

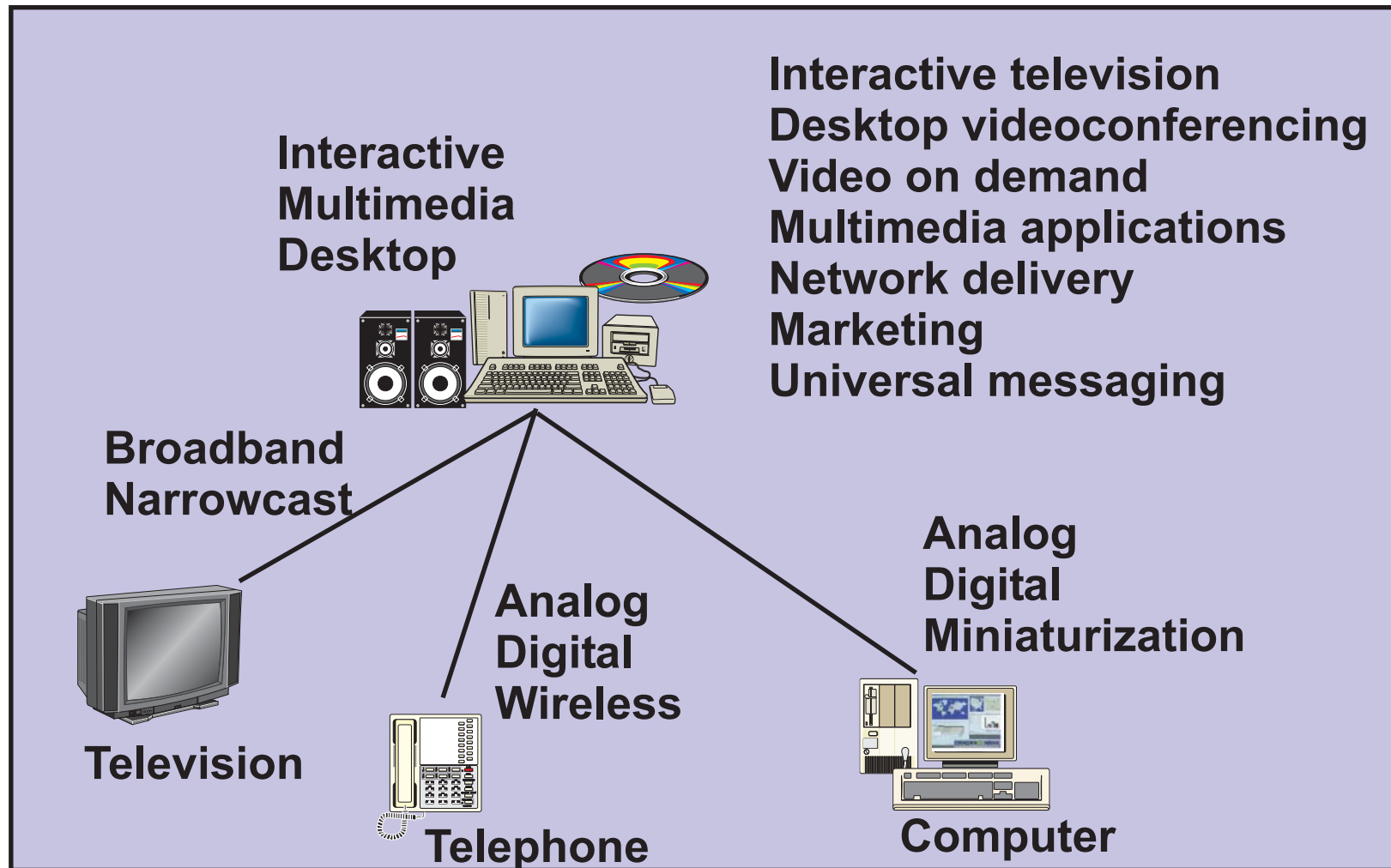
Voice Over IP

Session 134



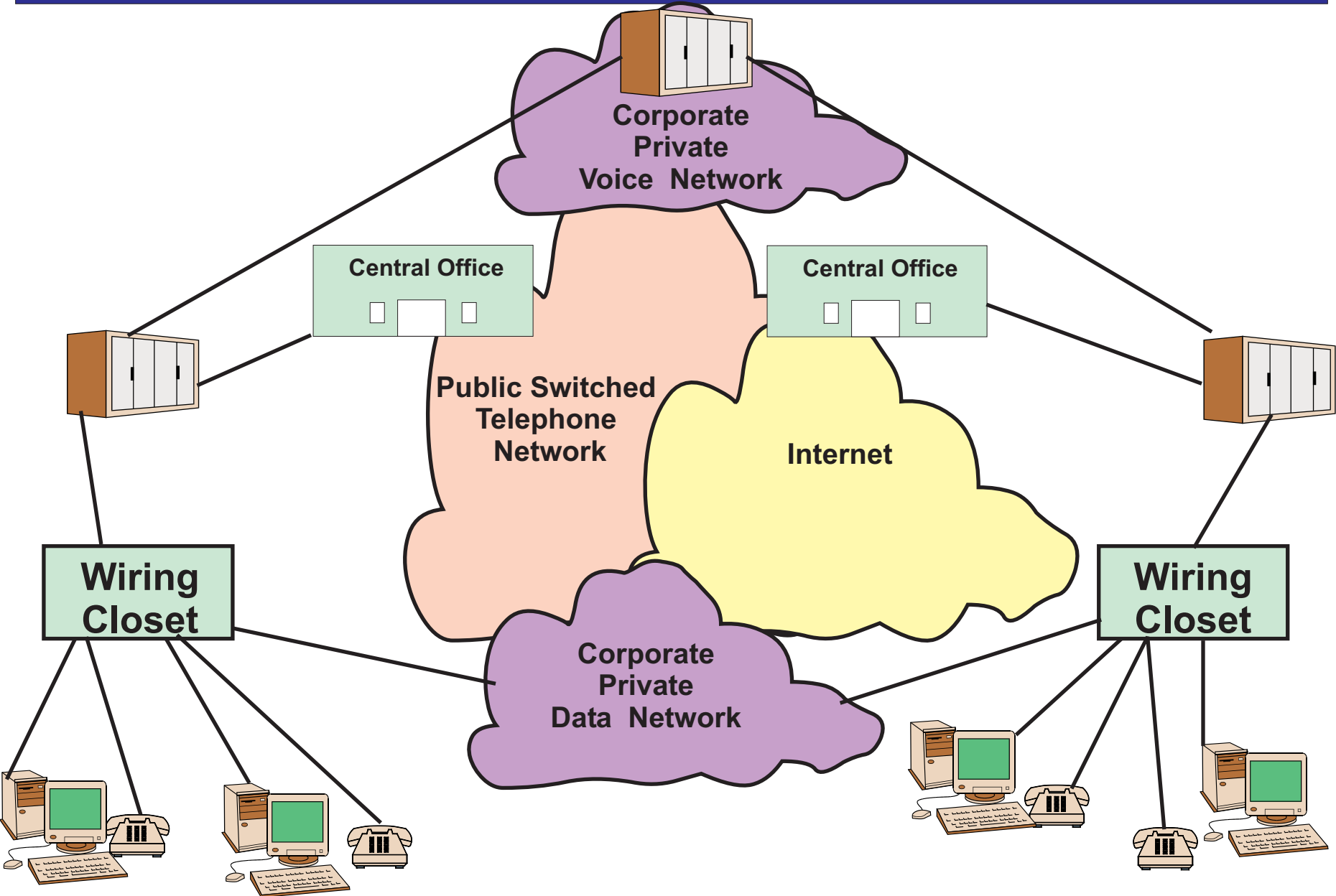
Laura Jeanne Knapp
IBM Technical Evangelist
1-919-224-2205
laura@lauraknapp.com

Technology Convergence



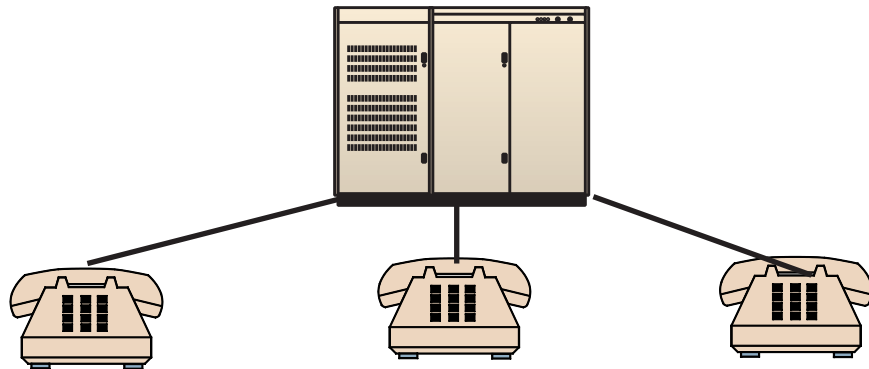
What about Network convergence?

Voice and Data Networks



Voice and Data Switch Similarities

Class 5 Switch



Handset aggregator

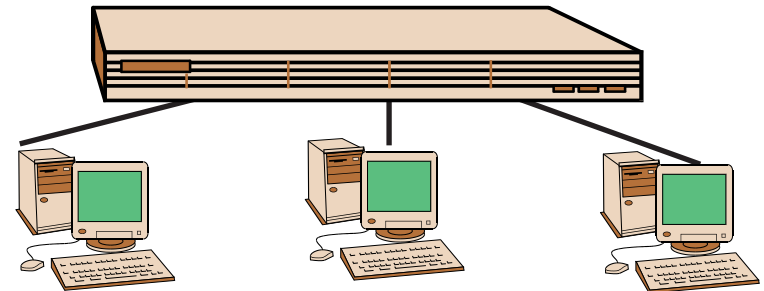
All telephones get a single analog/digital line

