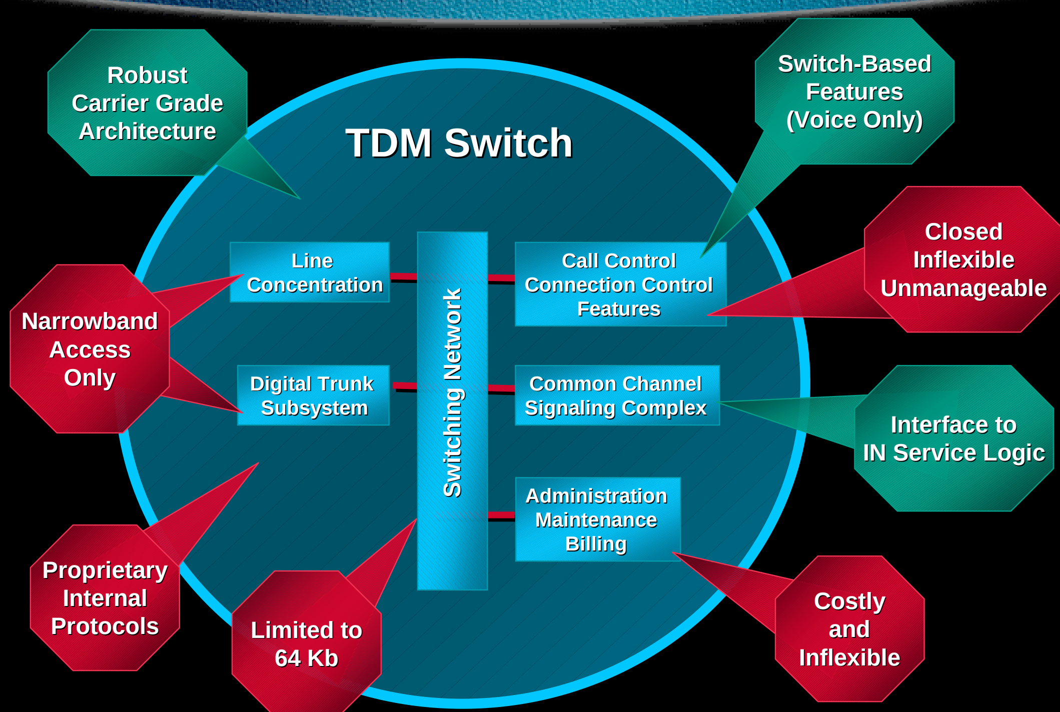
Agenda

- Introduction
- Technology Enablers
- Implementation Strategy
- Future Developments

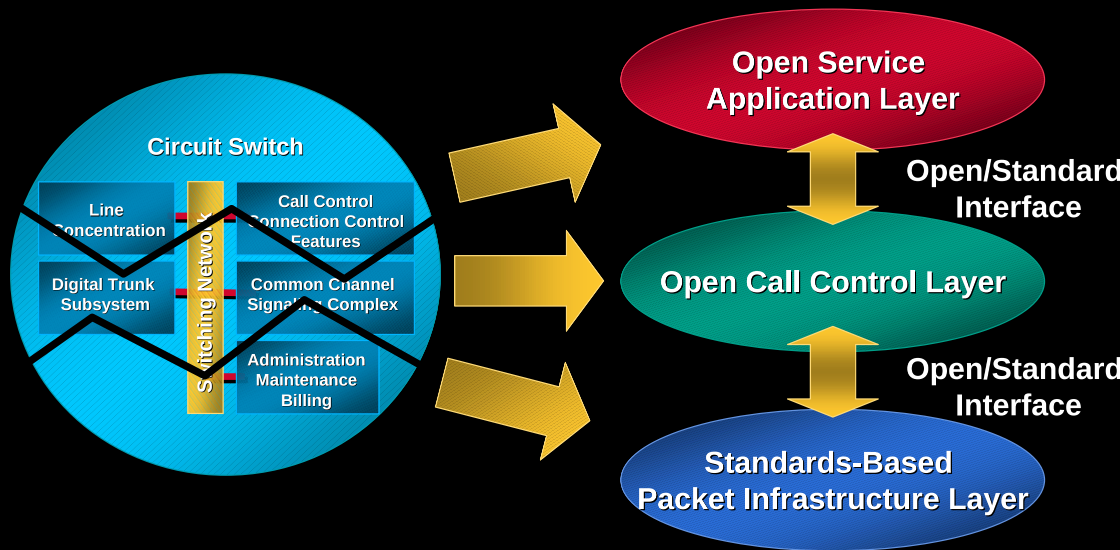
Agenda

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

Consider a Circuit Switch

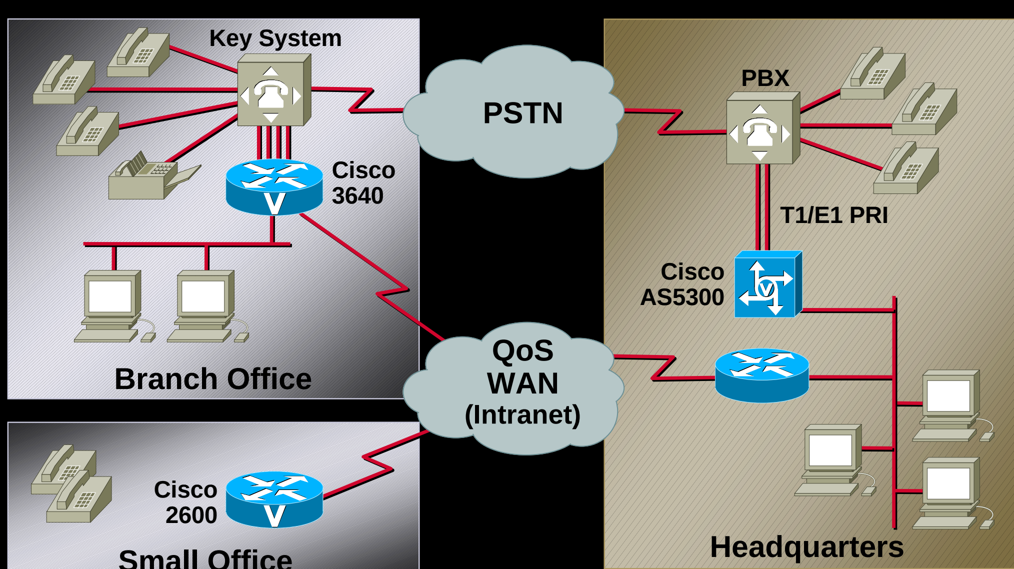


Open Packet Telephony Architecture

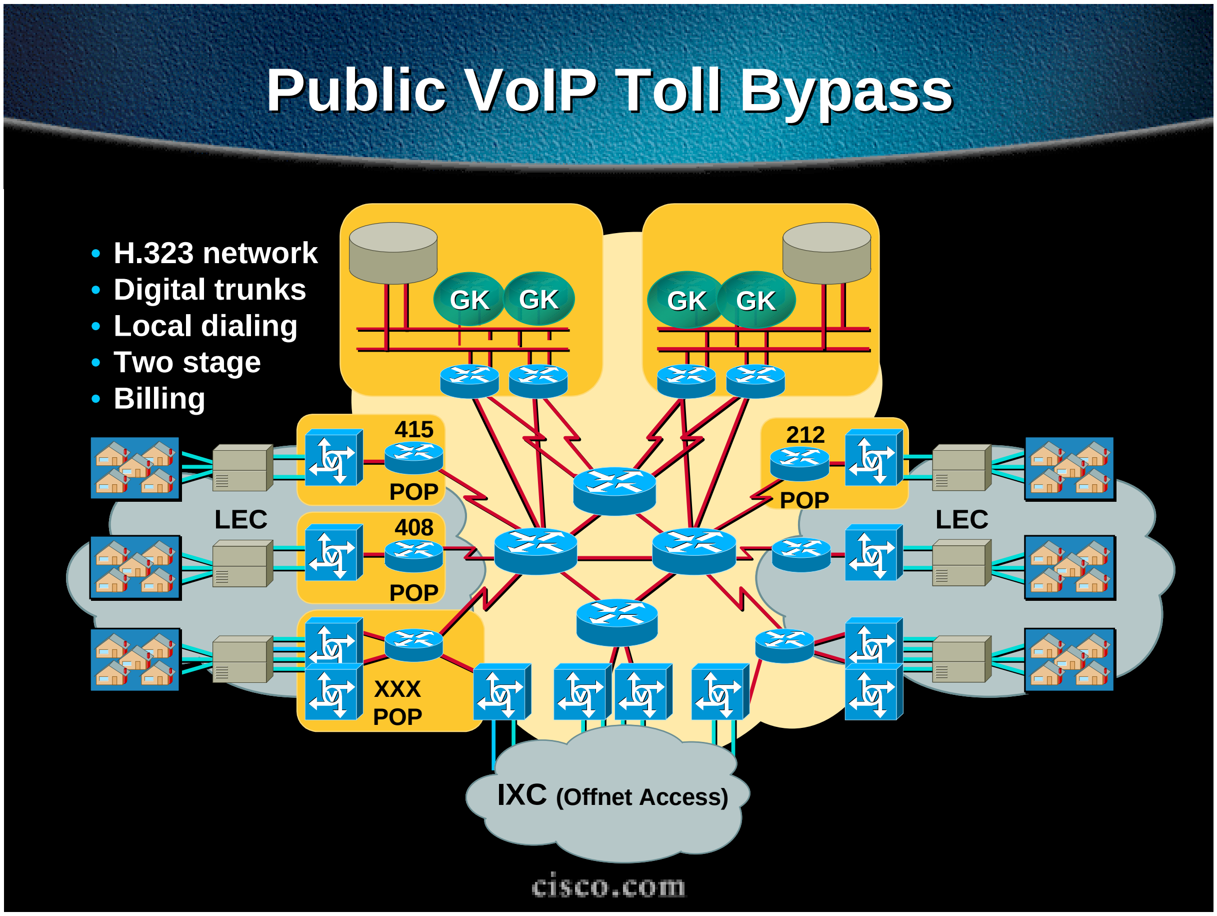


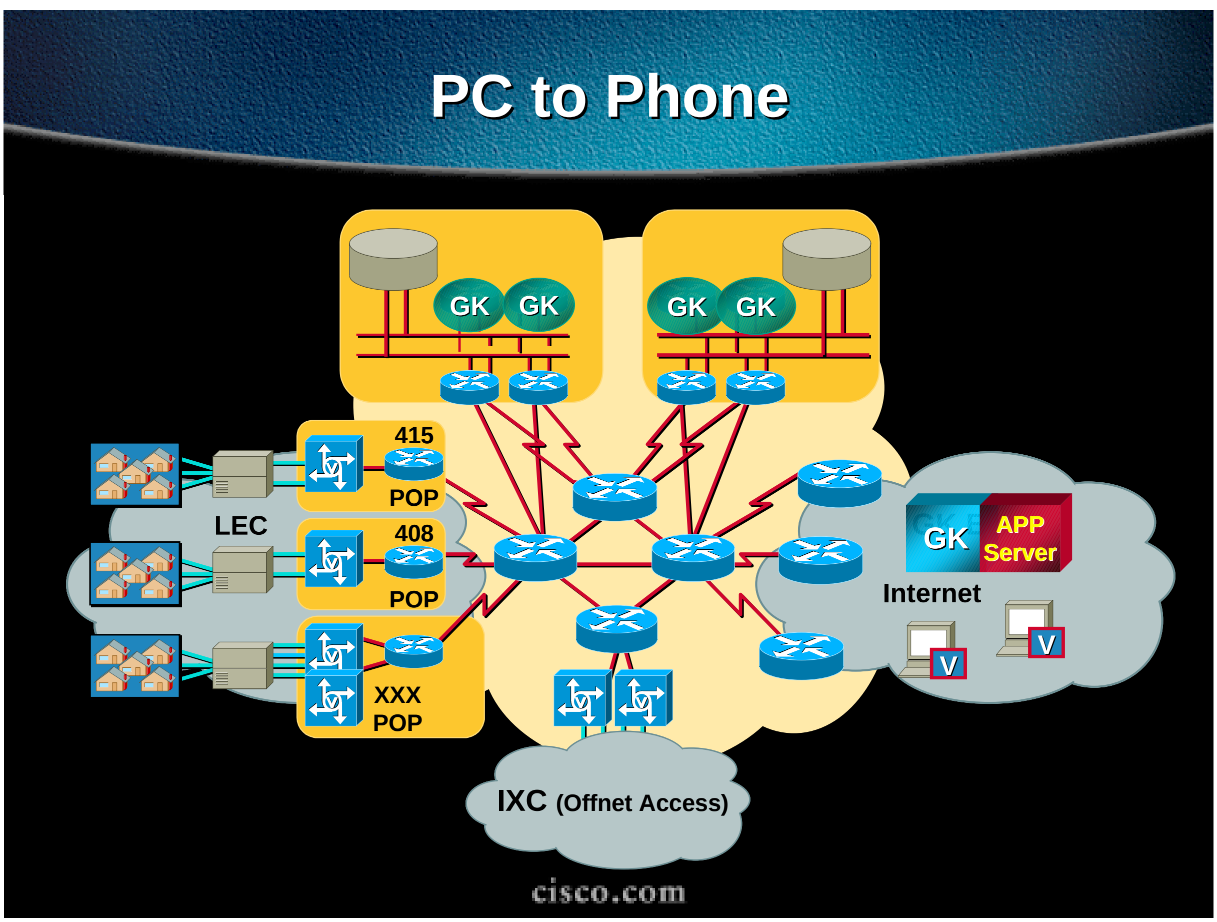
Open and Standardize
the Telephony Infrastructure

Private VoIP Toll Bypass



Enterprise Toll Bypass

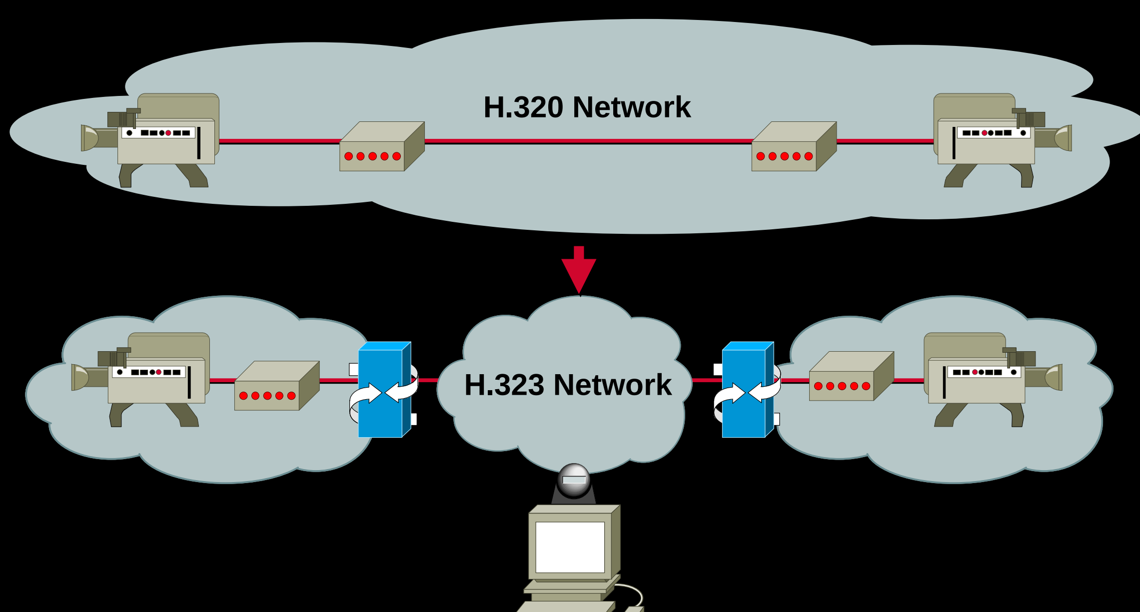

Public VoIP Toll Bypass
• H.323 network
• Digital trunks
• Local dialing
• Two stage
• Billing
GK
GK
GK
GK
415
POP
408
POP
XXX
POP
212
POP
LEC
LEC
IXC (Offnet Access)
cisco.com


PC to Phone
GK
GK
GK
GK
415
POP
408
POP
XXX
POP
LEC
GK
APP
Server
Internet
V
V
IXC (Offnet Access)
cisco.com

