VoIP
Standards and Status

Johannes Krohn (jkrohn@cisco.com)

Agenda

- Voice transport over IP
- Signalling Protocol Classification
- IETF Overview
- VoIP signalling protocols
  - H.323
  - MGCP
  - SCCP
  - SIP
- Myths / Facts

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- **Voice transport over IP**
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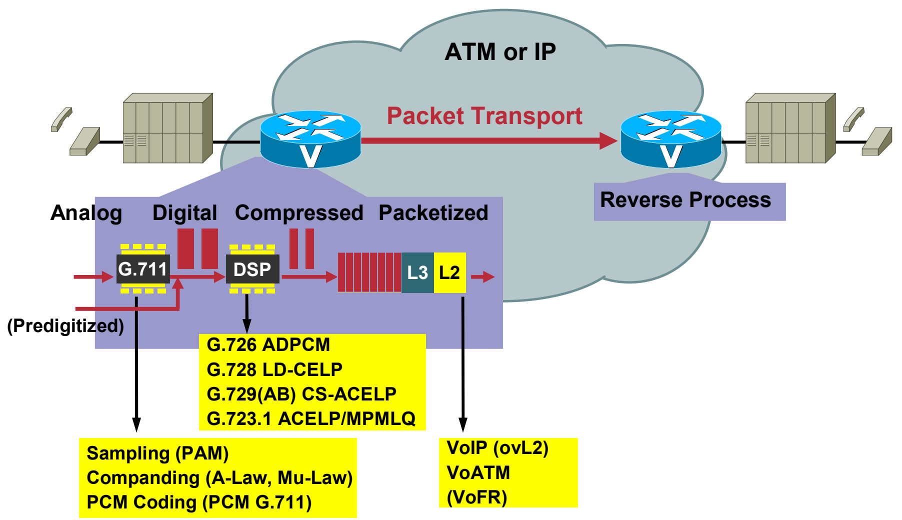
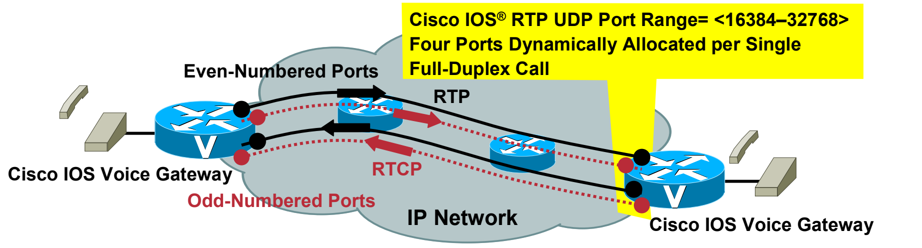
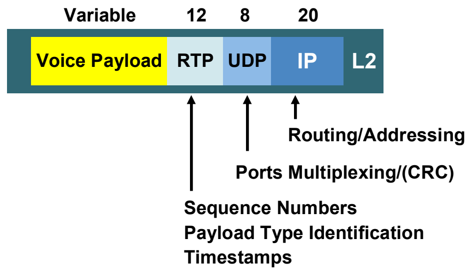
  - **MGCP**
  - **SCCP**
  - **SIP**
- **Myths / Facts**

Voice Packetization



Voice Encapsulation over IP



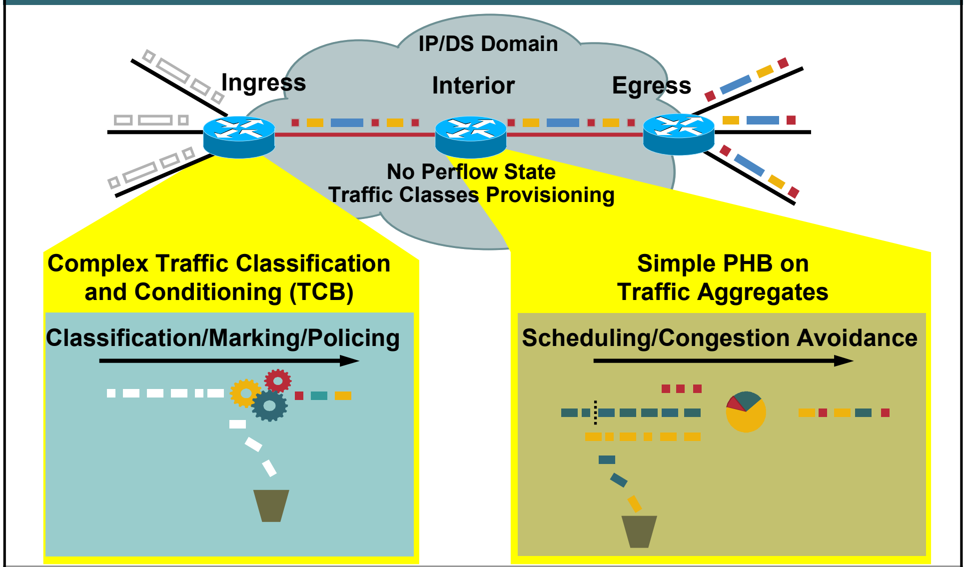


- Designed to carry real-time traffic on top of IP
- Real-Time Protocol (RTP)—media
- Real-Time Transport Control Protocol (RTCP)—form of signaling between RTP termination points
  - Watches the quality of underlying infrastructure
- RFC1889 and 1890