All devices have a phone number defined on the switch

Traffic engineering provides non-blocking

Path selection based on static least cost routing

Multilayer Switch



Computer aggregator

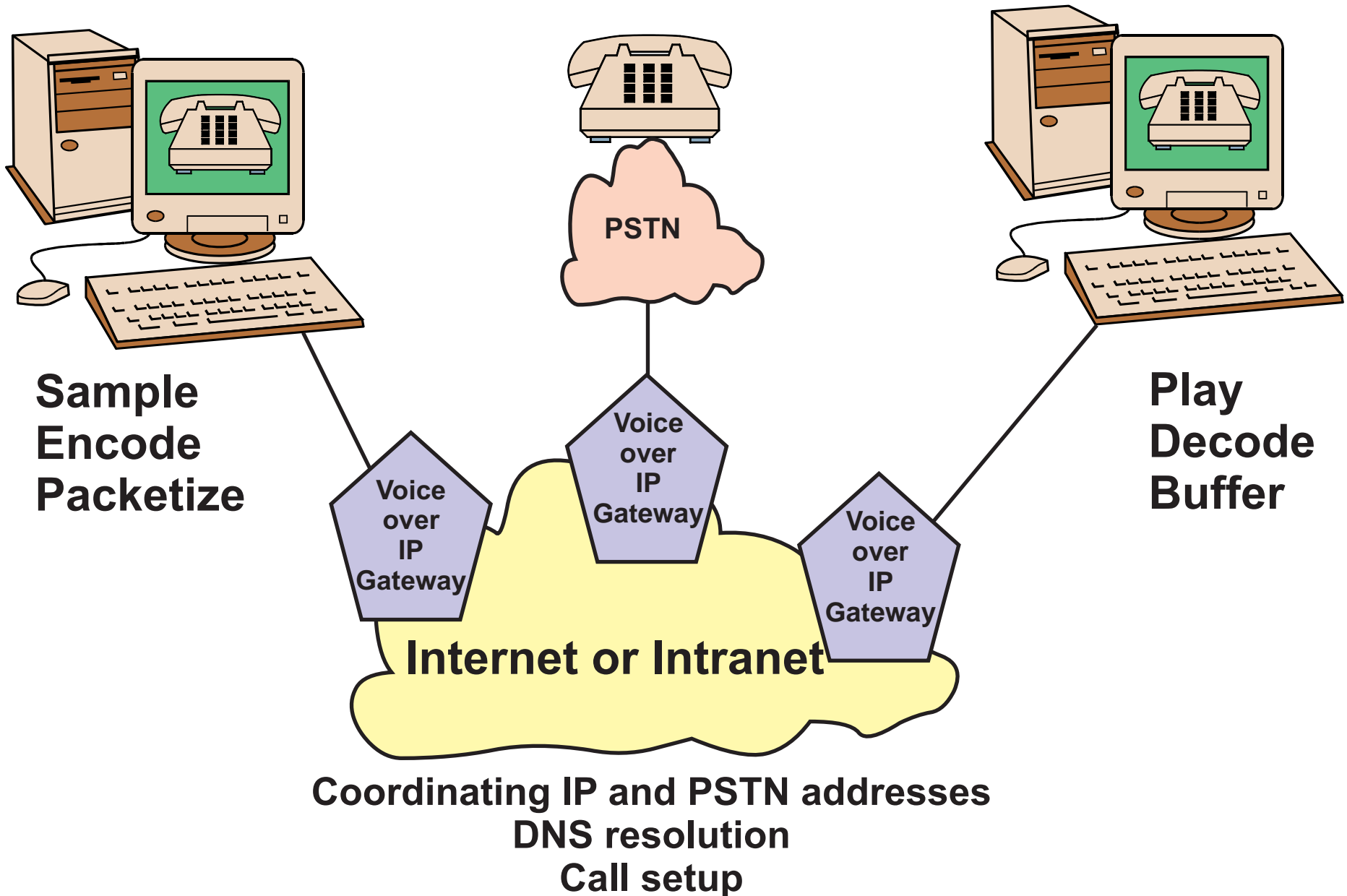
All devices get dedicated bandwidth

All devices have an IP address defined on the host

Non-blocking devices available

Path selection based on dynamic routing protocol

Providing Voice Functions



Choice of Solutions

H.323

SIP - Session Initiation Protocol

MGCP - Media Gateway Control Protocol

You need to understand what technologies vendors support as you make your decisions

What do their endpoints support for signaling?

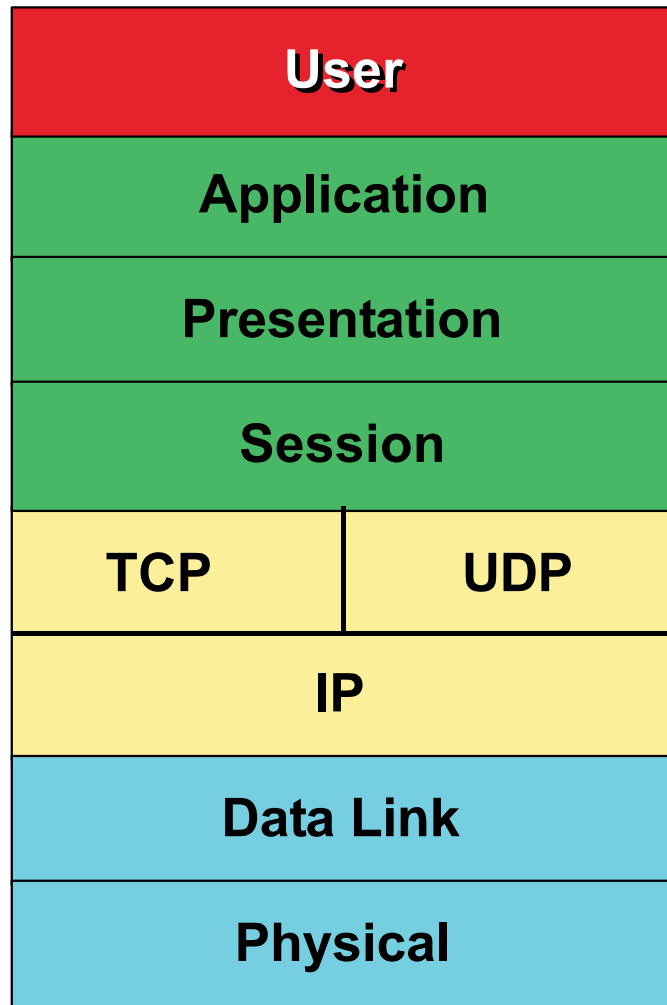
What do their gateways support for signaling?

What do their call centers support for signaling?