Agenda

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

H.323 Background

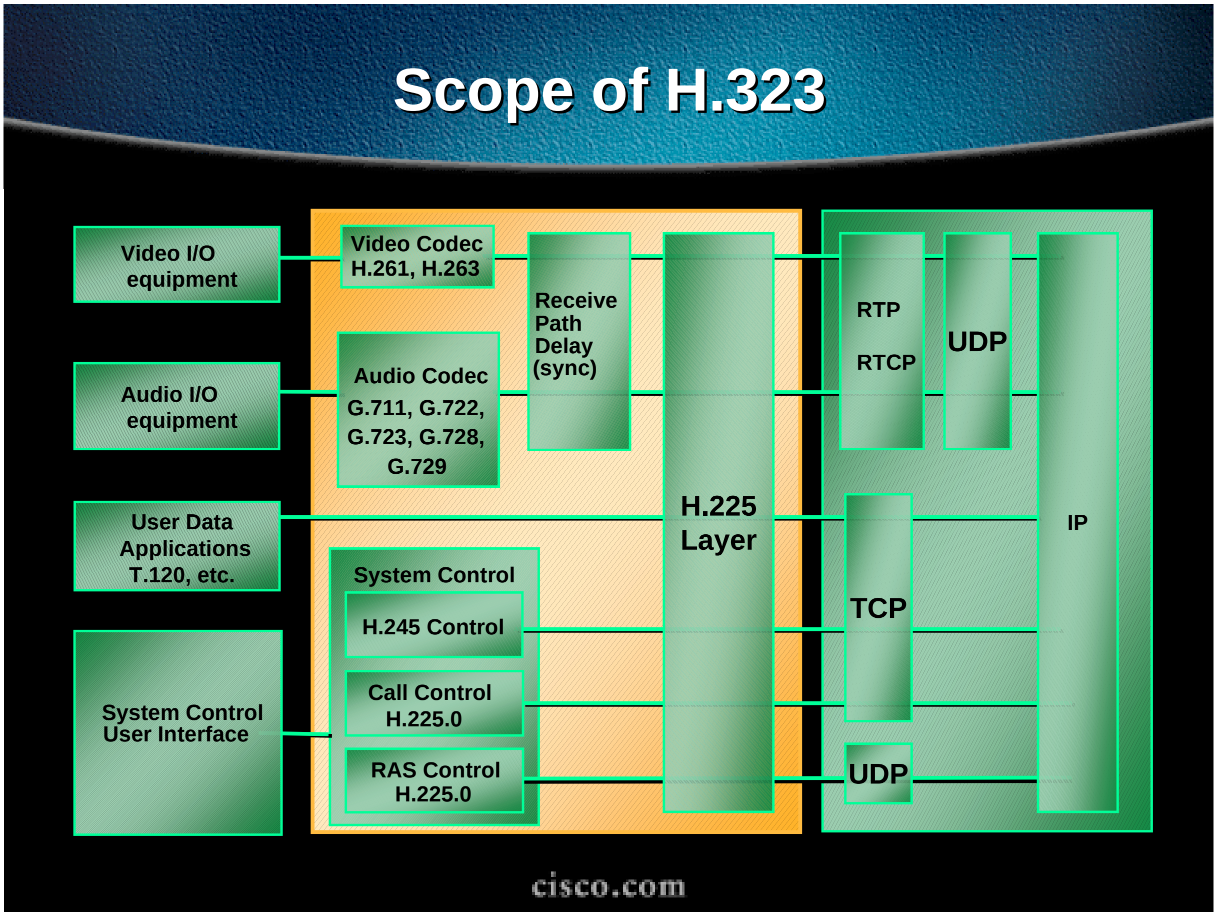


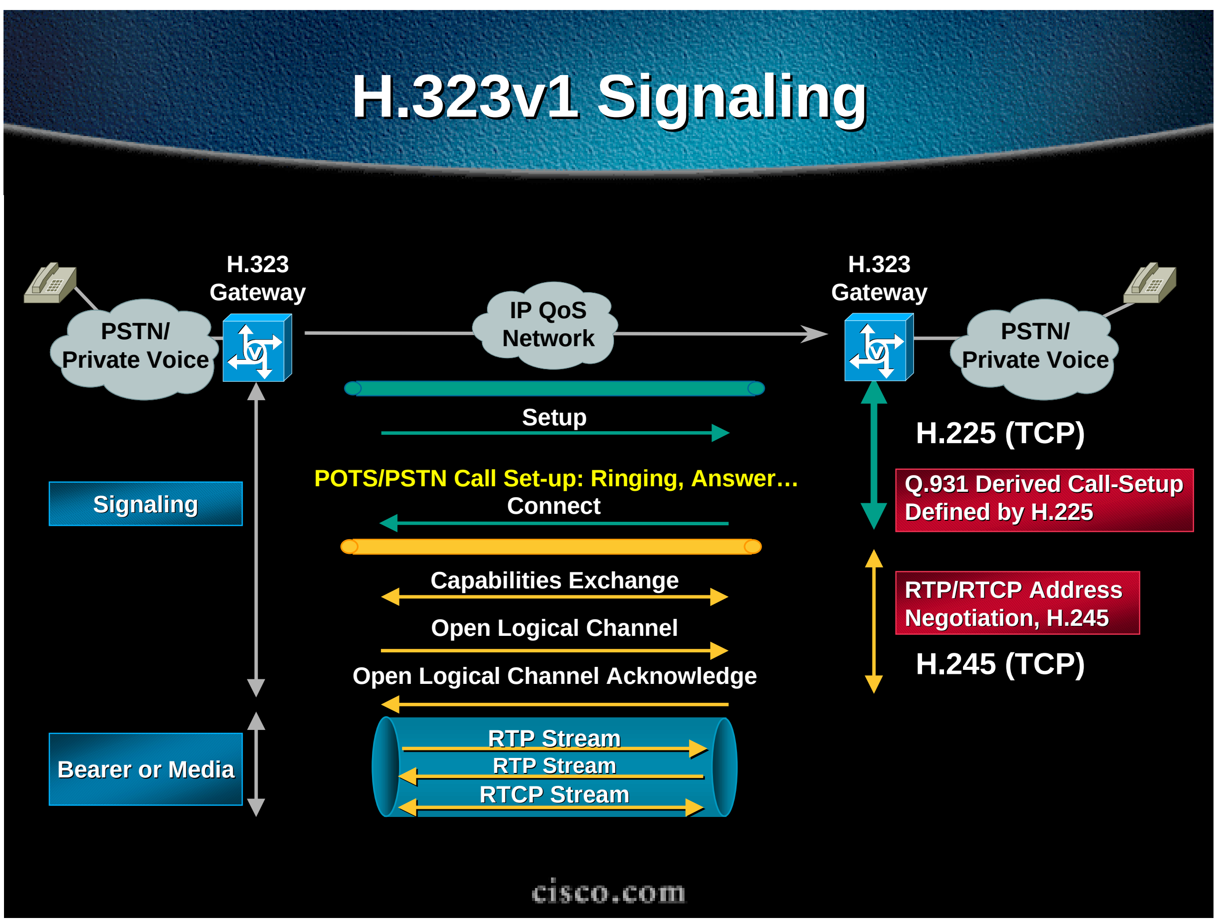
H.323 Background

- ITU recommendation
  - v1 approved in 1996, v2 in January 1998, v3 in September of 1999
- 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

Presentation	G.729(A)/G.723(.1)/G.71
Session	H.323
Transport	RTP/RTCP
Network	IP
Link	MLPPP/FR/ATM AAL1
Physical	---


Scope of H.323
Video I/O equipment
Audio I/O equipment
User Data Applications T.120, etc.
System Control User Interface
Video Codec H.261, H.263
Audio Codec G.711, G.722, G.723, G.728, G.729
Receive Path Delay (sync)
System Control
H.245 Control
Call Control H.225.0
RAS Control H.225.0
H.225 Layer
RTP
RTCP
UDP
TCP
UDP
IP
cisco.com


H.323v1 Signaling
H.323 Gateway
PSTN/ Private Voice
IP QoS Network
H.323 Gateway
PSTN/ Private Voice
Setup
H.225 (TCP)
Signaling
POTS/PSTN Call Set-up: Ringing, Answer…
Connect
Q.931 Derived Call-Setup Defined by H.225
Capabilities Exchange
RTP/RTCP Address Negotiation, H.245
Open Logical Channel
Open Logical Channel Acknowledge
H.245 (TCP)
Bearer or Media
RTP Stream
RTP Stream
RTCP Stream
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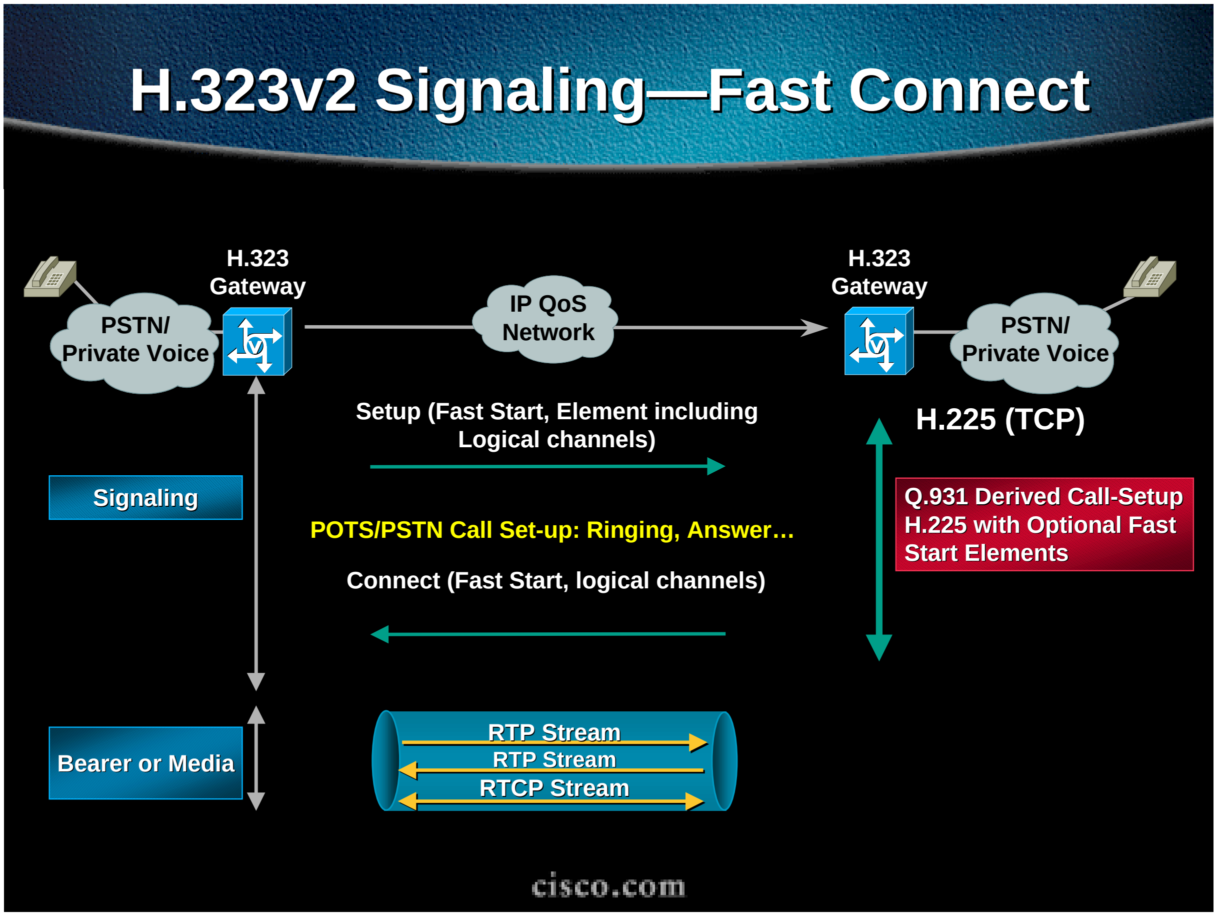
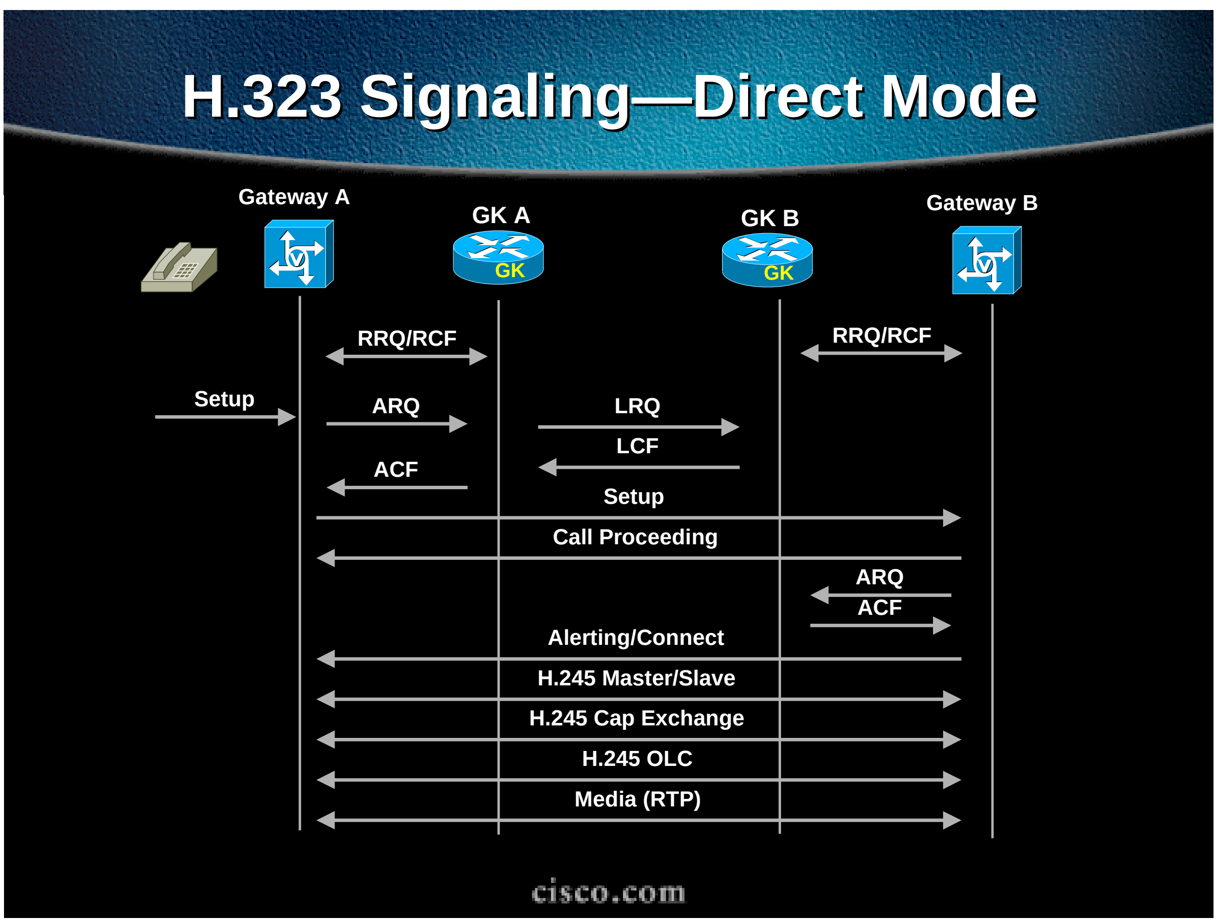
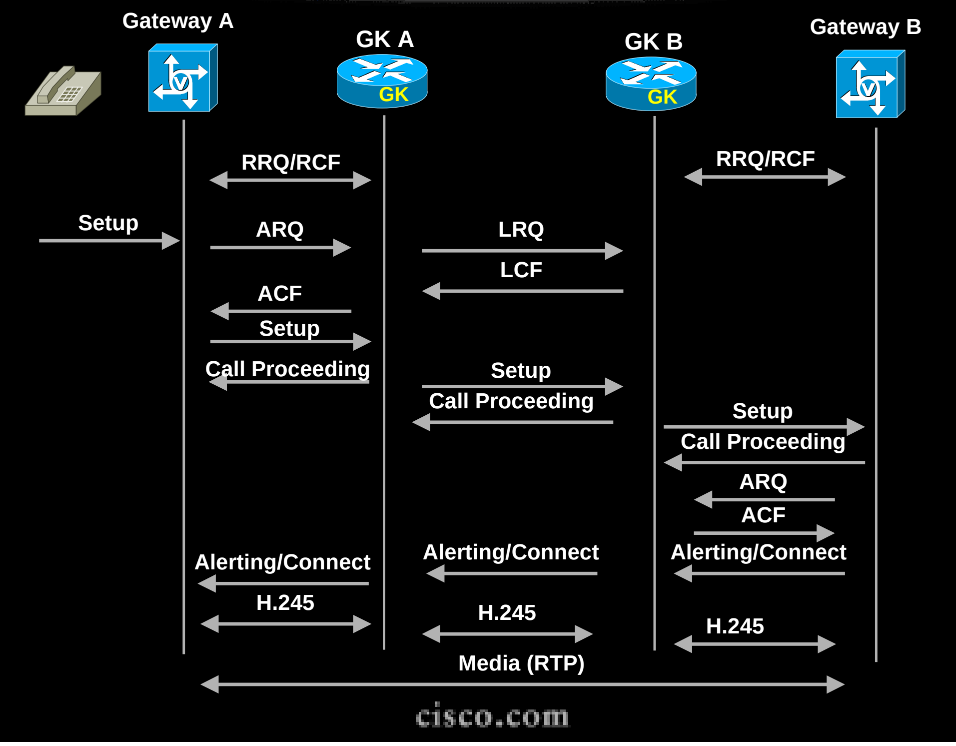
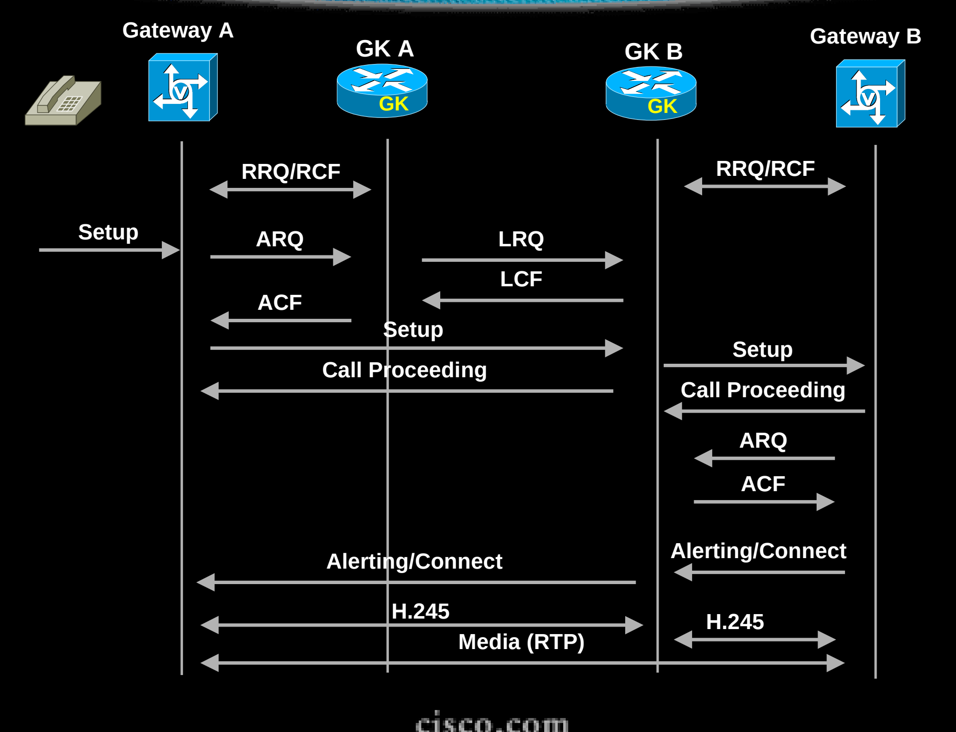