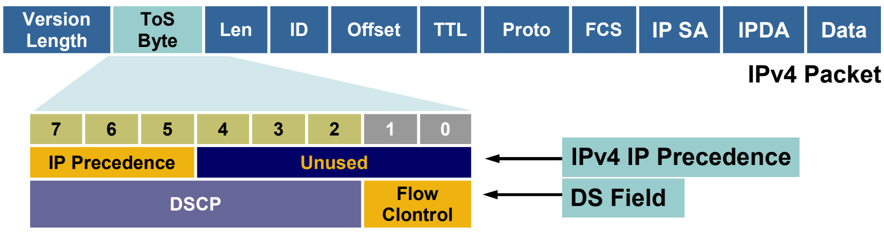
Voice QoS Network Requirements

- **Low delay**
  - Long delays cause the listener to start to talk before the speaker is finished
- **Low delay variation (jitter)**
  - Jitter causes gaps in the speech pattern that cause the quality of voice to sound “jerky”
- **Low packet loss**
  - Packet loss causes voice to sound “jerky” and annoying
- **Low echo**
  - Listener annoyed by hearing the speaker twice phase-shifted

Differentiated Services Architecture (RFC 2274, RFC 2275)



Encoding Traffic Classes (DSCP)

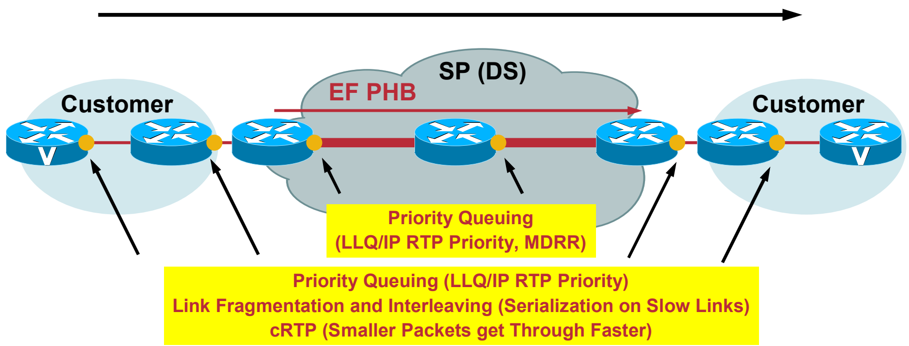


- DiffServ field: the IP version four-header ToS octet or the IP version six traffic-class octet when interpreted in conformance with the definition given in RFC2474
- DSCP: the first six bits of the DiffServ field, used to select a PHB (forwarding and queuing method)

Per-Hop Behaviors (PHB)

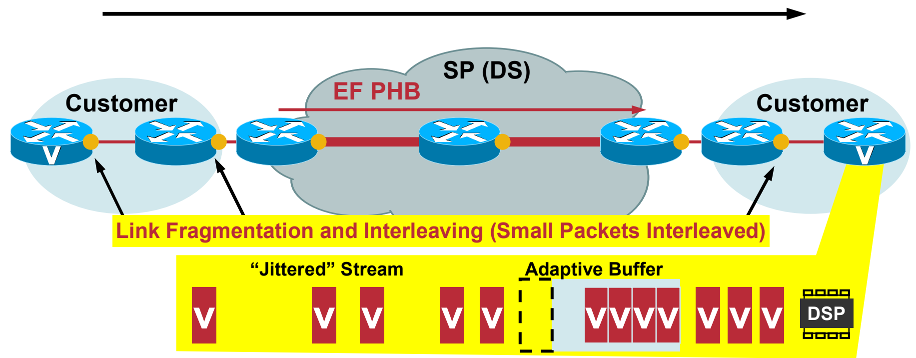
- **Expedited Forwarding (EF)**
  - Building block for low delay/jitter/loss (voice/video)
  - Served at a certain rate with short/empty queues
- **Assured Forwarding (AF)**
  - Designed for data
  - High probability of delivery if profile is not exceeded
  - Four classes and three levels of drop precedence
  - Specific resources (BW, buffer space) allocated to each class at each node
- **Best Effort (BE)**

End-to-End IP QoS for Voice Delay Mechanisms



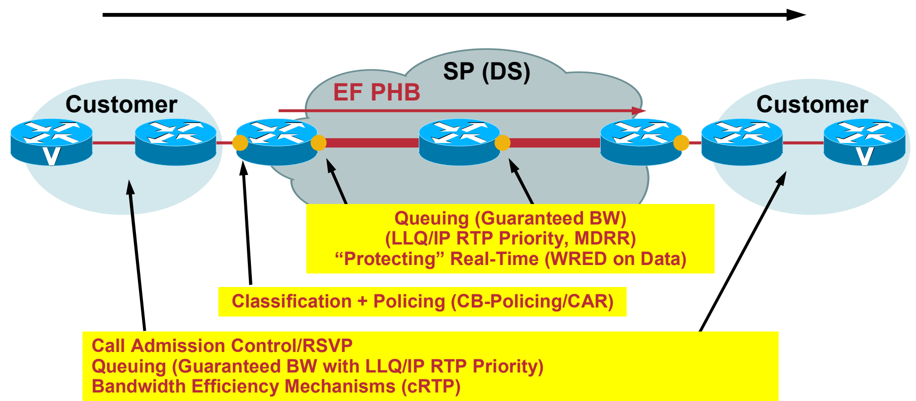
- Delay is a problem for us/humans, not the technology
- G.114 recommends less than 150-ms end-to-end, one-way delay
- Points of congestion are more susceptible to build delay
- Multiple-delay types
  - Codec-algorithmic, processing, serialization, propagation

