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Understanding Voice over IP Protocols

Cisco Systems—Service Provider Solutions Engineering

February, 2002

Topics to Discuss

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- History of VoIP
- VoIP—Early Adopters
- VoIP—Standards and Standards Bodies
- VoIP—Making Sense of the Protocols
- “The Great Voice Myth”
- VoIP—Protocol Challenges
- Summary

Why Move to VoIP?

- Cost savings—toll bypass
- Open standards—H.323, SIP, MGCP
- Multi-vendor interoperability
- Integrated IP voice and data networks

Cisco Packet Voice Architecture

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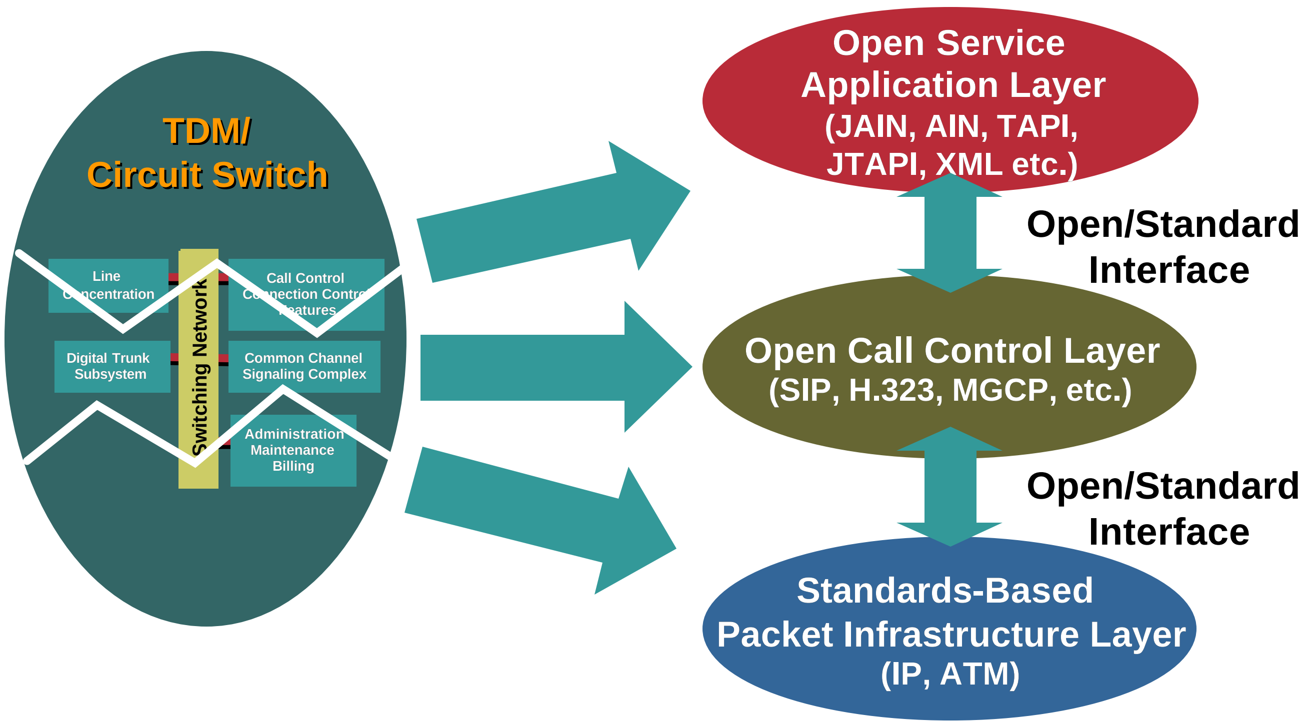

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Early Adopters—
Advanced Services and Toll-Bypass

- **Regulatory opportunities allowed for toll-bypass**
- **PC-to-phone, calling-card and international fax services**
- **Cisco-based carriers used standard protocols, but not all carriers implemented standards**
- **Inter-carrier connections had protocol interoperability challenges**

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Making the Rules for VoIP

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- **IETF (Internet Engineering Task Force)**

  The community of engineers that standardizes the protocols that define how the Internet and Internet Protocols work. http://www.ietf.org/

- **ITU (International Telecommunications Union)**

  An international organization within the United Nations System where governments and the private sector coordinate global telecom networks and services. http://www.itu.int/home/index.html

Defining the VoIP Protocols

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- **H.323**

  An ITU Recommendation that defines "Packet-based multimedia communications systems". H.323 defines a distributed architecture for creating multimedia applications, including VoIP

- **SIP**

  Defined as IETF RFC 2543. SIP defines a distributed architecture for creating multimedia applications, including VoIP

- **MGCP**

  Defined as IETF RFC 2705. MGCP defines a centralized architecture for creating multimedia applications, including VoIP

- **H.248**

  An ITU Recommendation that defines "Gateway Control Protocol". H.248 is the result of a joint-collaborate with the IETF. H.248 defines a centralized architecture, and is also known as "Megaco"

- **Megaco**

  Defined as IETF RFC 2885. Megaco defines a centralized architecture

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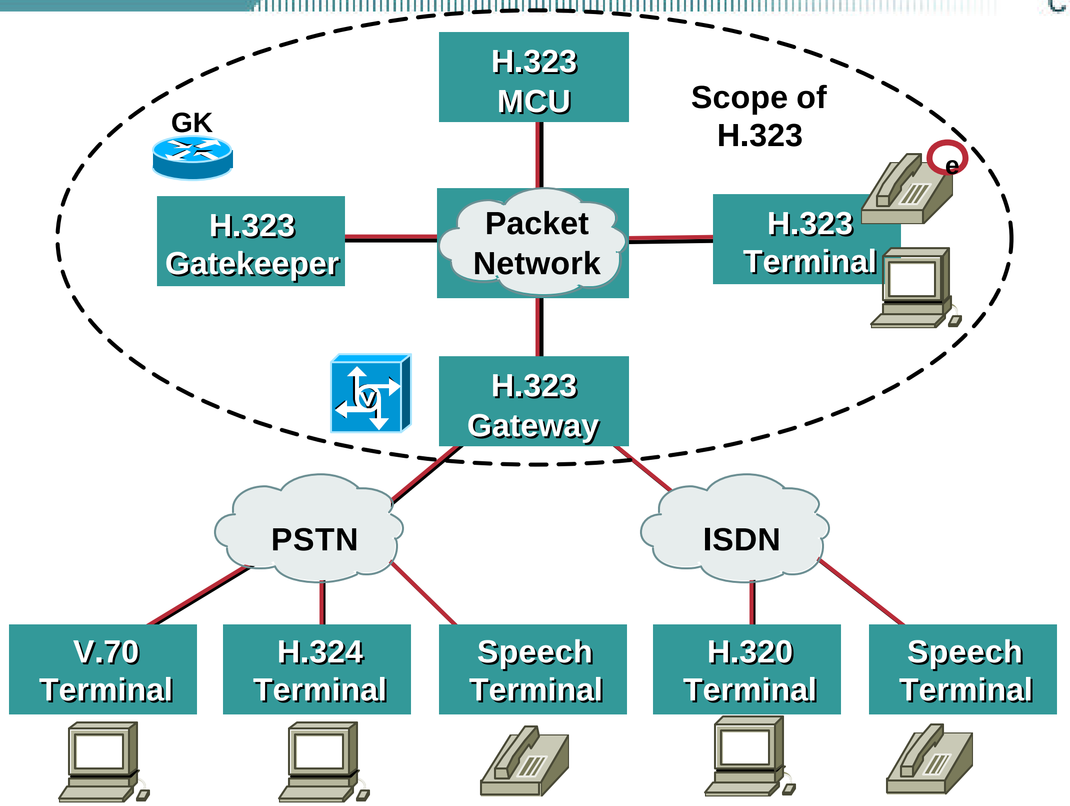
H.323 Components