H.323v2 Signaling—Fast Connect
H.323
Gateway
PSTN/
Private Voice
IP QoS
Network
H.323
Gateway
PSTN/
Private Voice
H.225 (TCP)
Setup (Fast Start, Element including
Logical channels)
Signaling
POTS/PSTN Call Set-up: Ringing, Answer…
Connect (Fast Start, logical channels)
Q.931 Derived Call-Setup
H.225 with Optional Fast
Start Elements
Bearer or Media
RTP Stream
RTP Stream
RTCP Stream
cisco.com


H.323 Signaling—Direct Mode
Gateway A
GK A
GK
GK B
GK
Gateway B
RRQ/RCF
RRQ/RCF
Setup
ARQ
LRQ
LCF
ACF
Setup
Call Proceeding
ARQ
ACF
Alerting/Connect
H.245 Master/Slave
H.245 Cap Exchange
H.245 OLC
Media (RTP)
cisco.com

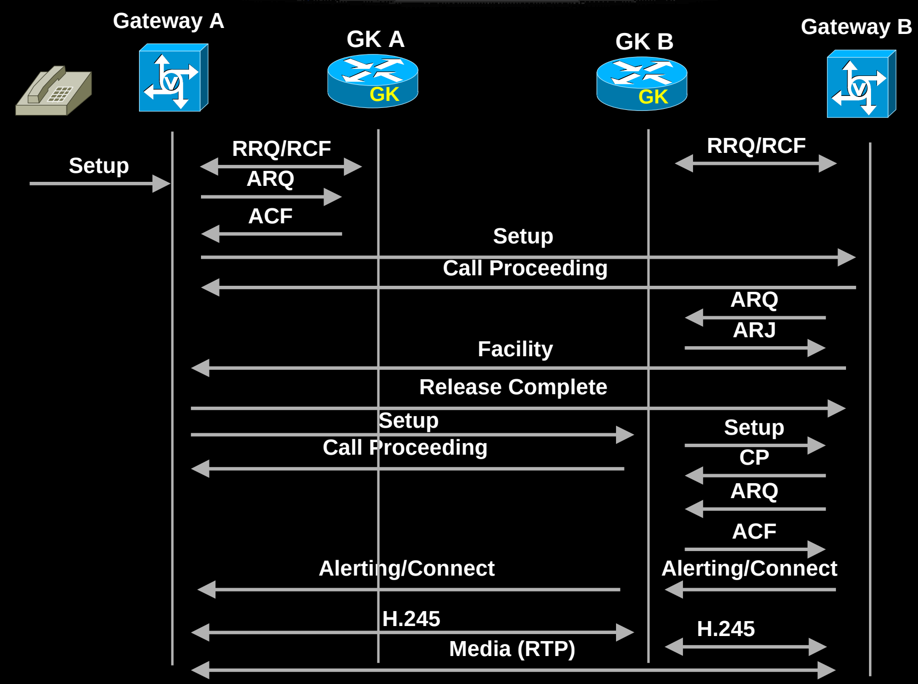
H.323 Signaling—Gatekeeper Routed



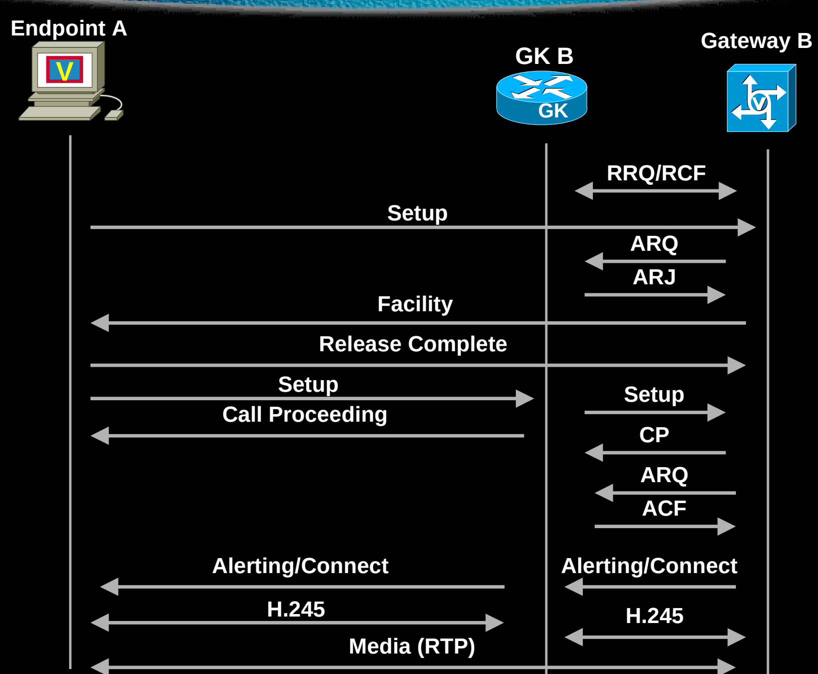
H.323 Signaling—Direct/Routed

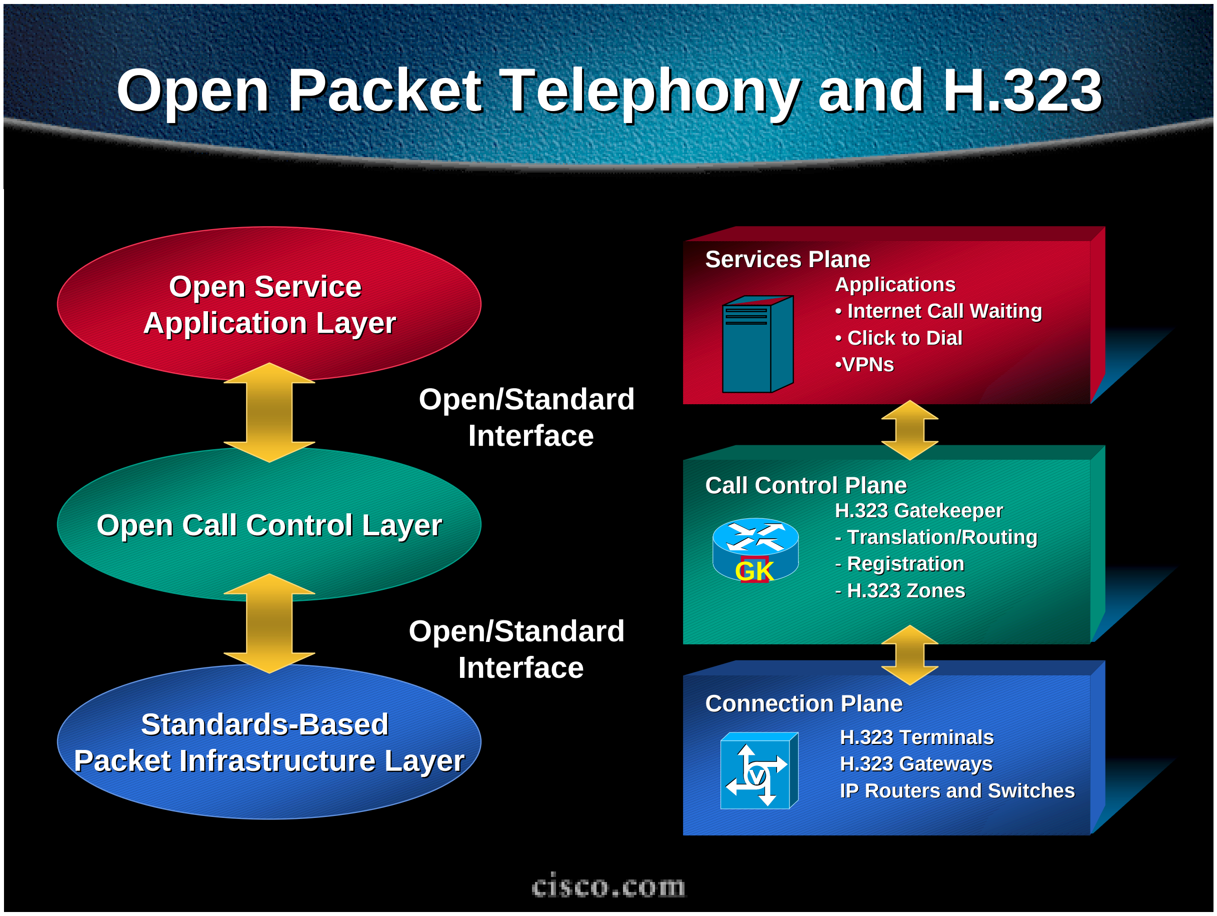


H.323 Signaling—Direct/ Routed



H.323 Signaling—Direct/Routed




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
GK
Connection Plane
H.323 Terminals
H.323 Gateways
IP Routers and Switches
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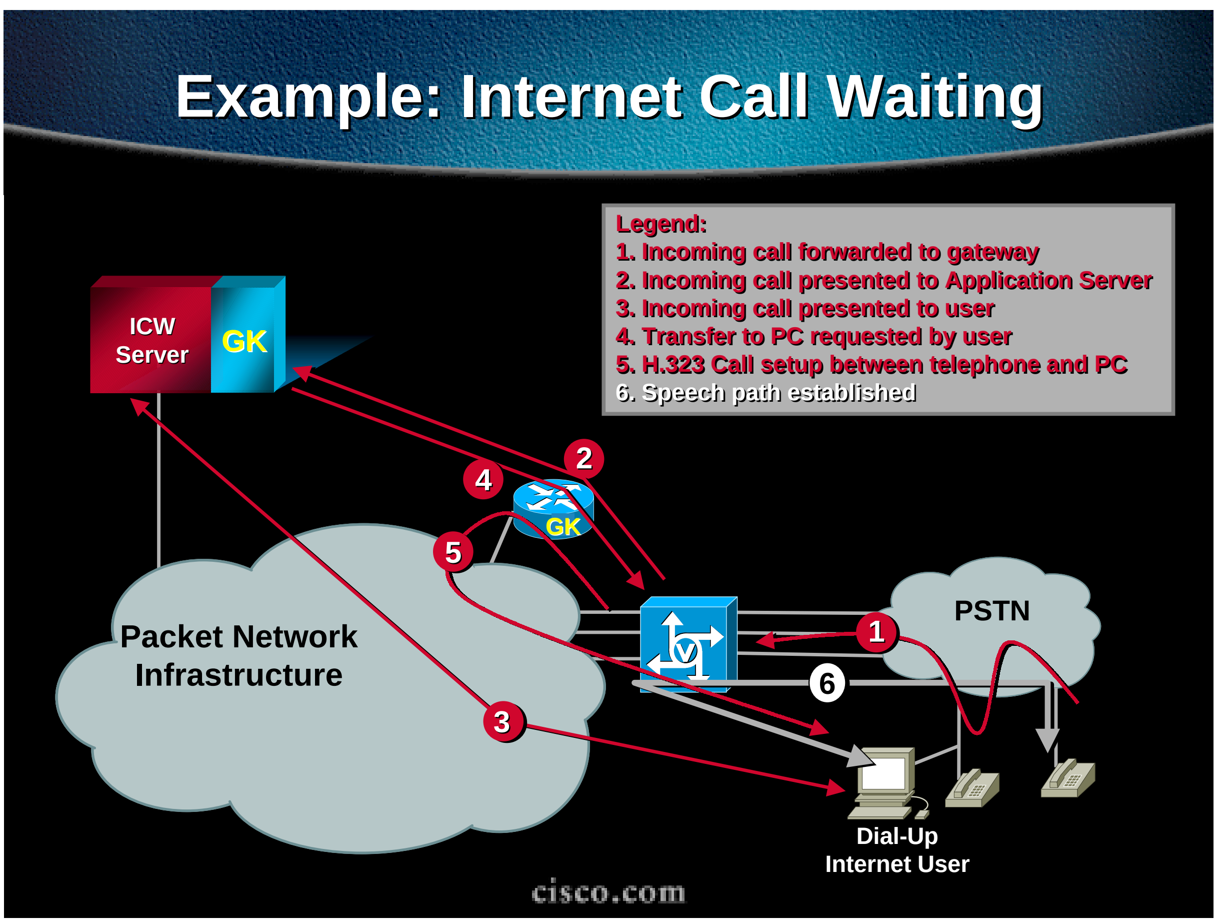
Example: Internet Call Waiting
Legend:
1. Incoming call forwarded to gateway
2. Incoming call presented to Application Server
3. Incoming call presented to user
4. Transfer to PC requested by user
5. H.323 Call setup between telephone and PC
6. Speech path established
ICW Server
GK
Packet Network Infrastructure
PSTN
Dial-Up Internet User
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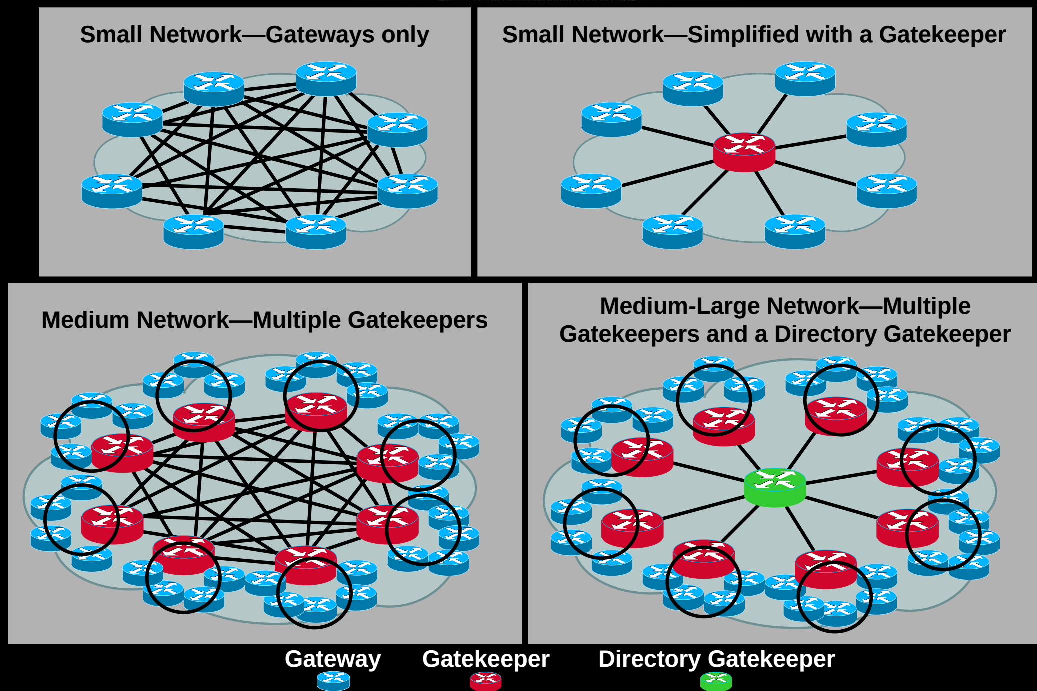
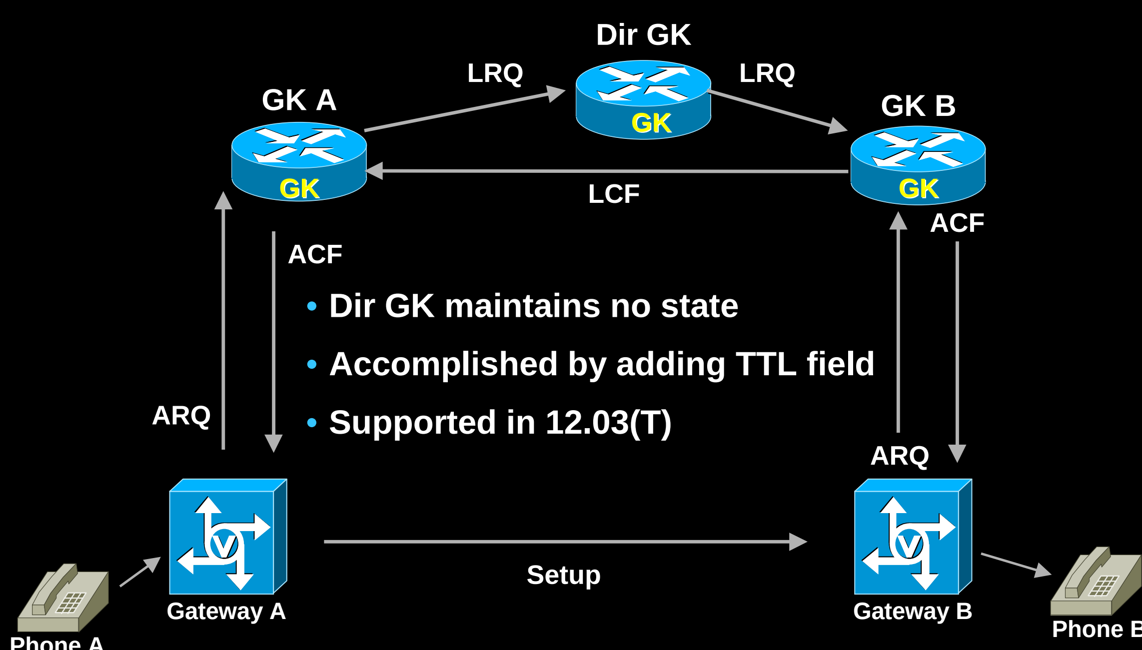
Directory Gatekeeper—Scaling