End-to-End IP QoS for Voice Delay Variation Mechanisms



- Delay variation causes buffer underruns → codec desynchronization
- Occurs due to mixing traffic of different types from different sources
- Different traffic types meet in output buffers
- Points of congestion are more susceptible to introduce delay variation

End-to-End IP QoS for Voice Packet Loss Mechanisms



- Voice can survive some packet loss—how much depends on codec
- Points of congestion drop packets due to buffer overflow
- Voice stream is not adaptive (UDP), does not react to WRED

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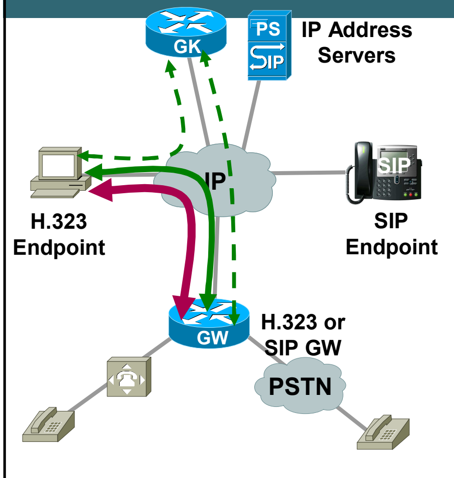
Protocol Architecture Models

- Terminals are managed by the switch/host and cannot talk directly to other terminals

- Peer endpoints can place calls without the presence of a call agent, but may consult call agents/proxies for name resolution/redirection

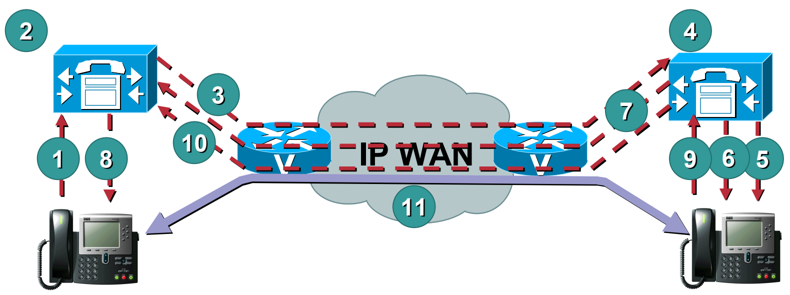
- Client endpoints cannot initiate calls without their call agent, but media streams flow peer to peer

Distributed Call Control: H.323 and SIP



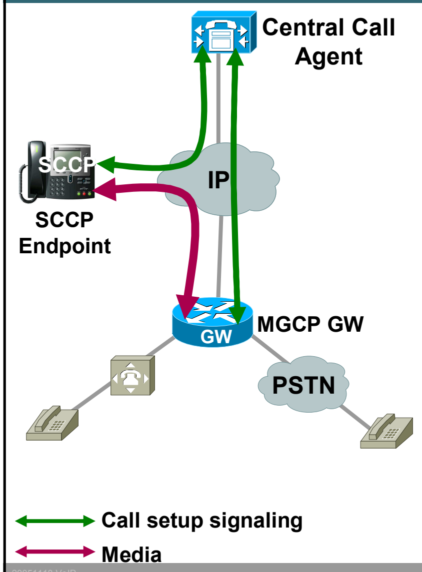
- All signaling messages and dialed digits are interpreted by the protocol stack on the Endpoint/Gateway
  - Gateway/Endpoint is an "intelligent" device
- Peer-to-peer call setup (dial plan/IP address servers are optional)
  - If IP Address Server is out of reach, the Endpoint/GW can choose an alternate route
- TDM signaling types supported is a function of the GW protocol stack
- Resilient over IP connectivity failures
- Scalable – distributed CPU power
- New applications deployed where needed w/o affecting rest of the network components (Internet model)
- Distributed configuration

Distributed Call Processing



1	Call Setup	5	Call Setup	9	Off Hook
2	E.164 Lookup	6	Ring	10	Call Connect
3	Call Setup	7	Alerting	11	Connect RTP Stream
4	E.164 Lookup	8	Ringback		