Scope of H.323 Recommendation

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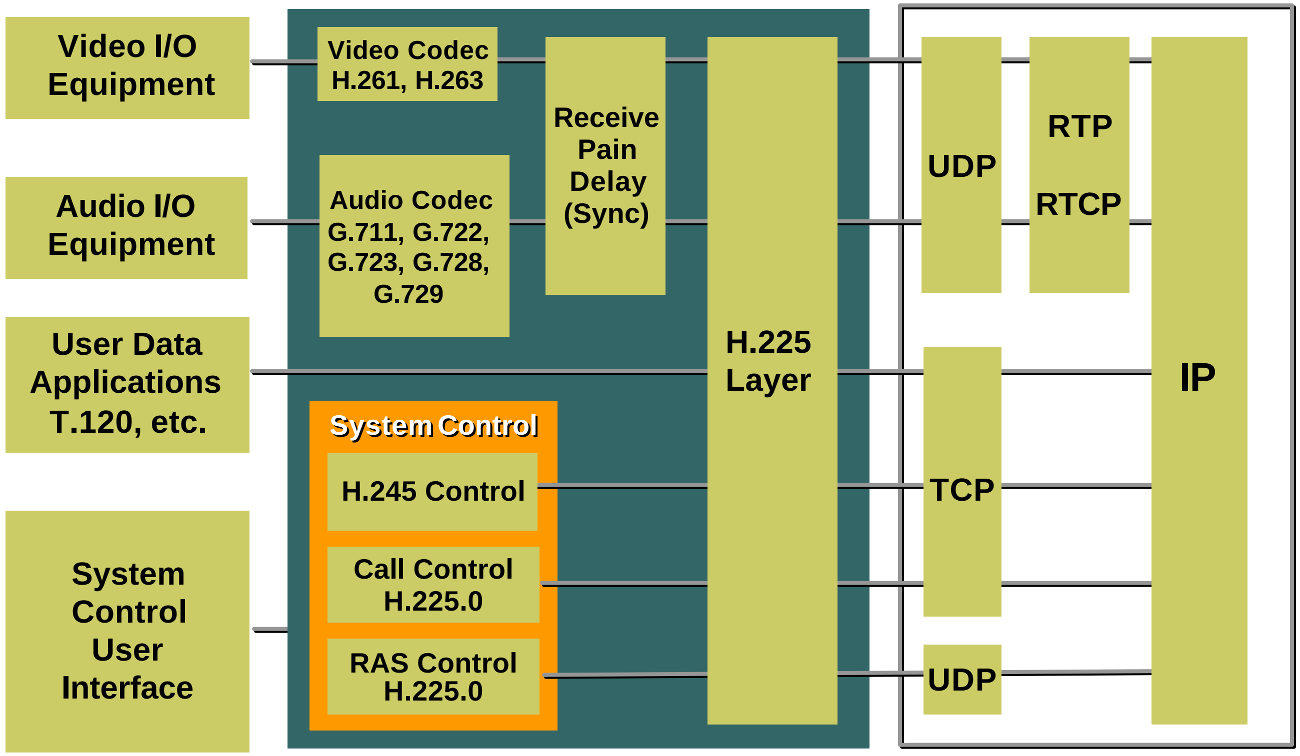
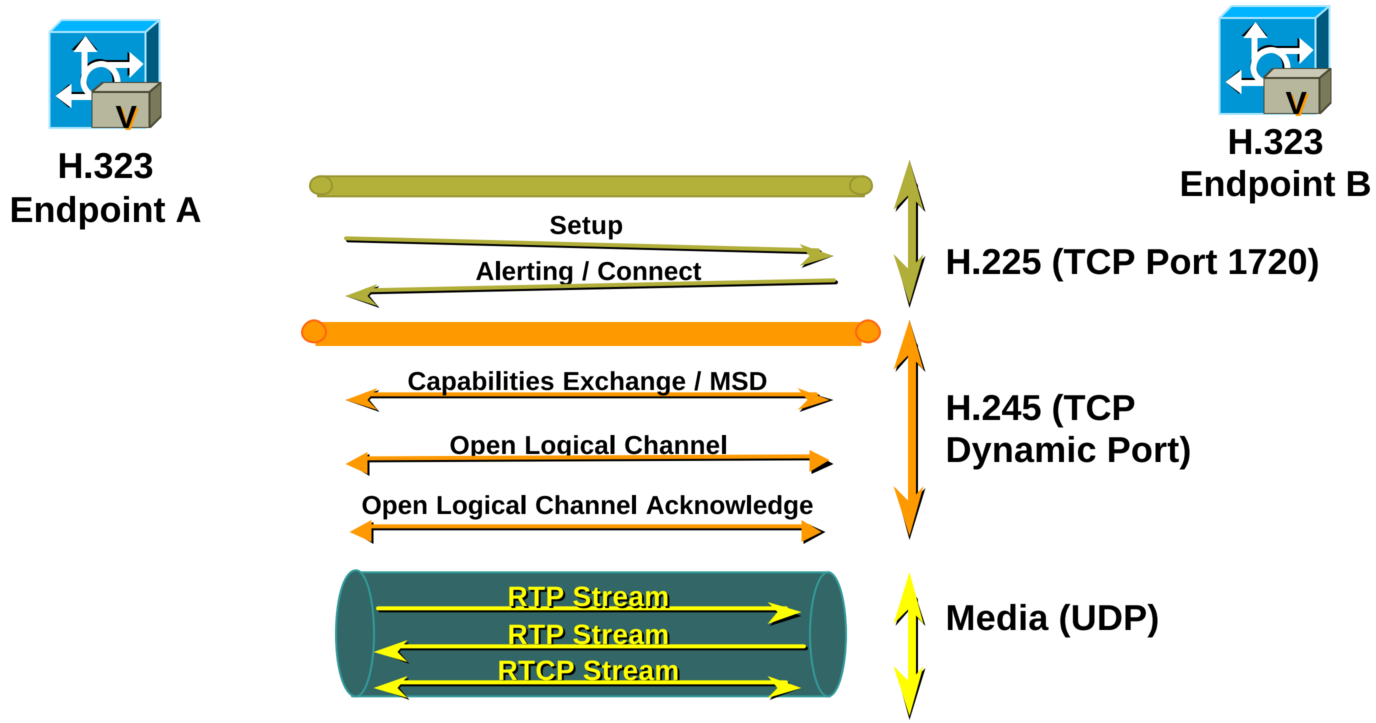