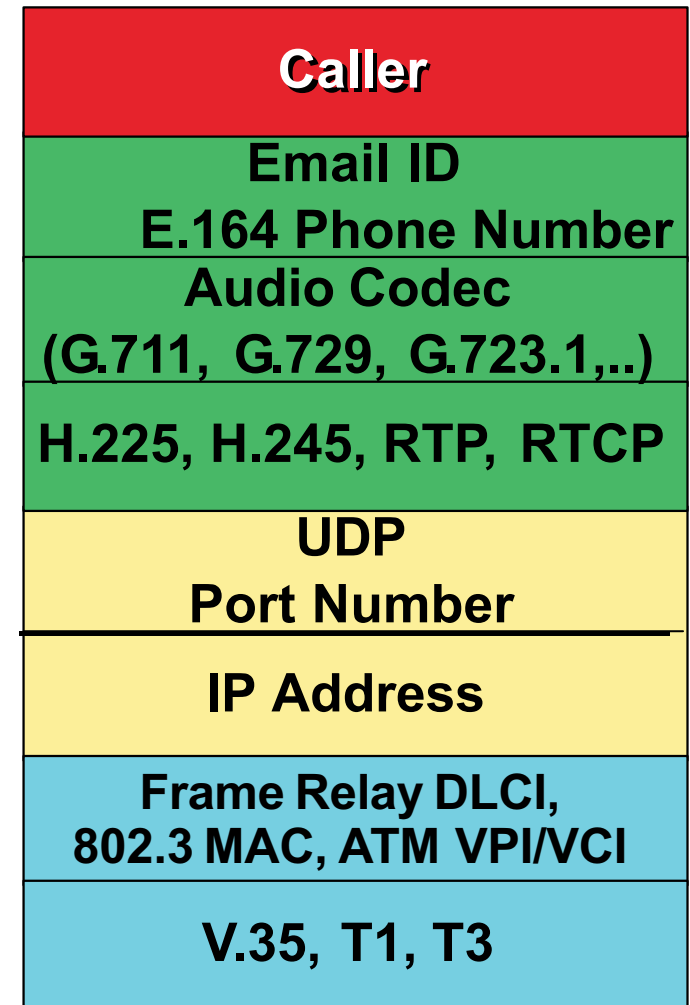


H.323 Structure and IP

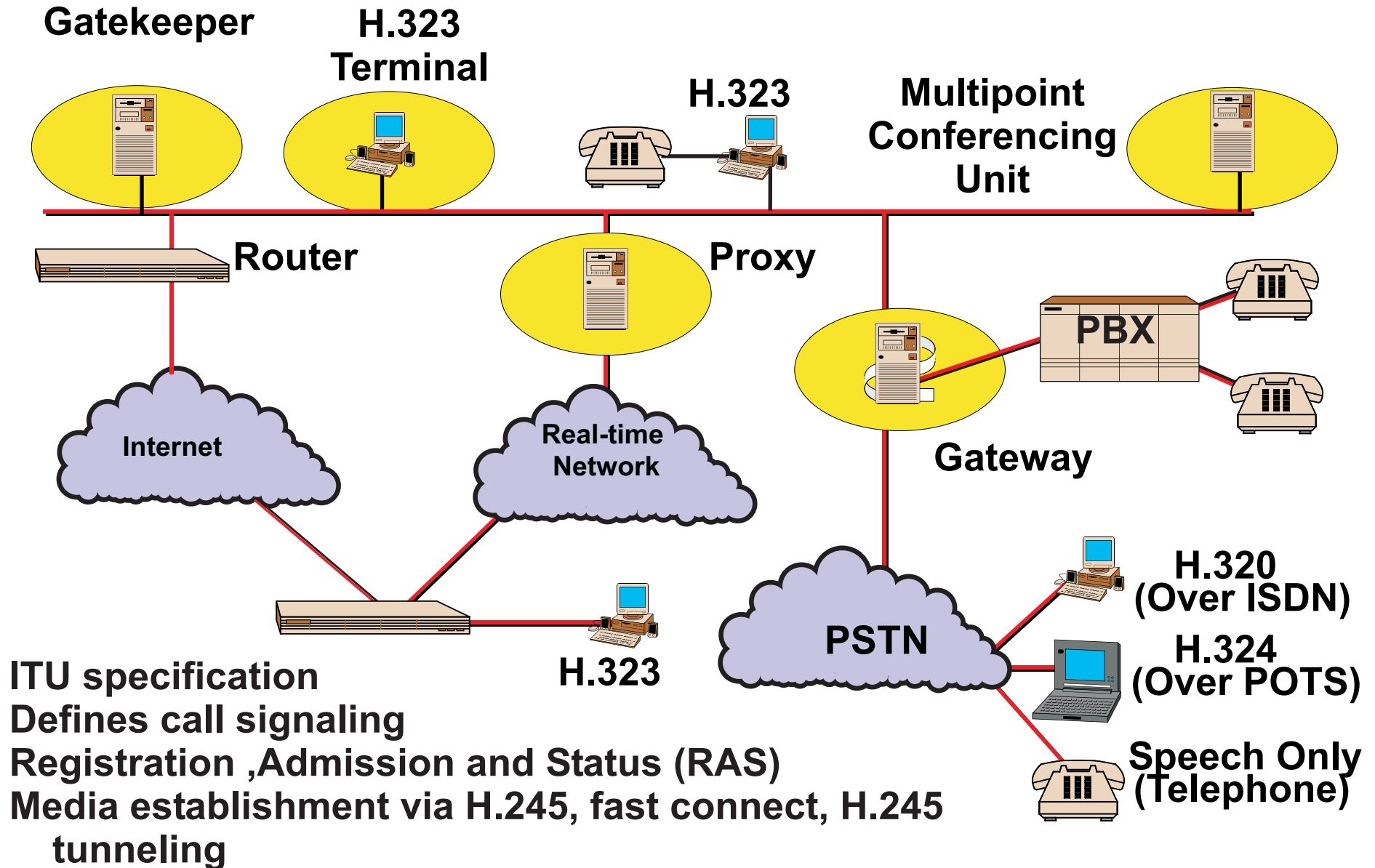
IP Layered Model



H.323 VoIP Model

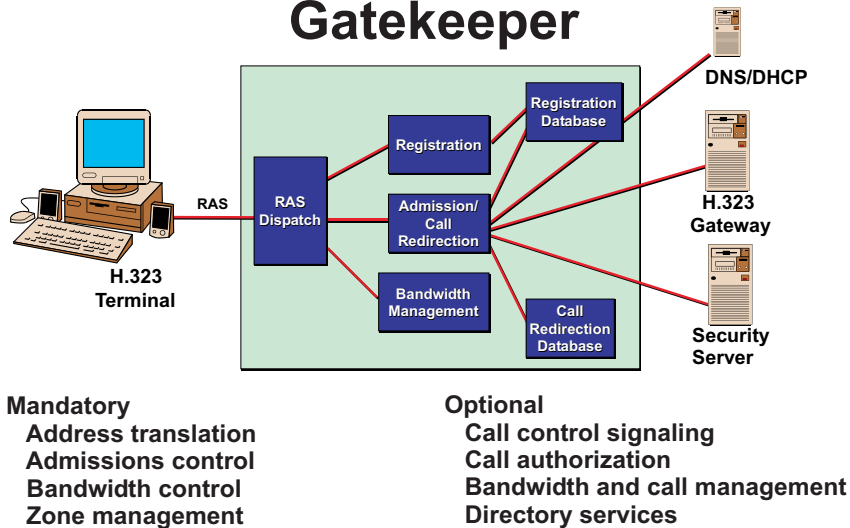


H.323 Infrastructure

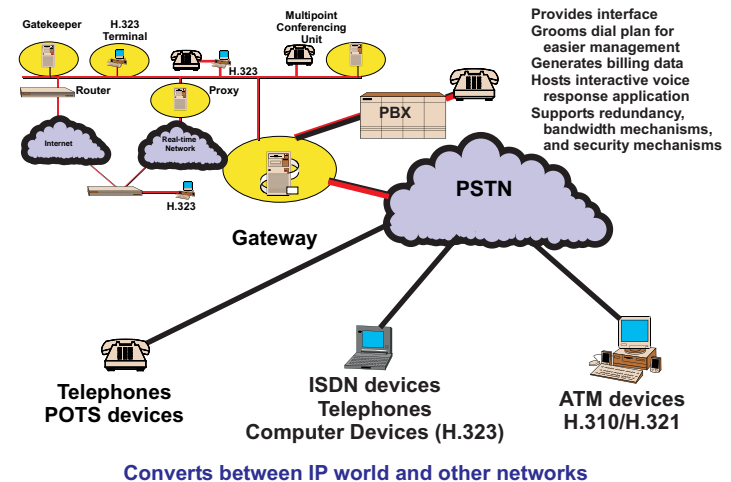


H.323 Components

Gatekeeper



Gateway

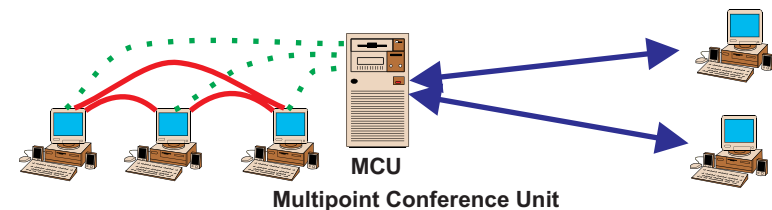


H.323 Terminal

| AV App | Terminal Control and Management | | | | Data App |
|----------------------------|---------------------------------|--|--------------------------|-------|----------|
| G.7XX Audio Codecs | RTCP | H.225.0 Terminal to Gatekeeper Signaling (RAS) | H.225.0 Call Signaling | H.245 | T.120 |
| RTP | | | Reliable Transport (TCP) | | |
| Unreliable Transport (UDP) | | | | | |
| Network Layer (IP) | | | | | |
| Link Layer | | | | | |
| Physical Layer | | | | | |

Discovery to find Gatekeeper to use
Registers with Gatekeeper
Microsoft Netmeeting is often used H.323 compliant client
H.261 and H.263 provide video access

Multipoint Conference Unit

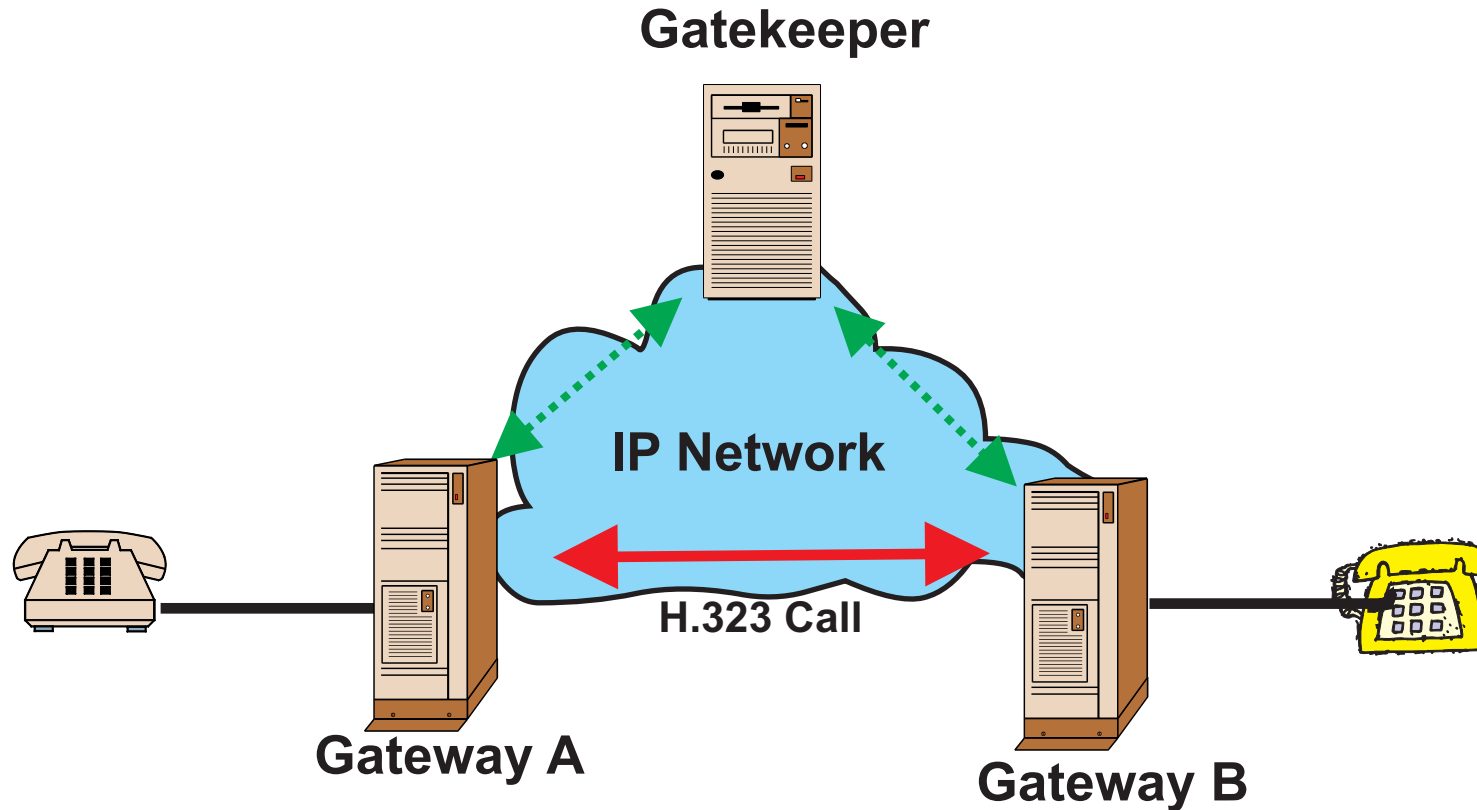


Decentralized site
Control processing
Terminal-terminal audio communication

Centralized site
Control processing
Sends audio to participating terminals

Negotiates capabilities to ensure common level of communications
Supports decentralized and centralized multipoint conferences
Can control resources in multicast operation

H.323 Registration



**Gateways register with gatekeeper
(Knew gatekeepers address via either static or auto configuration)**

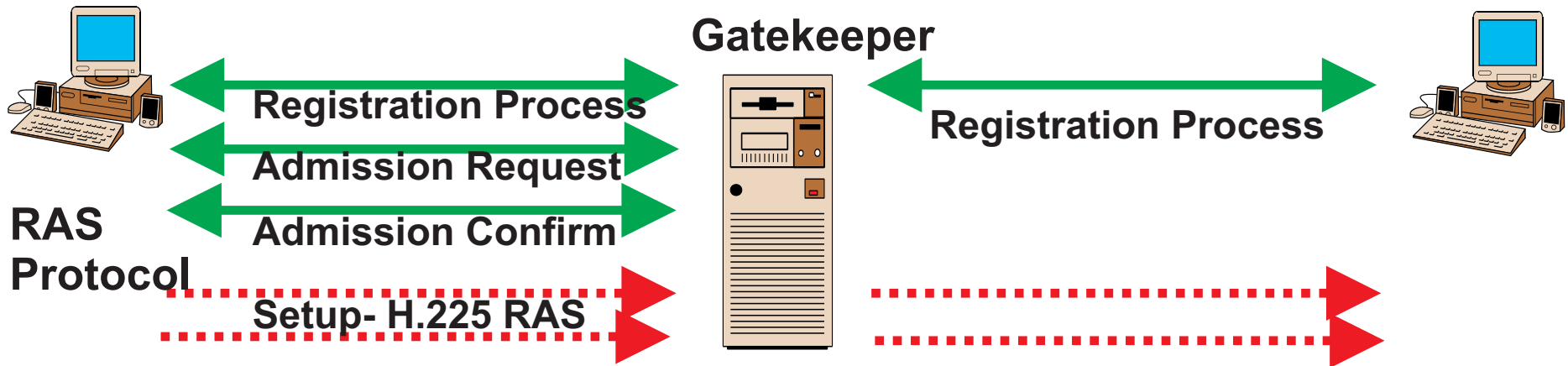
**Gateway A makes request to place a call, receives confirmation and
receives IP address of gateway B**