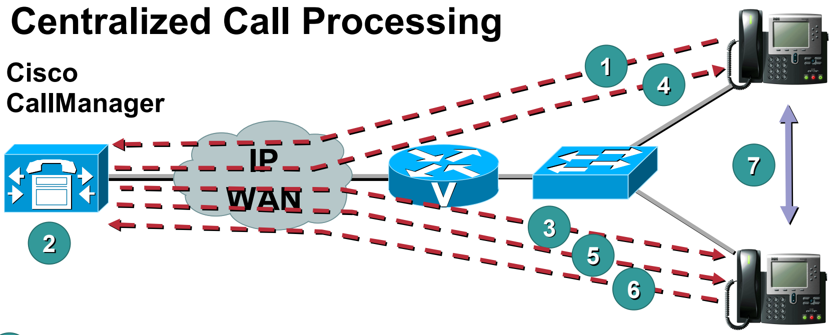
Centralized Call Control: MGCP and SCCP



- All signaling messages are "back-hauled" to the Call Agent
  - Gateway/Endpoint is a "dumb" device
- Call Agent arbitrates all call setup
  - If Call Agent is out of reach, the Endpoint/GW cannot function
- TDM signaling types supported is a function of the GW and Call Agent
- Dependent on IP network connectivity
  - Requires failover strategies
- Scalable – central Call Agent is a contention point
- New applications deployed requires Call Agent upgrade (CO model)
- Centralized configuration

Centralized Call Processing

Centralized Call Processing



1 Call Setup
2 E.164 Lookup
3 Call Setup
4 Ring Back
5 Ring
6 Off Hook
7 Connect RTP Stream

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IETF Standards Process

- Let's examine the IETF Document Process
- SIP has an RFC number assigned to it [3261], it's a "Standard" right?
- 6 Classes of documents
  – Standards Track
  – Best Current Practice (BCP) - 1 shot Standard
  – Informational - not intended to Standardized
  – Experimental - not sufficiently studied to Standardize
  – Historical - not to be coded to anymore
  – Obsoleted - revised by another RFC

IETF Standards Process (Cont.)

There are 2 Categories of Standards in the IETF:

- Best Current Practice – 3 stage process
  - Most often references other RFCs and Standards to put an "Architecture" together
  - Or, could state what "shouldn't be done"
  - Doesn't generally have new feature descriptions (called "normative text")
- Standards Track – 5 stage process

IETF Standards Process (Cont.)

Standards Track Documents

- 5 Document levels (called a "Status")
  - First: the Individual "Internet Draft"
    - Can be written by anybody, or asked for by a WG Chair(s), or can be the result of a "Design Team" collaboration
    - ASCII Formatted only
    - Valid for up to 6 months (with an electronic cutoff)
    - Can be revised/renewed (theoretically indefinitely)
    - Must become an Official Working Group Item at some point
    - WG Chair submits a version to the WG as WGLC for "consensus"
    - WG Chair submits WG consensus ID to IESG for RFC "acceptance"
    - Assigned an RFC number and published
    - It's now a Standard, right?

IETF Standards Process (Cont.)

Standards Track Documents (cont'd)

- **Proposed Standard (PS)**
  - Core of this stage is the "Request for Comments" aspect
  - 18 months to 2 years here (best case... usually 3-5 yrs)
  - "MUST" have two independent interoperable "completely" compliant implementations demonstrated to IESG to become DS
- **Draft Standard (DS)**
  - Considered "worthy" for Standardization
    - Generally should have a 3rd implementation to become FS
- **Standard (FS or just STD)**
  - Only 62 of these exist !

But there are more than 3825 RFCs... How does that work?!

IETF Standards Process (Cont.)

Which Category are these Protocols/Apps in?

IPv4	ARP	RADIUS	RSVP	MPLS
ICMP	DNS	HTTP/1.1	IPsec	IKE
UDP	PPP	IPv6	GRE	COPS
TCP	POP3	DHCP	Diffserv-EF	CIDR
Telnet	OSPFv2	MIME	Diffserv-AF	SDP
FTP	RIPv2	BGP-4	LDAPv3	L2TP
TFTP	SIP	BOOTP	TLS	SMTP
SNMPv3	MGCP	RTP	IMAPv4	DVMRP

IETF Standards Process (Cont.)

Which Category are these Protocols/Apps in?

Standard

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March 2003 -- SNMPv3 to Full STD

IETF Standards Process (Cont.)

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Standard		Draft Standard	Proposed Standard*	
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*No Protocol Extensions can progress until ALL normative references have progressed

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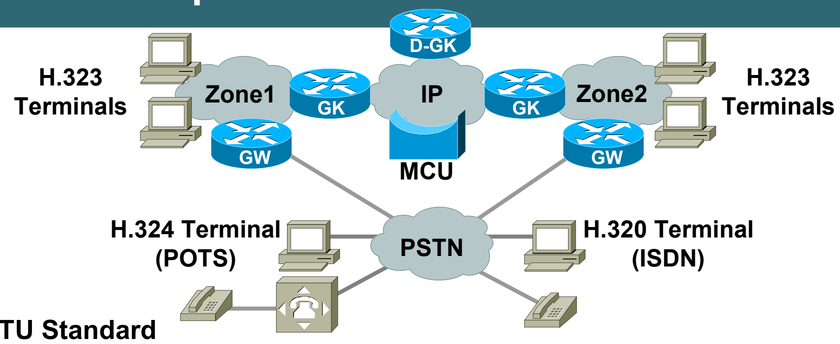
H.323 Background