H.323 Signaling



Basic H.323 Call

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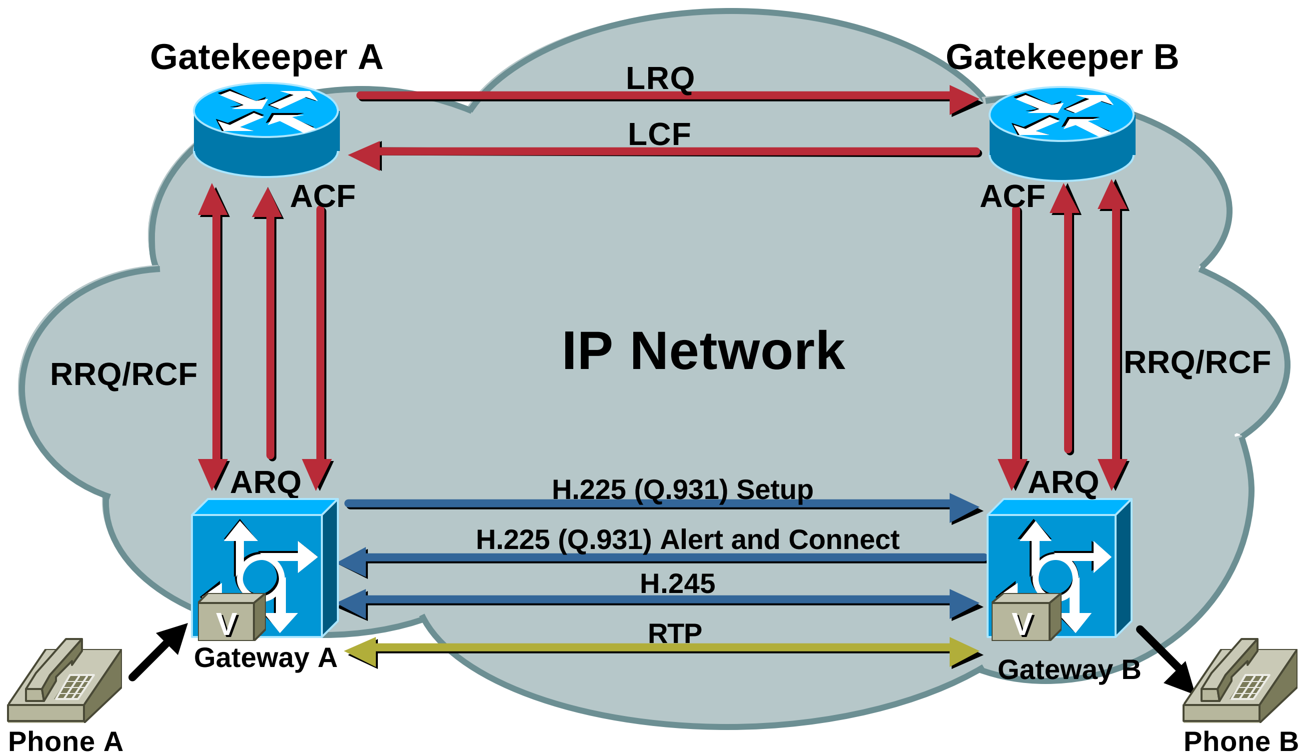

Deploying H.323 Networks

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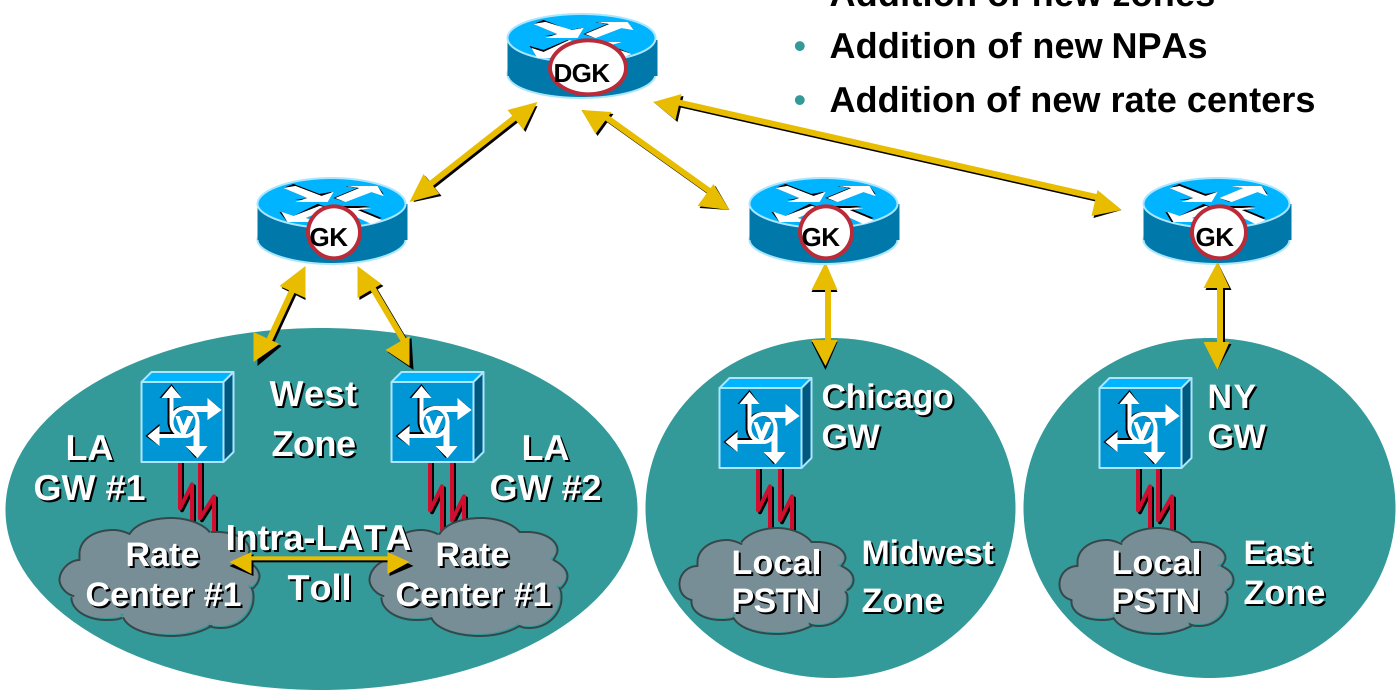
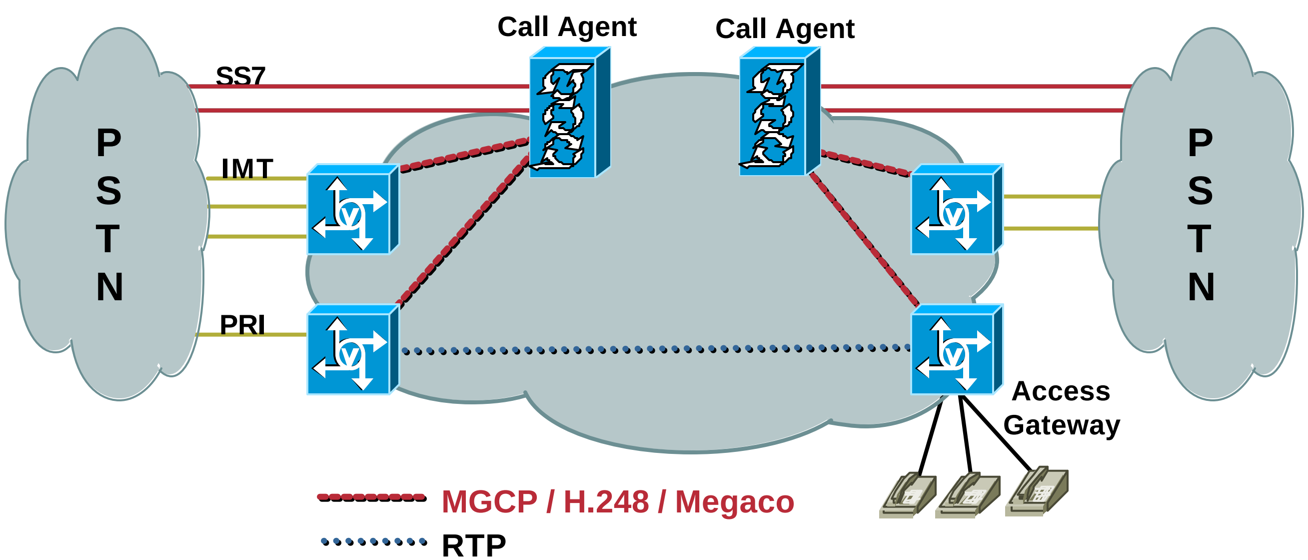
- Minimizes GK configuration
- Addition of new zones
- Addition of new NPAs
- Addition of new rate centers