Gateway A places call to gateway B

H.323 Signaling

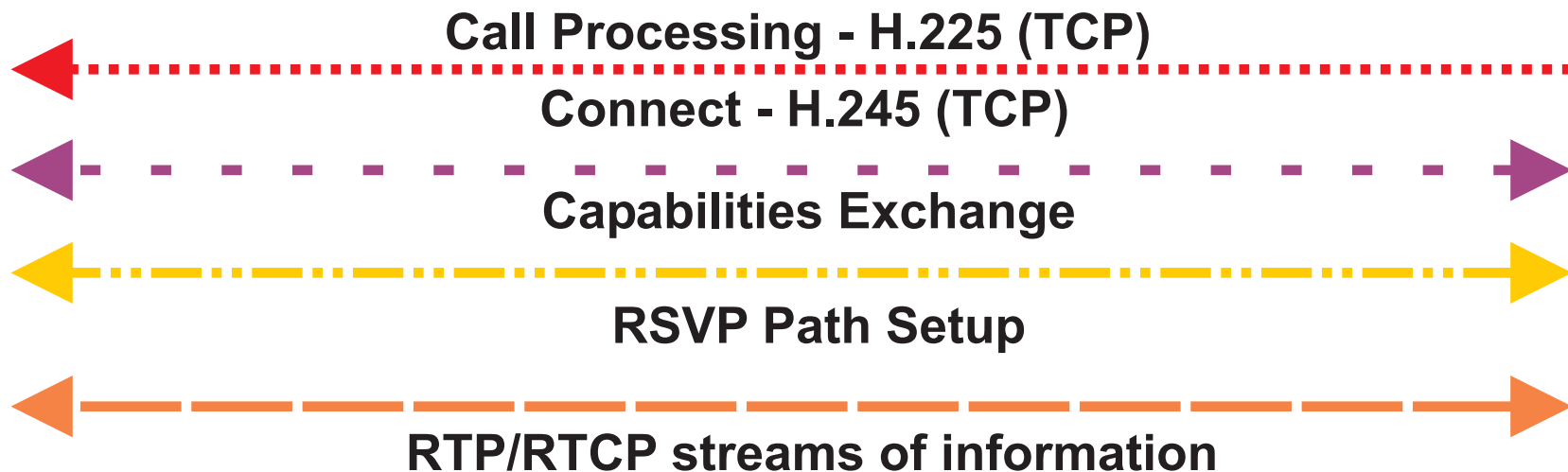
IP Telephony Application

IP Telephony Application

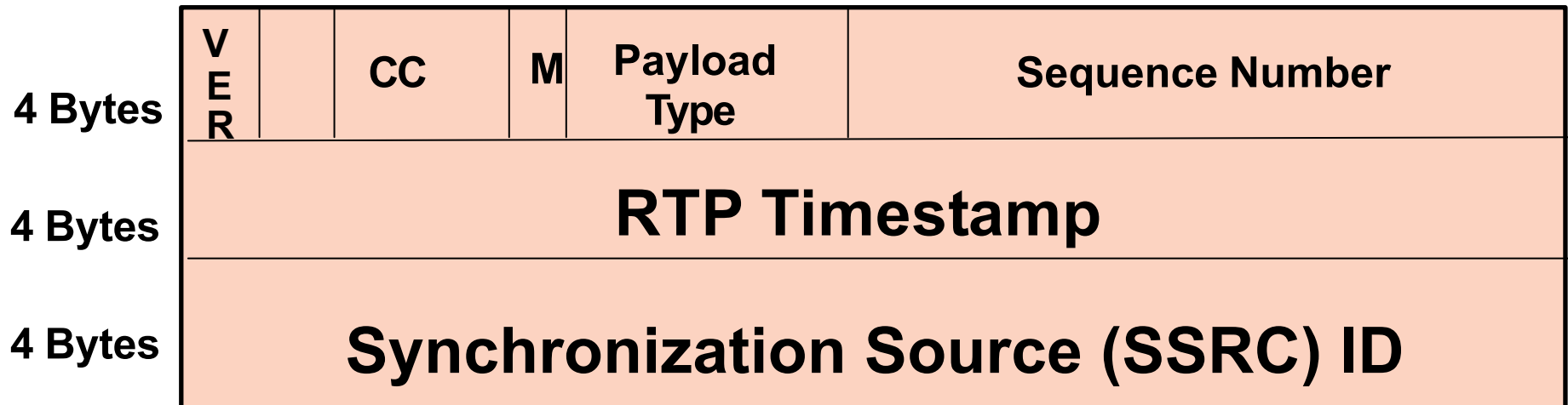


RAS
Protocol

This process can take as many as 20 requests/replies to set up the call!
How long will people wait to hear a "ring"?
Reduced with version 2, but led to development of other solutions



Real Time Protocol with Compression



Provides feedback on quality of data distribution

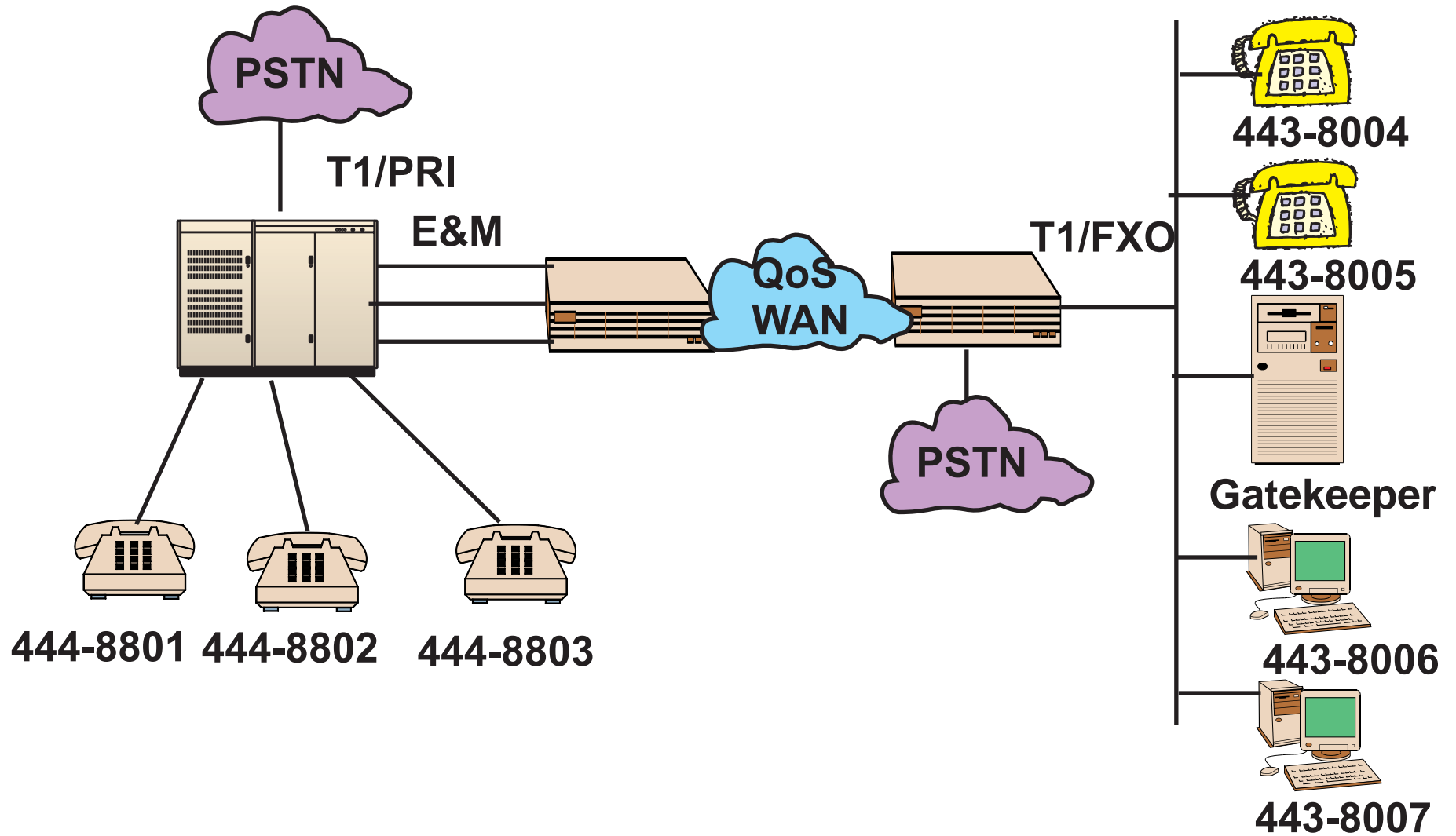
Tracks participants in an RTP session

Limits feedback send rate

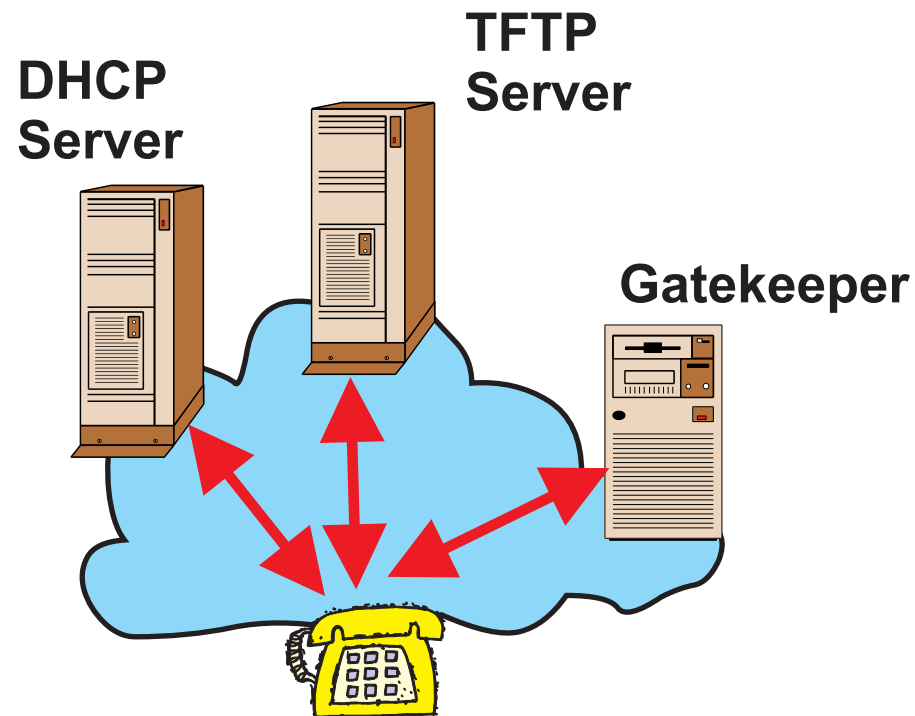
Conveys minimal session control information

**Header compression: requires 20 byte payload
(20 ms @ 8kbps yields 20 byte payload)
(IP header =20; UDP header=8; RTP header=12)**

IP Telephone



IP Telephone Call Setup



Phone makes DHCP request for IP information

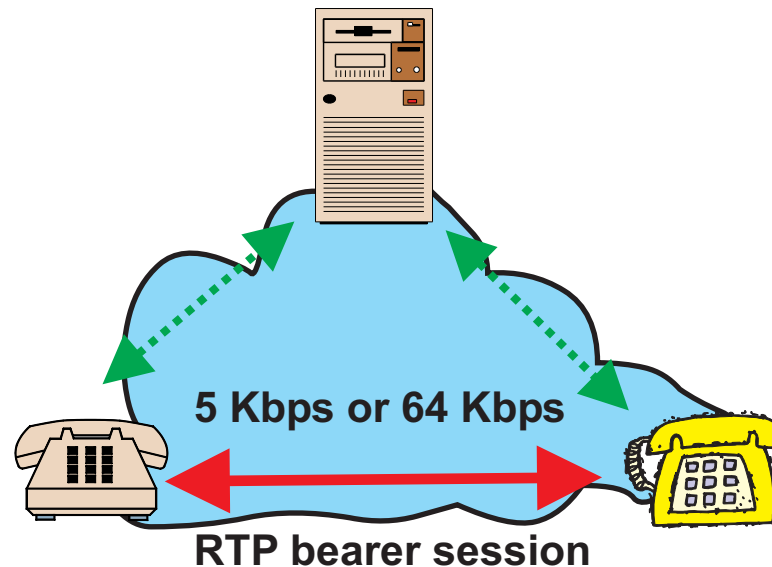
Phone makes TFTP boot file request to get gatekeeper