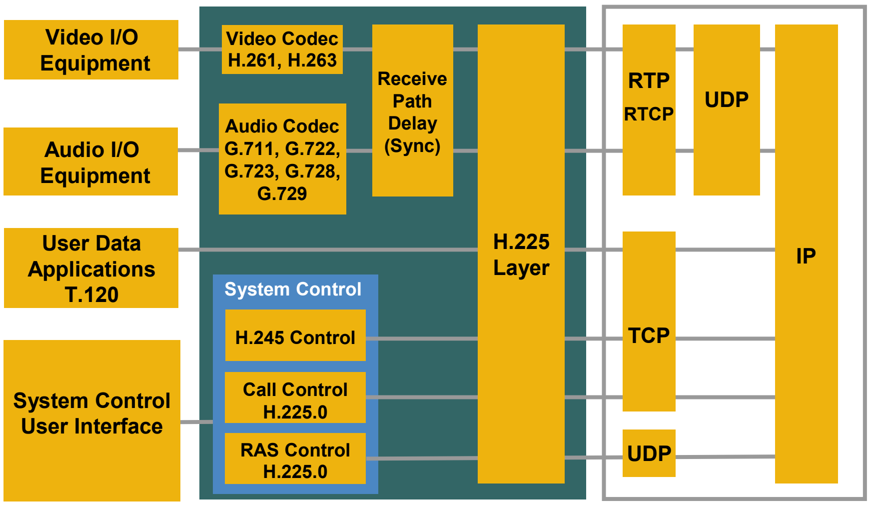
- International (ITU) standard for:
  - Packet-based multimedia communications systems
- Originated from H.320 (multimedia over ISDN)
- ITU recommendation
  - v1 approved in 1996, v2 in January 1998, v3 in September 1999, v4 in November 2000, and v5 in July 2003
- Leverages existing standards, i.e., H.320, Q.931, H.450, etc.
- Wide market acceptance
- Facilitates interoperability between vendors

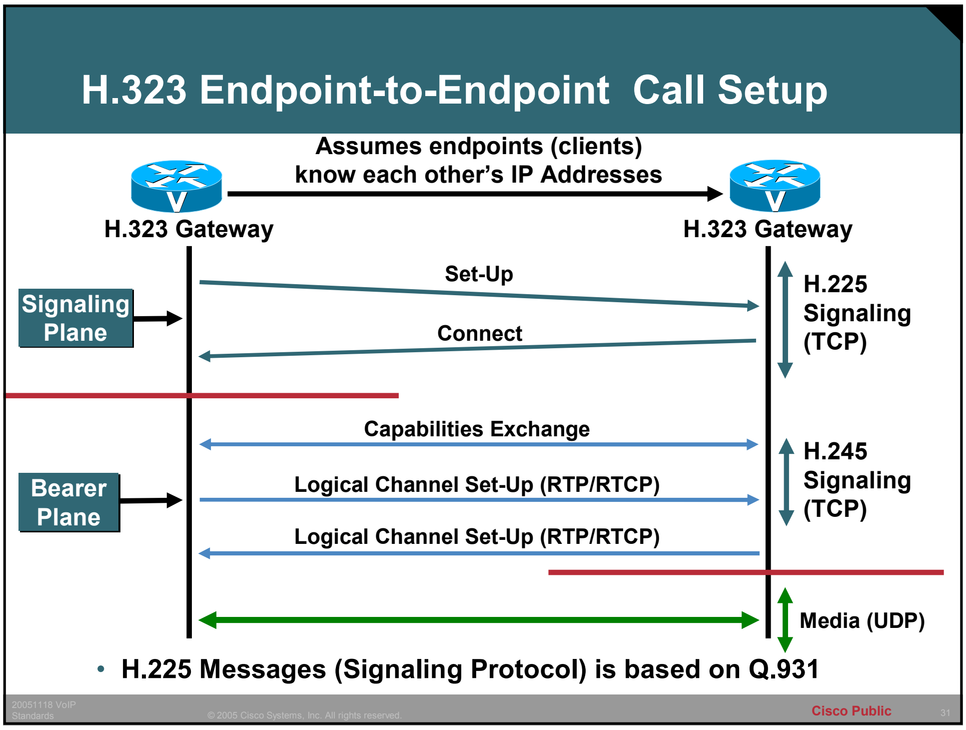
H.323 Components



- ITU Standard
- Terminal—Endpoint
- Gateway (GW)—TDM to IP conversion
- Gatekeeper (GK)—Phone number and name to IP address lookup and CAC bandwidth management
- Directory Gatekeeper (D-GK)—Tiered hierarchy of GKs
- MCU—Multipoint Control Unit to mix audio and replicate video