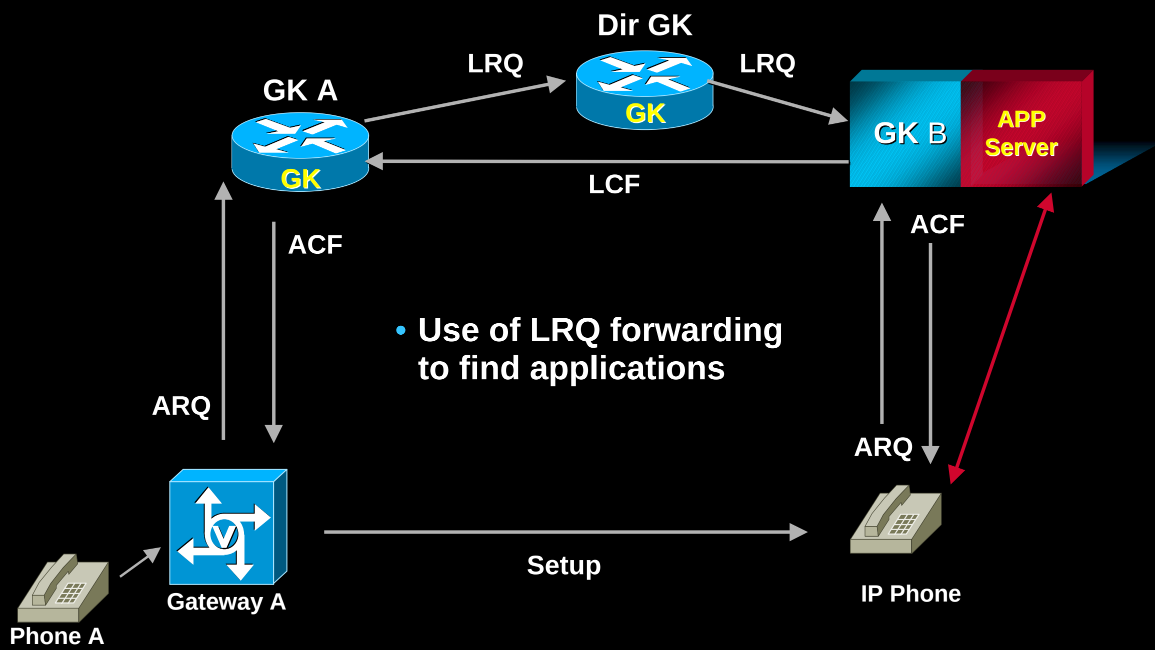


LRQ Forwarding



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

LRQ Forwarding



- Use of LRQ forwarding to find applications

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

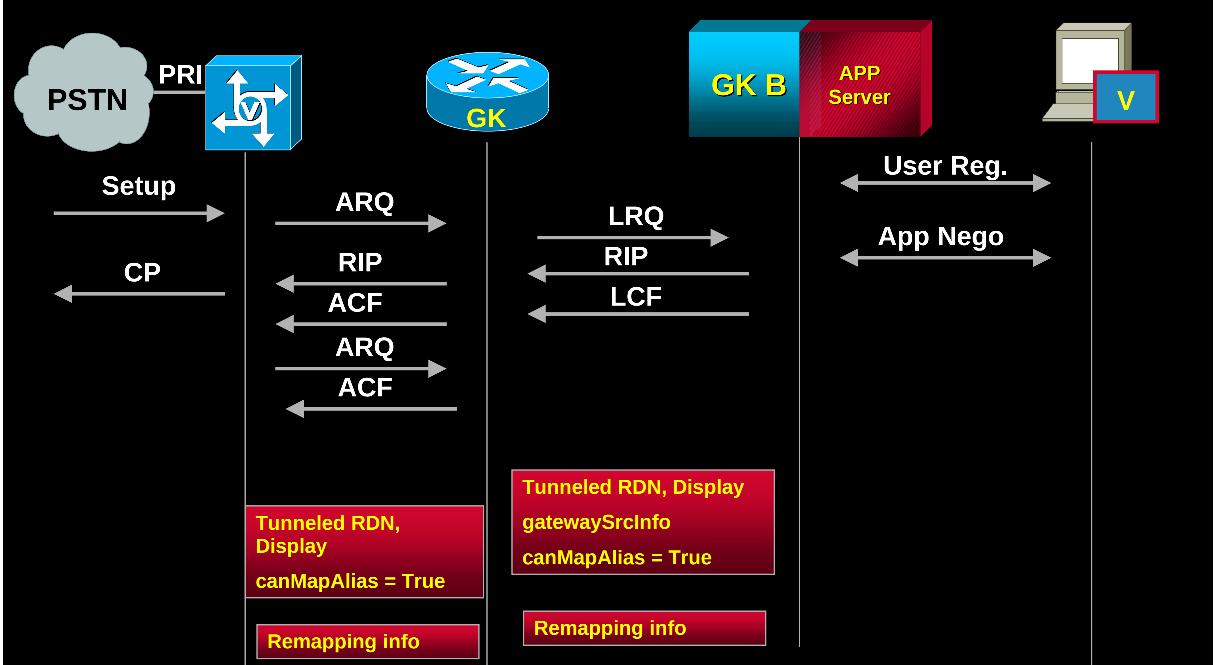
ARQ

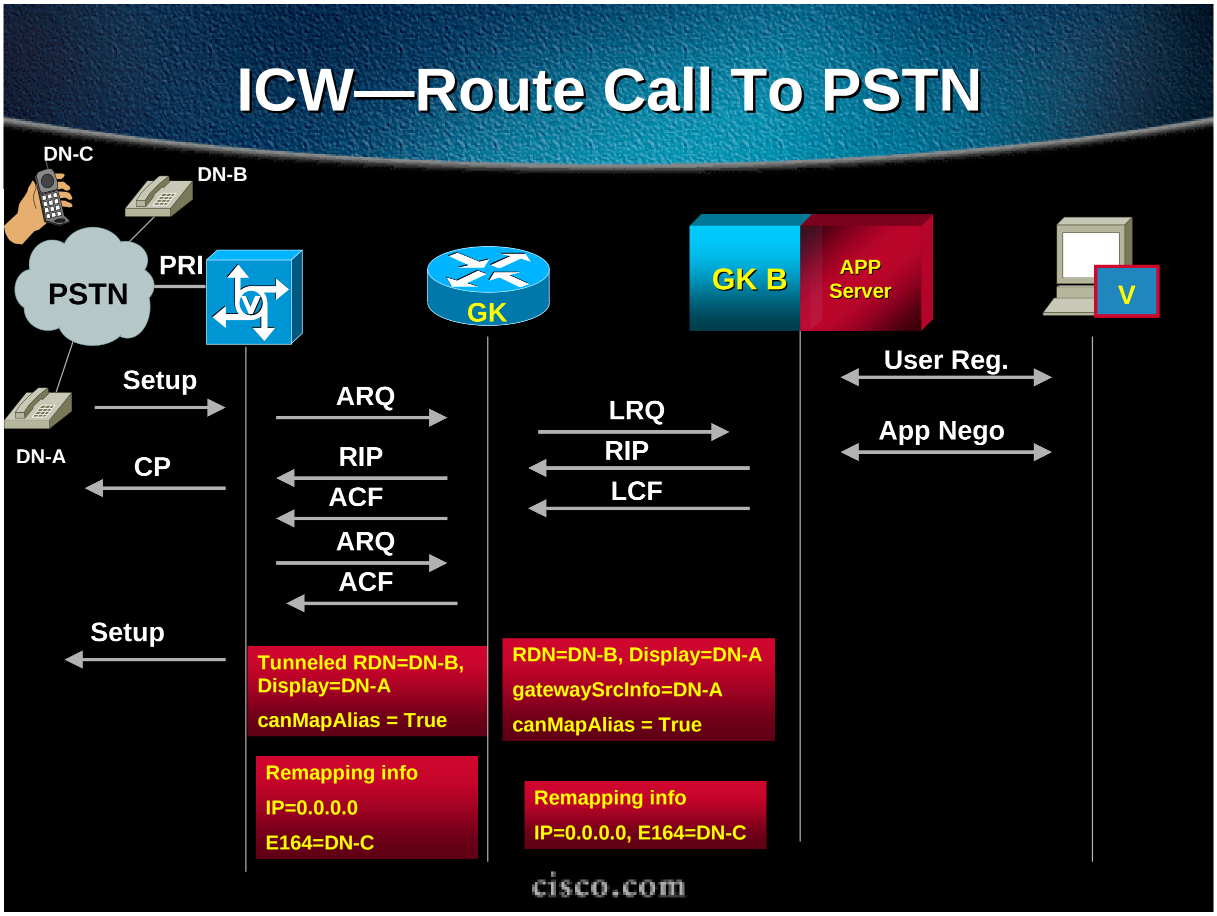
- redirectIEInfo
- callingOctet3a
- displayInformationElement
- interfaceSpecificBillingID
- interfaceDescription