Phone registers with gatekeeper

Phone displays gatekeeper time/date

Phone is now ready to receive/place calls

IP Telephone Control



Calling phone sends off-hook message to gatekeeper

Gatekeeper directs phone to play dial-tone

Phone sends dialed digits to Gatekeeper

Gatekeeper rings called phone and accepts off-hook message

Calling phone initiates VOIP RTP session

Gatekeeper is notified of disconnect and records call details

H.323 and Other Advances

H.323 originally designed for LAN

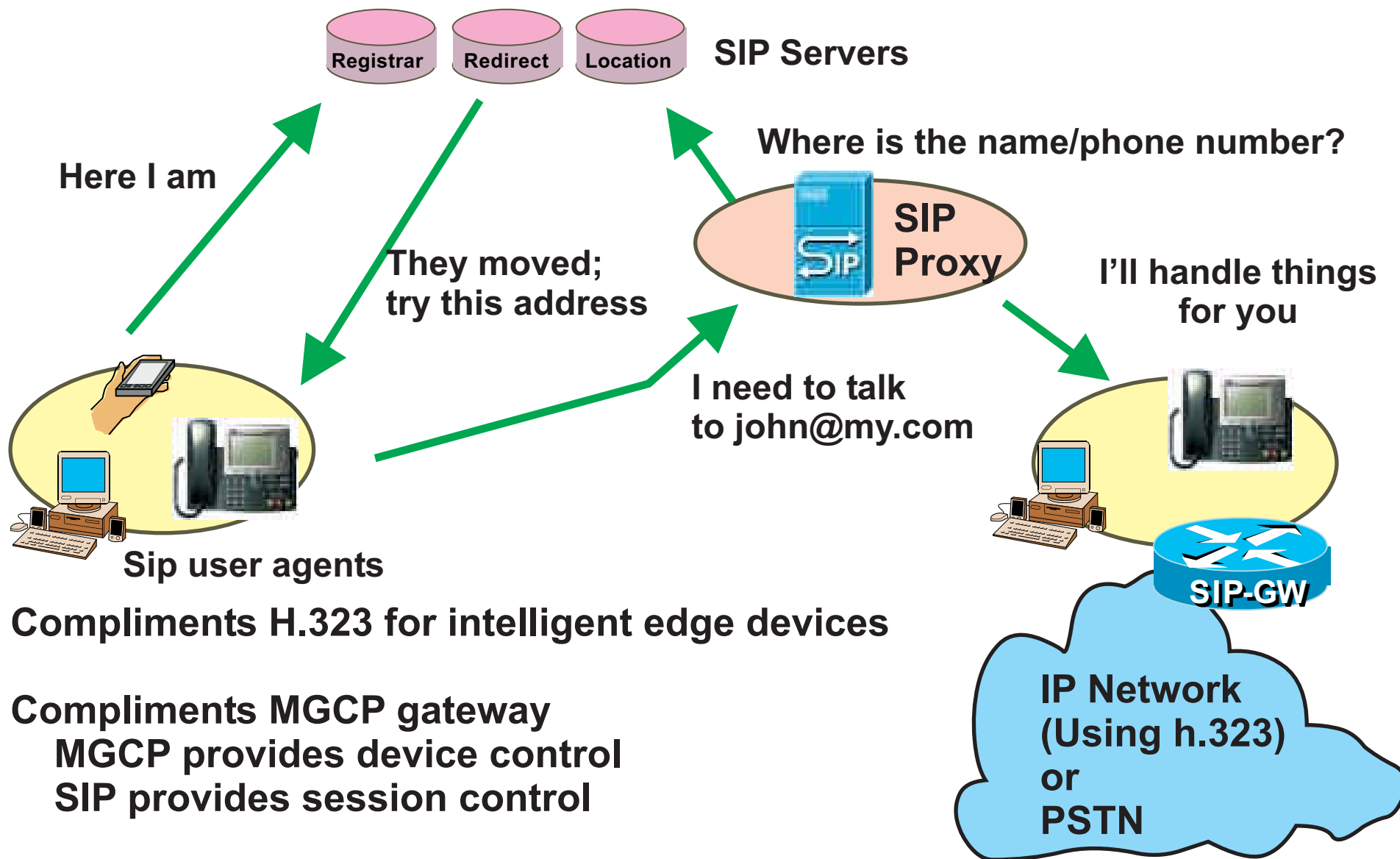
**No concept of Internet
wide area addressing
loop cancellation
call setup time**

**H.323v2
fast connect
call waiting
RAS
call transfer**

**H.323v3
maintain and reuse
multiplexing
interdomain transfers
call hold
park
pick-up
message waiting**



Session Initiation Protocol



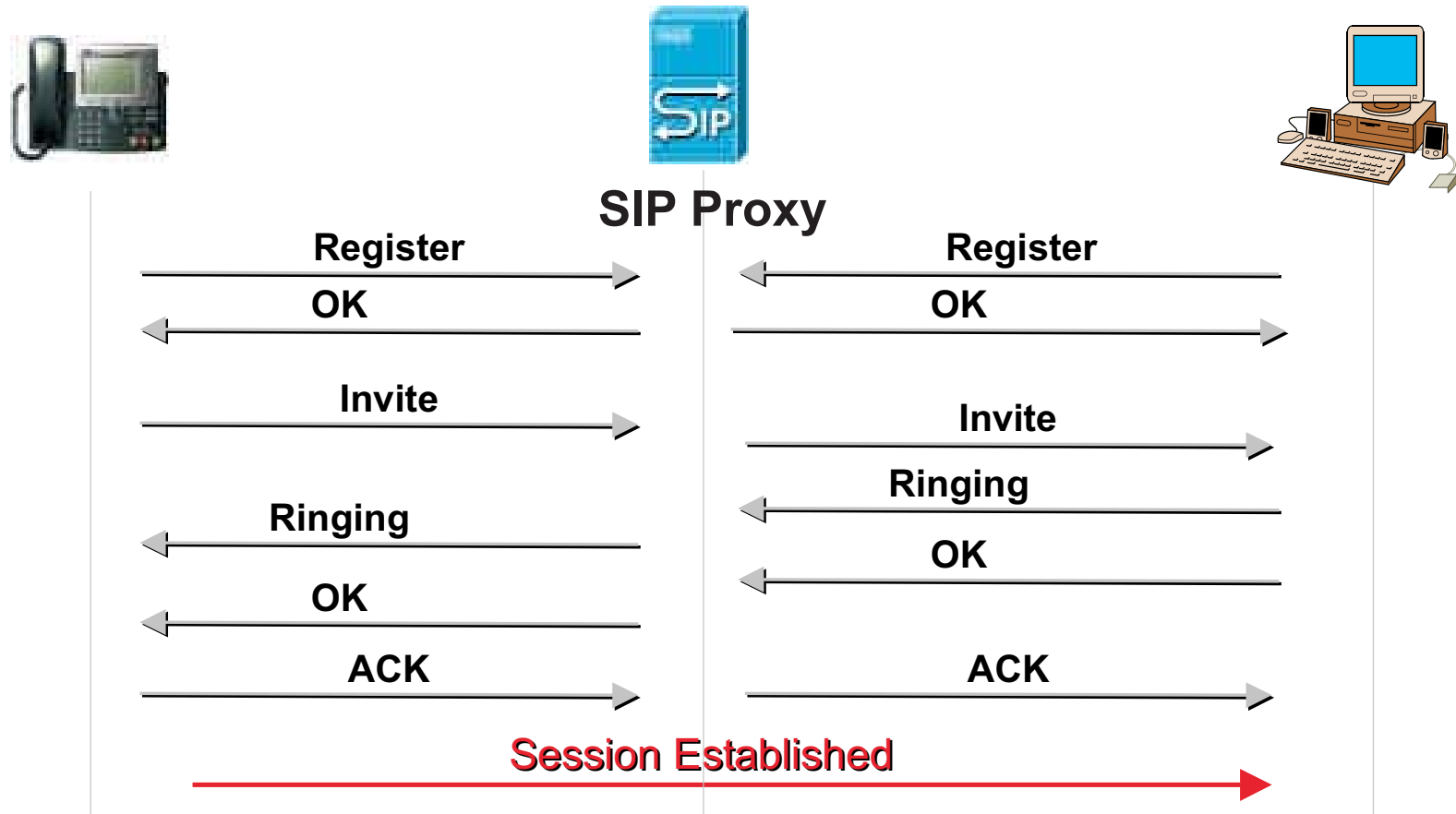
Compliments H.323 for intelligent edge devices

Compliments MGCP gateway
MGCP provides device control
SIP provides session control

Uses either E.164, DNS, or E-mail for addressing

SIP - Session Initiation Protocol

RFC 2543



SIP defined by IETF RFC 2543

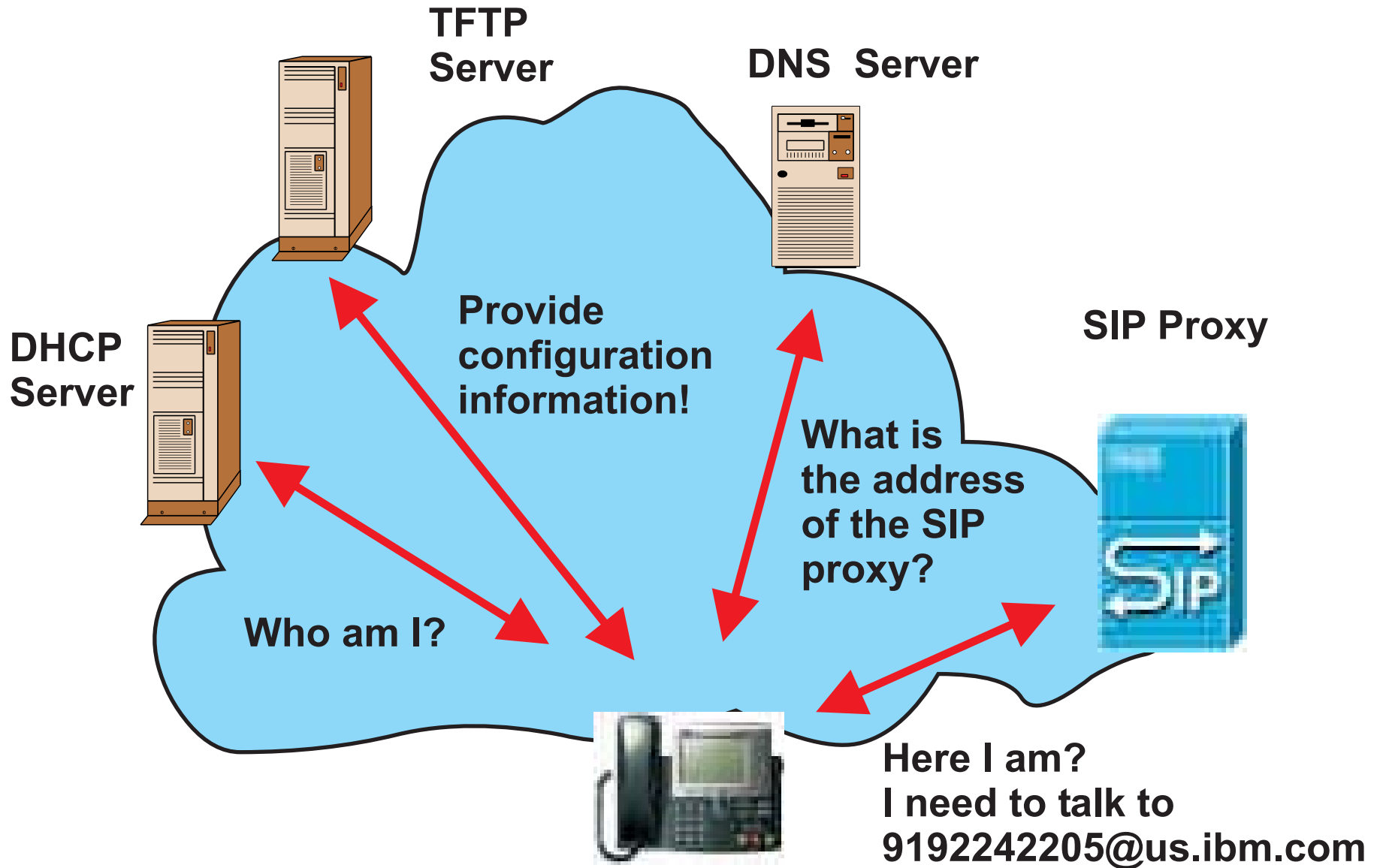
Defines transactions between clients and servers