Scope of H.323 Recommendation




H.323 Endpoint-to-Endpoint Call Setup
Assumes endpoints (clients)
know each other's IP Addresses
H.323 Gateway
H.323 Gateway
Signaling Plane
Set-Up
Connect
H.225 Signaling (TCP)
Bearer Plane
Capabilities Exchange
Logical Channel Set-Up (RTP/RTCP)
Logical Channel Set-Up (RTP/RTCP)
H.245 Signaling (TCP)
Media (UDP)
• H.225 Messages (Signaling Protocol) is based on Q.931
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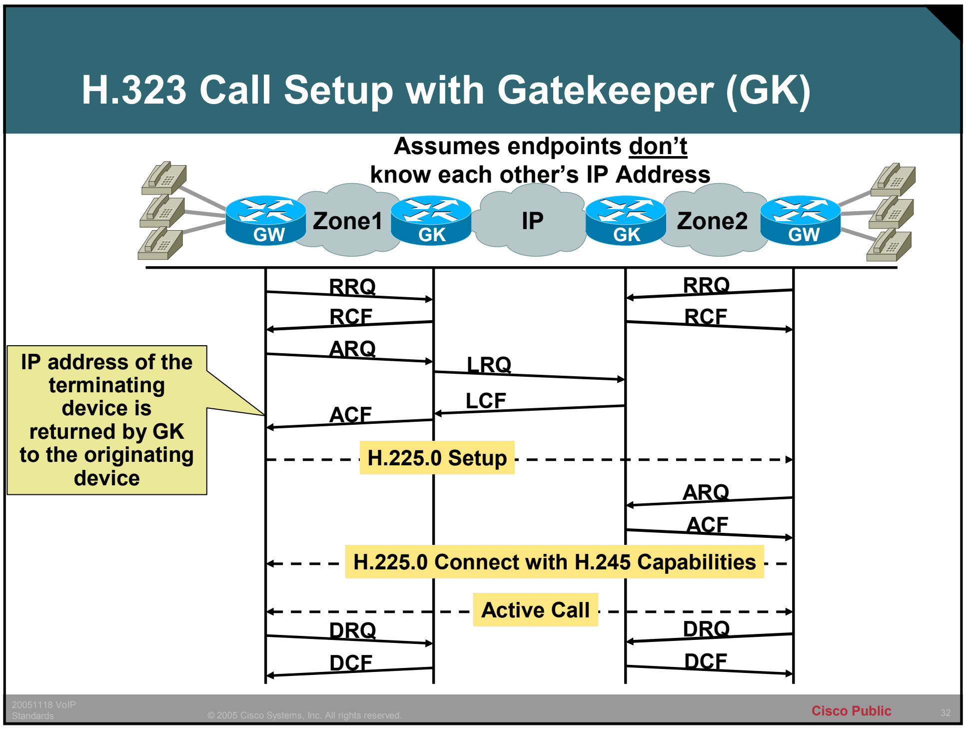
H.323 Call Setup with Gatekeeper (GK)
Assumes endpoints don't
know each other's IP Address
GW
Zone1
GK
IP
GK
Zone2
GW
RRQ
RCF
ARQ
LRQ
LCF
ACF
RRQ
RCF
IP address of the terminating device is returned by GK to the originating device
H.225.0 Setup
ARQ
ACF
H.225.0 Connect with H.245 Capabilities
Active Call
DRQ
DCF
DRQ
DCF
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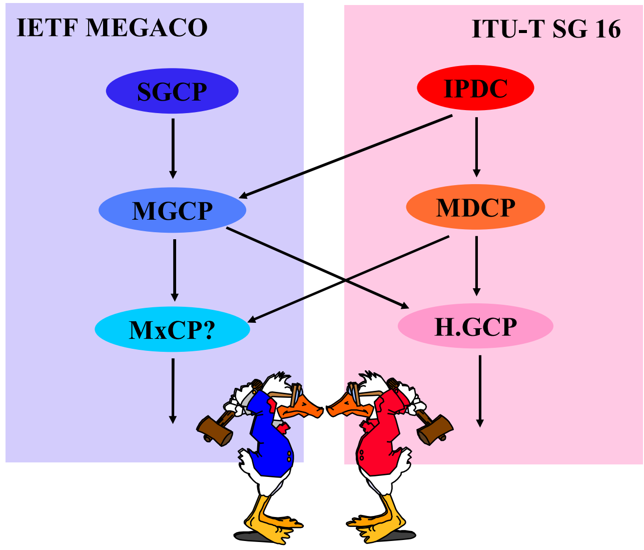
Background - Media Gateway Control

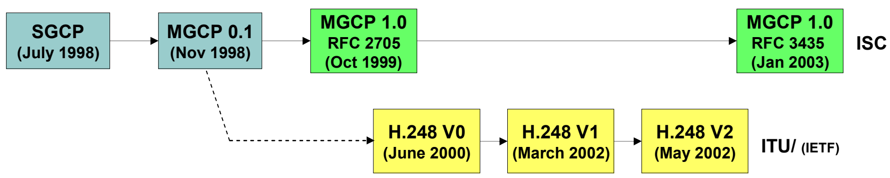
SGCP (July 1998) → MGCP 0.1 (Nov 1998) →

- And then it was proposed to the IETF and ITU, and...

