Advanced Networking

Voice over IP: Introduction and H.323 standard

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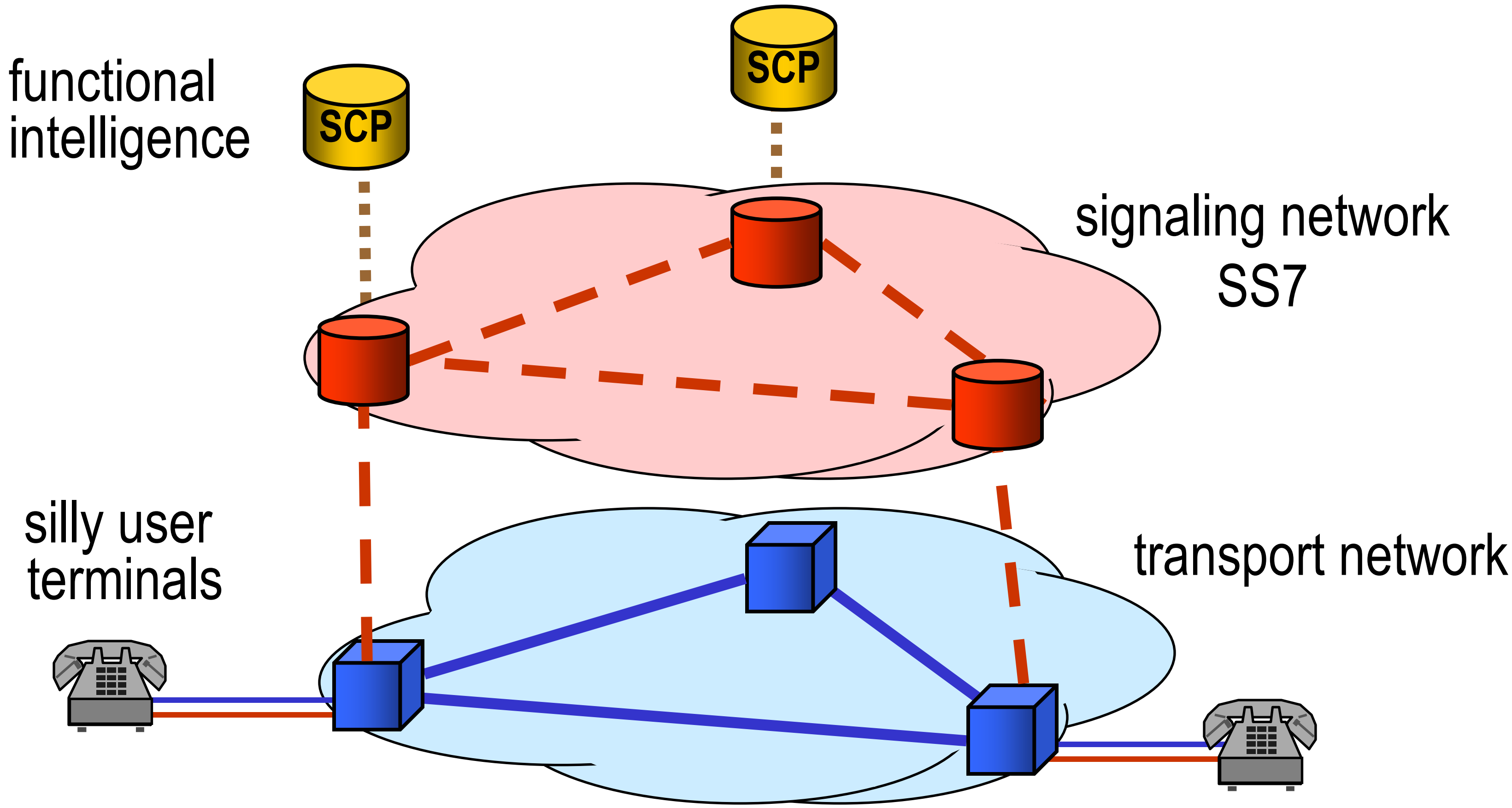
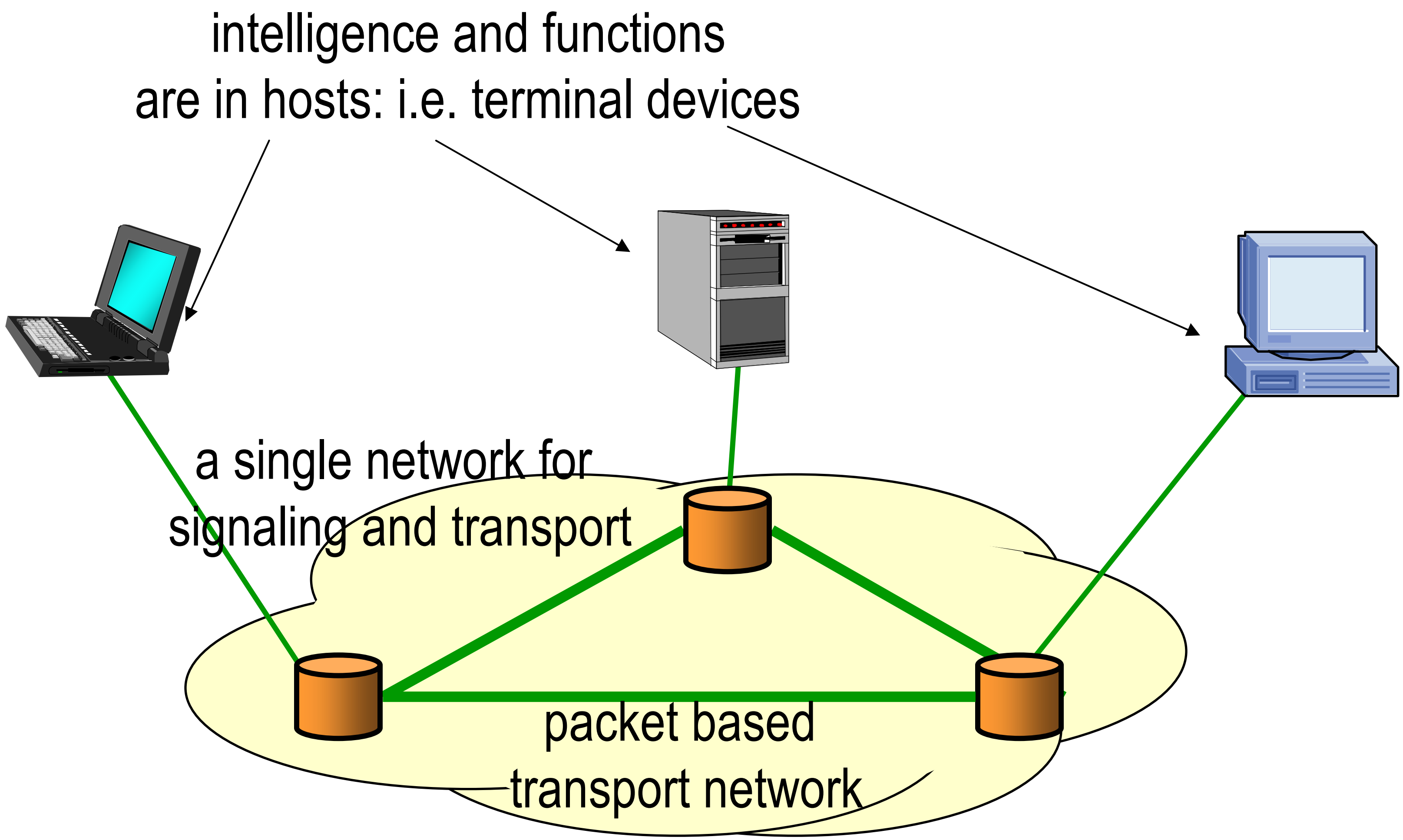
VoIP: Integrating Services

- Voice on IP Networks is just "another application"
- Nothing "special" or "specialized" as traditional telephony, where the network and the service are joint, coupled and sinergic
- VoIP is realized through end-to-end application level protocols, normally not strictly tailored for voice
- Is QoS required?

Plain Old Telephone Service (POTS)



IP services



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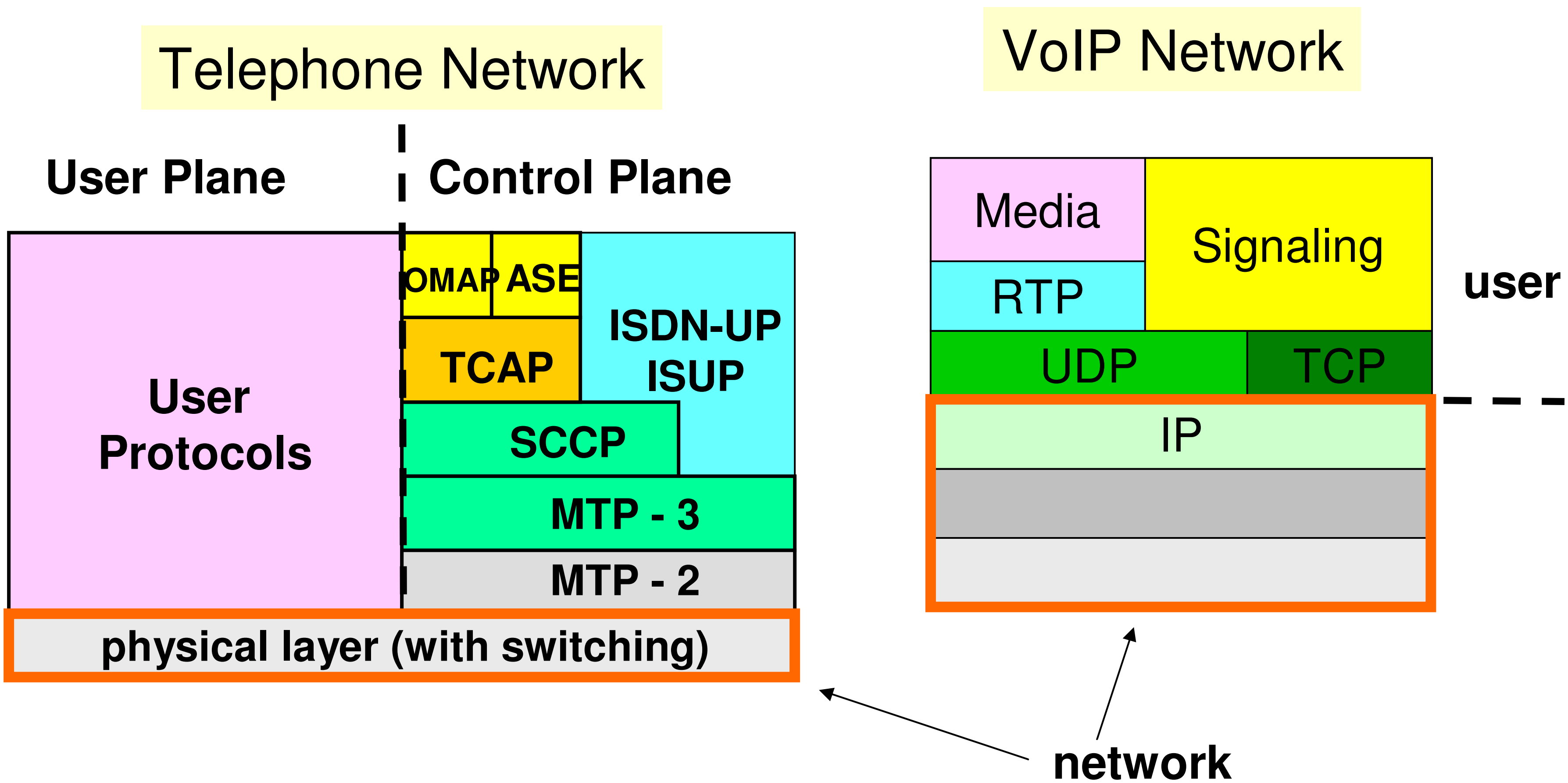
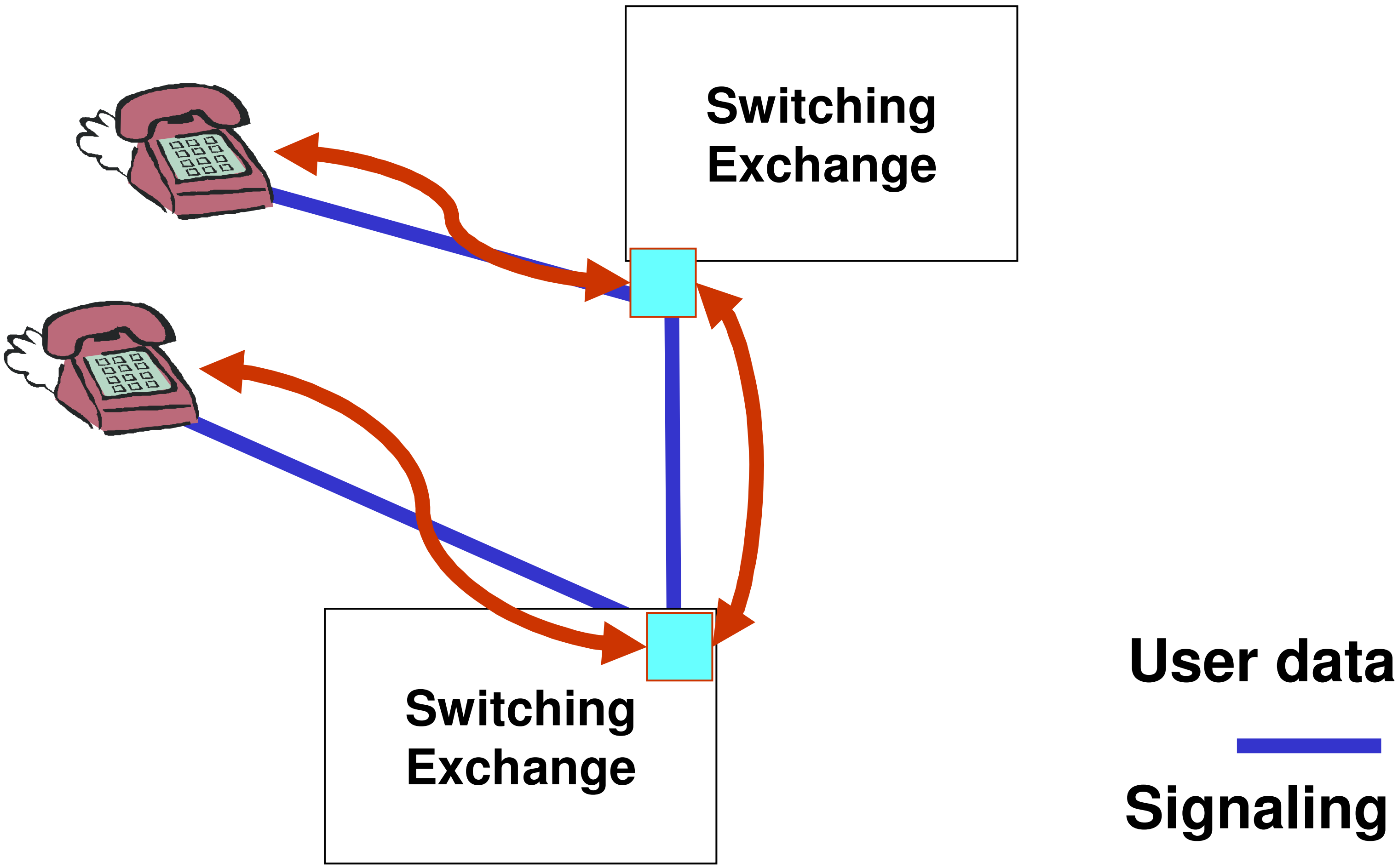
Architectural difference