Uses URL style addresses and syntax via E.164, E-mail, DNS

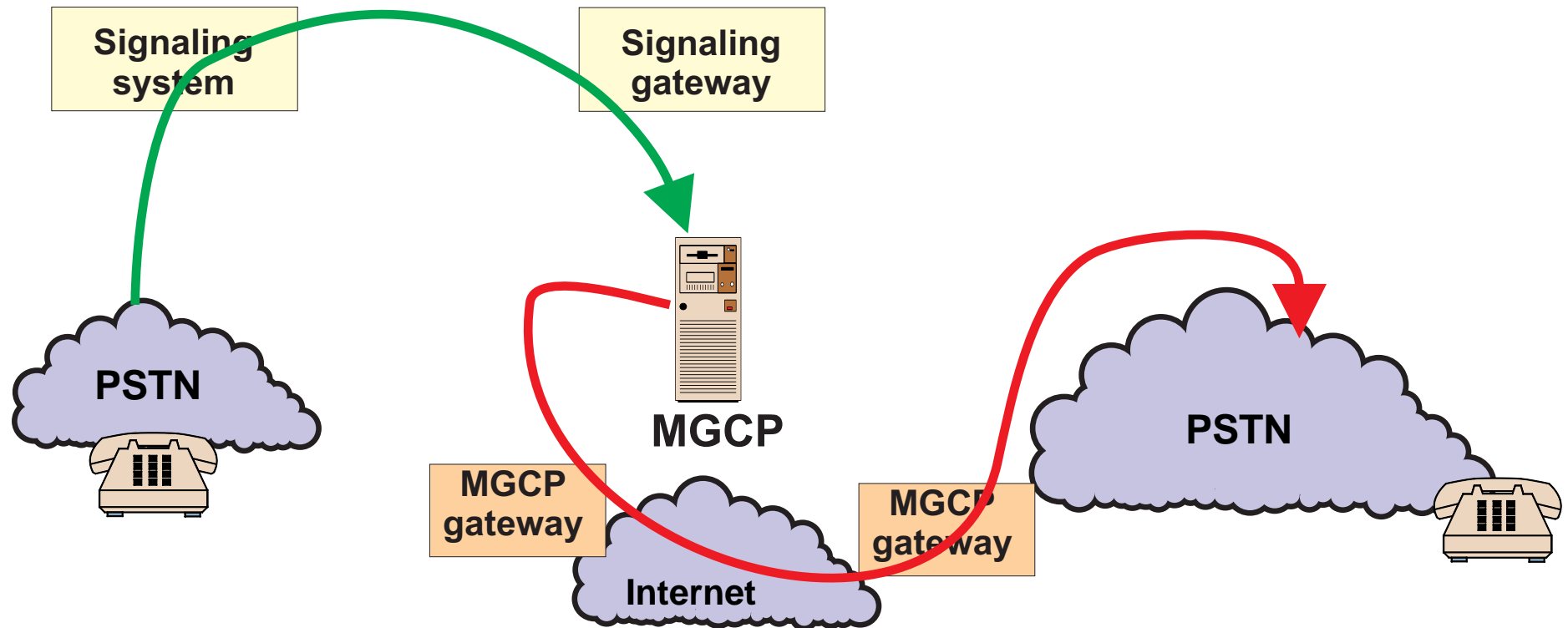
Simple extensible protocol

Works with both UDP and TCP (most implementations use UDP)

SIP - Phone



MGCP - Media Gateway Control Protocol



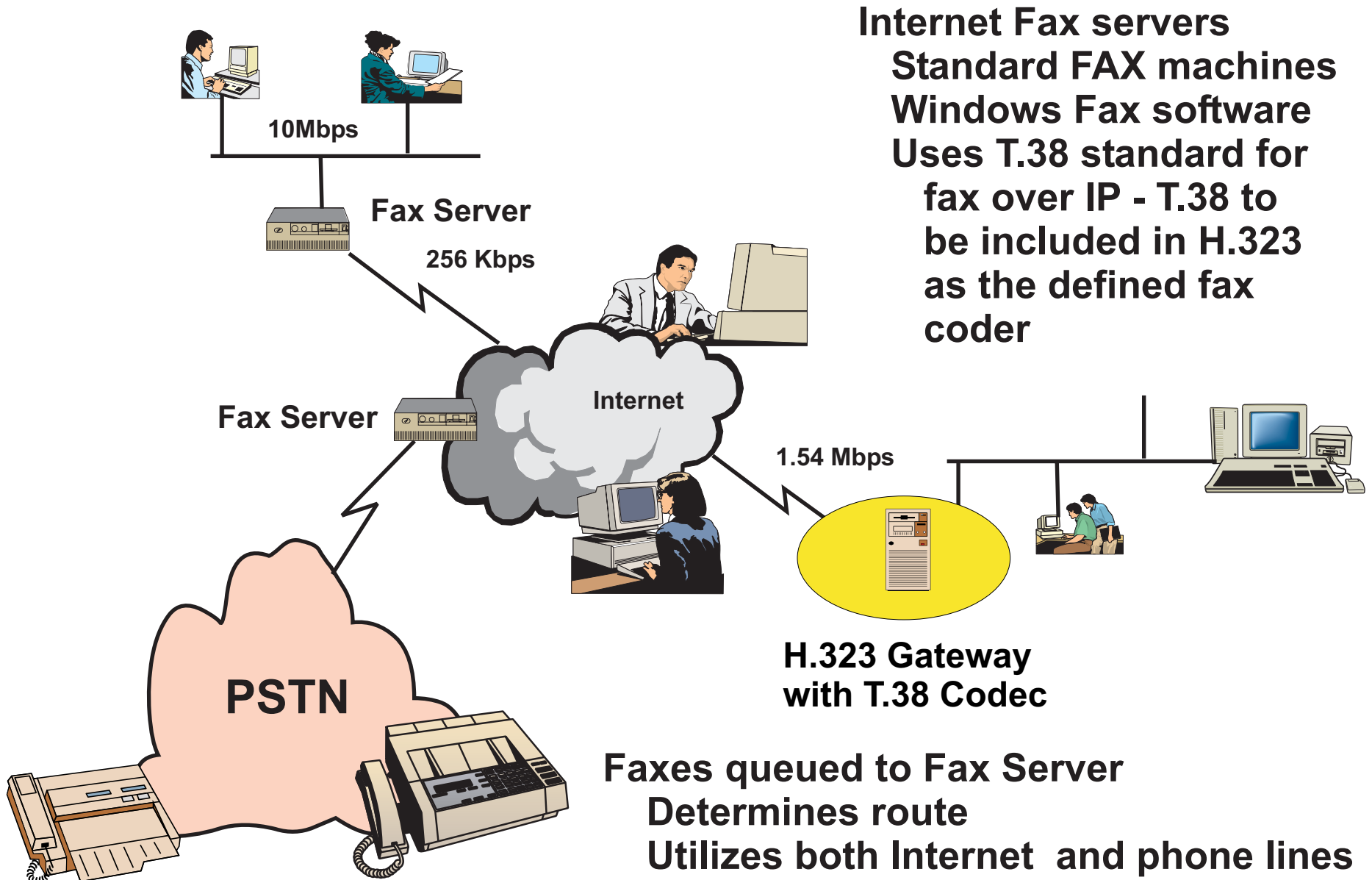
Designed for non-intelligent devices
Allows remote control of various devices
Create, modify and delete connections
Generate and detect events(tones)
Track resource states
Fits in well with H.323

Telcos support technology (trunking, residential and access gateways)

H.323 or SIP or MGCP: Confused?

| | H.323 | SIP | MGCP |
|-------------------------|----------------------------------|---------------------------|--------------------------------|
| Scope | Intelligent devices | Intelligent device | Non-intelligent devices |
| Call set-up | 2 round trips* | 1 round trip | 2 round trips |
| Call control | Control multiple elements | Session control | Device control |
| Transport | TCP | UDP | UDP |
| Complexity | High | Low | Medium |
| Functionality | High | Low | Medium |
| Billing/Security | High | Low | Medium |

Internet FAX

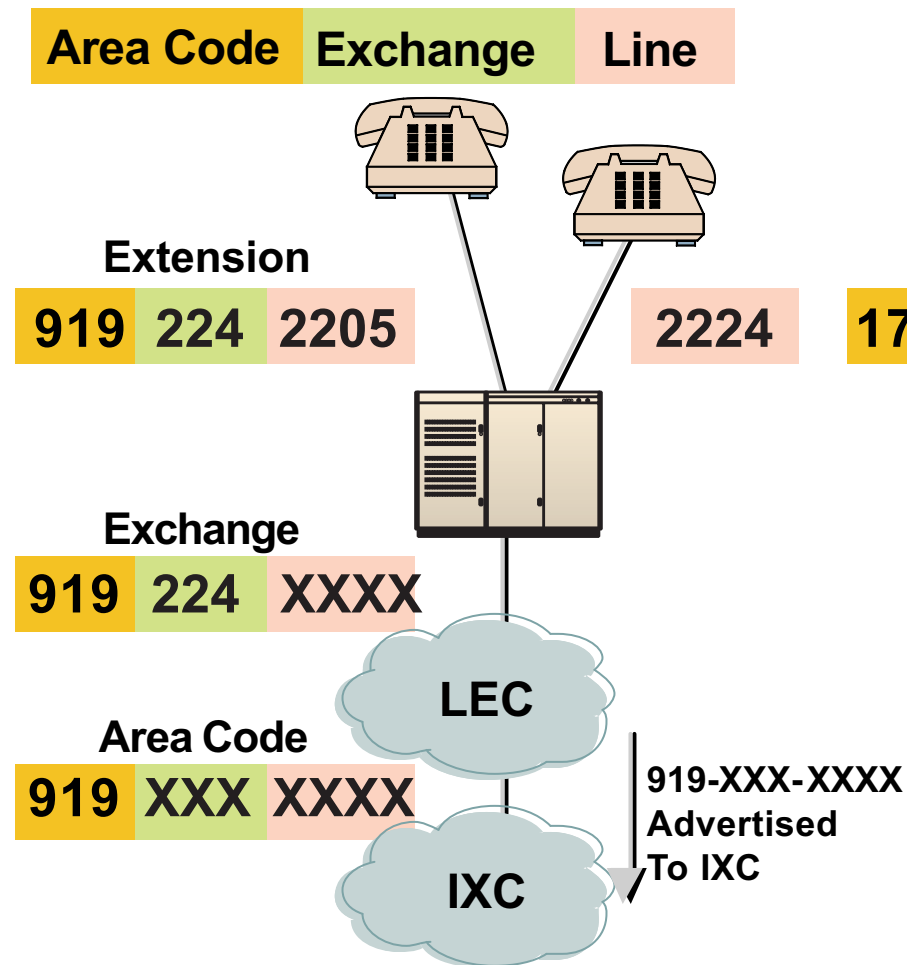


Internet Fax servers
Standard FAX machines
Windows Fax software
Uses T.38 standard for fax over IP - T.38 to be included in H.323 as the defined fax coder