Background - Between 11/98 and 3/99



Background - Media Gateway Control

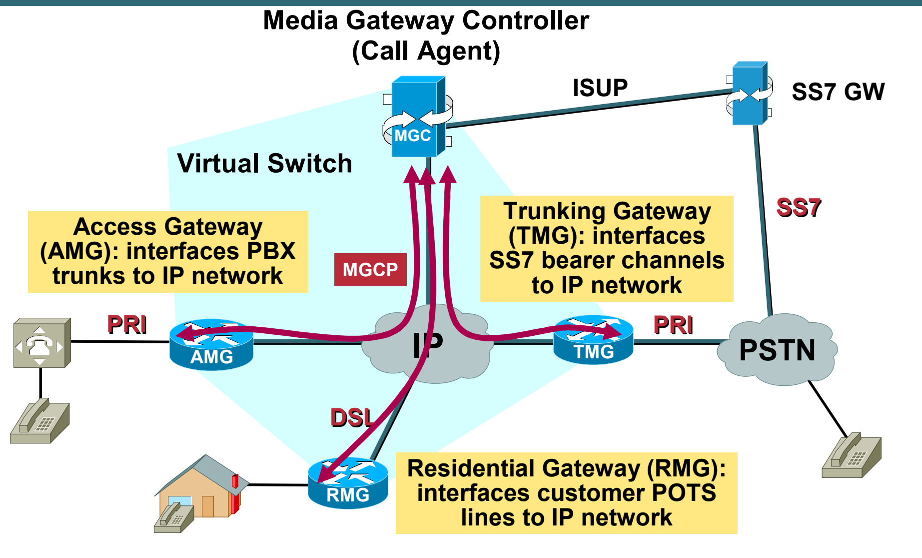


- International SoftSwitch Consortium (ISC) continued with MGCP: development, extensions, Interop testing
- ITU/IETF H.248 work was initiated at a point when initial MGCP development was fairly complete
- Both Use Session Description Protocol (SDP, RFC 2327) to describe media capabilities - Just as SIP does

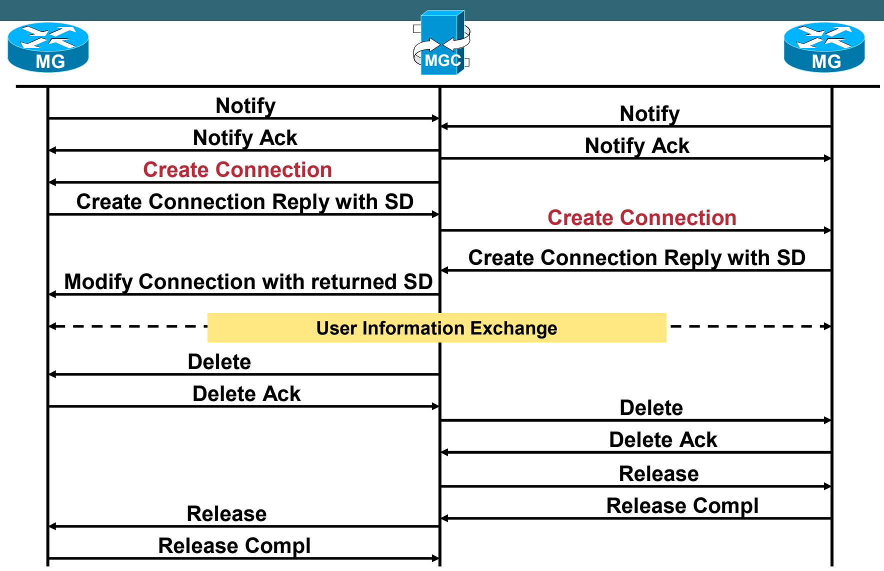
MGCP Concept

> “MGCP is designed as an internal protocol within a distributed system that appears to the outside as a single VoIP gateway.”
>
> RFC3435