A Telephone network ...



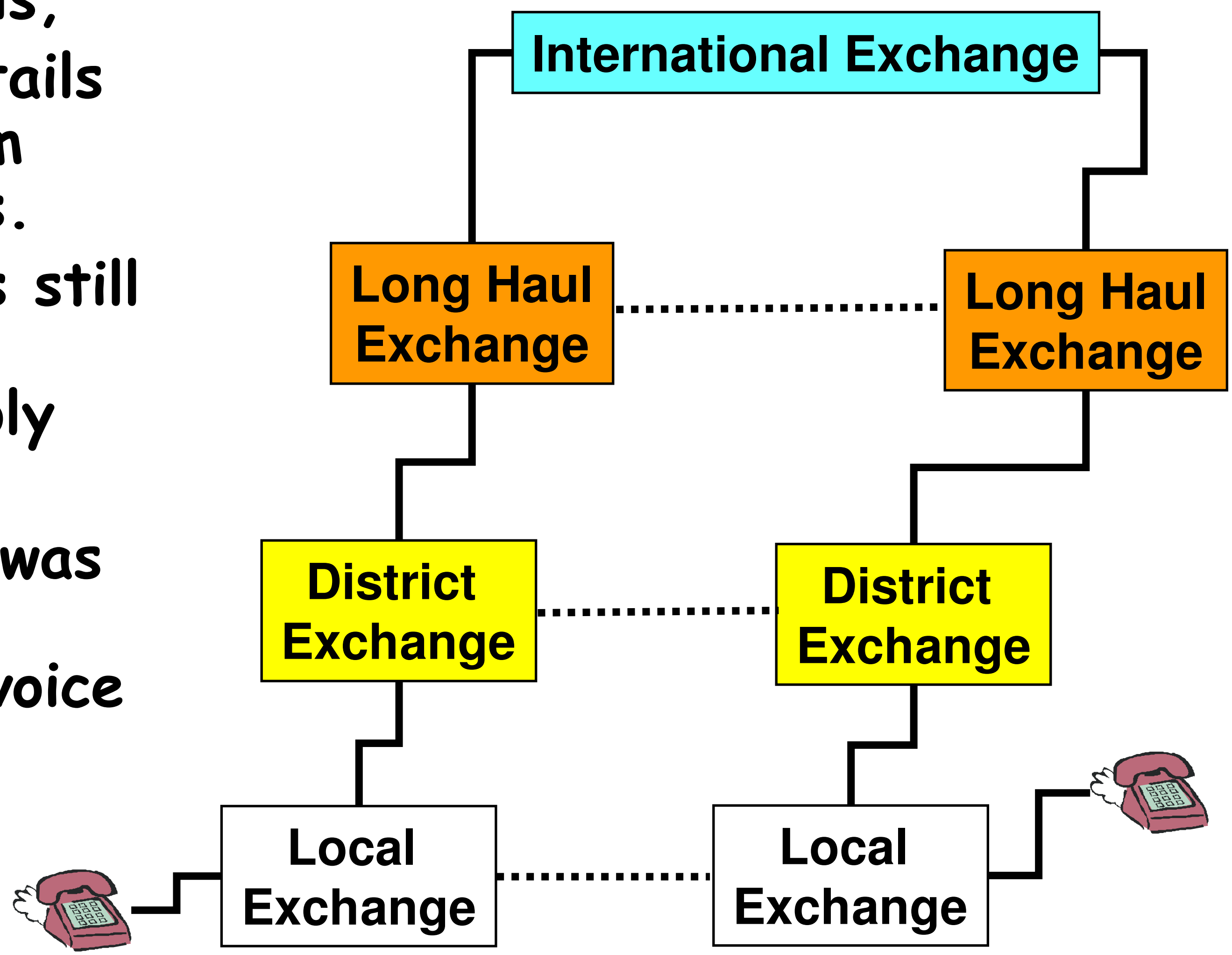
Hierarchical organization

Hierarchy levels,

Names and details are not uniform across coutries.

Architecture is still biased by the original monopoly system

The structure was tailored and optimized for voice transport



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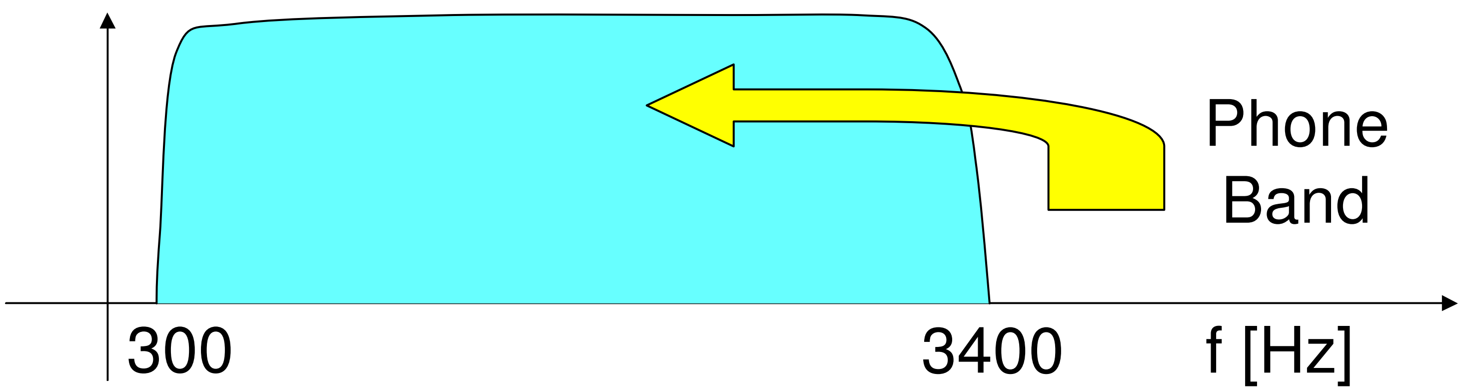
Voice and its transmission

- The voice signal is transmitted with analogic technique on the local loop, filtered between 300 and 3400 Hz to allow direct curret for powering the phone and to limit the signal bandwidth to a known extent
- The local exchange immediately convert the analogic signal to digital PCM



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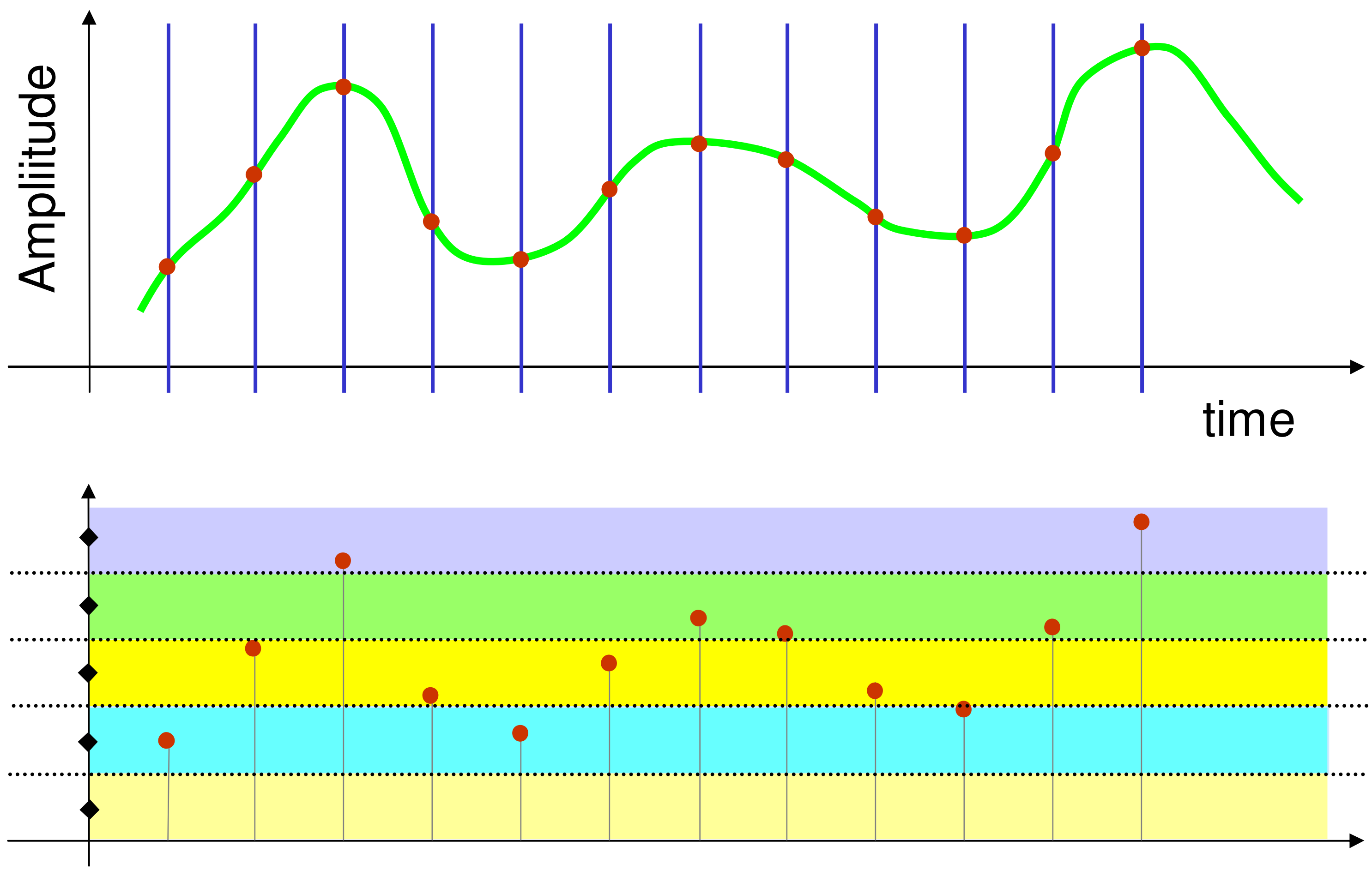
PCM: Sampling and Quantizing