LRQ

- redirectIEInfo
- callingOctet3a
- gatewaySrcInfo
- displayInformationElement

RAS Extensions




ICW—Route Call To PSTN
DN-C
DN-B
PSTN
PRI
GK
GK B
APP Server
V
DN-A
Setup
CP
ARQ
RIP
ACF
ARQ
ACF
Setup
LRQ
RIP
LCF
User Reg.
App Nego
Tunneled RDN=DN-B, Display=DN-A
canMapAlias = True
Remapping info
IP=0.0.0.0
E164=DN-C
RDN=DN-B, Display=DN-A
gatewaySrcInfo=DN-A
canMapAlias = True
Remapping info
IP=0.0.0.0, E164=DN-C
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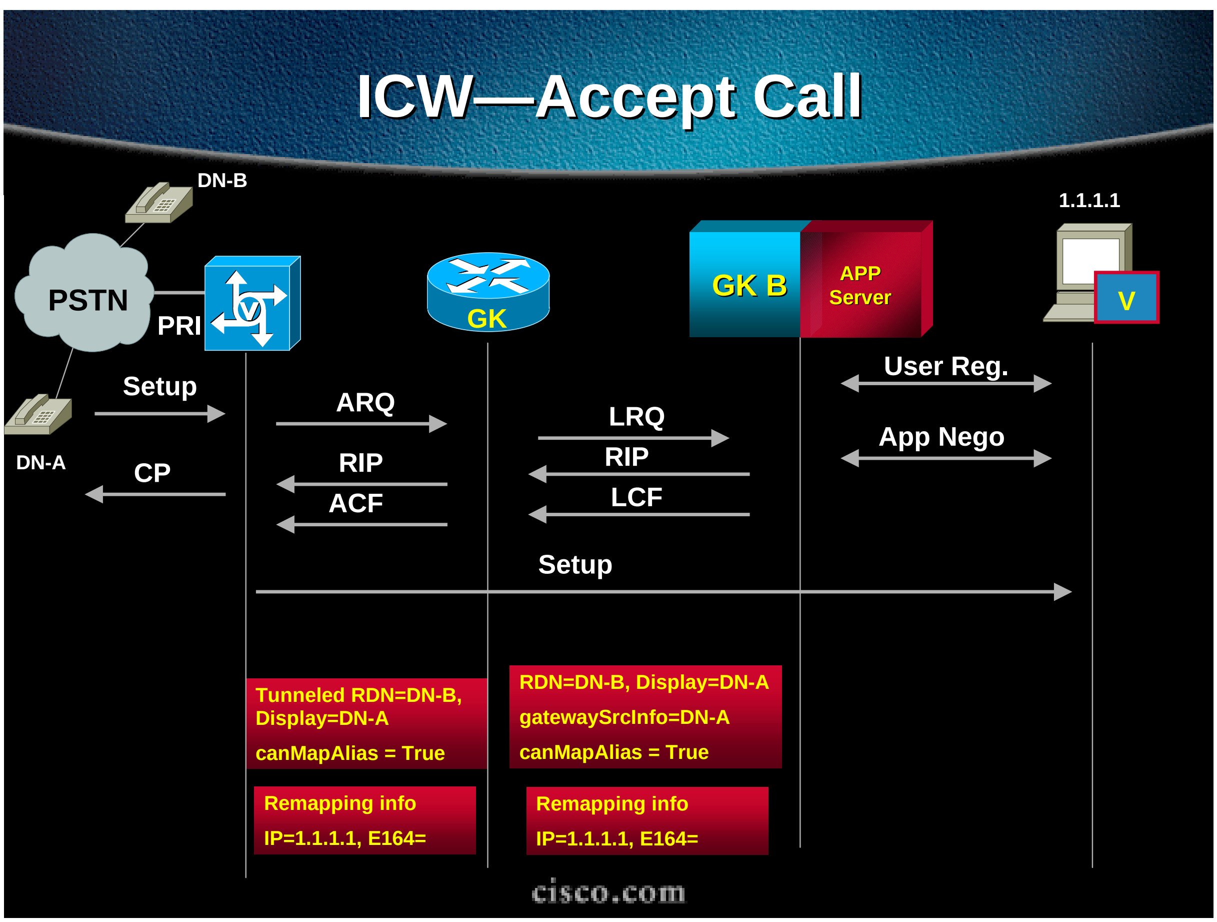
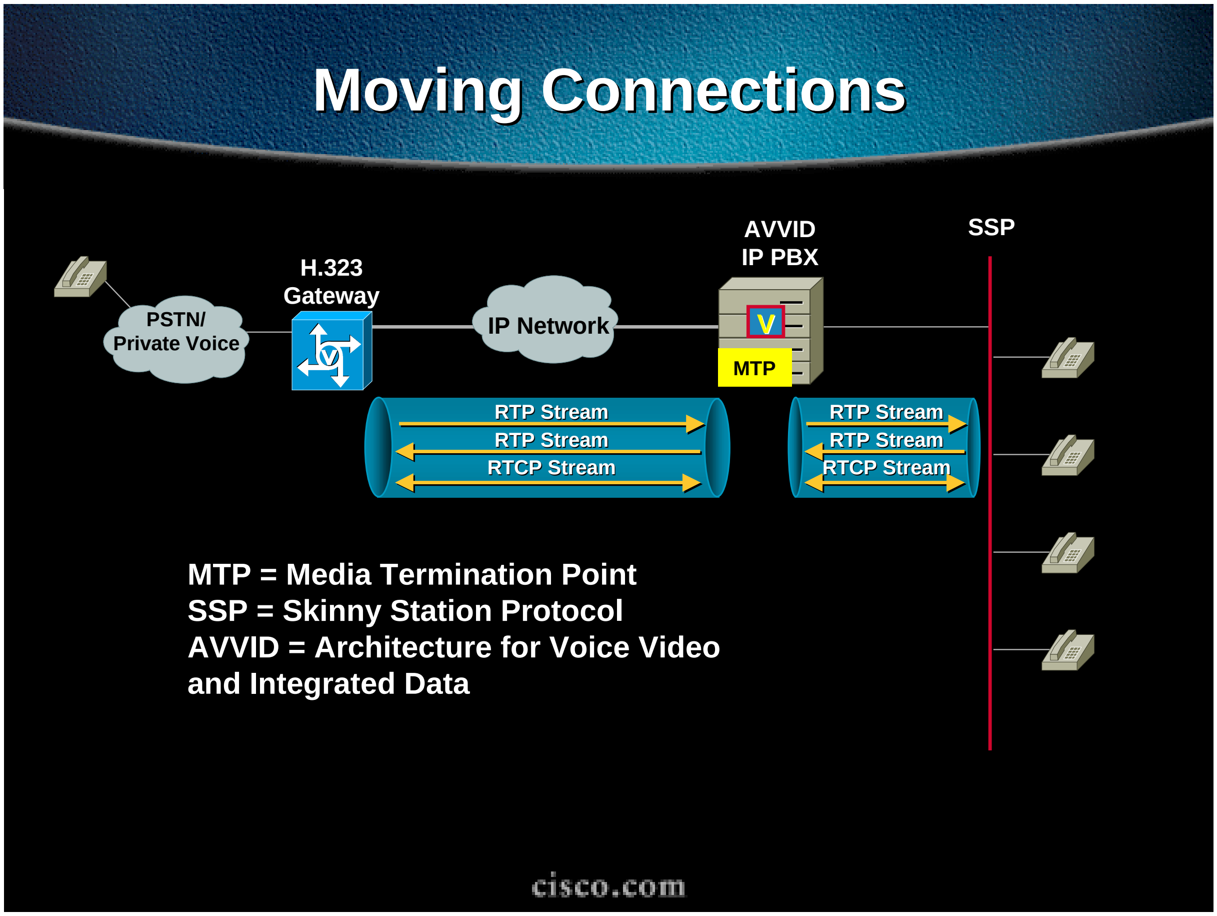
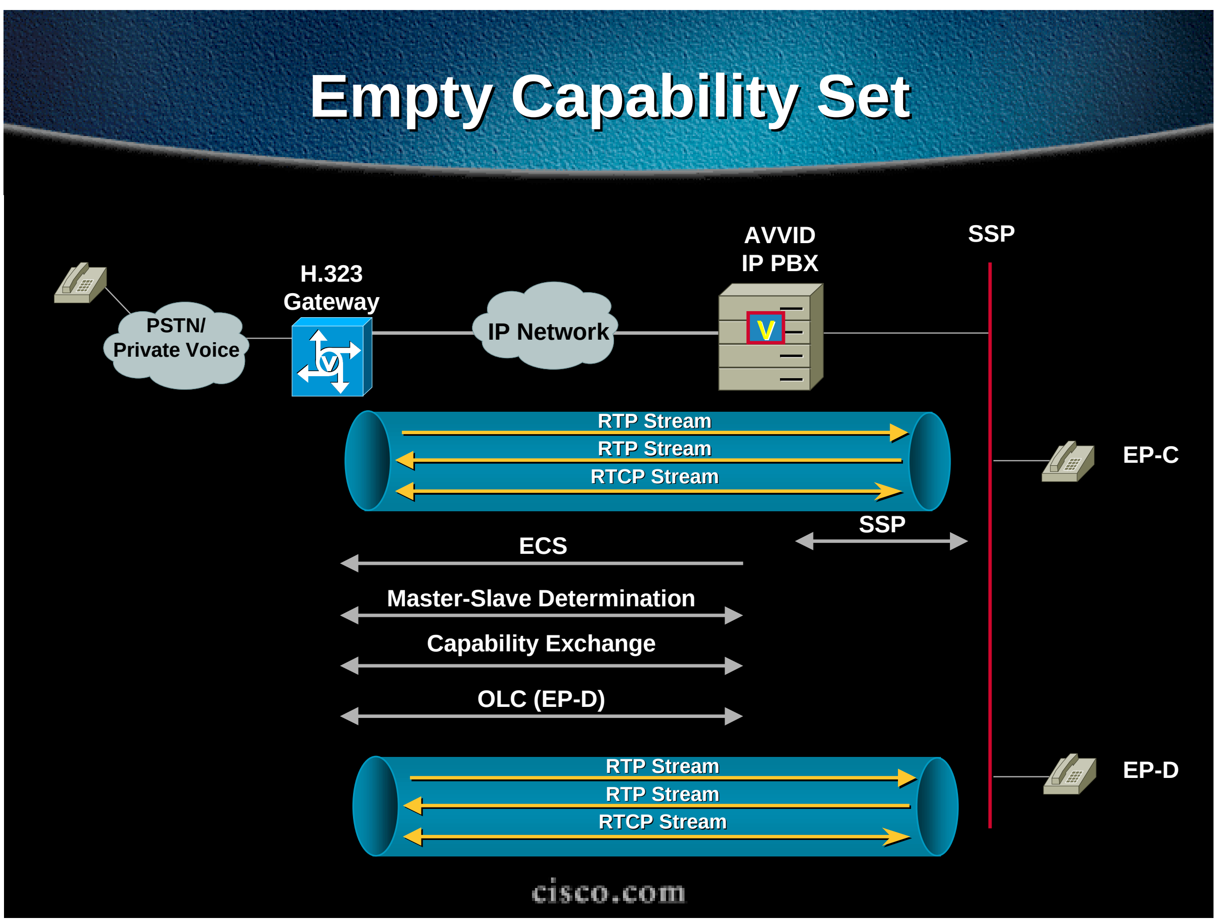
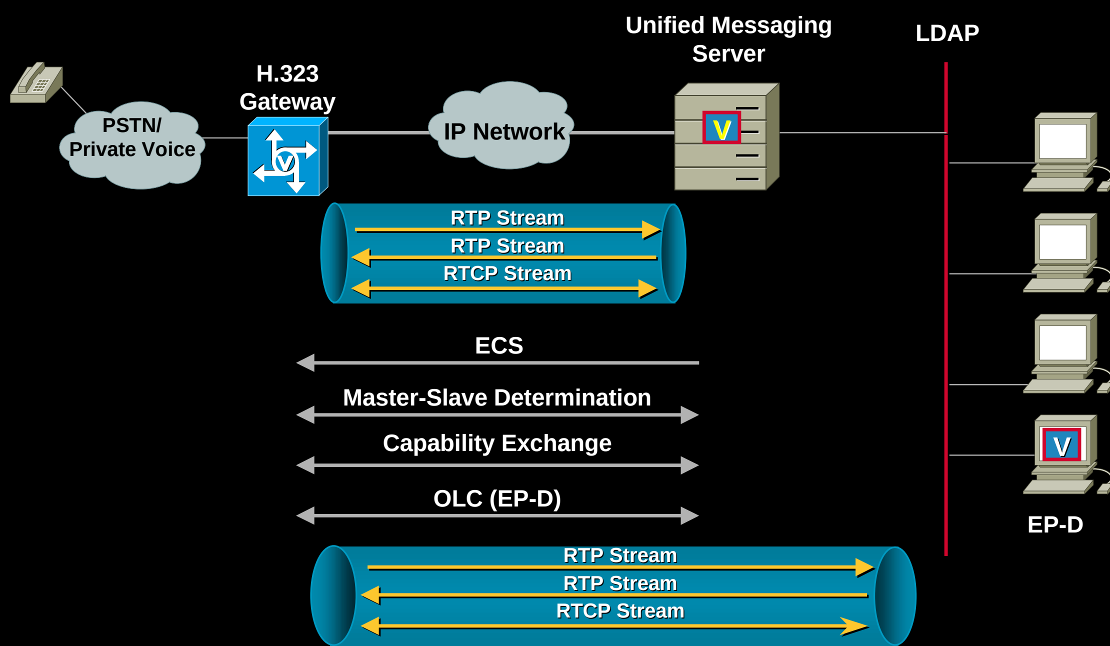

ICW—Accept Call
DN-B
1.1.1.1
PSTN
PRI
GK
GK B
APP Server
V
DN-A
Setup
CP
ARQ
RIP
ACF
LRQ
RIP
LCF
User Reg.
App Nego
Setup
Tunneled RDN=DN-B, Display=DN-A
canMapAlias = True
Remapping info
IP=1.1.1.1, E164=
RDN=DN-B, Display=DN-A
gatewaySrcInfo=DN-A
canMapAlias = True
Remapping info
IP=1.1.1.1, E164=
cisco.com


Moving Connections
AVVID
IP PBX
SSP
H.323
Gateway
PSTN/
Private Voice
IP Network
V
MTP
RTP Stream
RTP Stream
RTCP Stream
RTP Stream
RTP Stream
RTCP Stream
MTP = Media Termination Point
SSP = Skinny Station Protocol
AVVID = Architecture for Voice Video
and Integrated Data
cisco.com

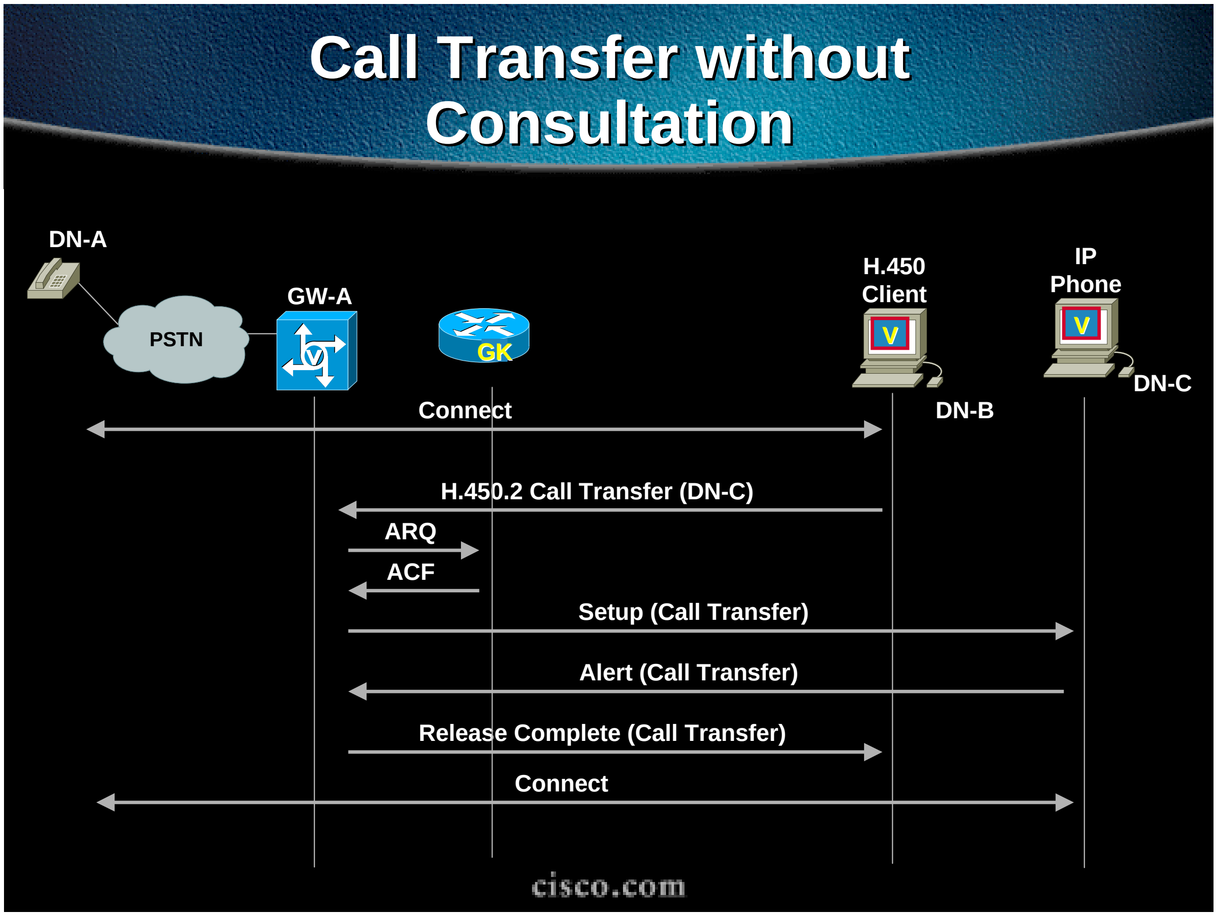

Empty Capability Set
AVVID
IP PBX
SSP
H.323
Gateway
PSTN/
Private Voice
IP Network
V
RTP Stream
RTP Stream
RTCP Stream
EP-C
SSP
ECS
Master-Slave Determination
Capability Exchange
OLC (EP-D)
RTP Stream
RTP Stream
RTCP Stream
EP-D
cisco.com