MGCP—Components



MGCP Call Setup



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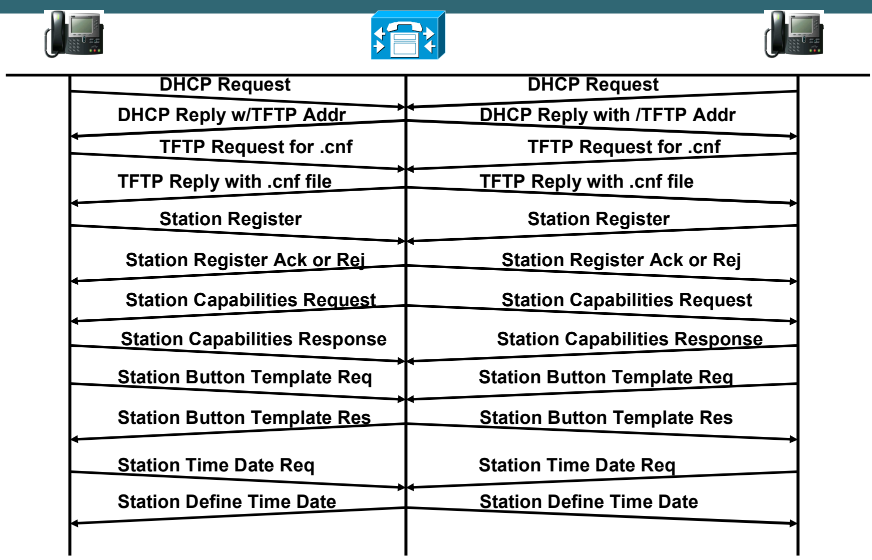
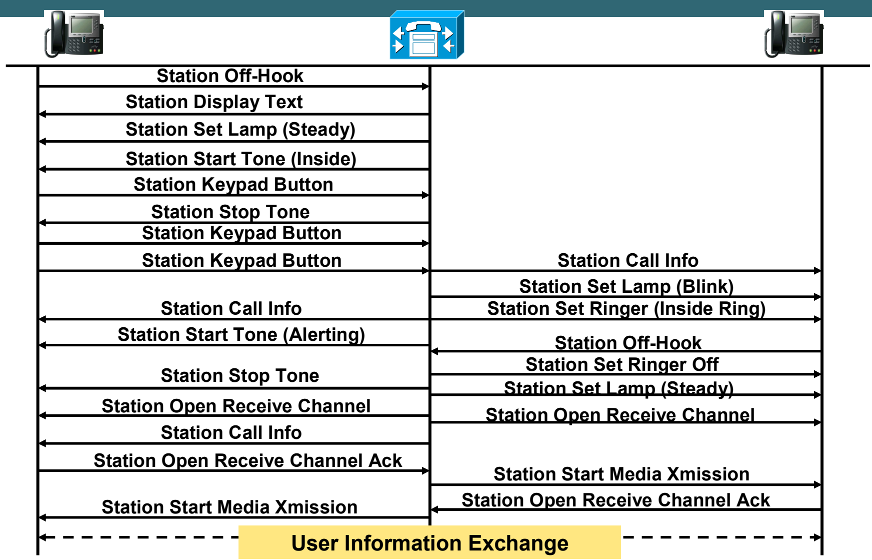
SCCP-Skinny Client Control Protocol

- Client – Server protocol
- used in Cisco Enterprise IPT solution (CallManager)
- available to external developers
- stimulus based
- no intelligence in the endpoints

AVVID SCCP Client Registration



AVVID SCCP Client Call Connect



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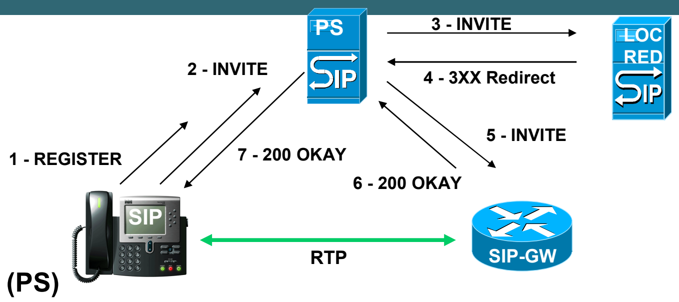
SIP Components



- IETF Standard (PS)
- SIP Proxy Server (PS)
  - Registration Server (REG) – Accepts registration requests from UAs
  - Redirect Server (RED) – Maps SIP request to one or more addresses
  - Location Server (LOC) – Provides information on a callees locations
- User Agent (UA)
  - SIP Gateway (SIP-GW)
  - IP Phones (SIP)

Factors, that delayed SIP in the Enterprise

- SIP is relatively new, having just been published in 1999
- SIP was and is still evolving, and certain issues still need to be resolved.
- Solutions using SIP were and are generally not full featured, but rather a subset of the standard features required in enterprise networks today.
- Some offerings use SIP as an encapsulation protocol, thereby negating the interoperability and application benefits of SIP. This implementations out there are propriatary.

The Reality: SIP is Ready, SIP is Mature

- **The Core SIP Specifications and many of its extensions are at an excellent level of maturity** (most of what you need is DONE)
- **A lot of very hard problems are solved**
- **This is ready to implement and folks are implementing now**
- **Many unpublished specifications are basically ready, but waiting for process overhead**

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Will SIP solve everything .... ?

- Endpoints of vendor A will operate with basic functionality on Vendor B`s „IP-PBX“ and vice versa.
- There will be vendor dependant extensions that will not interoperate with others.

Myths and Facts About SIP

SIP is the future and the whole industry is moving toward it.

Myth

SIP is growing in popularity on roadmaps, but it’s not widely deployed. Many issues still need to be solved.

Fact

The great Voice Myth

"The Great Voice Myth" states that there is only one way to build voice networks, and that there should be only one voice protocol for each function in a packet voice network.

IP is the paradigm shift, not SIP

The ability to choose protocols is fundamental to

the value of IP Communications !!!

VoIP is about services not about protocols

The question that companies must ask is not

"Which protocol is best ?“

but

"Which services do we want to deploy and which VoIP protocols best support those services?"

Conclusion

- Customers need vendors that are committed to support open standards within their products, and are actively developing voice strategies that consider interoperability with all VoIP protocols.
- Without this commitment, VoIP systems are in danger of becoming as proprietary as legacy voice systems.
- Customers need products that support multiple protocols. This way, if a company finds that it needs to migrate its systems or add products that support a different protocol, it will not be required to perform upgrades to the network.

Q and A

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