PCM

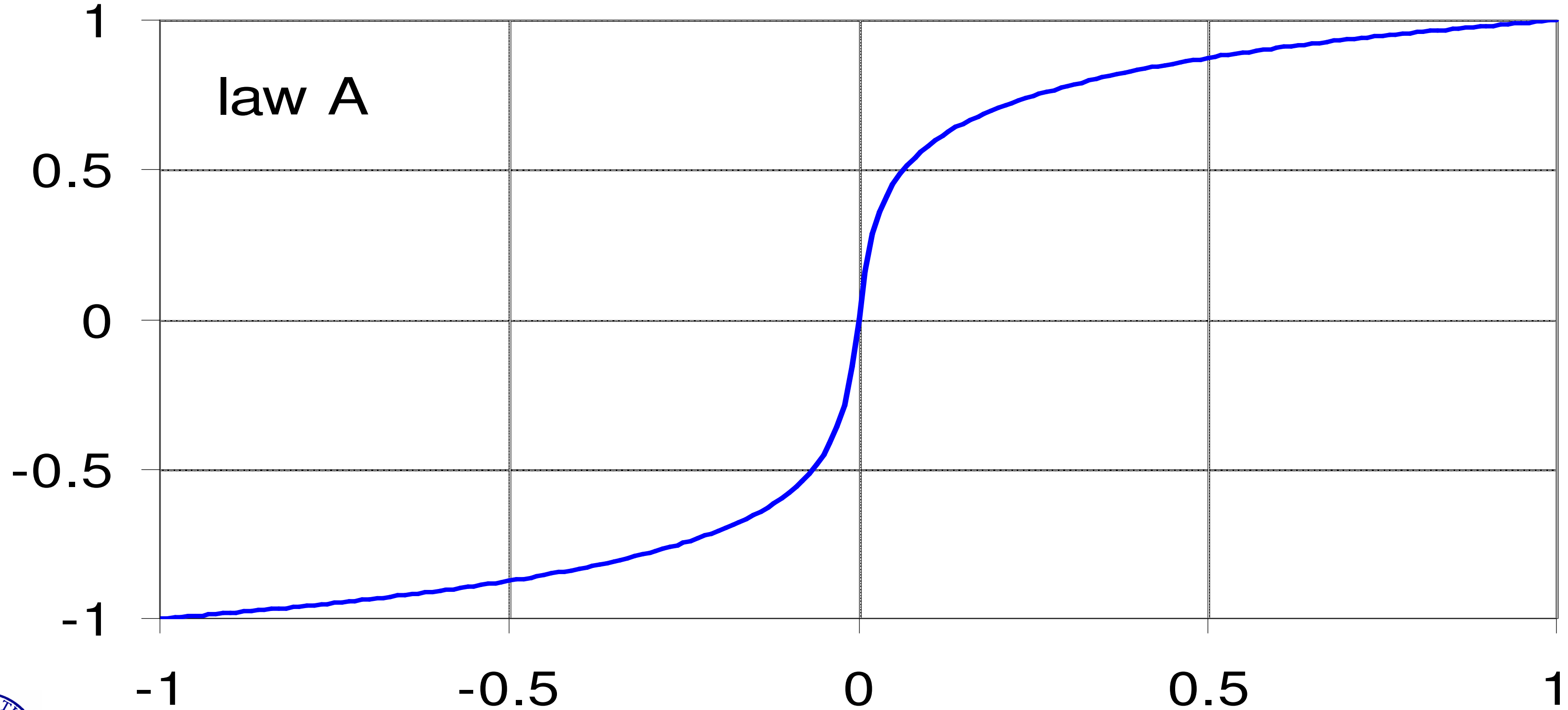
- PCM (Pulse Code Modulation) encoding is nothing else than sampling and quantizing (with non-linear quantization for telephone networks)
- Linear quantizing means equal intervals; non linear (companding) means different intervals as a function of amplitude
  - **Linear PCM: CD (~44 kHz, 16 bit =~ 1.5 Mbit/s)**
  - **Companding PCM: telephones (8kHz, 8 bit = 64kbit/s). Based on the fact that human ear sensitivity is logaritmic**

"A" Compression Law

$$Y = \begin{cases} \dfrac{A}{1+\ln(A)} X & X < 1/A \\ \dfrac{\mathrm{sgn}(X)}{1+\ln(A)} \left(1+\ln|AX|\right) & 1/A < X < 1 \end{cases}$$

$$A = 87.6$$

$$X = V / V_{\max}$$



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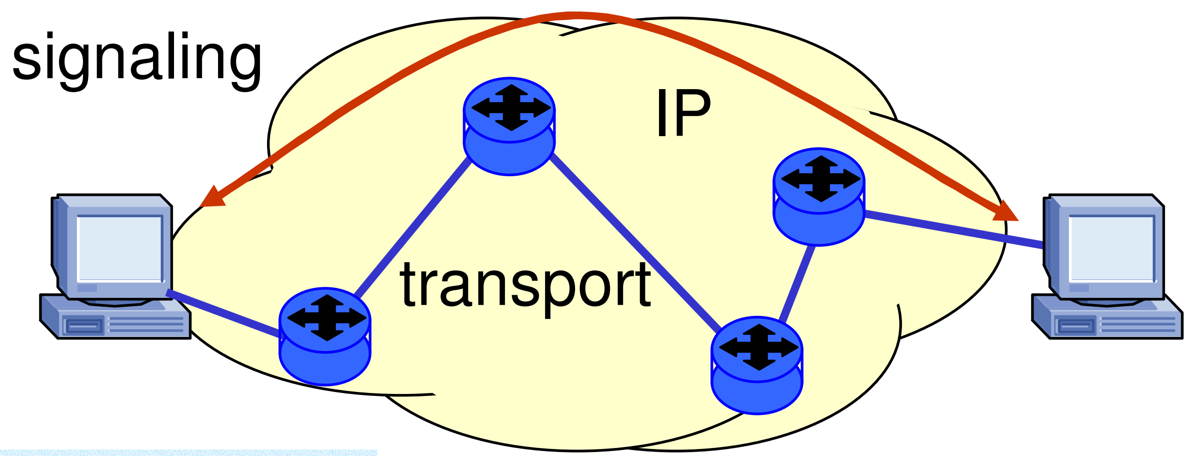
Service-specific problems

- Voice is "just another service", but ...
- Is it possible to realize e-t-e conversational services without involving the network layer?
- Signaling in telephony has application-level functionalities
  - access
  - callee identification
  - negotiation of characteristics and quality
  - billing and accounting ...
- But also control function on the transport channel
  - routing and setup
  - resource finding and reservation

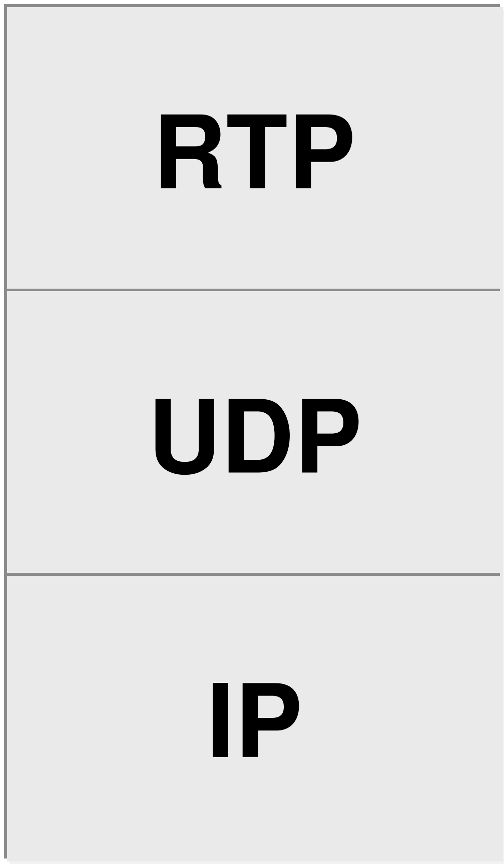
Service-specific problems

- Application level signaling are simplified by the IP e-t-e approach ☺
- Network services for the control of the channel (e.g. QoS) simply do not exist in IP ☹
- Routing is not controllable (no alternate routing), hot-swap reliability is not present, QoS control is almost impossible unless by “circuit-like” dimensioning. ☹



Real-Time Transport in IP

- Real Time (Transport) Protocol
- Developed by Audio Video Transport Working Group of IETF
- RFC 1889 obsoleted by 3550/3551
- It is an add-on to UDP building a connection-oriented unreliable channel
- Adds and header with information for:
  - Multimedia data management (coding, timestamping, etc.)
  - Error and QoS control (feedback on the reverse channel)



RTP: characteristics and functionalities

- Independent from the PHY (obvious!!!)
- Scalable
  - Unicast e multicast
- Defines separate logical channels for data and control
  - indeed a "pair" of protocols RTP-RTC$_{\text{ontrol}}$P
- Packet reordering at destination
- Delay jitter equalization with buffers (in addition to the playout buffer of the application)
- Sender identification
- Intra-media synchronization
- No predefined Port, but must be even

RTP: header format

Ver.	P	X	CSRC ct.	M	payload type	sequence number
Timestamp						
synchronization source identifier (SSRC)						
content source identifiers (CSRC)						
Extension Headers (optional)						
Data						

The RTP header (12 bytes)

- Ver.(2 bits): Version of the protocol. Current is 2
- P (1 bit): Indicate if there are extra padding bytes at the end of the RTP packet.
- X (1 bit): Extensions to the protocol used (ELH present)
- CC (4 bits): Number of CSRC identifiers that follow the fixed header
- M (1 bit): If set means that the current data has some special relevance for the application defined in a profile (external to the protocol)
- PT (7 bits): Format of the payload and its interpretation by the application
- SSRC: Indicates the synchronization source and timing
- Extension header: Length of the extension (EHL=extension header length) in 32bit units, excluding the 32bits of the extension header

RTCP

- Real Time Control Protocol
- Functionalities:
  - Data Distribution Control
  - Session information advertisement (during the session, not for setup)
  - QoS feedback
  - Error reporting
  - ...
- RTCP messages are sent on RTP-port+1

VoIP: Signaling Protocols

Standardized

- H.323
- SIP
- H.248

Proprietary

- skype
- wengo – connecting your phone to the future
- Asterisk (IAX, open source)
- YAHOO! MESSENGER
- AOL Instant Messenger
- Cisco Systems (Skinny)

… and many many more…

Applications based on “VoIP” protocols

Brief History of VoIP (1)

- Sharing expensive lines (end of ’90)
  - VoIP enters the enterprise market as a way to save telecom (transmission) cost by using excess data capacity for Voice
  - using the same lines for data and voice communication
    - utilizing existing Local Area Networks (LANs) and WAN connectivity for voice communication, i.e. reduce enterprises’ bill from PSTN operator
  - ITU-T promotes H.323 as protocol (ISDN-style VoIP protocol)

Brief History of VoIP (2)

- Network Convergence (beginning of ‘00)
  - network convergence:
    - data over ISDN was initially successful in some countries (Ger, J) but usage price was high and bandwidth was soon too limited
    - when Internet bandwidth became abundant – VoIP success started
  - IETF completes standardization of an Internet-style VoIP signaling protocol: Session Initiation Protocol (SIP) and media transport protocol (RTP)
  - Internet (IP) becomes the new Integrated Services Digital Network
  - Operator’s convergence began with VoIP in the backbone
  - only lately moving to the access (2005+)

Brief History of VoIP (3)