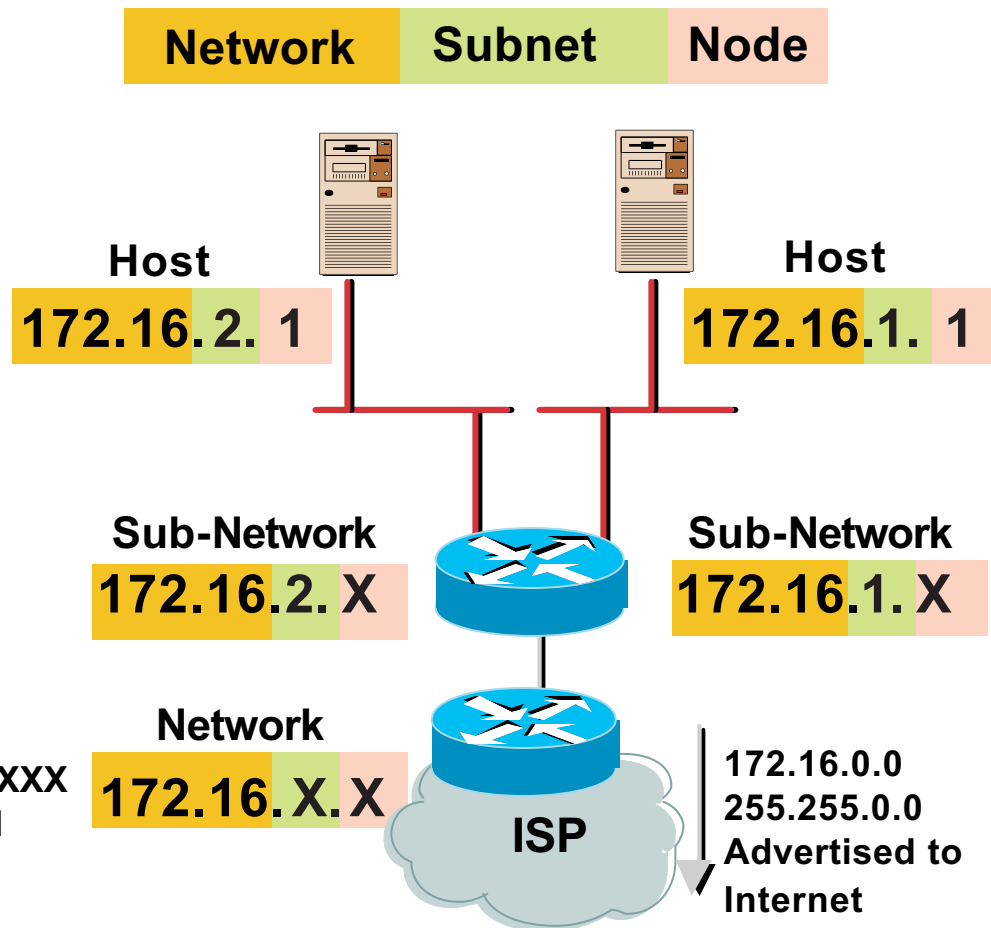
Faxes queued to Fax Server
Determines route
Utilizes both Internet and phone lines

Voice Numbers and DNS Convergence

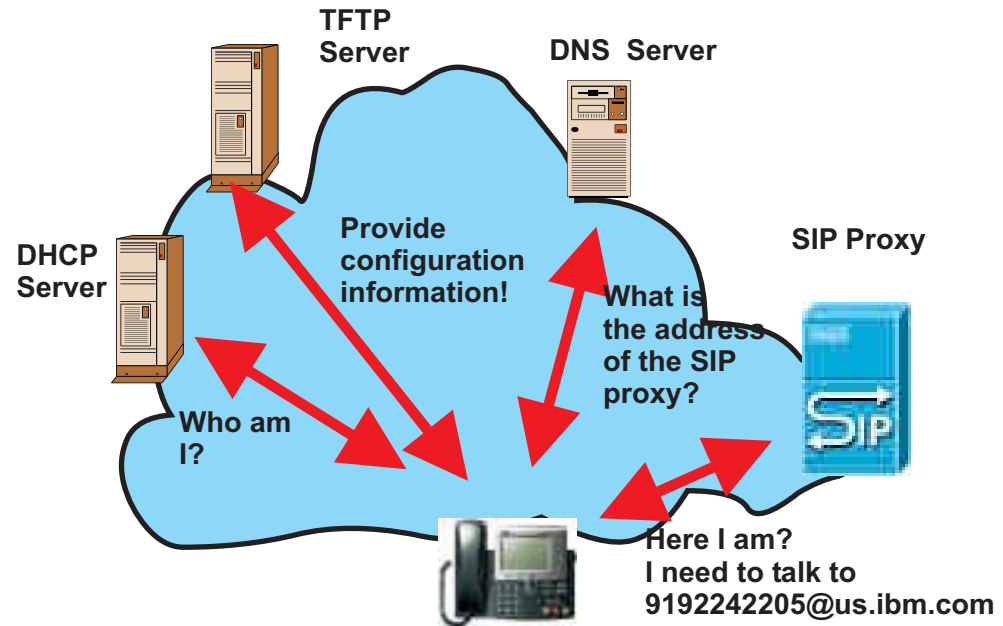
North American numbering plan direct inward dial



Internet Protocol Addressing



ENUM: Phone Numbers Meet the Net



DNS modified by RFC 2915
to add NAPTR field

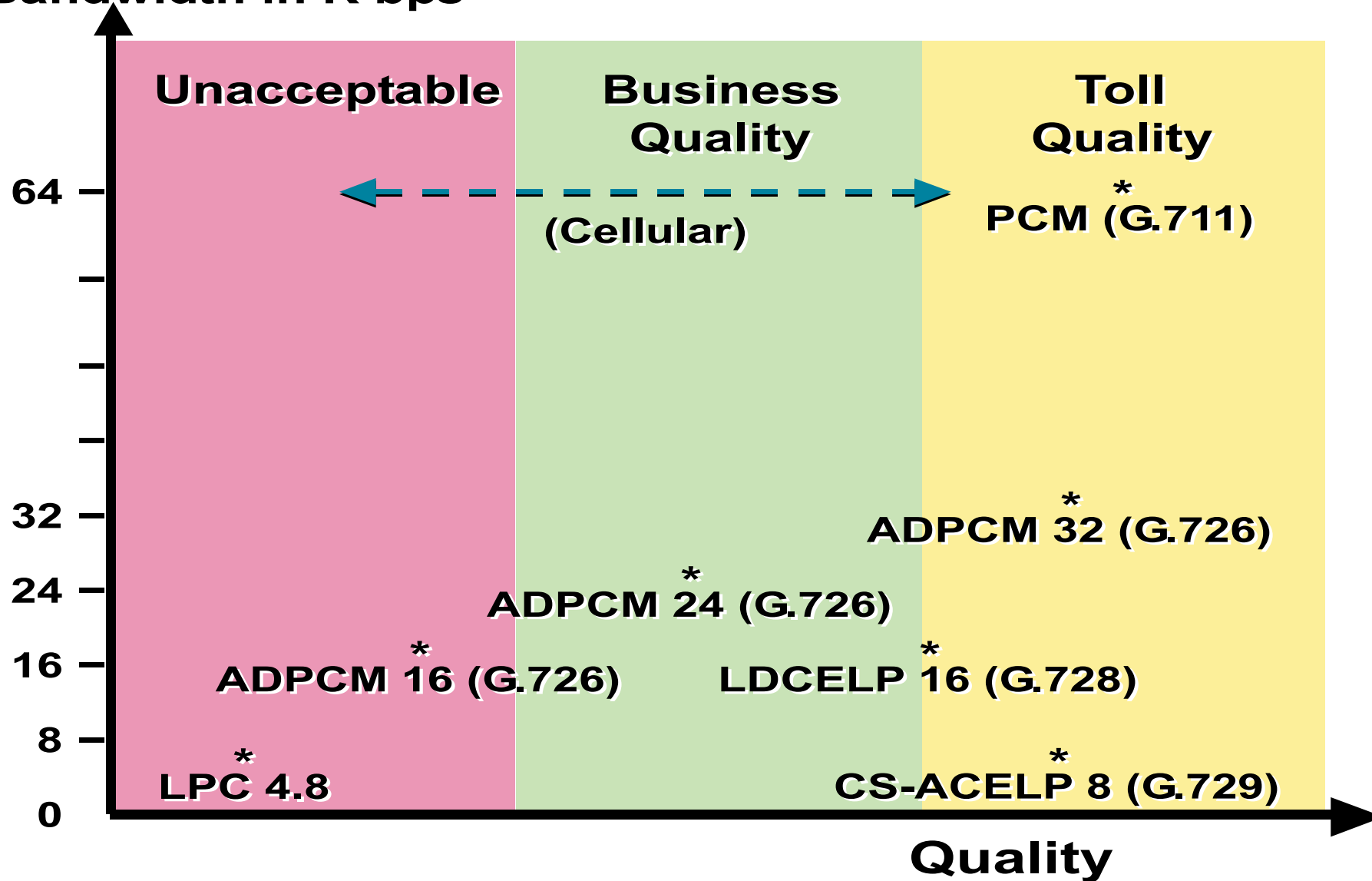
Input to DNS: 5.0.2.2.4.2.2.9.1.9.1.e164.arpa

Output from DNS

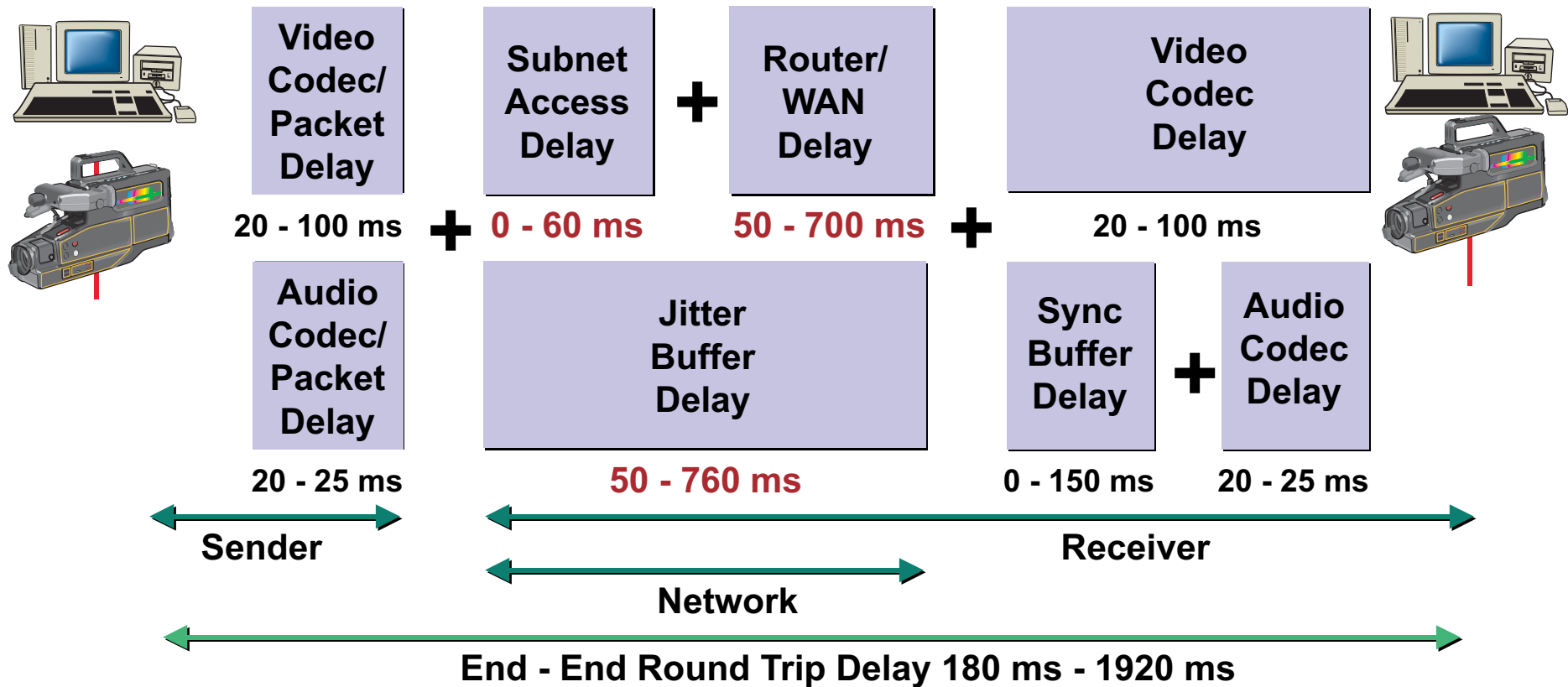
| | |
|-----------------------------------|--------------------------------------|
| IN NAPTR 102 10 "u" "sip+E2U" | "!^."\$!sip:laura@lauraknapp.com!". |
| IN NAPTR 102 10 "u" "mailto+E2U" | "!^."\$!mailto:ljknapp@us.ibm.com!". |
| IN NAPTR 102 10 "u" "fax-t37+E2U" | "!^."\$!tel+18002495324!". |
| IN NAPTR 102 10 "u" "tel+E2U" | "!^."\$!tel:+19192242205!". |