MGCP/H.248/Megaco—Architectures



Deploying MGCP/H.248/Megaco Networks

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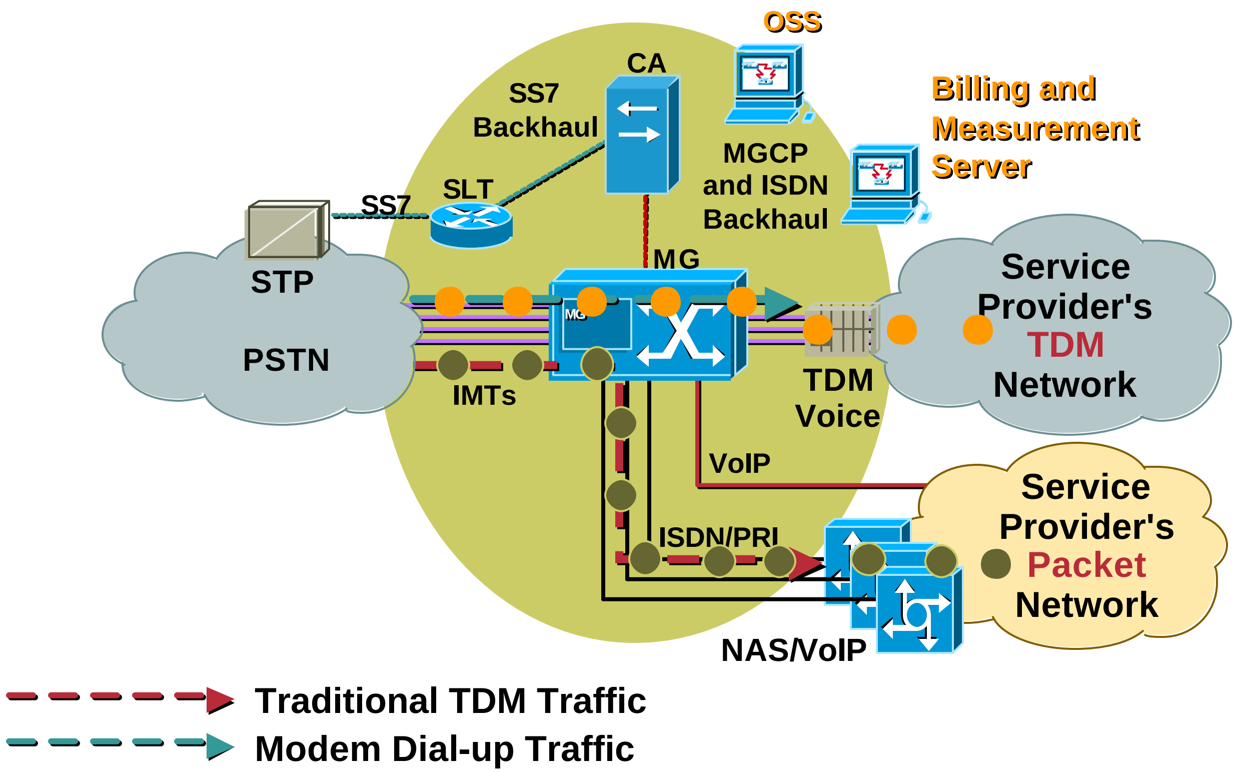
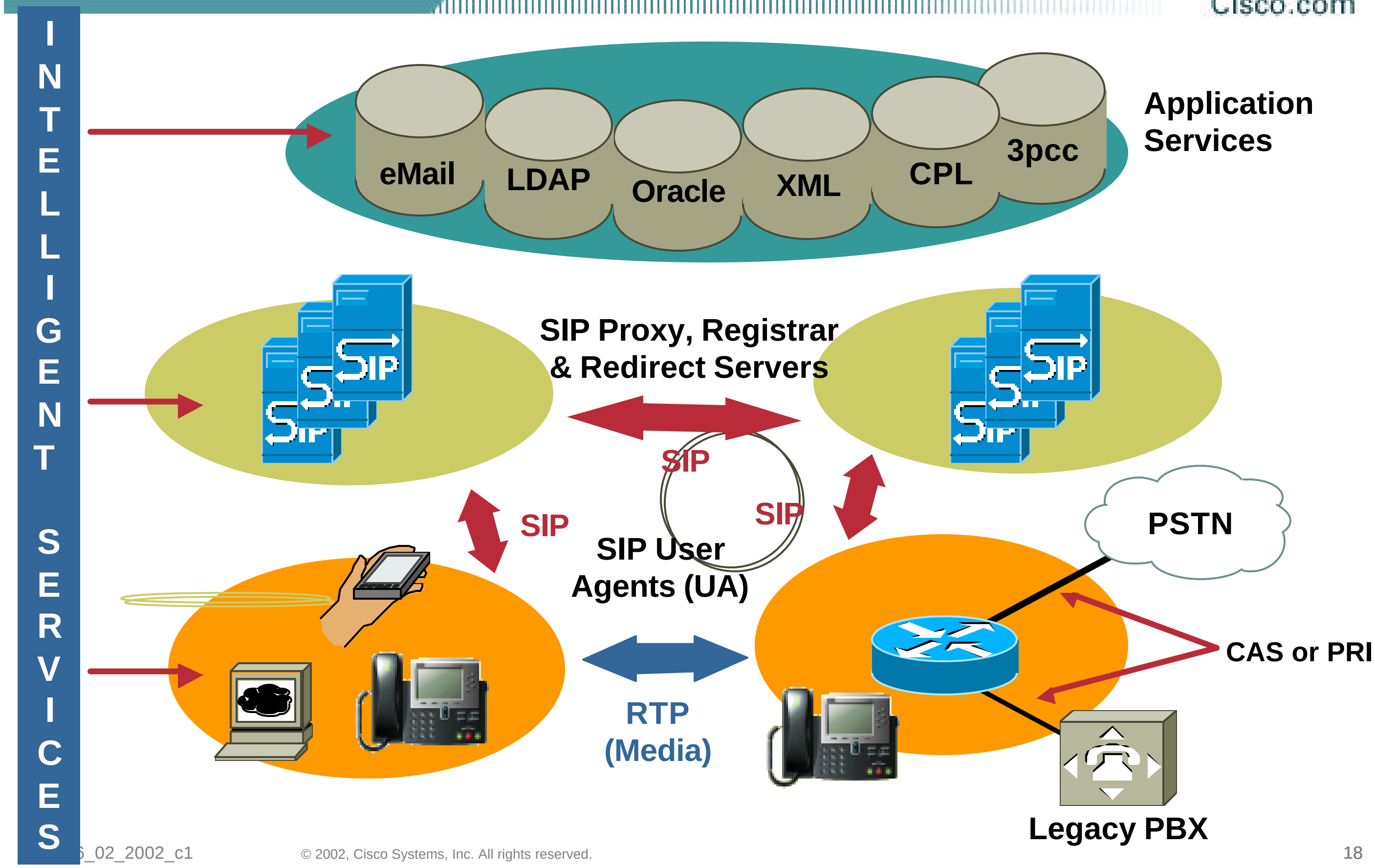
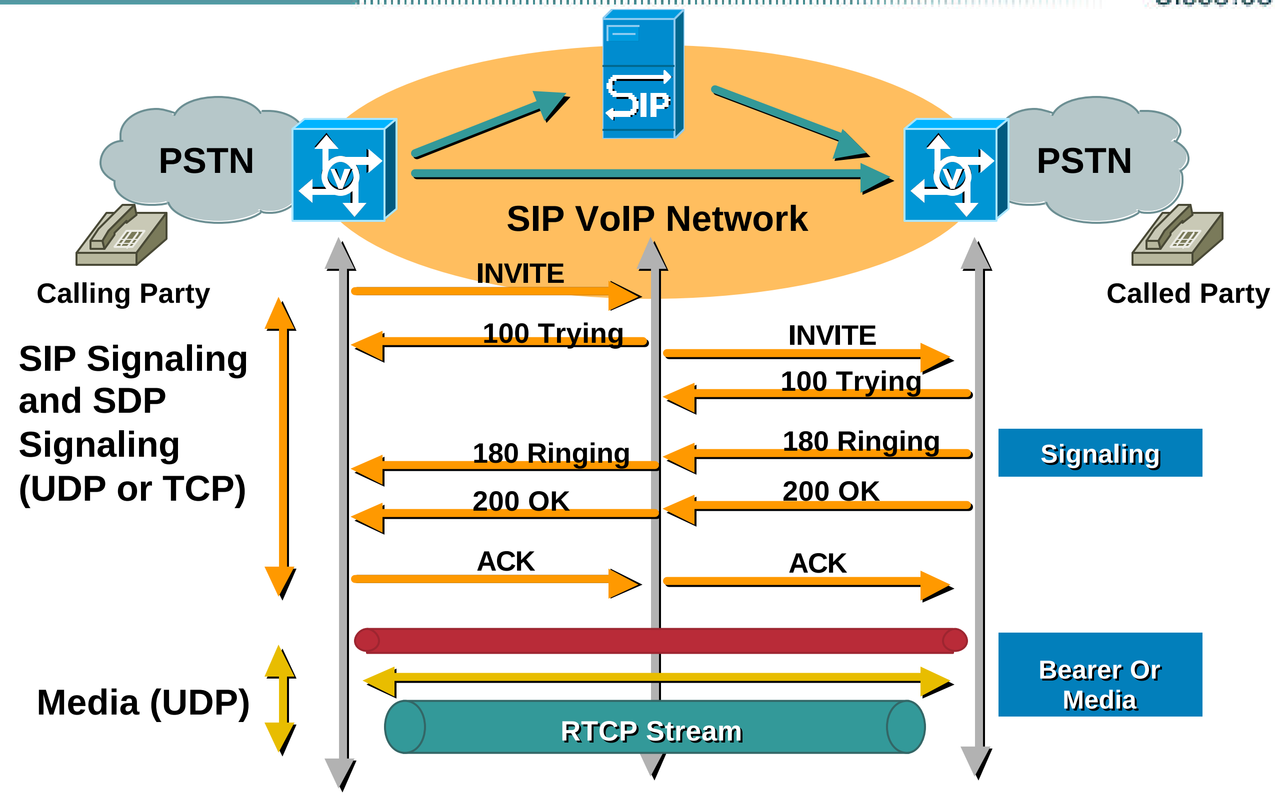