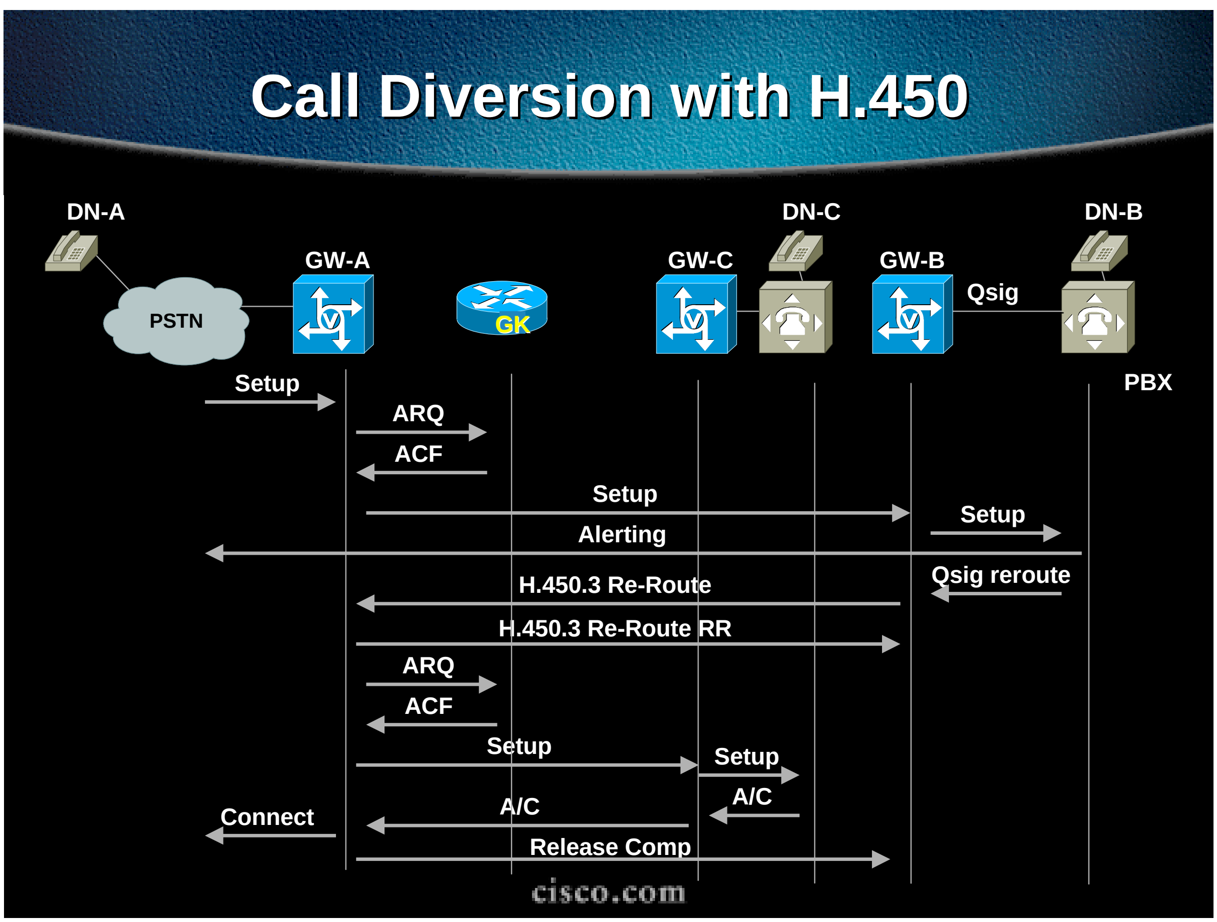
ECS with Unified Messaging



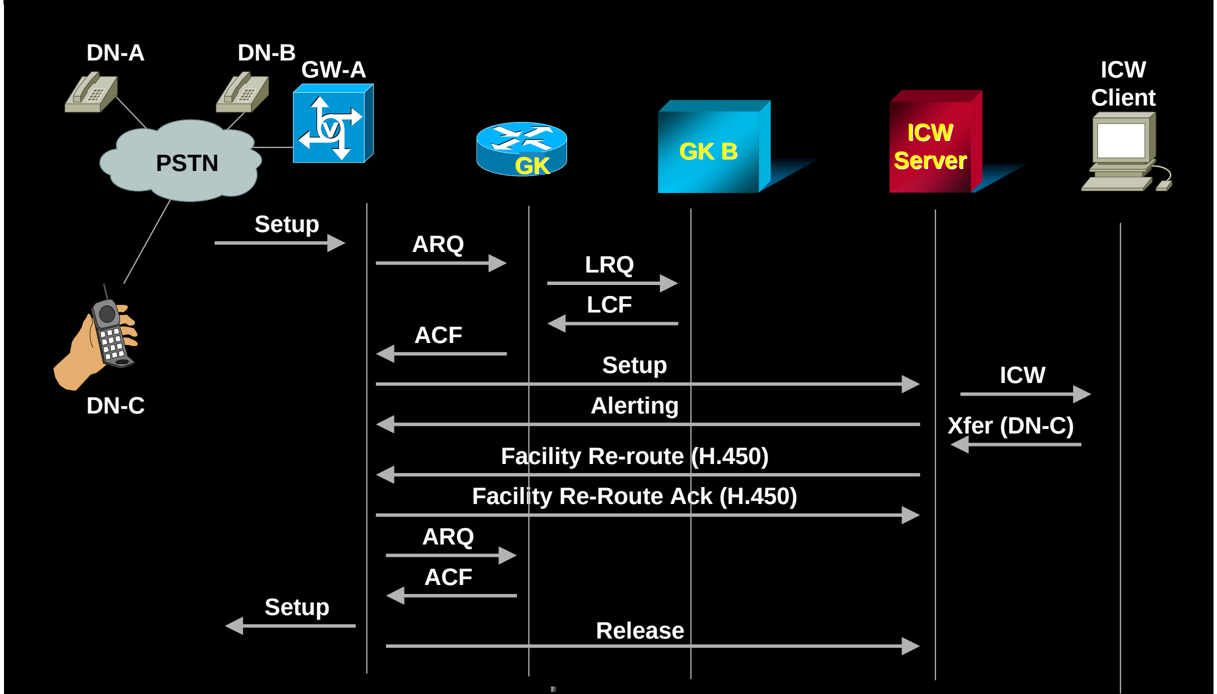
Supplementary Services

- 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

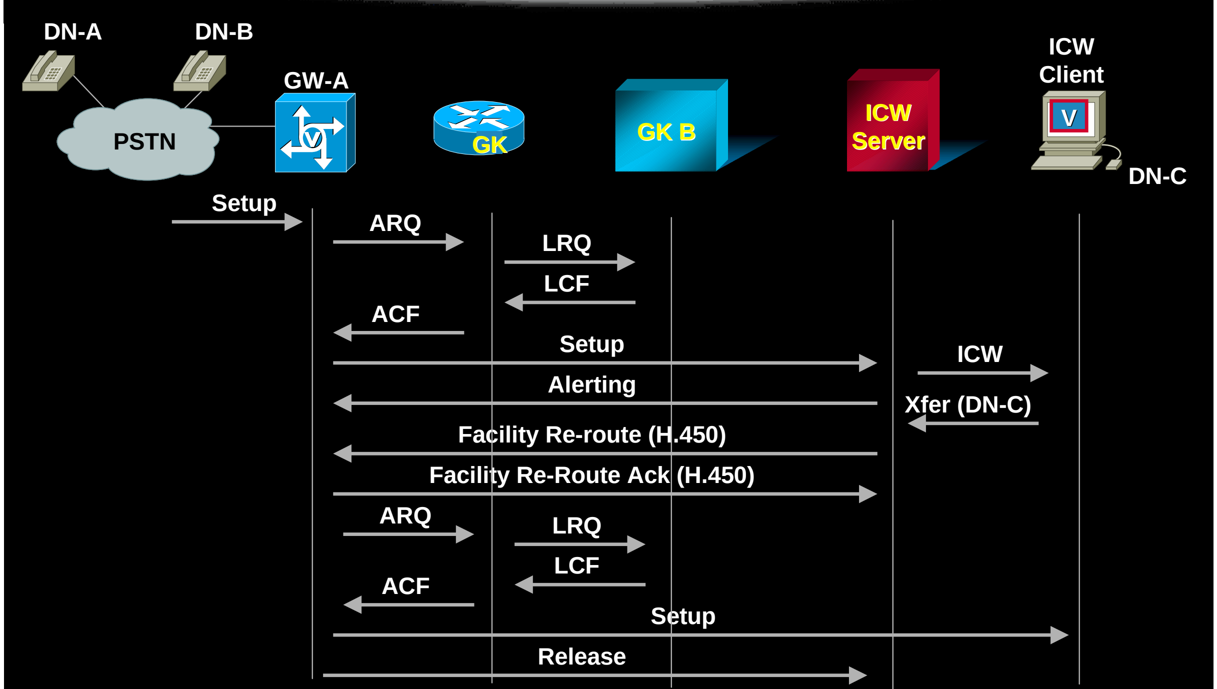

Call Transfer without Consultation
DN-A
PSTN
GW-A
GK
H.450 Client
IP Phone
DN-B
DN-C
Connect
H.450.2 Call Transfer (DN-C)
ARQ
ACF
Setup (Call Transfer)
Alert (Call Transfer)
Release Complete (Call Transfer)
Connect
cisco.com


Call Diversion with H.450
DN-A
PSTN
GW-A
GK
GW-C
DN-C
GW-B
Qsig
DN-B
PBX
Setup
ARQ
ACF
Setup
Setup
Alerting
H.450.3 Re-Route
Qsig reroute
H.450.3 Re-Route RR
ARQ
ACF
Setup
Setup
A/C
A/C
Connect
Release Comp
cisco.com

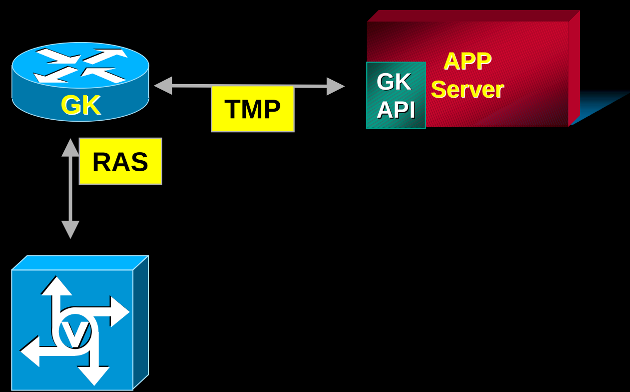
ICW with H.450.3 Example



ICW with H.450.3 Example



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

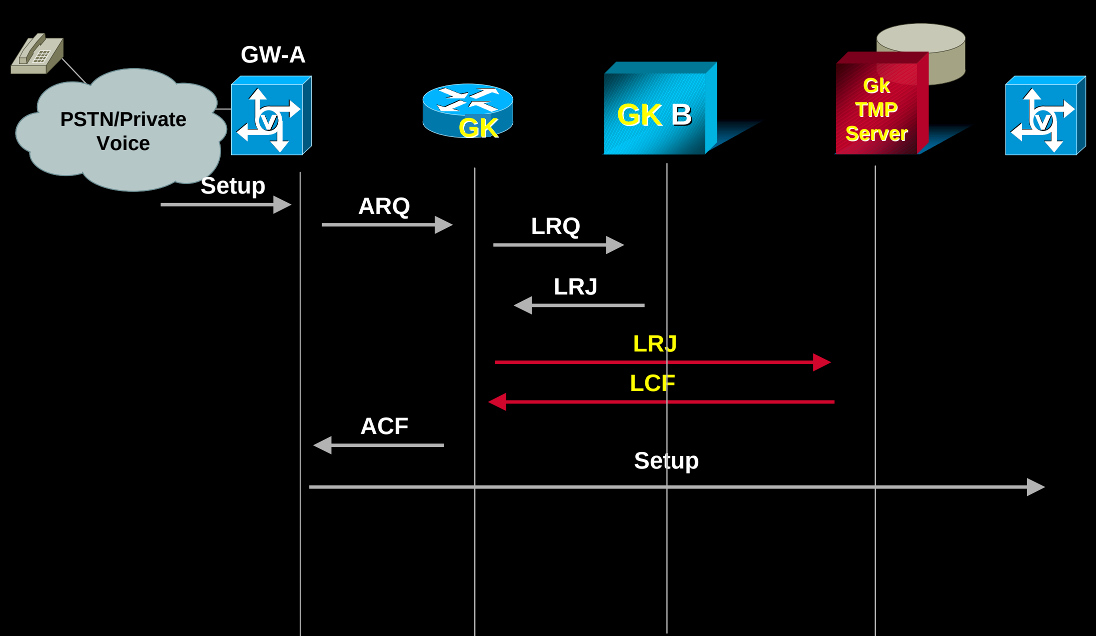
RRJ—Registration Reject

URQ—Unregistration Request

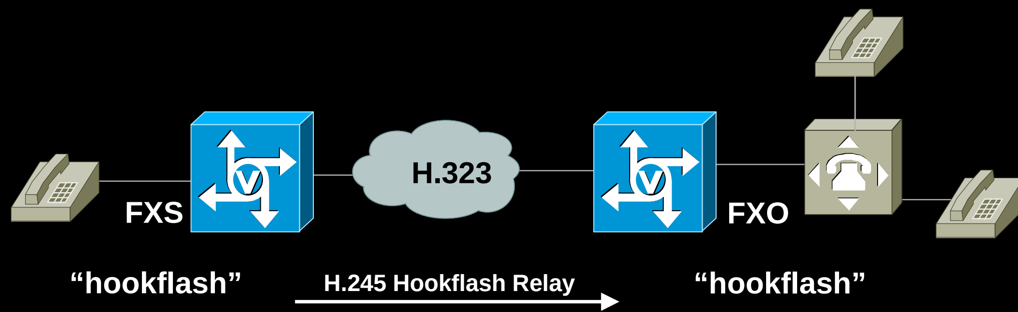
GK TMP Message Interpretation

- 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



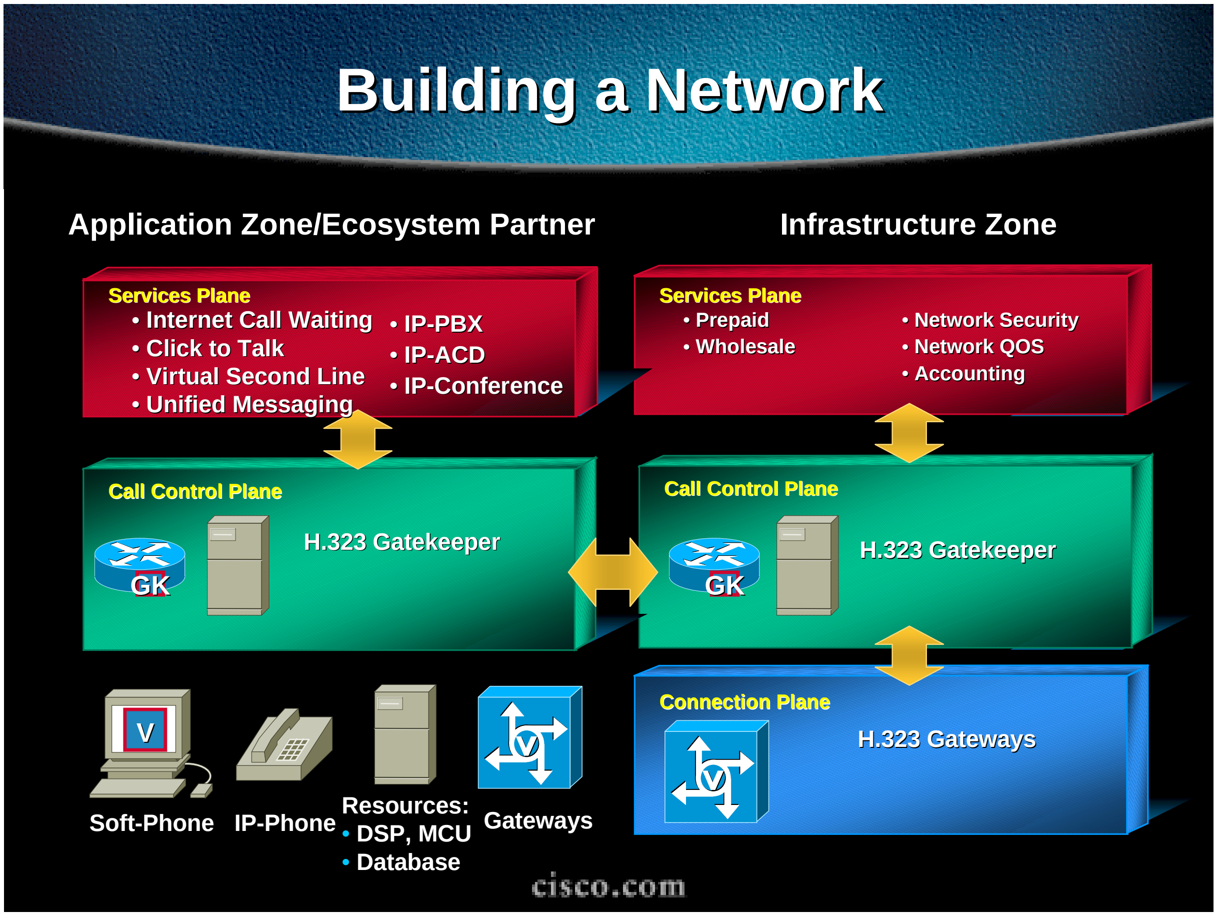
Hookflash Relay

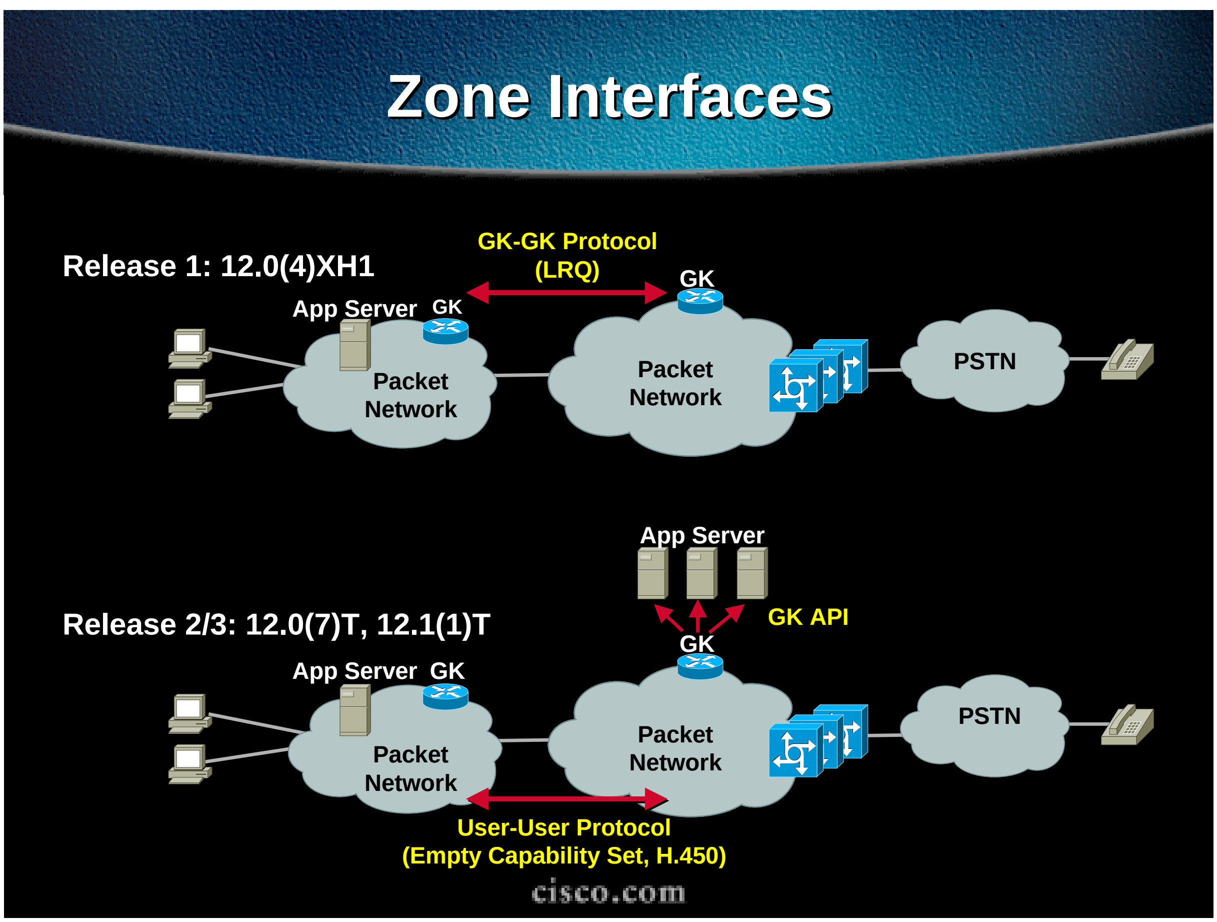


- Supported with H.245 DTMF relay (alpha or numeric, but not with Cisco-RTP DTMF relay; Also works with Fast Connect