- SIP becomes the dominant VoIP protocol (‘00 until now)
  - H.323 had the earlier start, but more oriented towards local networks
  - ISDN-style H.323 was more liked by traditional operators
  - SIP is a text-based protocol on to of IP, much like HTTP and XML
    - therefore easy to understand for IP and/or web experts
    - SIP better suited for large scale application
    - efficiency is poor
    - security threats
  - but SIP became the choice of Internet community
    - Standardized by IETF

Brief History of VoIP (4)

- **breakthrough:** SIP chosen by 3GPP as basis for IMS, i.e., all multimedia services (including VoIP) in 3G
- The consumer segment becomes aware of VoIP
  - Skype clients are widespread
    - using proprietary protocols
  - consumer market is not interested in standards – only costs
  - the business model of Skype – owned by ebay – is "the whole world can talk for free" – revenue is made through arbitrage:
    - Skype out / Skype in – Gateway to PSTN
    - advertising
  - advertisements, lack of privacy/security, quality are the price consumers pay

Today's Situation

Three VoIP market segments

1. **enterprise**
2. **public operators**
3. **consumer**

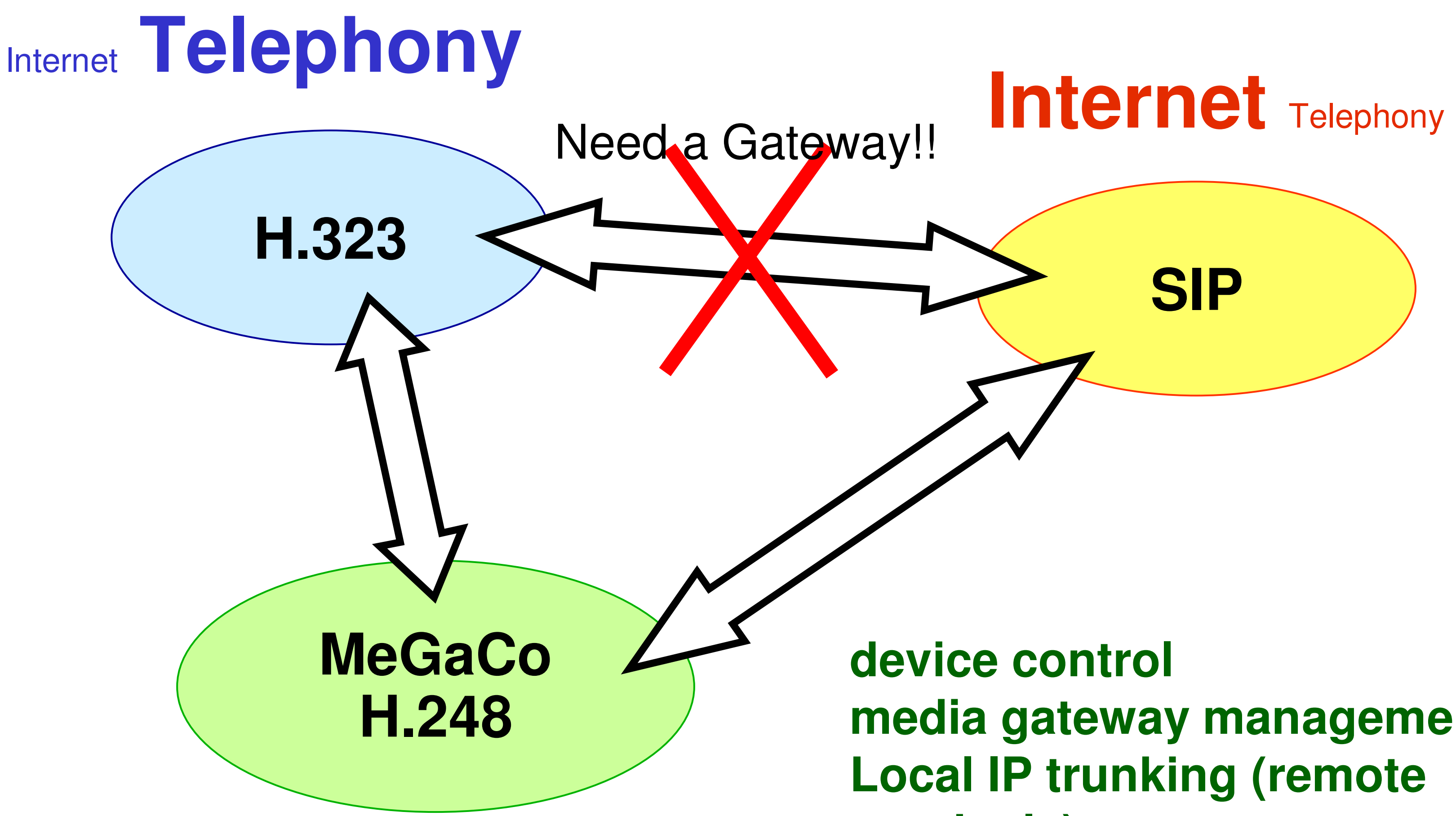
What about protocols?

- H.323 is still in the market but will probably die sooner or later ⇨ no point to get into H.323 market in 2006/7/8/9/....
- SIP is already dominating
  - today new investments are based on SIP
  - SIP large scale deployment still in the beginning
  - already dominating the corporate market
  - entering the operator market
- Proprietary protocols, e.g. Skype, are competing in consumer market only

Signaling and Service Protocols

- H.323
  - Vertical, Hierarchic, Complex, Rigid, Omni-comprehensive "LAN oriented", not easy to integrate with PSTN
- SIP
  - Horizontale, Flat, Simple, "WAN oriented", impossible to integrate with PSTN
- MeGaCo (H.248)
  - Vertical, Hierarchical, Complementary to H.323/SIP, Separates data and signaling for management, easy support for soft-switches PSTN-oriented, used locally to control media-gateways, not ment as "entire system"

Protocolli



device control
media gateway management
Local IP trunking (remote terminals)

Protocols "philosophy"

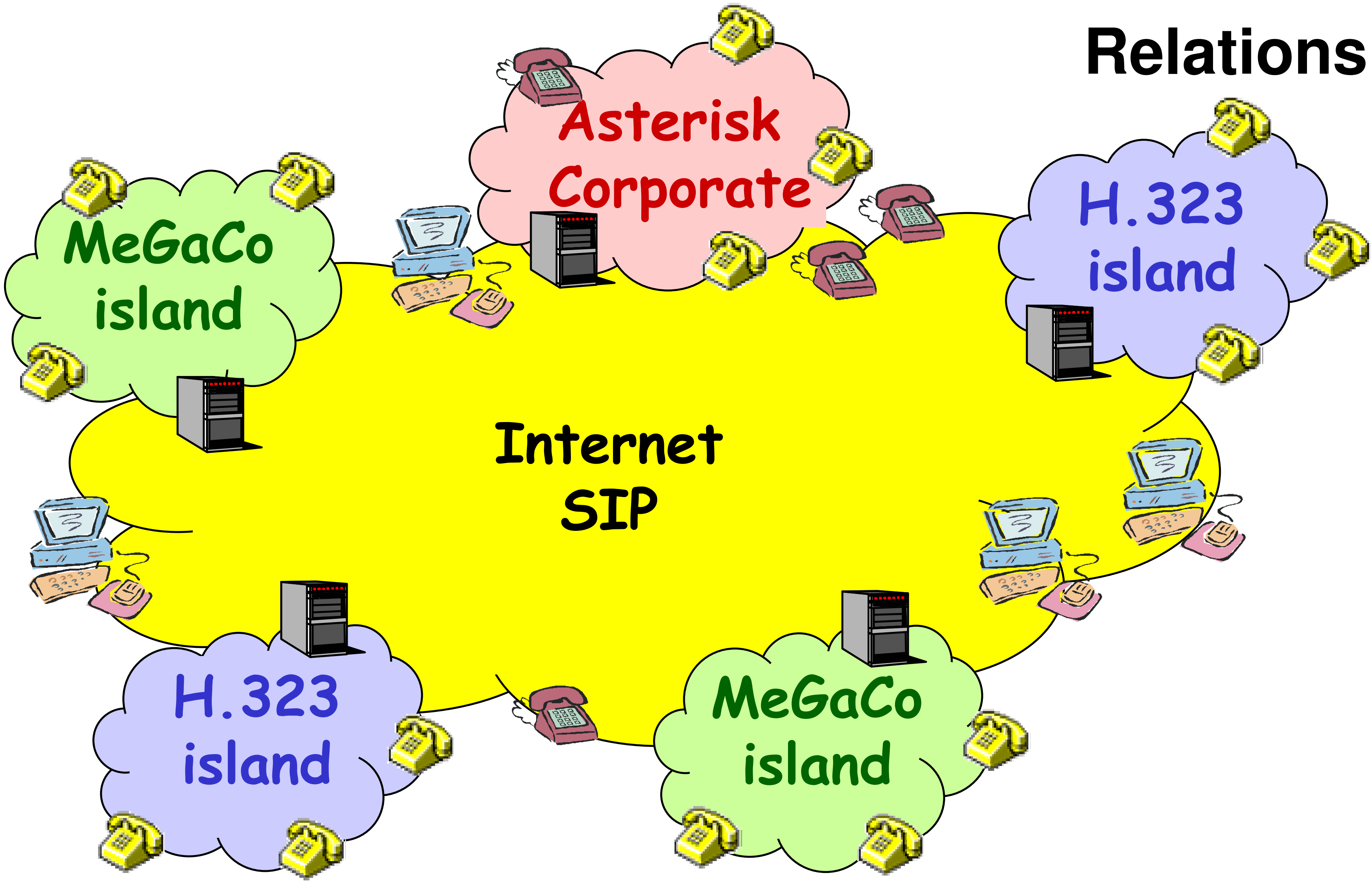
Internet **Telephony**

- voice oriented, try to emulate POTS on top of IP
- "ISDN-like" signaling, protocol piles separated for signaling and data
- aims at the integration with SS7

Internet Telephony

- VoIP ⇒ **Y.A.I.S.** (Yet Another Internet Service) like -casting, conferencing, ...
- voice will be a tiny fraction of the traffic
- integrates voice with mail, web, etc.
- telephone is just a particular case of voice, which is a particular case of media, and sessions can be multi-media

Protocols & Relations



Standard protocols: H.323

- H.323: “Packet-based multimedia communications systems”
  - recommendation from ITU-T
  - used to establish, modify, terminate multimedia sessions (e.g. VoIP calls)
  - it is based on H.320 (ISDN Videoconferencing)
    - multistage signaling
    - good interoperability with PSTN
    - it inherits its complexity
    - recent recommendations extend it to wide deployments
  - some operators deployments are still H.323-based
    - many operators have already SIP in their core network

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Standard protocols: SIP

- SIP: Session Initiation Protocol
  - IETF standard
  - used to establish, modify, terminate multimedia sessions (e.g. VoIP calls)
  - it is based on HTTP (light protocol)
  - it inherits its vulnerabilities
  - easily extensible
- It supports name mapping and redirection services transparently
  - personal mobility: one single externally visible identifier regardless of the network location
- Where is SIP used?
  - corporate deployments
  - 3GPP IMS (PS signaling protocol)
  - TISPAN NGN will be based on core IMS and thus on SIP as well

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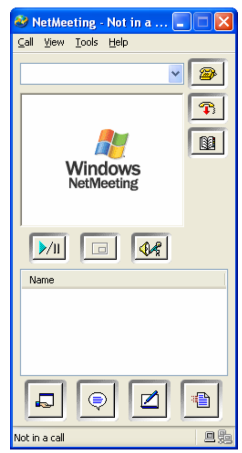
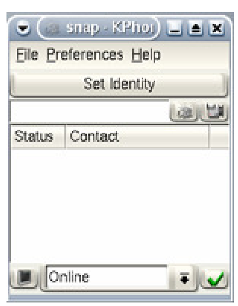
VoIP: architectural elements (I)

- Terminals (end-points)
  - hardware clients
  - software clients
  - optional
    - video codec
    - data transmission
    - instant message
    - presence