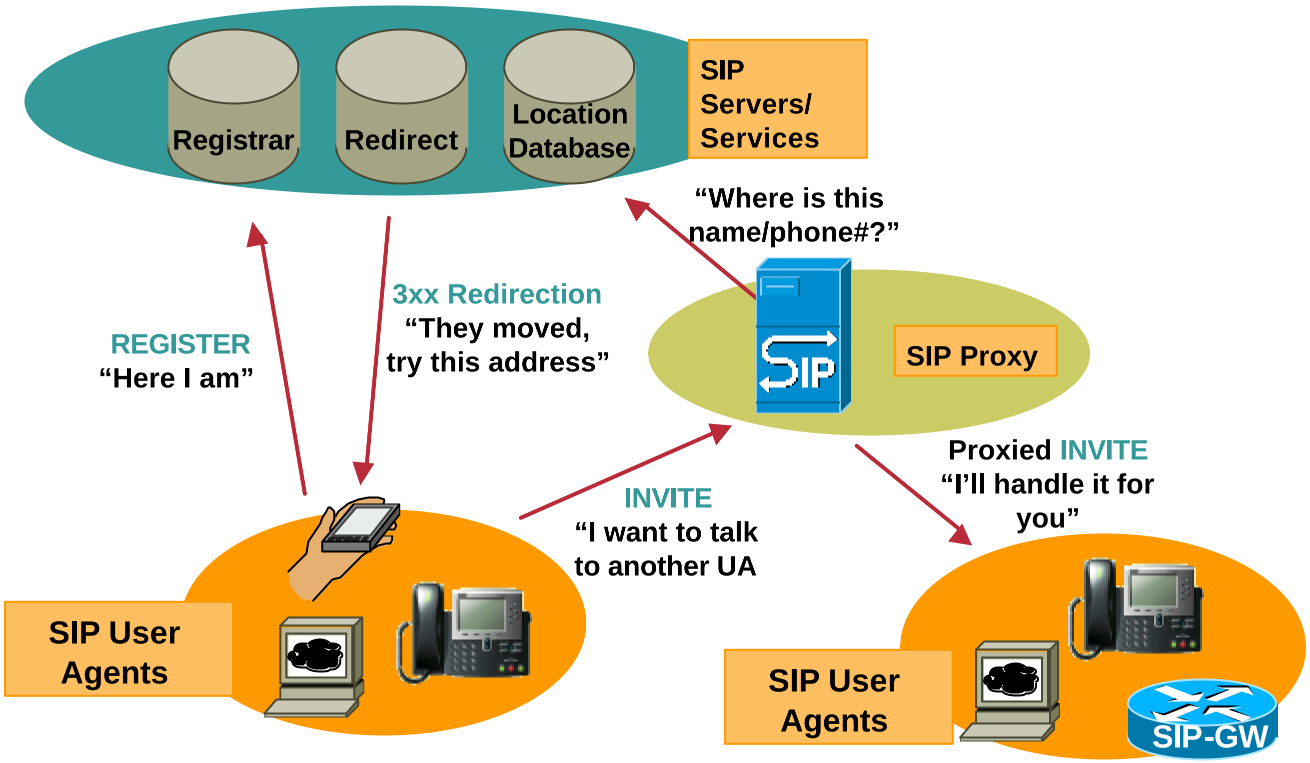
SIP Architecture



SIP Signaling



SIP Servers/Services



Deploying SIP Networks

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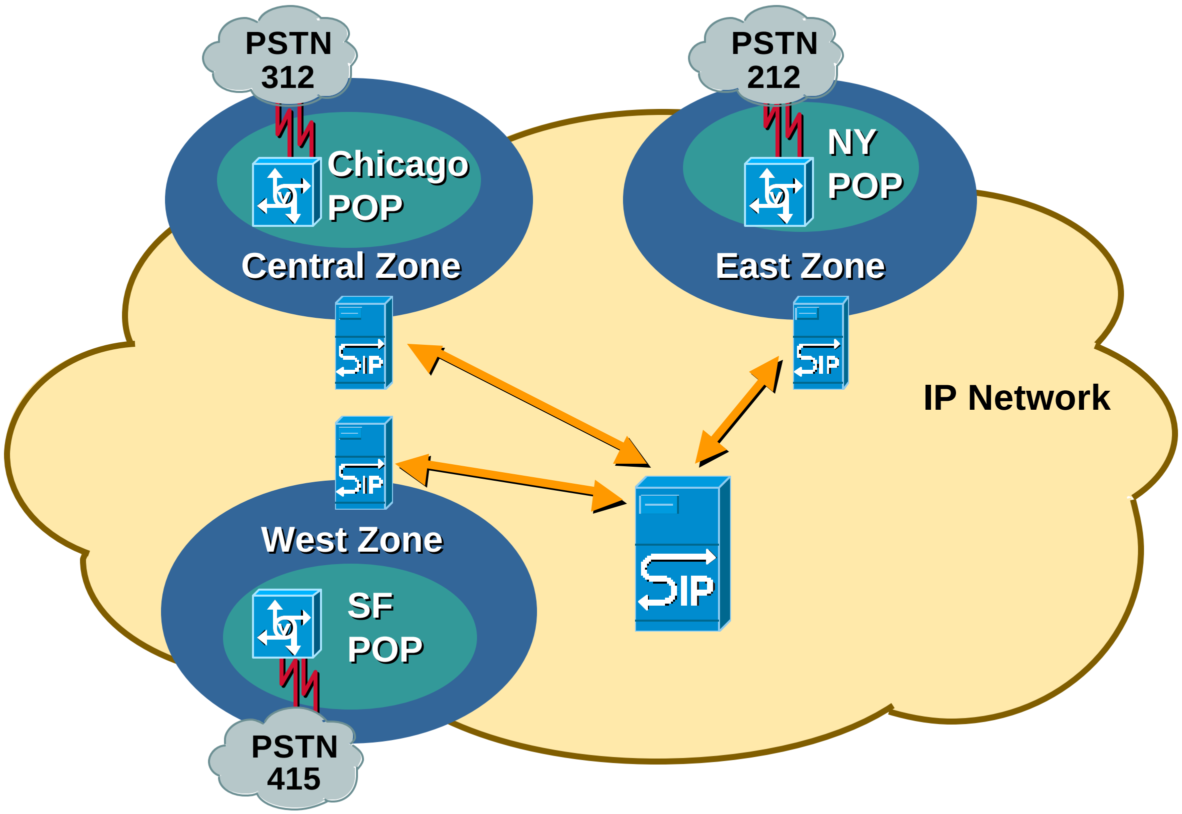

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Voice Myths

Myths

- Networks can only be built one way
- Networks will only use one protocol
- All networks will converge

Facts

- VoIP allows centralized or distributed architectures
- H.323, SIP, MGCP and H.248/Megaco will all be present in VoIP networks
- Networks will converge to IP