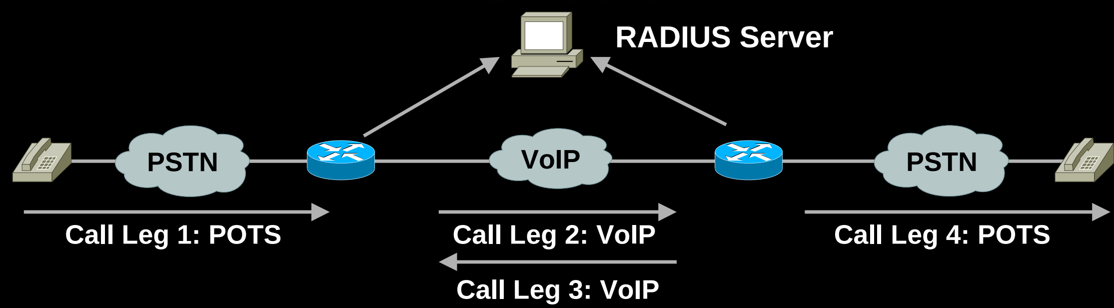
Agenda

- Introduction
- Technology Enablers
- Implementation Strategy
  - Infrastructure
  - Application Zones
  - Billing
  - Security
- Future Developments


Building a Network
Application Zone/Ecosystem Partner
Infrastructure Zone
Services Plane
• Internet Call Waiting
• Click to Talk
• Virtual Second Line
• Unified Messaging
• IP-PBX
• IP-ACD
• IP-Conference
Services Plane
• Prepaid
• Wholesale
• Network Security
• Network QOS
• Accounting
Call Control Plane
GK
H.323 Gatekeeper
Call Control Plane
GK
H.323 Gatekeeper
V
Soft-Phone
IP-Phone
Resources:
• DSP, MCU
• Database
V
Gateways
Connection Plane
V
H.323 Gateways
cisco.com


Zone Interfaces
Release 1: 12.0(4)XH1
GK-GK Protocol
(LRQ)
App Server
GK
GK
Packet
Network
Packet
Network
PSTN
Release 2/3: 12.0(7)T, 12.1(1)T
App Server
GK API
GK
App Server
GK
Packet
Network
Packet
Network
PSTN
User-User Protocol
(Empty Capability Set, H.450)
cisco.com

Billing



- 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)

Billing

Standard Supported RADIUS Attributes (RFCs 2138 and 2139)

Attribute	Value	Description RFCs 2139, 2139
NAS- IP-Address	4	IP address in 4 hex octets (ASCII string)
NAS-Port-Type	61	4 octets (used for MLPPP)
User-Name	1	ASCII string field up to 63 ostet
Called-station-Id	30	1 or more octets (DNIS phone number, ASCII string)
Calling-Station-Id	31	1 or more octets (ANI phone number, ASCII string)
Acct-Status-Type	40	4 octets (hex ASCII number)
Service-Type	6	4 octets (hex ASCII number)
Acct-Session-Id	44	“overloaded” for CDR ASCII string up to 256 bytes
Acct-Input-Octets	42	4 octets stop records only (hex number)
Acct-Output-Octets	43	4 octets stop records only (hex number)
Acct-Input-Packets	47	4 octets stop records only (hex number)
Acct-Output-Packets	48	4 octets stop records only (hex number)
Acct-Session-Time	46	4 octets (hex number rep. Seconds) stop records only
Acct-Delay-Time	41	4 octets (hex number in seconds)

Billing

<session id>/<call leg setup time>/<gateway id>/<connection id>/<call origin>/<call type>/<connect time>/<disconnect time>/<disconnect cause>/<remote ip address>

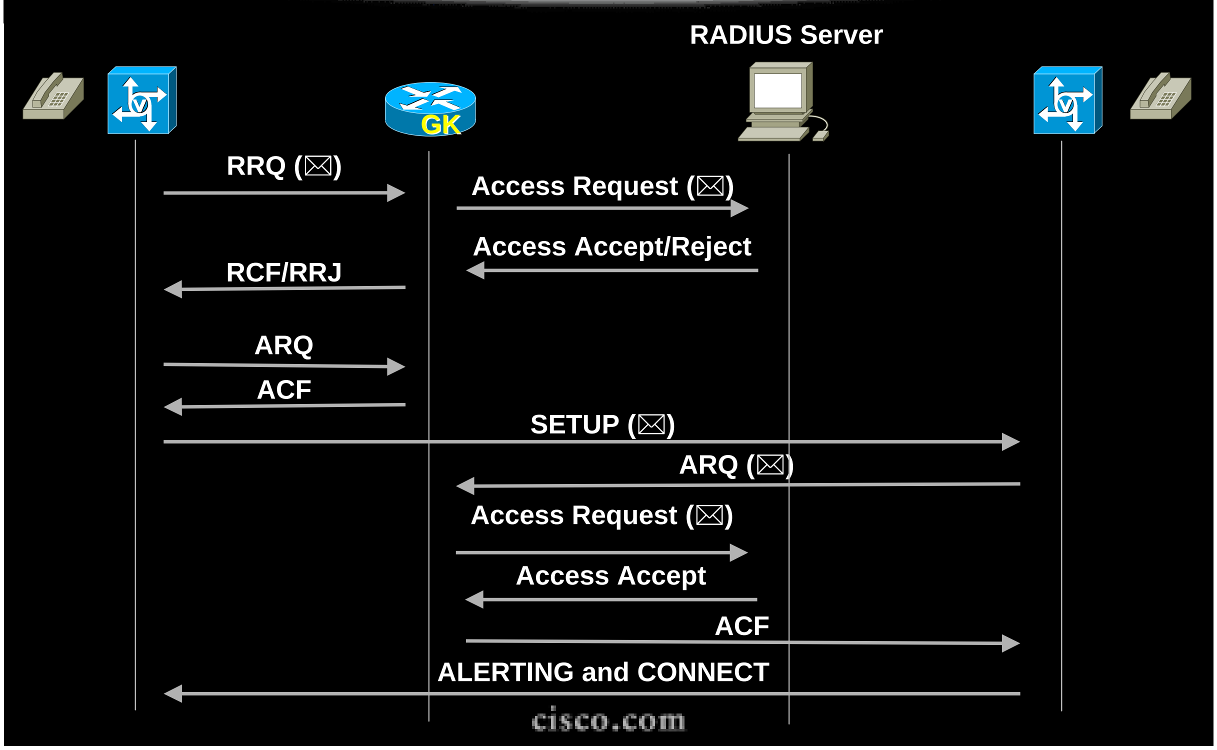
Overloaded *Acct-Session-Id* Field Descriptions:

Field	Description
session id	The standard (RFC 2139) RADIUS *account-session-id*
call leg setup time	The Q.931setup time for this connection in NTP format.
gateway id	The name of the underlying gateway. Name string is of form "gateway.domain_name"
connection id	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.
call origin	Indicates origin of the call relative to the gateway. Possible values are "originate" and "answer".
call type	Indicates call leg type. Possible values are: "Telephony" and "VoIP."
connect time	The Q.931 connect time for this call leg in NTP format. (stop only)
disconnect time	The Q.931 disconnect time for this call leg in NTP format. (stop only)
disconnect cause	Documented in Q.931 specification. Can be in the range of 1-160. (stop only)
remote IP address	IP address of the remote gateway used in this connection (stop only)

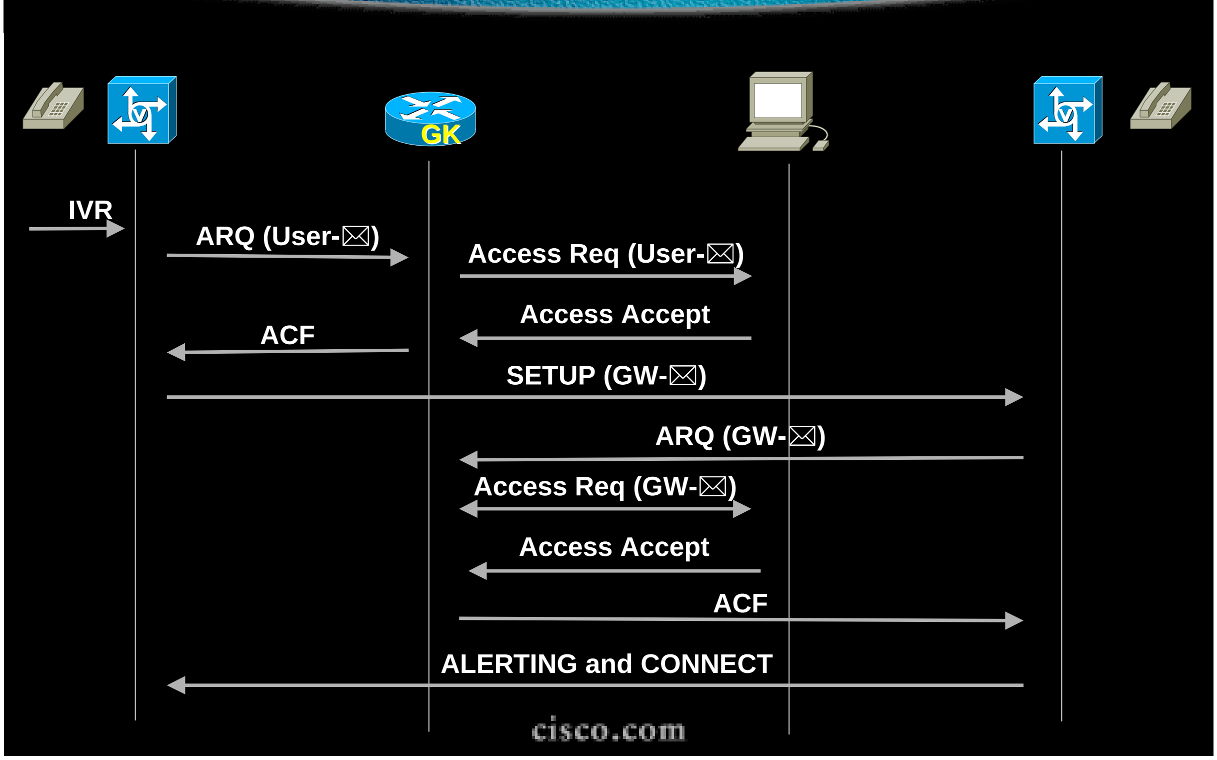
Security

- 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

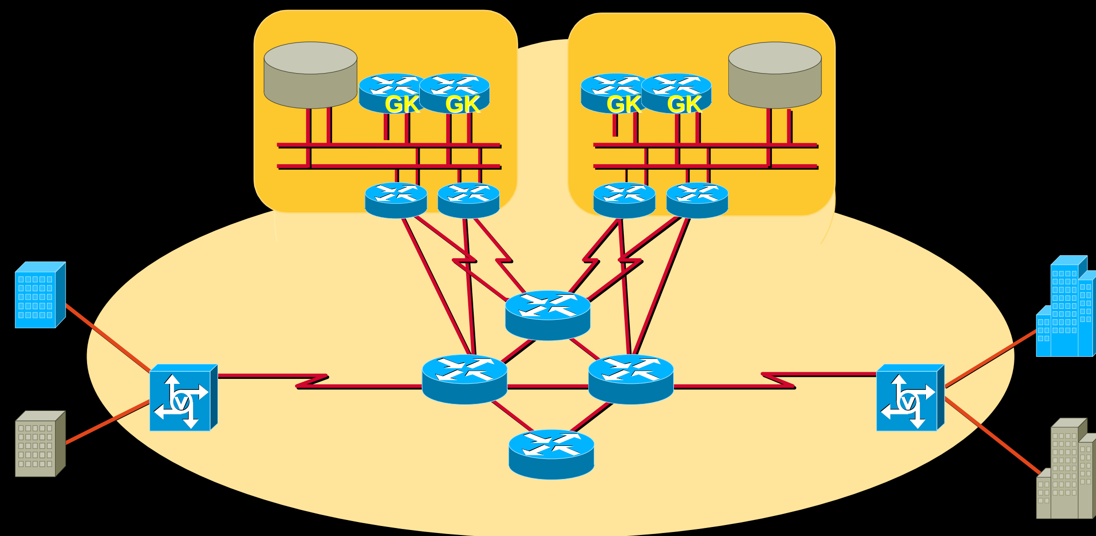
H.235 Security—Registration



H.235 Security—Admission

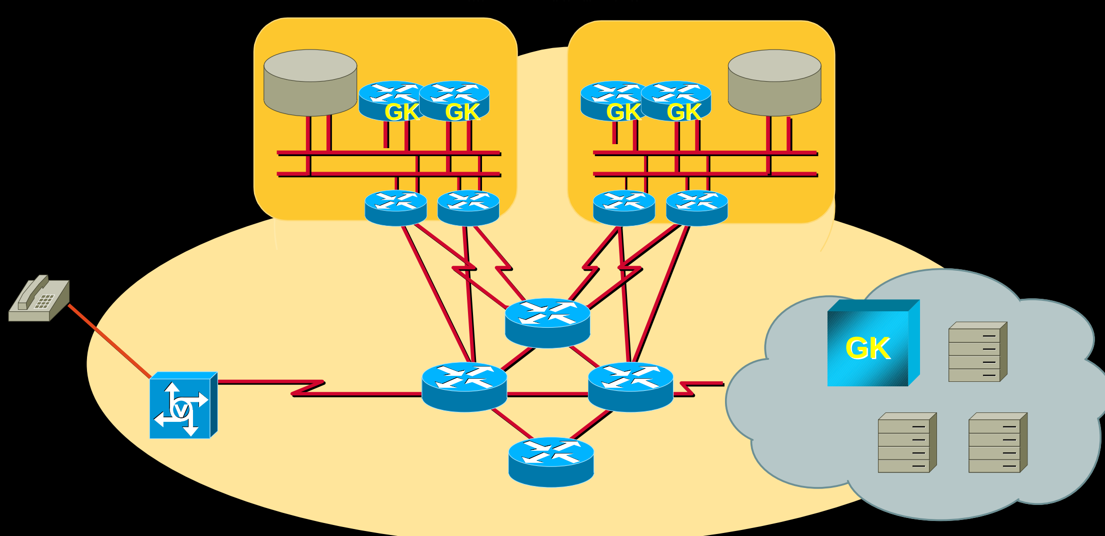


Implementation Strategy

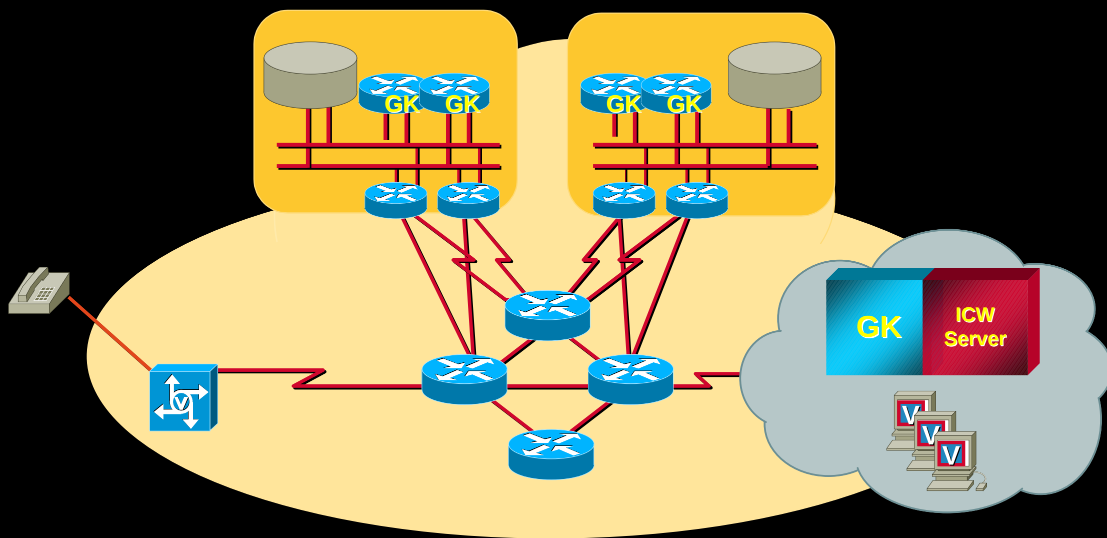


- Start with infrastructure example—VPN with overlapping dial plan

Adding Unified Communications



Internet Call Waiting



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Agenda

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

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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)