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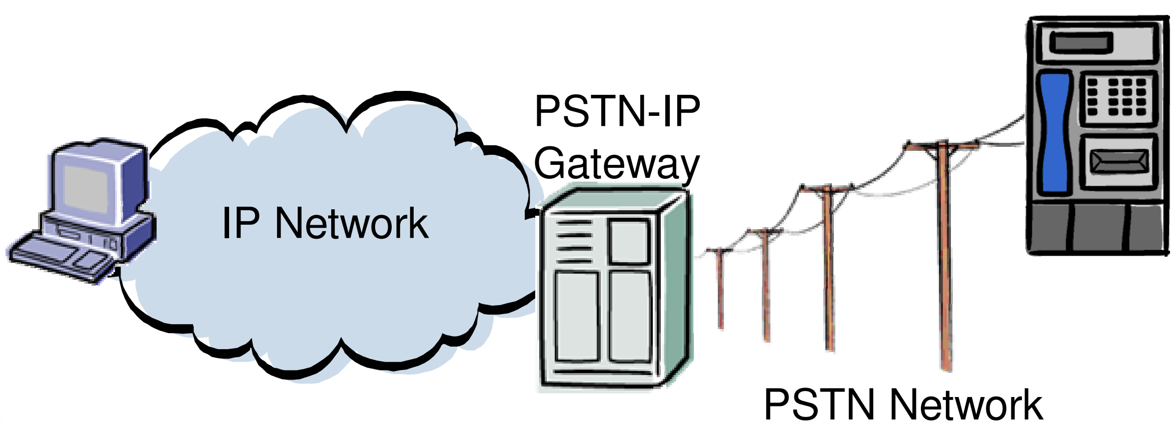
VoIP: architectural elements (II)

- Gateway
  - generic: an interface between two worlds
  - specific: interface between packet-based networks and circuit switched networks or between different architectures in packet-based networks (e.g. SIP—Skype)



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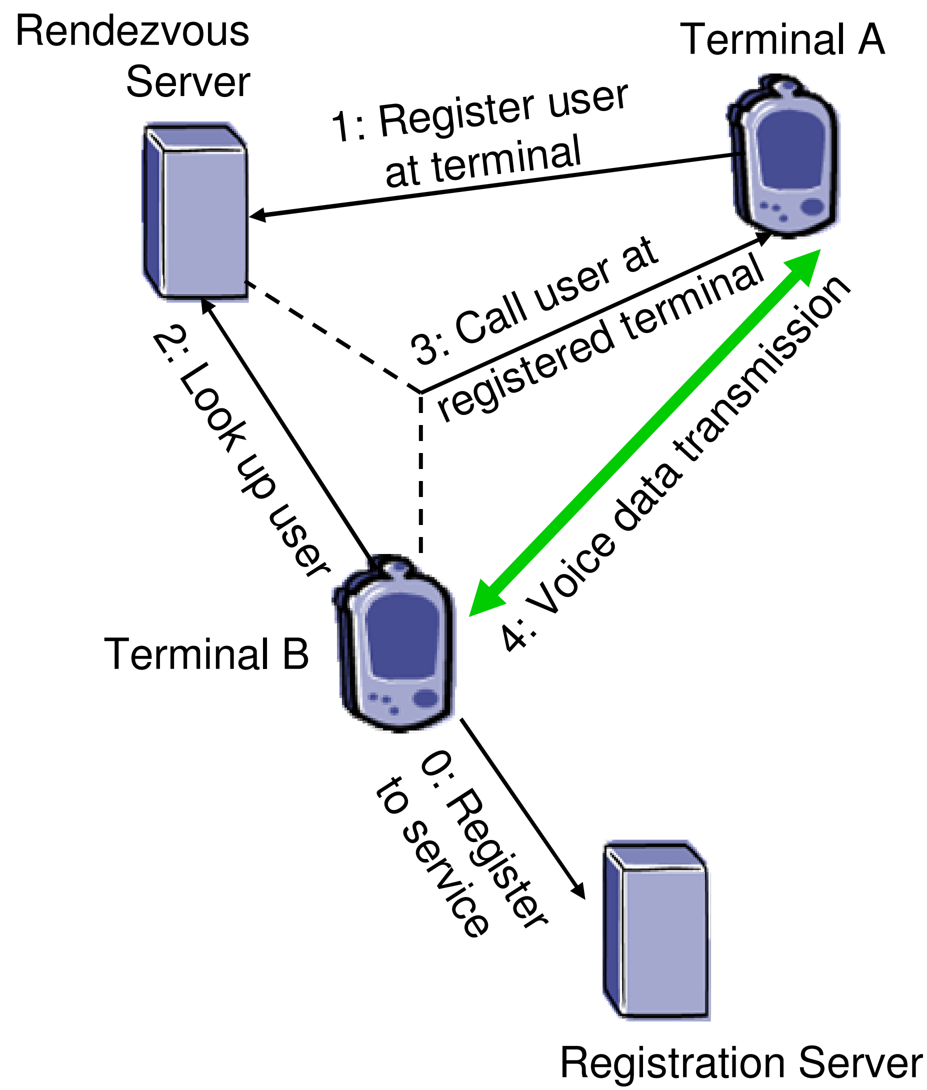
VoIP: architectural elements (III)

- Rendezvous server
  - H.323 world: Gatekeeper
  - SIP world: Proxy server
  - Main functionalities
    - Managing entities in its domain
      - Endpoint registration
    - Address translation
      - user identity to to terminal location
    - Call routing
      - Next hop location
- Additional servers
  - application servers
    - registration servers
    - conferencing server
    - presence server
    - etc.



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H.323: Delving deeper

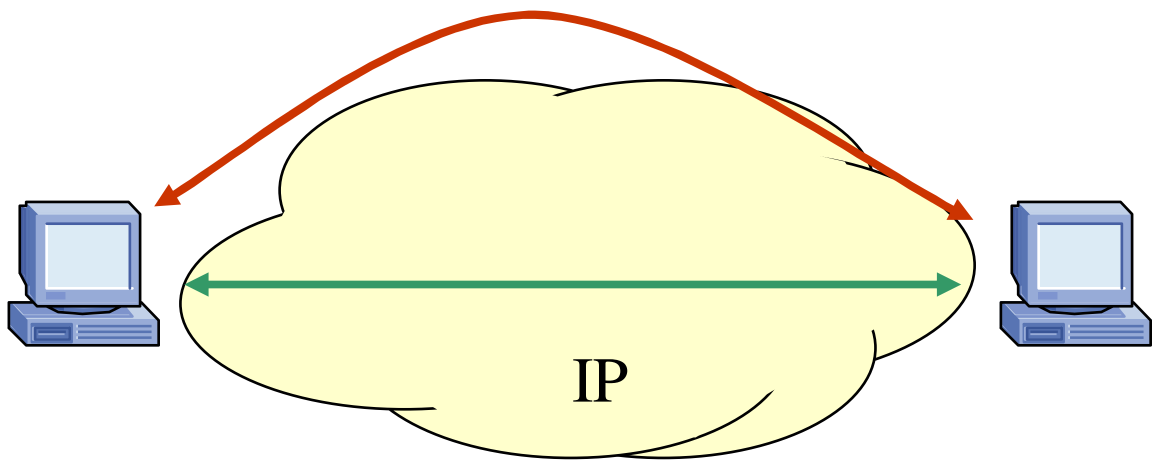
- It’s the first architecture developed fo audio/video services on packet (not necessarily IP!!) networks
- Is has been defined in the “telco” (ITU-T) world, it’s probably still the most diffused protocol for VoIP ....
- .... but just because it was the first one ....

H.323

- The architecture is derived from video-conferencing in LANs services defined in the '80s and early '90s
  - Version 1 (1996) Multimedia over LAN
  - Version 2 (1998) Telephony over IP
  - Version 3 (1999) + Communications across administrative domains
  - Version 4 (2000) + Supplemental services + web-based service creation
  - Version 5 (2003) + Use of URLs and DNS + Video conferencing support + ...
  - Version 6 (2006) + Security + ...

H.323 architecture

- Enables direct end-to-end signaling
  - terminal interconnection
  - logical channels (for the media) set up
- Uses directly the IP address and the TCP/UDP ports



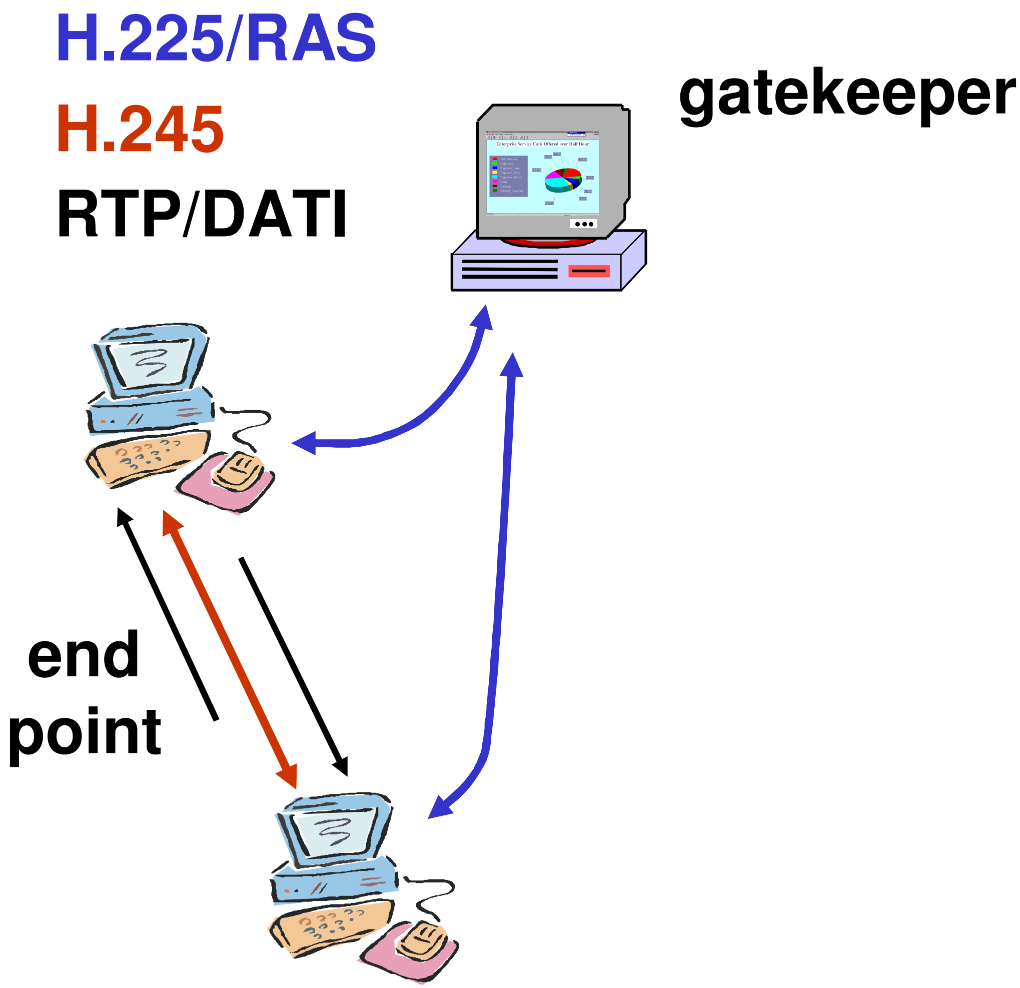
H.323 elements (logical devices)

- End-point: terminals enabled for communications
- Gateway: inter-working unit with other networks (PSTN/ISDN and SIP in particular)
- Gatekeeper: controls communications (central office)
- MCU (Multipoint Control Unit): multicast communications (conferencing) and supplemental services

H.323: compulsory components

- H.225 (connection and status control):
  - Q.931 user signaling
  - RAS (Registration, Authentication and Status) endpoint to gatekeeper signaling
- H.245: e-t-e signaling on terminal capabilities and "media" that support information
- RTP/RTCP: transport and flow control
- G.711: mandatory coding (64 kbps) all other codecs are optional!!