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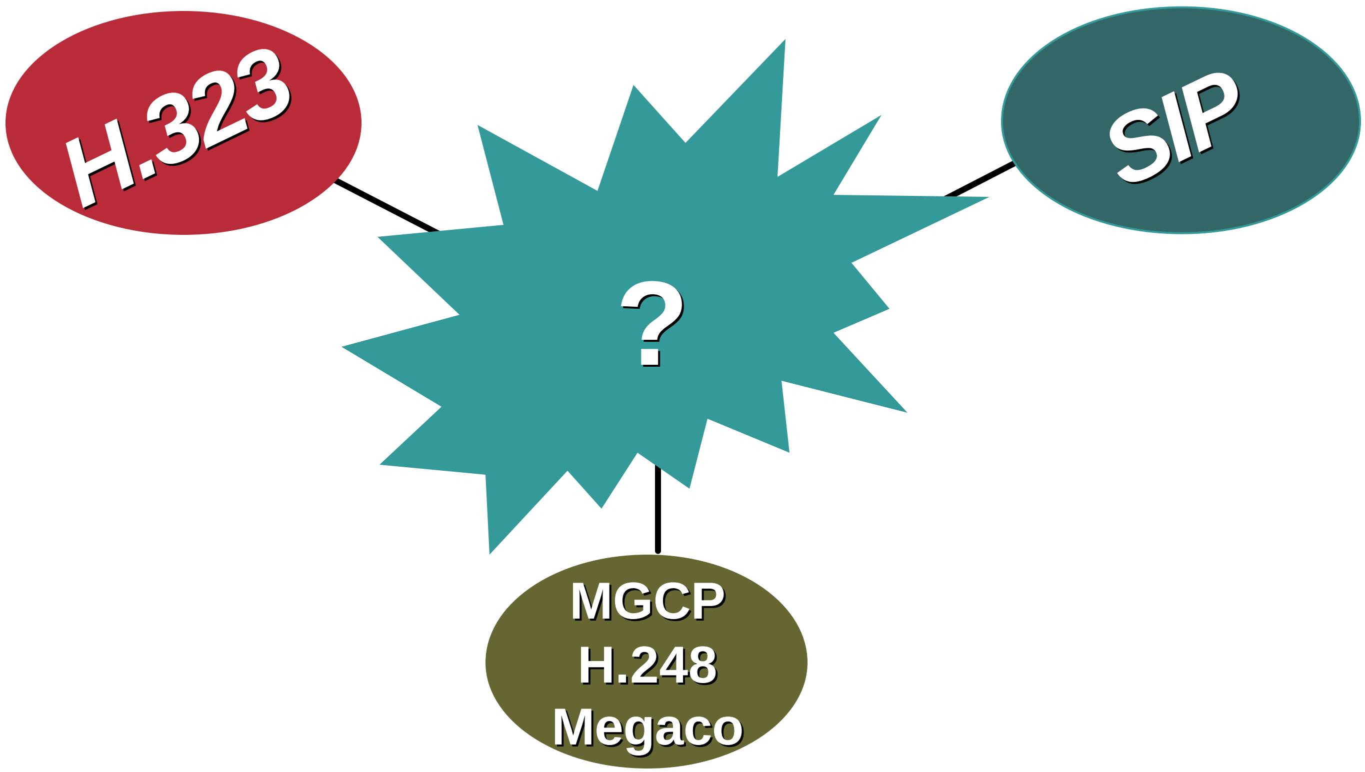
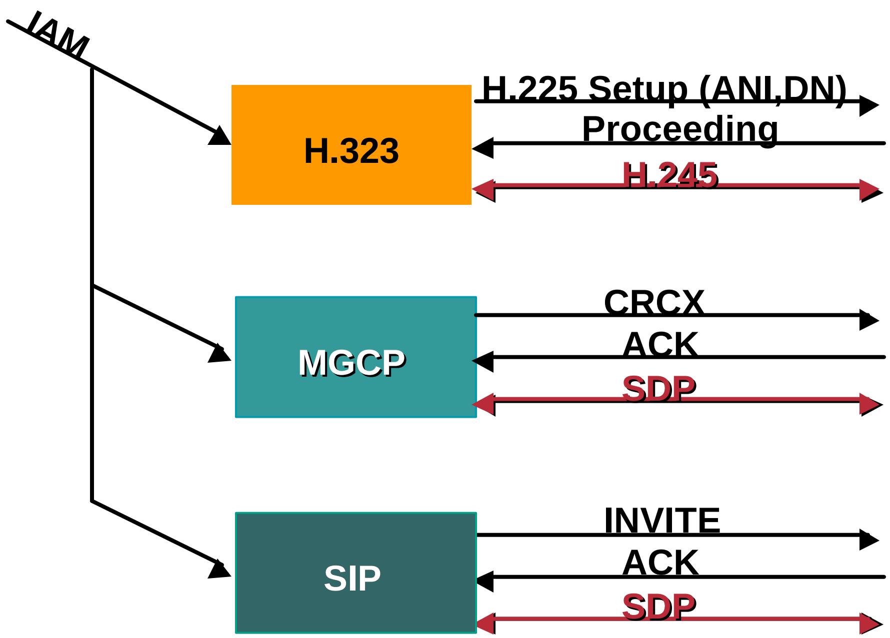
Interconnecting VoIP Networks



Connecting VoIP to SS7/C7 Networks



VoIP Interworking Issues

- **Service interworking**
  - **E.g.: H.450 <-> SIP <-> MGCP**
- **Media interworking**
  - **End-to-end codec negotiation**
- **Bearer interworking**
  - **End-to-end fax, modem, DTMF**

VoIP Interworking

- **Bearer level**
  - Modem (relay/passthru)
  - Fax (relay/passthru)
    - T.38
    - T.37
  - DTMF (relay/passthru)
- **Media level**
  - Codec (negotiation, selection)
- **Service translation issues**
  - Call deflection
  - Park/hold
- **Signal issues**
  - SDP
  - H.245

Fax and Modem Passthru Mechanisms

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- Modem and fax are control mechanisms based on PLL (Phase Locked Loops)
- They are both time sensitive
- Highly sensitive to packet network impairments:
  - Jitter
  - Packet loss
  - Delay
- Susceptible to clock slew (clock sync differences between gateways)

Passthru Simplified

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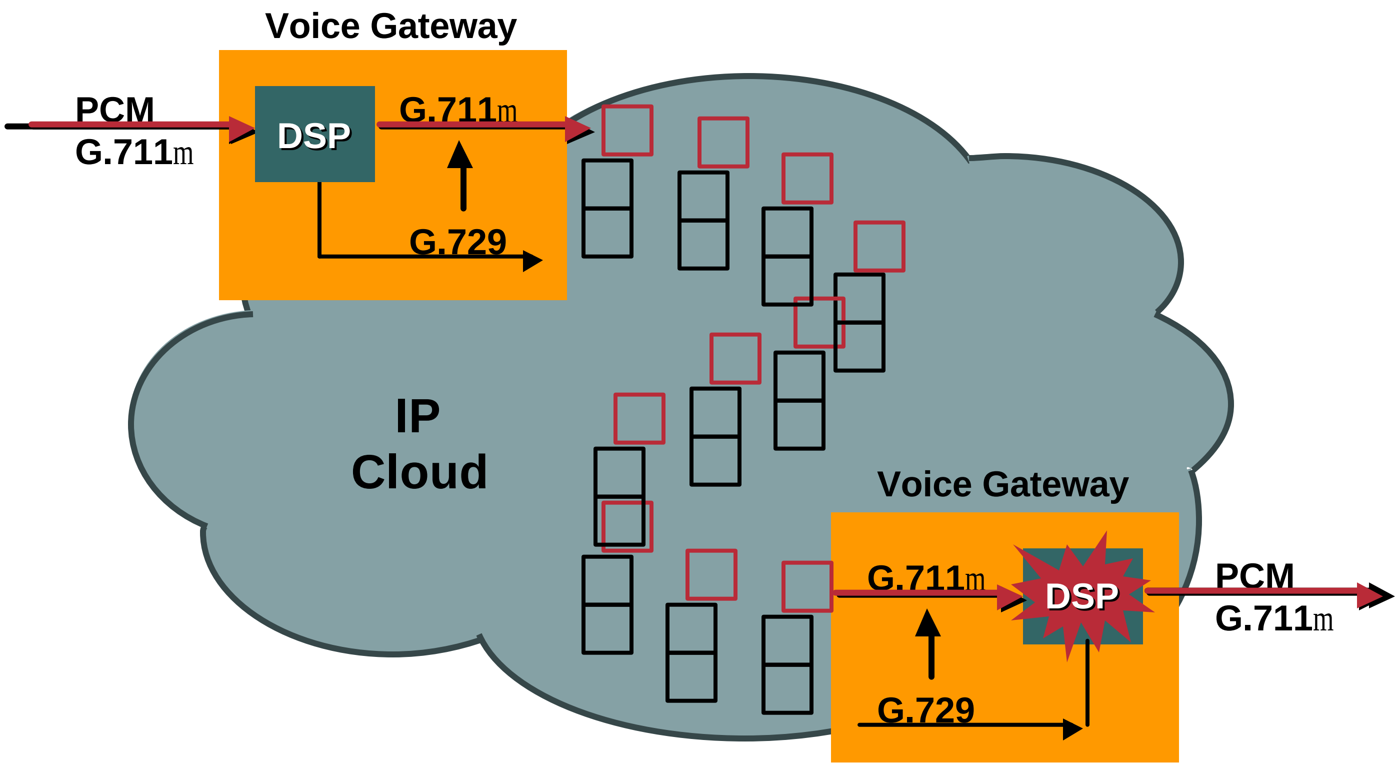