Voice Over IP Vcoders

Bandwidth in K bps



Multimedia Delay Breakdown



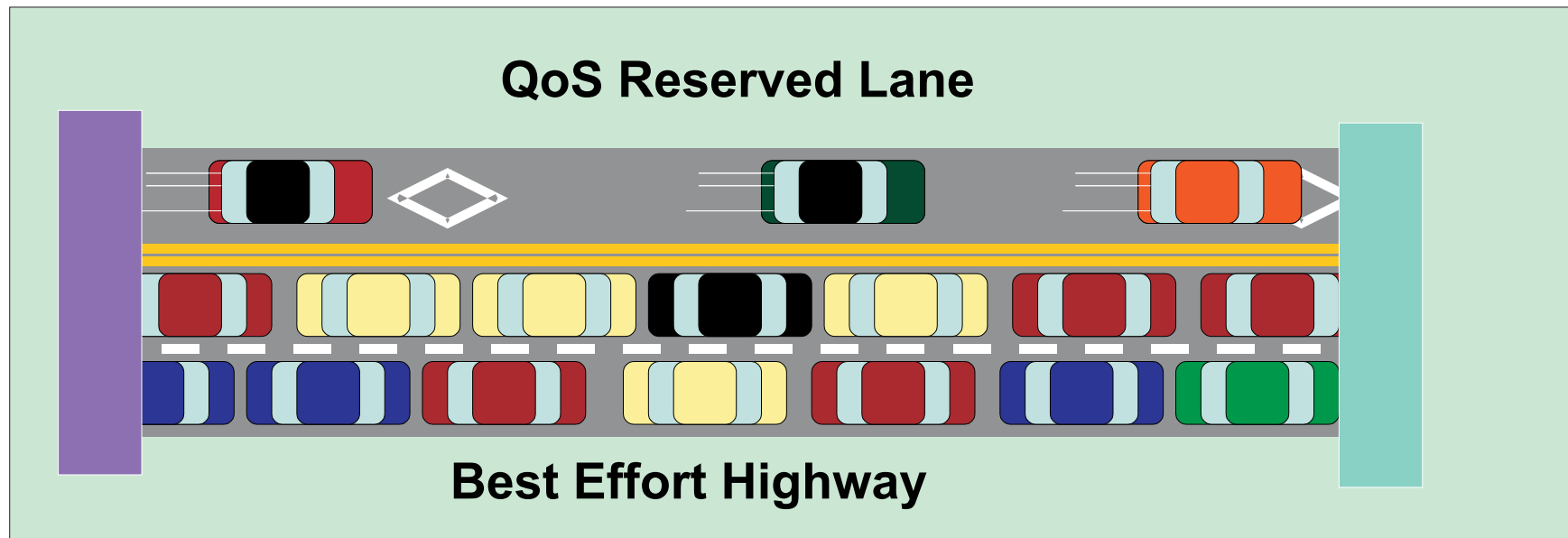
Humans will tolerate only about 200 millisecond delay before hearing the response from the other person
Why everyone is talking about QOS (Quality of Service) on the Internet!

Quality of Service

Let application request resources
bandwidth
latency
priority

Applications need to
reserve bandwidth
prioritize packets

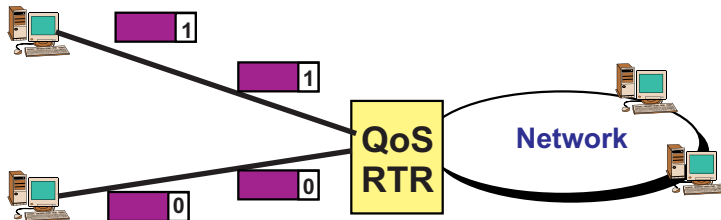
Network administrators
police network usage
provide service guarantees



DiffServ, RSVP, and MPLS will be used to provide QoS for IP data

Protocol Advances

DiffServ Differentiated Services



- Use bits in IP header to identify packet's level of service
- Packets can be marked at source or other parts of network
- Does require routers and switches that are Differentiated Services enabled
- Does not require end-to-end signaling
- Does not require per-flow state

RSVP Resource Reservation Protocol

- End user applications Reserve bandwidth Lock in quality of service

Software for routers, hosts and applications

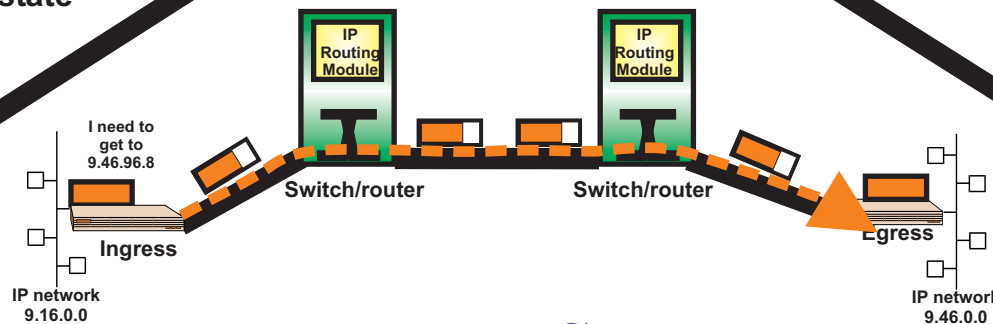
Quality of service requests repeated or reservation will lapse

Changes in network topology will revoke reservation

Applications must re-establish service levels

- Reservations Shared (audio) Non-shared (video)

Quality of Service Guaranteed Controlled load



MPLS Multiprotocol Label Swapping

What to Look for in Products

Form factor

Architecture

Circuit switched interface

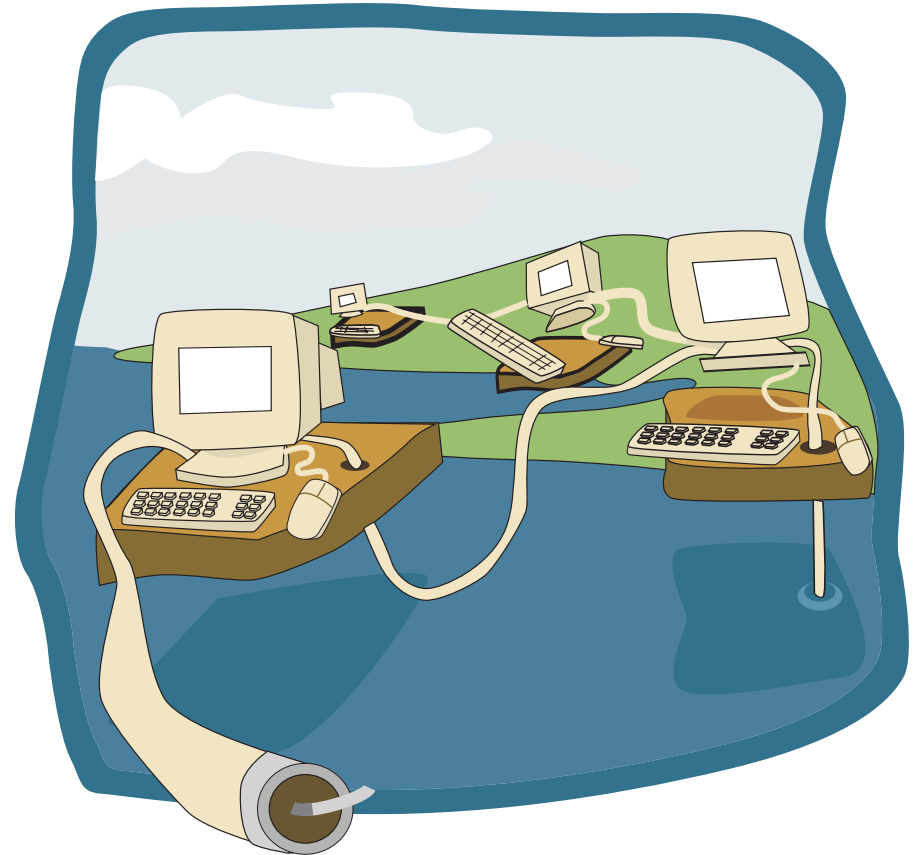
Max phone ports

H.323/H.323 gatekeeper

Prioritization scheme

Encoding scheme

Maximum compression



Sooner or Later !!!

Pieces emerging to finally allow for converged voice/data networks?

Complex

How will billing be handled?

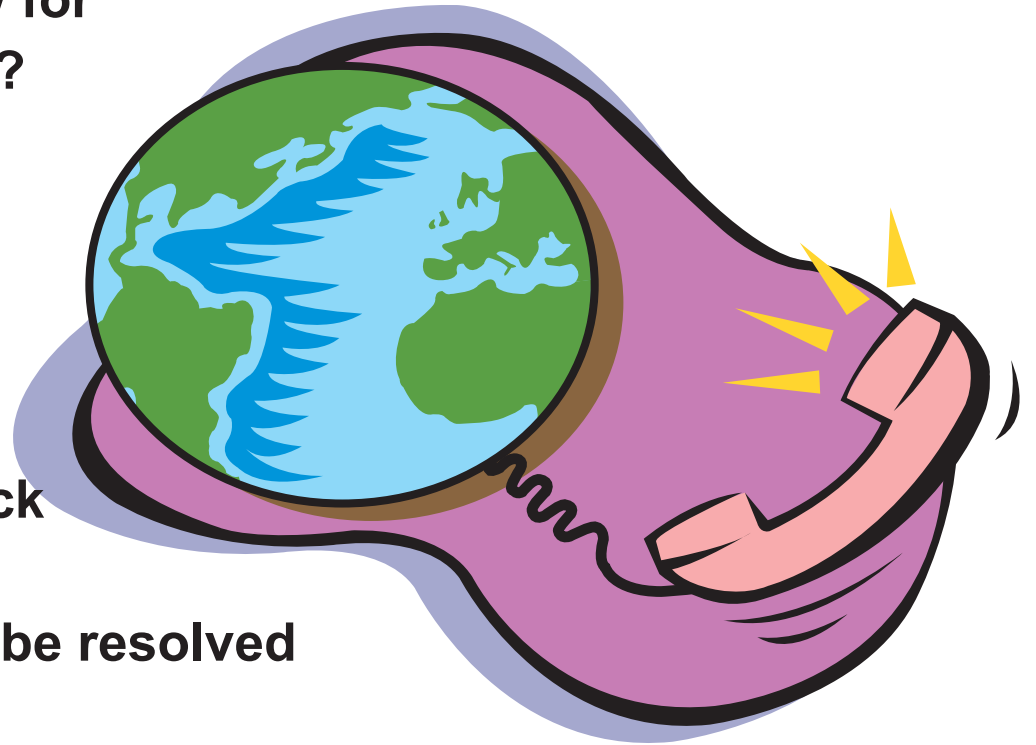
QoS still a major stumbling block

Many application issues still to be resolved

Interoperability issues being resolved, but it will take time

Increasing government regulation

Over time.....should change our communications significantly



Background Information

SIP www.cs.columbia.edu/~hgs/sip/
SIP www.ietf.org
SIP www.greycouncil.com/sipug
SIP www.sipcenter.com
H.323 www.itu.int
H.323 www.imtc.org
H.323 www.packetizer.com
MGCP.... www.sofswitch.org
All www.etsi.org
ENUM.... www.cybetele-com.com
ENUM.... www.enum.org
ENUM.... www.enumworld.com
ENUM.... www.netnumber.com