H.323 communication between "internal" terminals



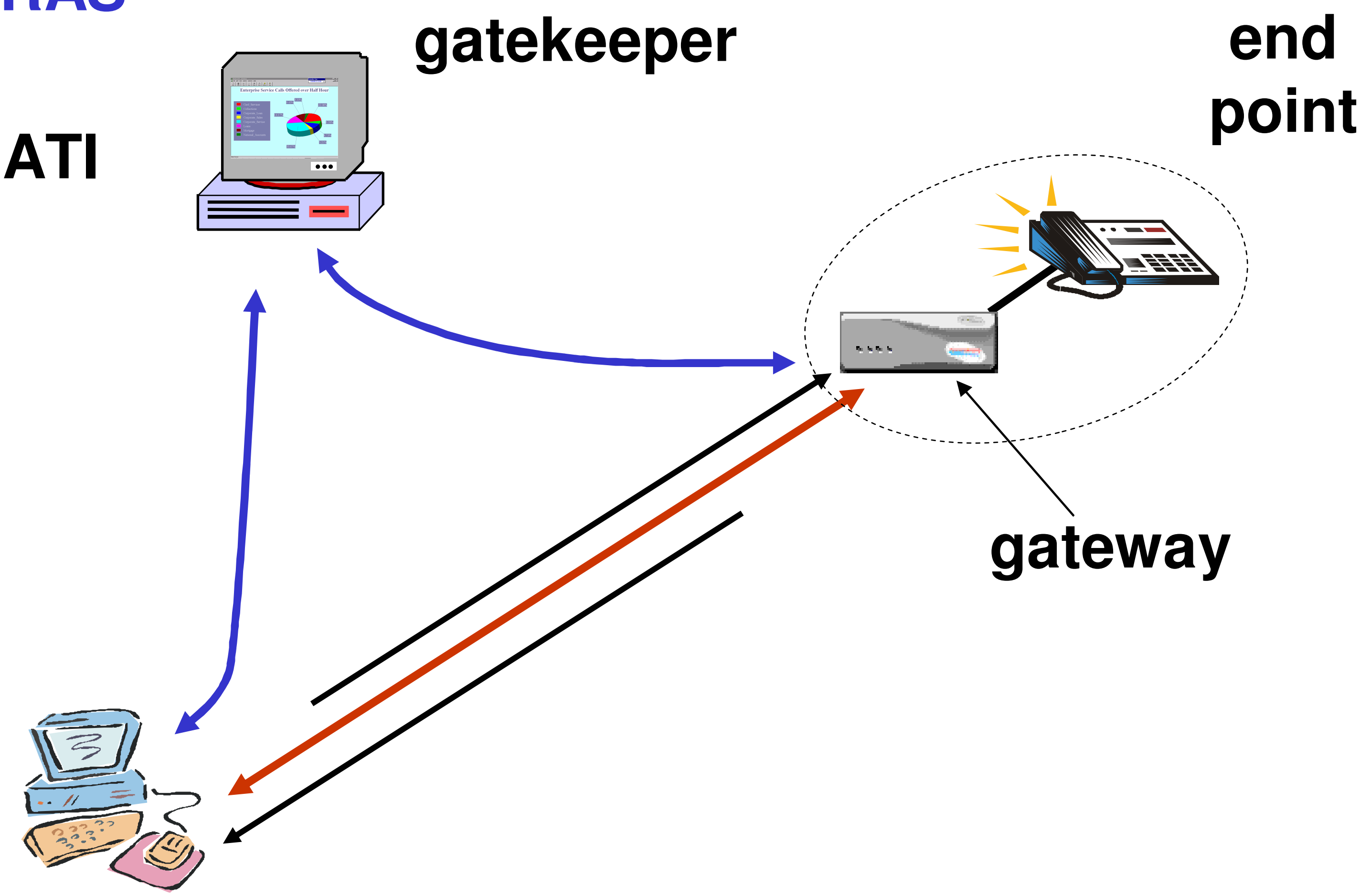
H.323

communication between 1 internal and 1 external terminal

H.225/RAS

H.245

RTP/DATI



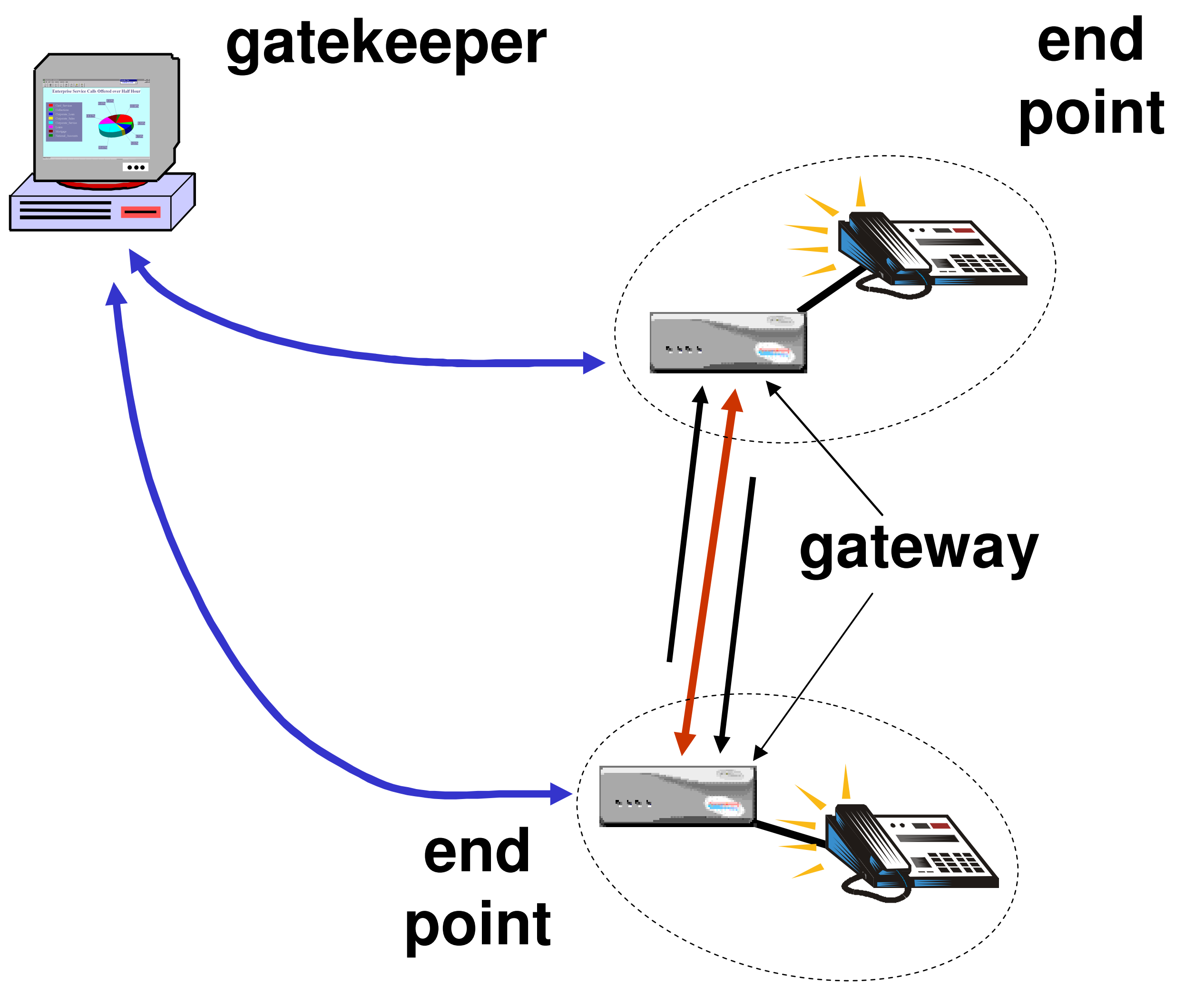
H.323

communication between external terminals

H.225/RAS

H.245

RTP/DATI



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H.323 architecture

- A H.323 network is composed by one or more "zones"
- One zone is a logical ensable of H.323 devicews managed by a single gatekeeper
- Zone boundaries can be based on administrative limits, addressing structures, geography, etc.
- Calls involving more zones are managed involving more gatekeepers, a working mode defined in Version 3 and available in devices 2001-02

Gatekeeper

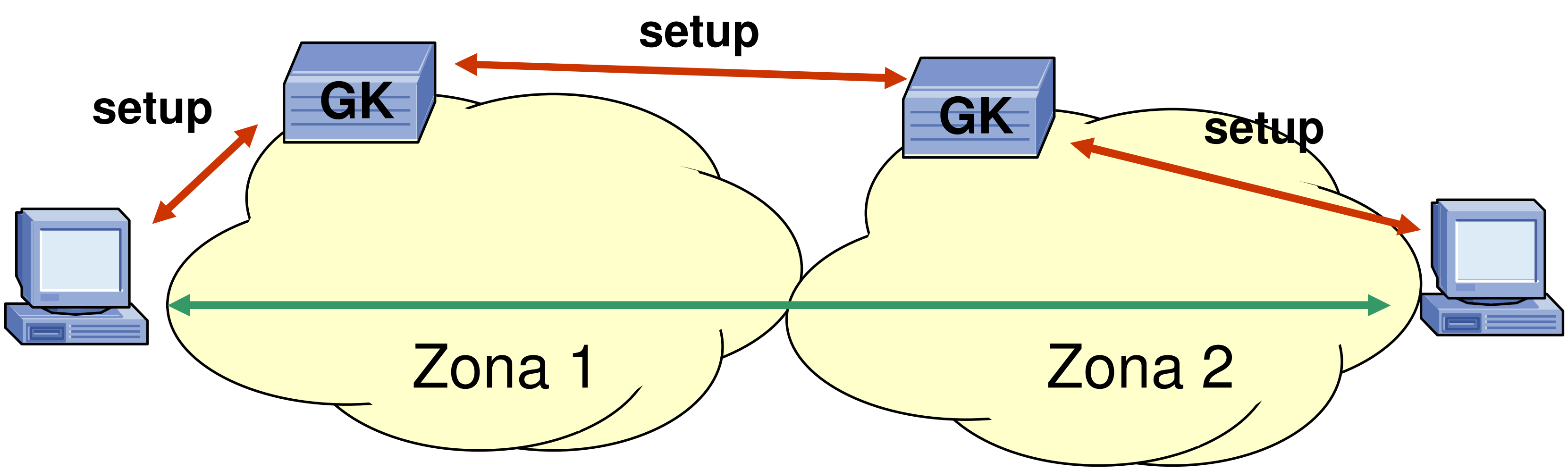
- It's the "intelligent" device of H.323 architecture and services
- Each gatekeeper manages a "zone" (a collection of end-points, gateways, MCUs)
- It has the following compulsory functionalities:
  - Admission Control (verification of end-points authrization to place and receive calls)
  - Address translation (telephone alias <-> IP)
  - Bandwidth control (if required by the call)
  - Zone management

Gatekeeper

- May implmenen optional functions and features
  - Autorization
  - Resource Management
  - Call control signalling (act as randevouz point also for terminal-to-terminal signaling –H.245)
  - Resource Reservation (for end-point not able to run reservation protocols like RSVP)
  - Call management (multimedia calls and complex services)
  - Gatekeeper management information (remote management via SNMP on standard MIBs)
  - Directory services

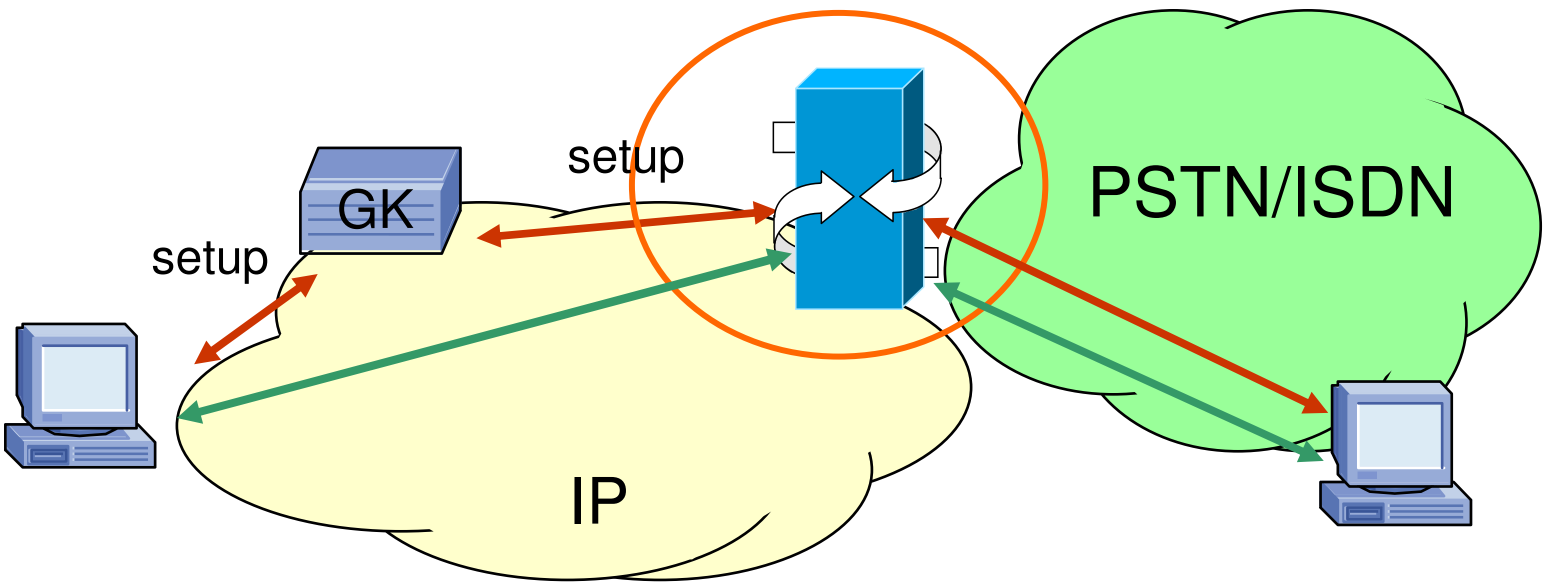
H.323 architecture

- Il Gatekeeper can be a proxy signaling
- May be the interface toward additional services
- May also force data-flow switching, behaving as a traditional PBX (computationals and traffic burden)