What Is Modem Passthru?

- **It is the transport of modem signals (modulation, error correction and compression) through a packet network using PCM encoded packets**

Modem Passthru (Cont.)

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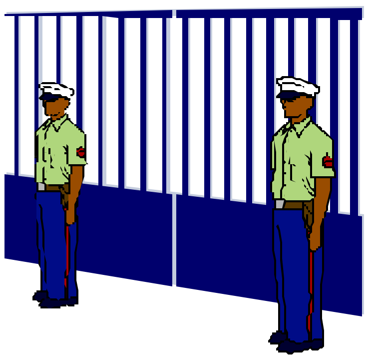
- Modem tone detection (<= V.90)
- Switchover signaling
- No VAD
- EC off
- RTP payload redundancy (10ms packetization) RFC2198 (optional)

Modem Passthru Issues

- Consecutive packet drops (loss) cause retrain
- Consecutive drops during retrain causes disconnect
- Variation of delay (jitter) has quite an effect
- Jitter (at 10%) is a conservative estimate—Since jitter mostly impacts performance with packet loss



What Is Modem Relay?

- Modem relay involves demodulating the modem signal at ingress gateway
- Passing this data as packet data to terminating gateway
- Re-modulating the data and passes it to the receiving modem

Fax Relay—T.38

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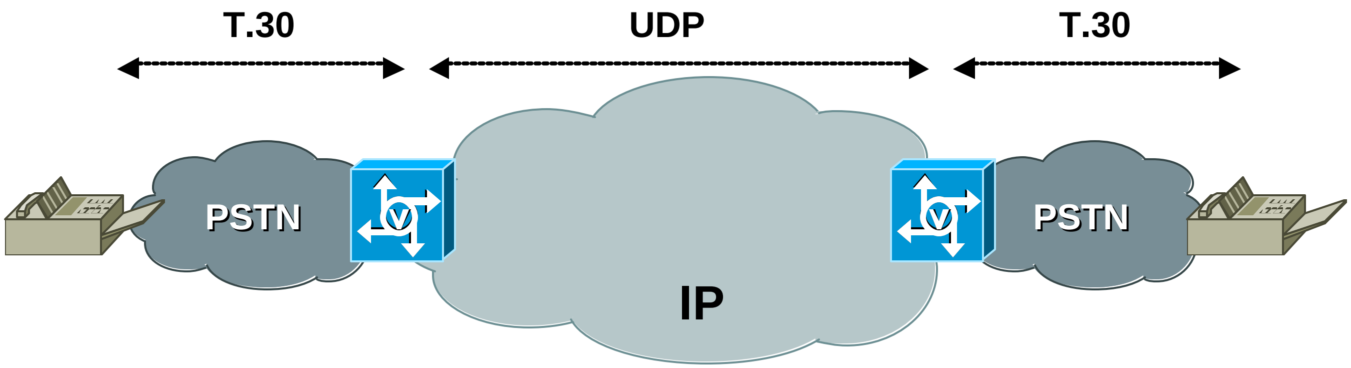

- **Real-time**
- **Also called demod/remod**
- **Can be used in H.323/MGCP/SIP signaling**
- **Delivers fax data over UDP streams (uses same RTP port)—reuses voice UDP ports**
- **Fallback to proprietary mode**
- **Method of encoding the T.30 and T.4 into packets**

DTMF

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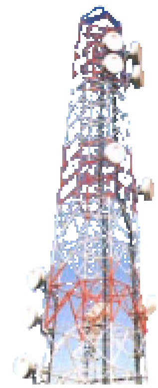
- **What is DTMF**
- **Why is it required? and where is it used?**
- **How do you transport it in IP?**
- **DTMF implementation**



DTMF (Cont.)

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- In TDM world, all voice traffic is sent as uncompressed 64Kbs PCM streams; anything sent on that circuit is an untouched stream of bits; (e.g., voice speech, modem tones, fax tones, and DTMF digits)
- DSP codecs designed to interpret human speech, can distort DTMF tones (machine-tones)
- High b/w codecs less likely to distort
- Distortion causes problems with voicemail and IVR systems

DTMF Schemes with VoIP Protocols

	H.323	MGCP, H.248, Megaco	SIP
In-Band	In-Band	In-Band	In-Band
Out-of-Band	Cisco RTP, H.245 Alphanum, H.245 Signal, AVT Tones RFC2833	Cisco RTP, NSE, NTE,RFC2833	RFC2833

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Summary

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- **Understand the possibilities and the issues**
- **Avoid protocol/product based bias**
- **Decide on application**
- **Consider market and business drivers**
- **Deploy what's possible today**
- **Choose signaling protocol depending on services intended to be offered**
- **Many possibilities—stay tuned**

Crystal Ball on VoIP

- All three protocols (or its variations) are here for the long run
- Changes/enhancements will be made
- IP will be the core

Reference URLs

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- **ITU:** **www.itu.org**
- **IETF:** **www.ietf.org**
- **SIP:** **www.cs.columbia.edu/~hgs/sip/**
- **H.323:** **www.packetizer.com/iptel/h323/**
- **MGCP:** **www.softswitch.org/asp/techlibrary_protocol.asp?page=techlibrary**

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Empowering the Internet Generation