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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

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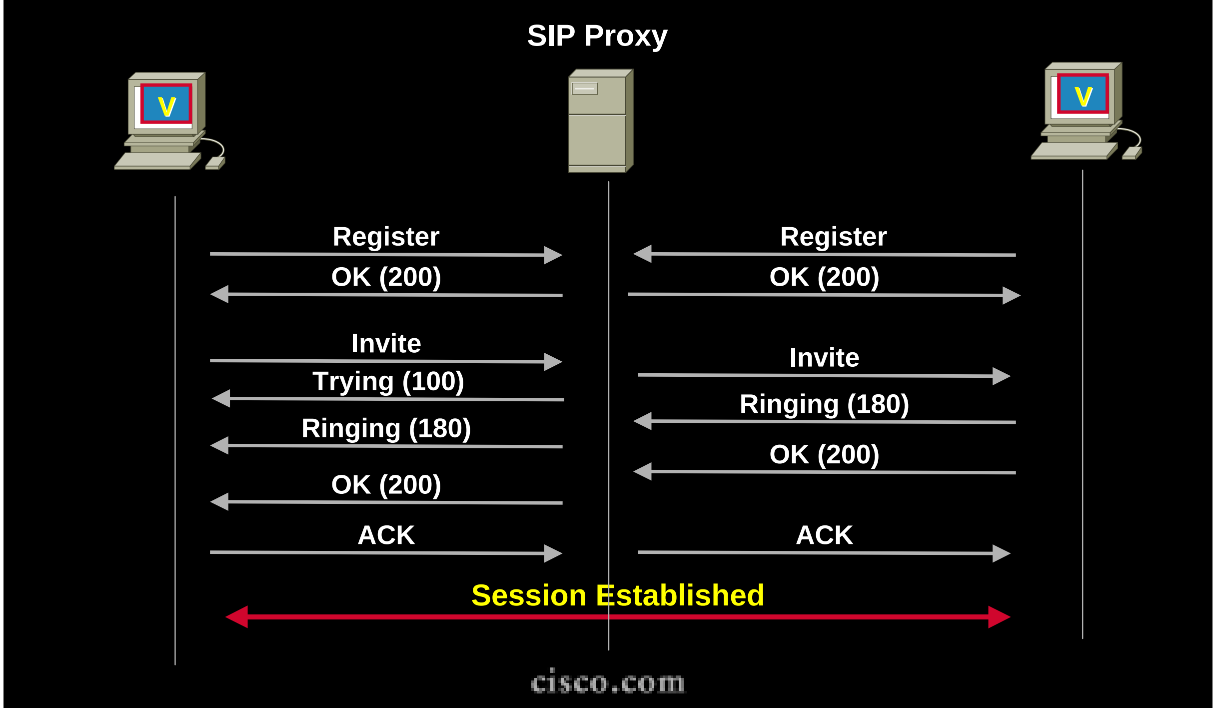
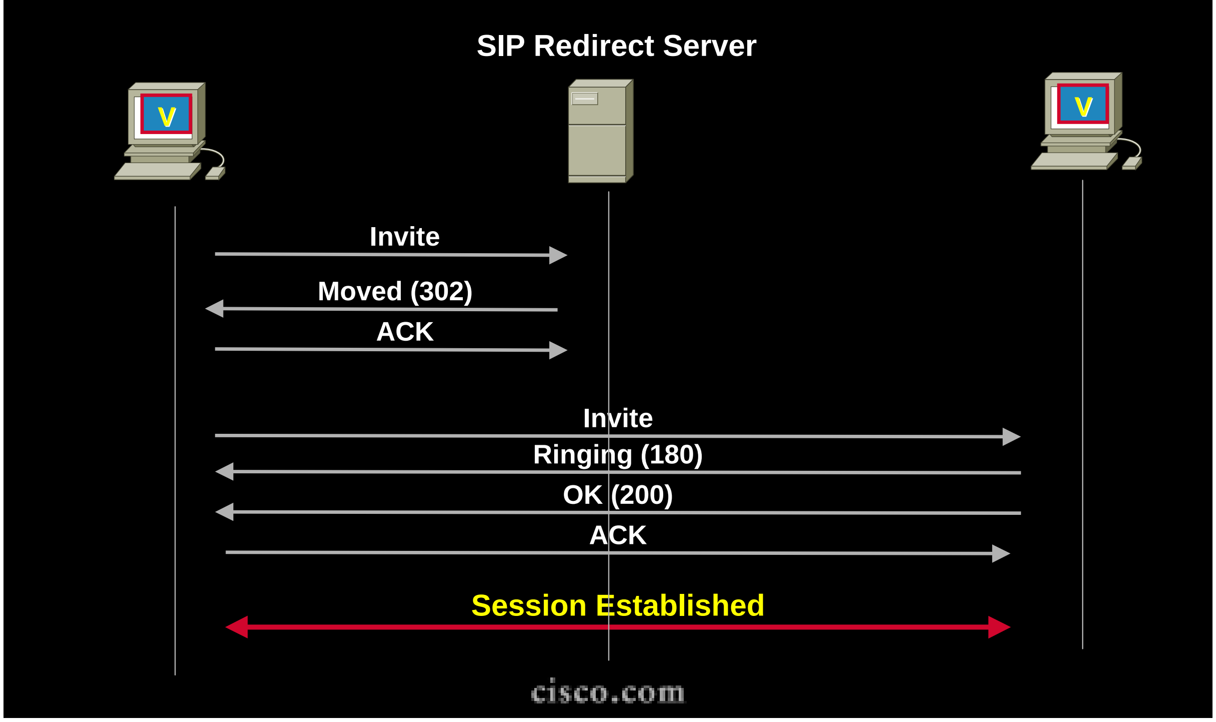
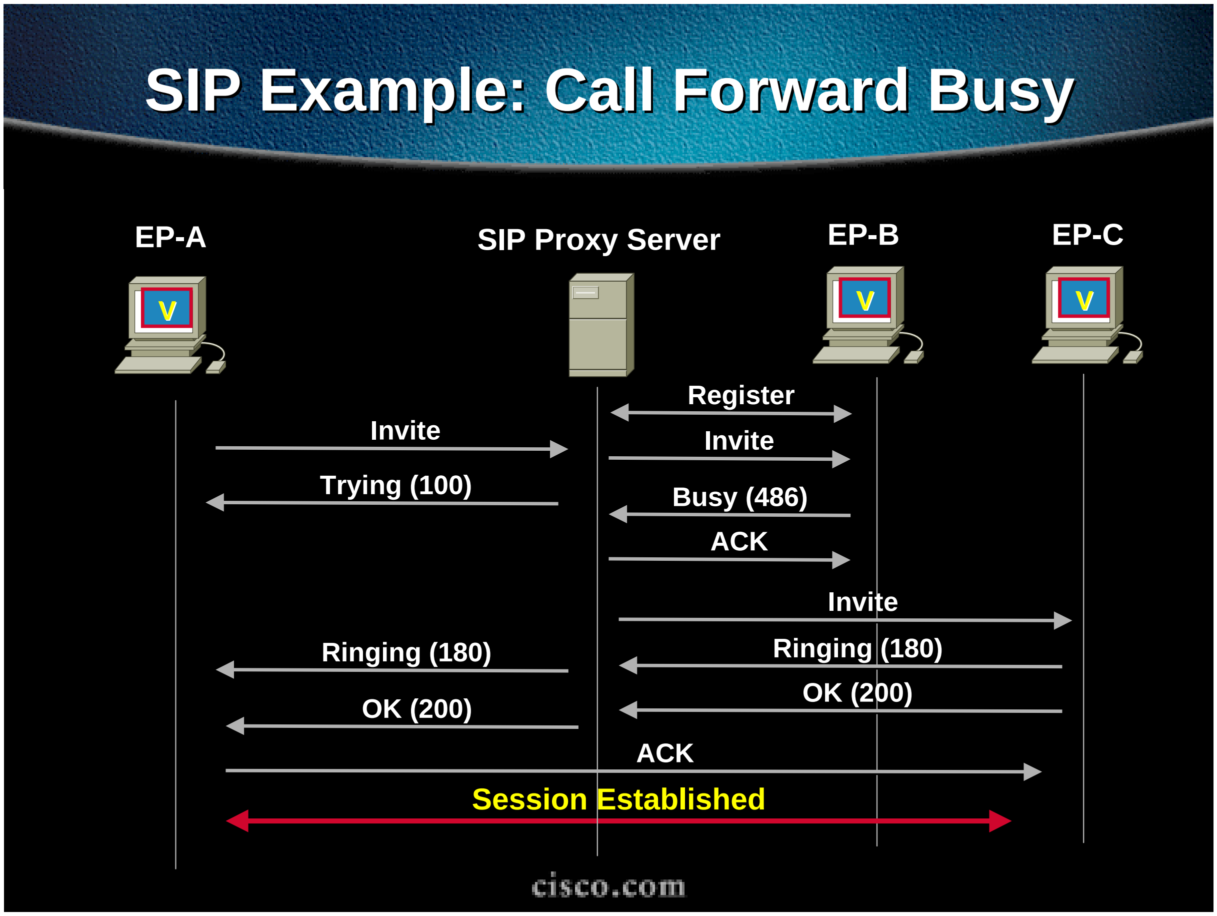
SIP Call Flow with Proxy



SIP Call Flow with Redirect




SIP Example: Call Forward Busy
EP-A
SIP Proxy Server
EP-B
EP-C
Register
Invite
Invite
Trying (100)
Busy (486)
ACK
Invite
Ringing (180)
Ringing (180)
OK (200)
OK (200)
ACK
Session Established
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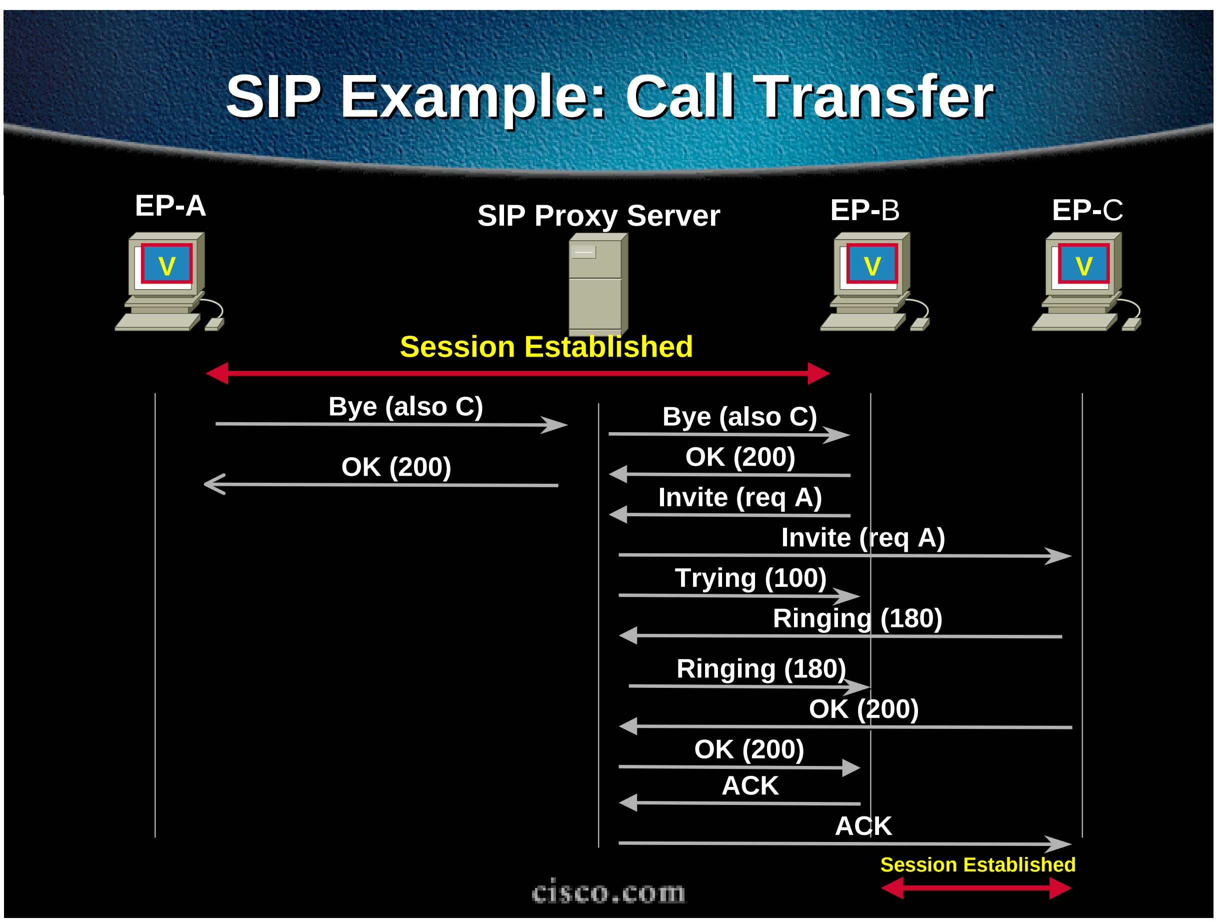
SIP Example: Call Transfer
EP-A
SIP Proxy Server
EP-B
EP-C
Session Established
Bye (also C)
Bye (also C)
OK (200)
OK (200)
Invite (req A)
Invite (req A)
Trying (100)
Ringing (180)
Ringing (180)
OK (200)
OK (200)
ACK
ACK
Session Established
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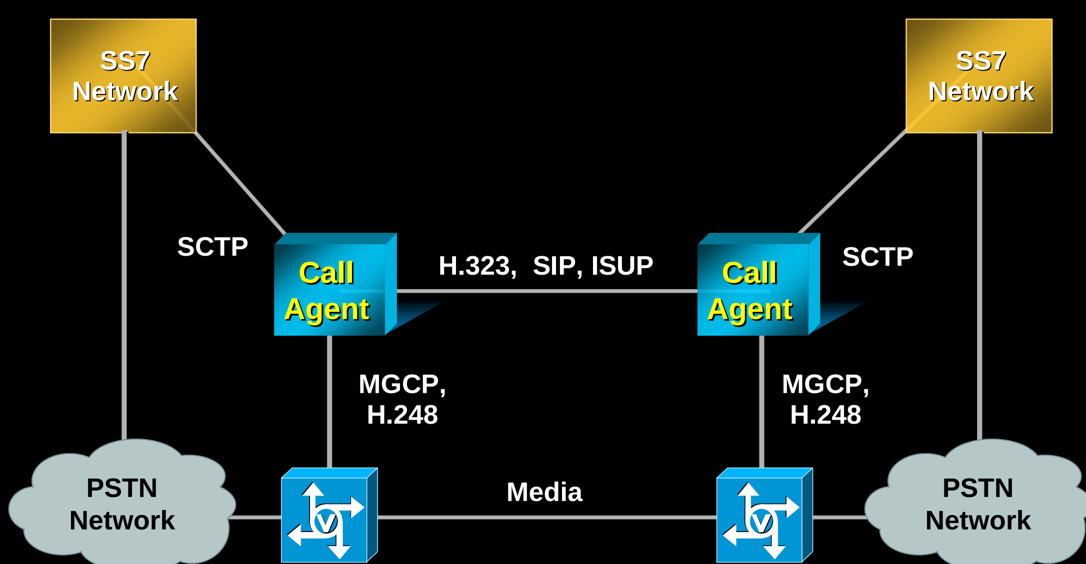
Media Gateway Control

- 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



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

Advanced Voice over IP Enhanced Services

Session 2105

Please Complete Your Evaluation Form

Session 2105

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