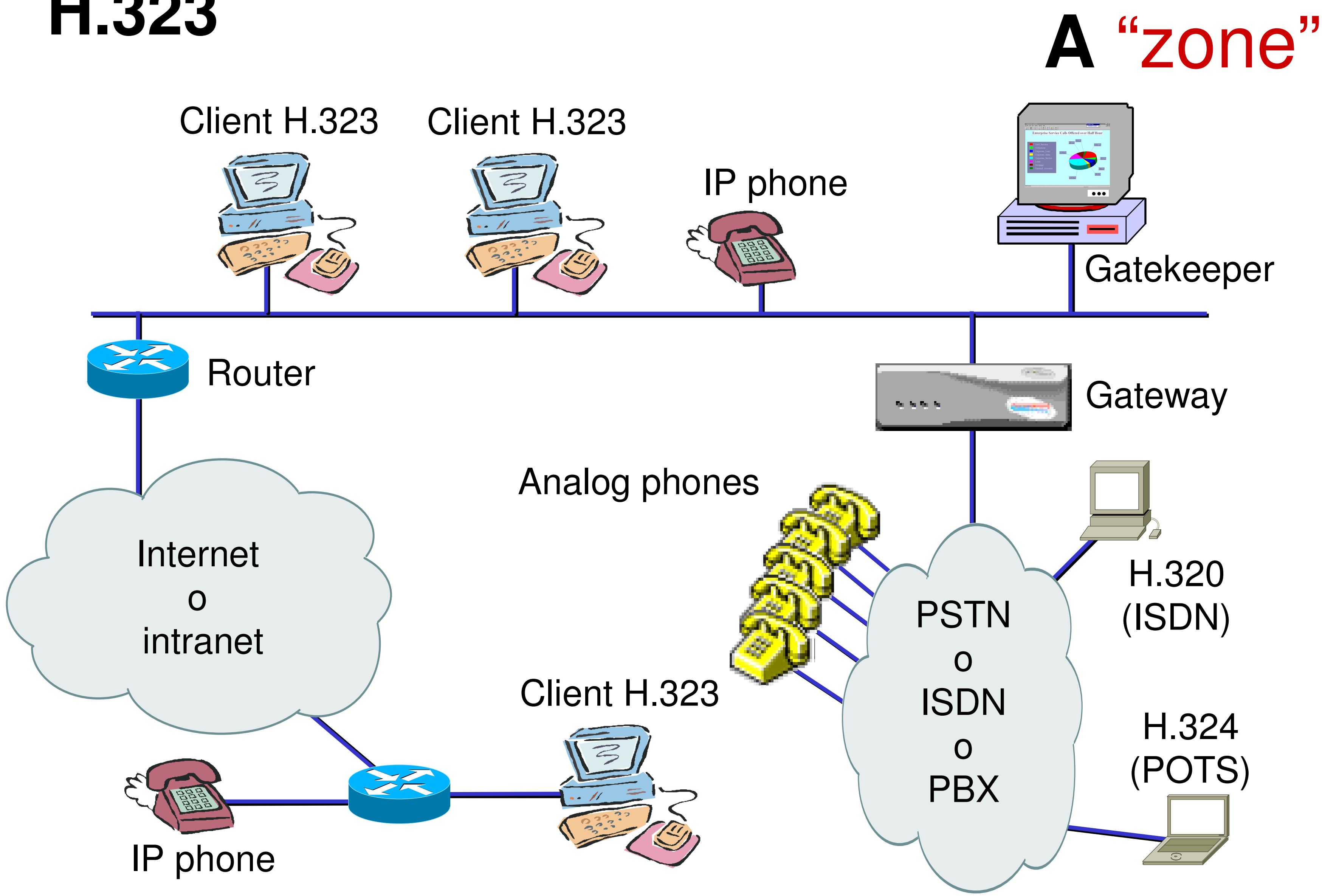


H.323 architecture

- **Gateways** are devoted to interworking with other architectures, and specifically with PSTN
- Also other VoIP archietctures (SIP, Skinny, Astersk (IAX), skype and other proprietary protocols)



H.323



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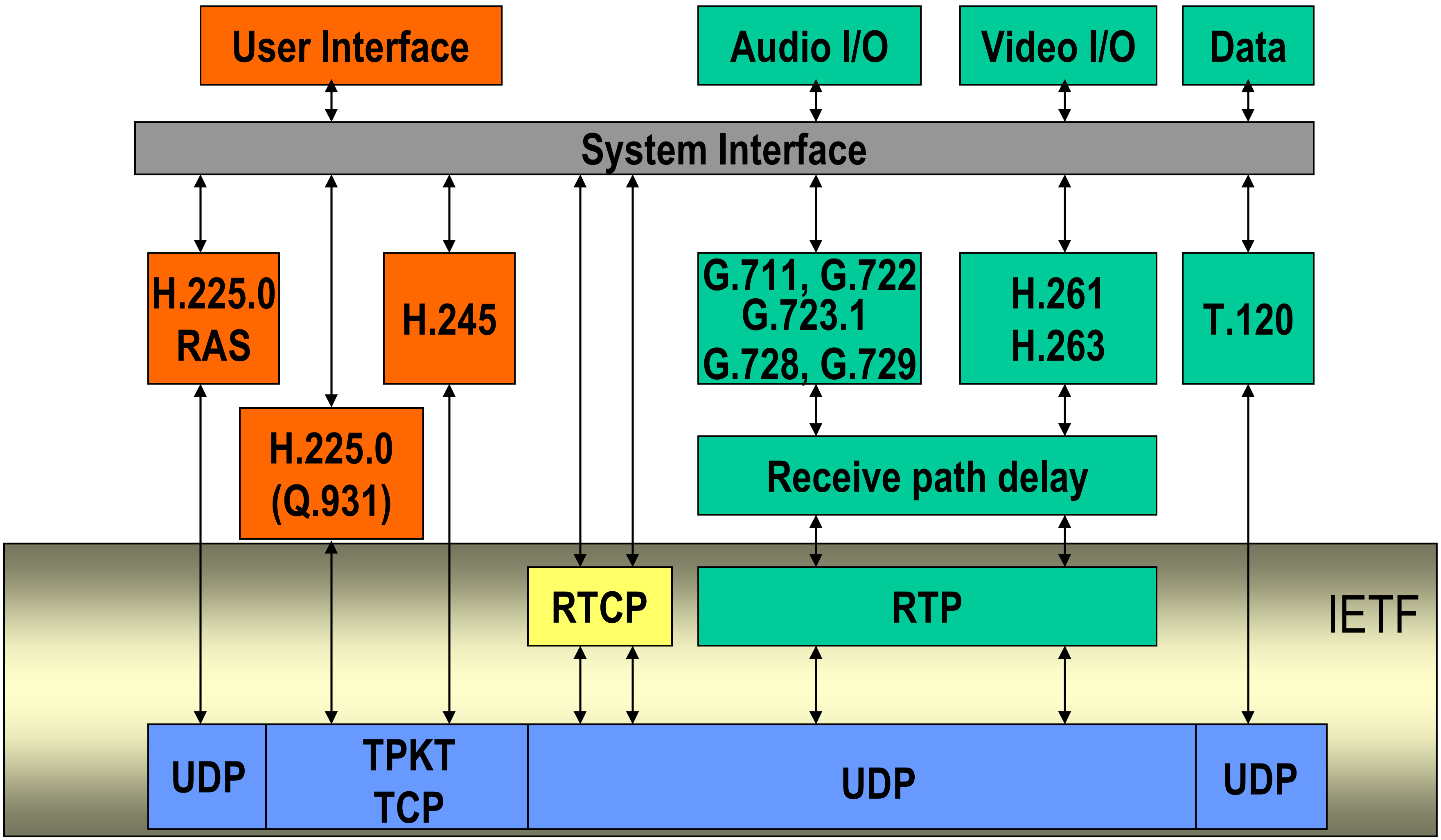
H.323 protocol stack



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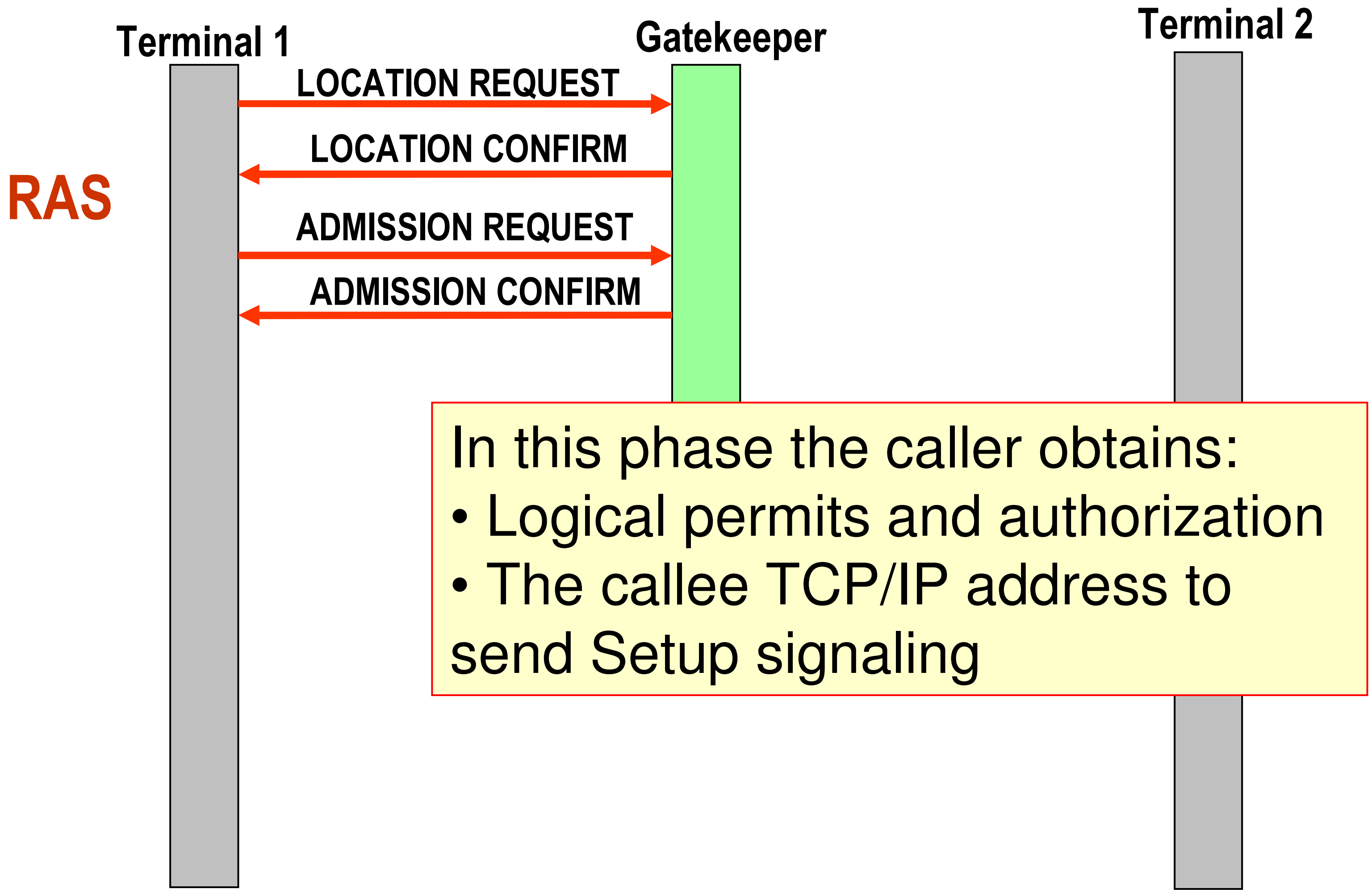
H.323: RAS



In this phase the caller obtains:
- Logical permits and authorization
- The callee TCP/IP address to send Setup signaling

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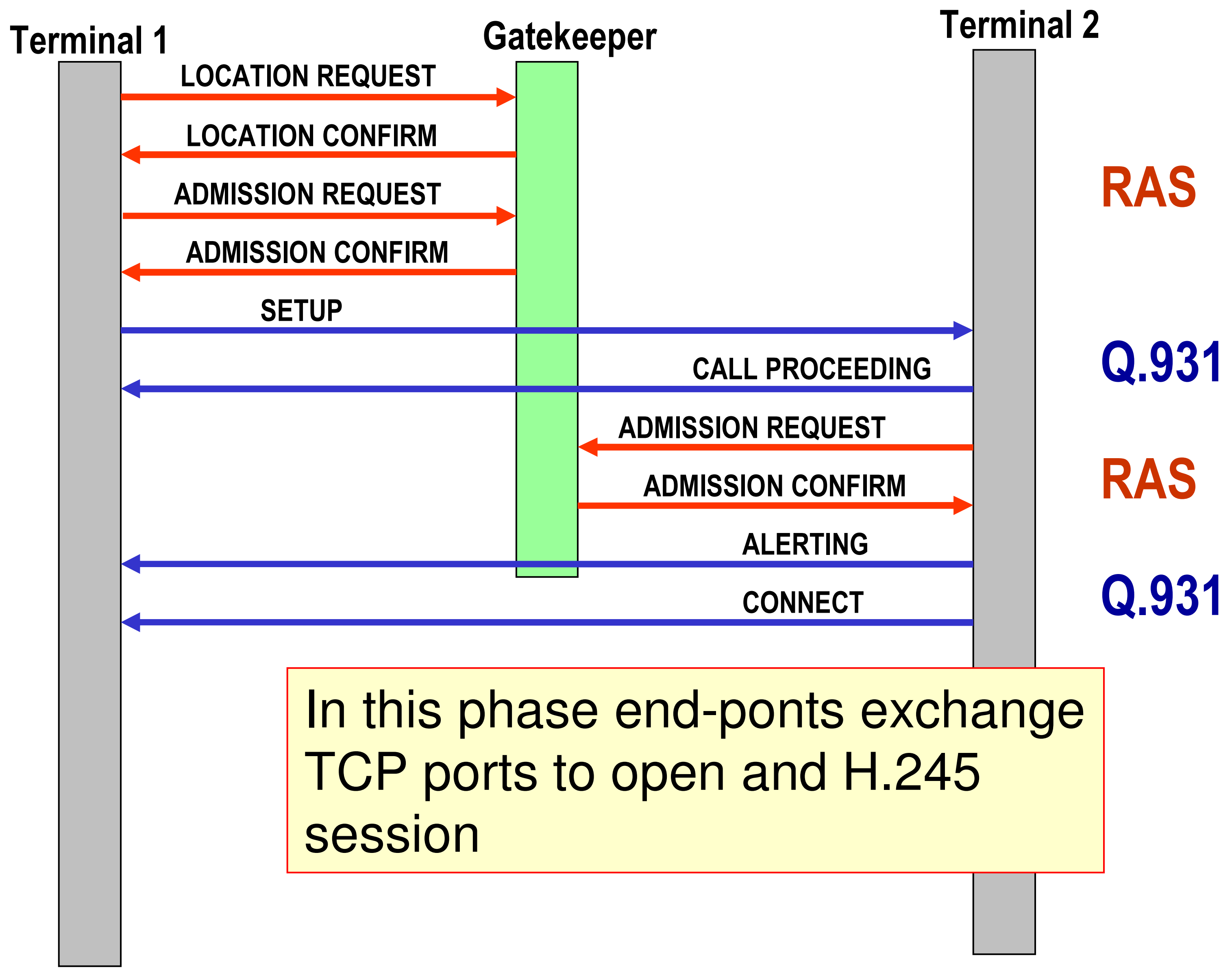
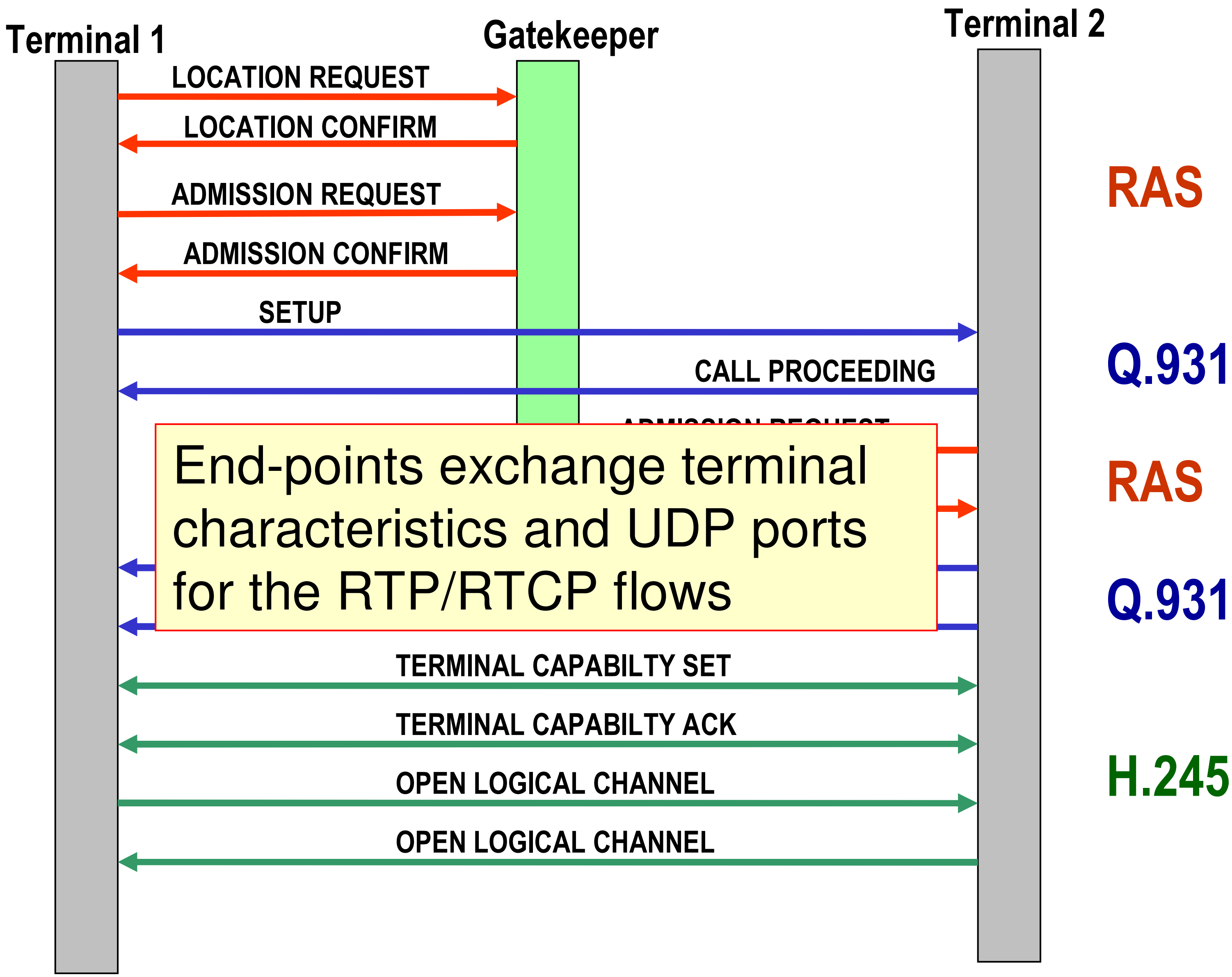
H.323: Q.931phase

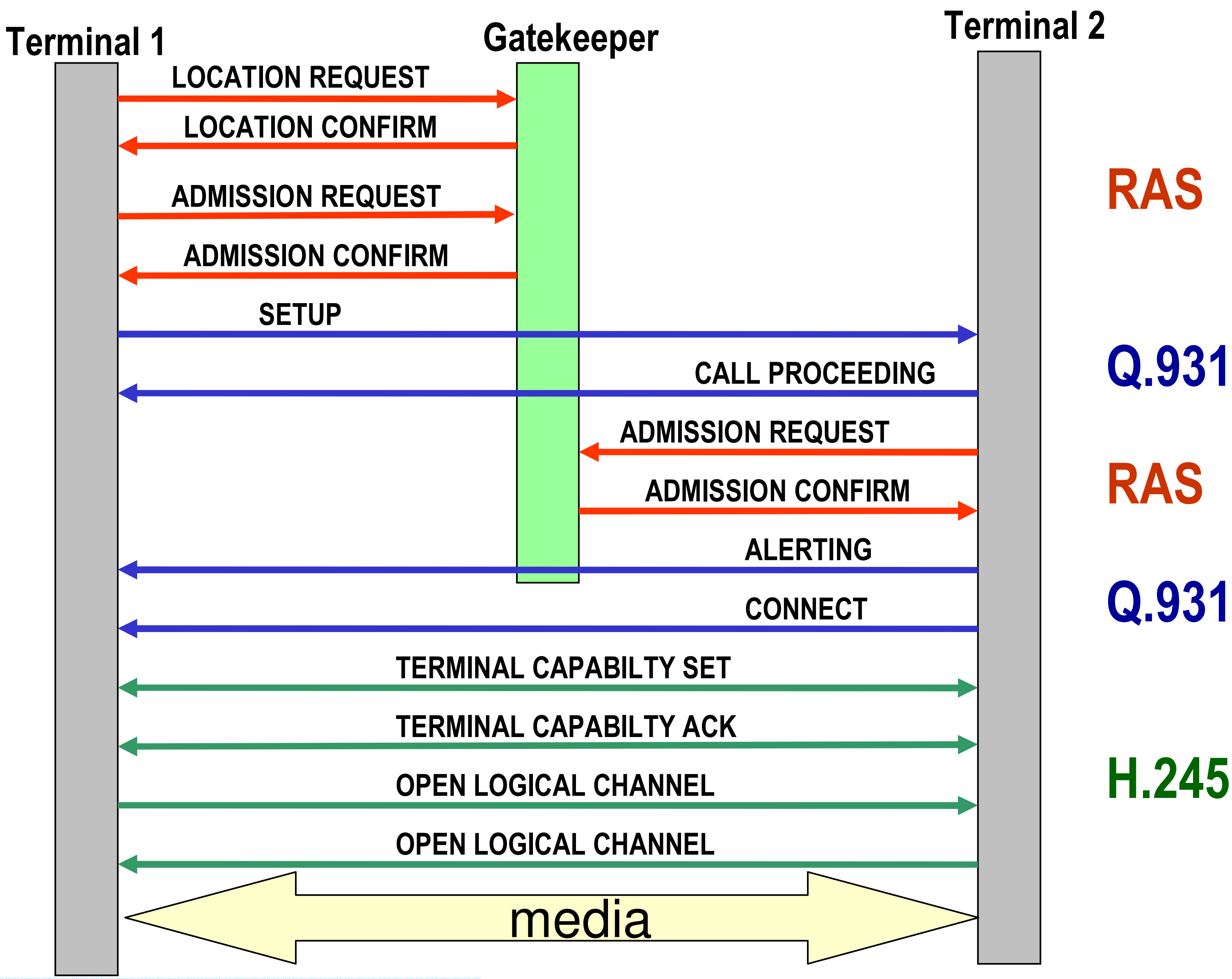


In this phase end-ponts exchange TCP ports to open and H.245 session

H.323: H.245 phase



H.323: media exchange phase

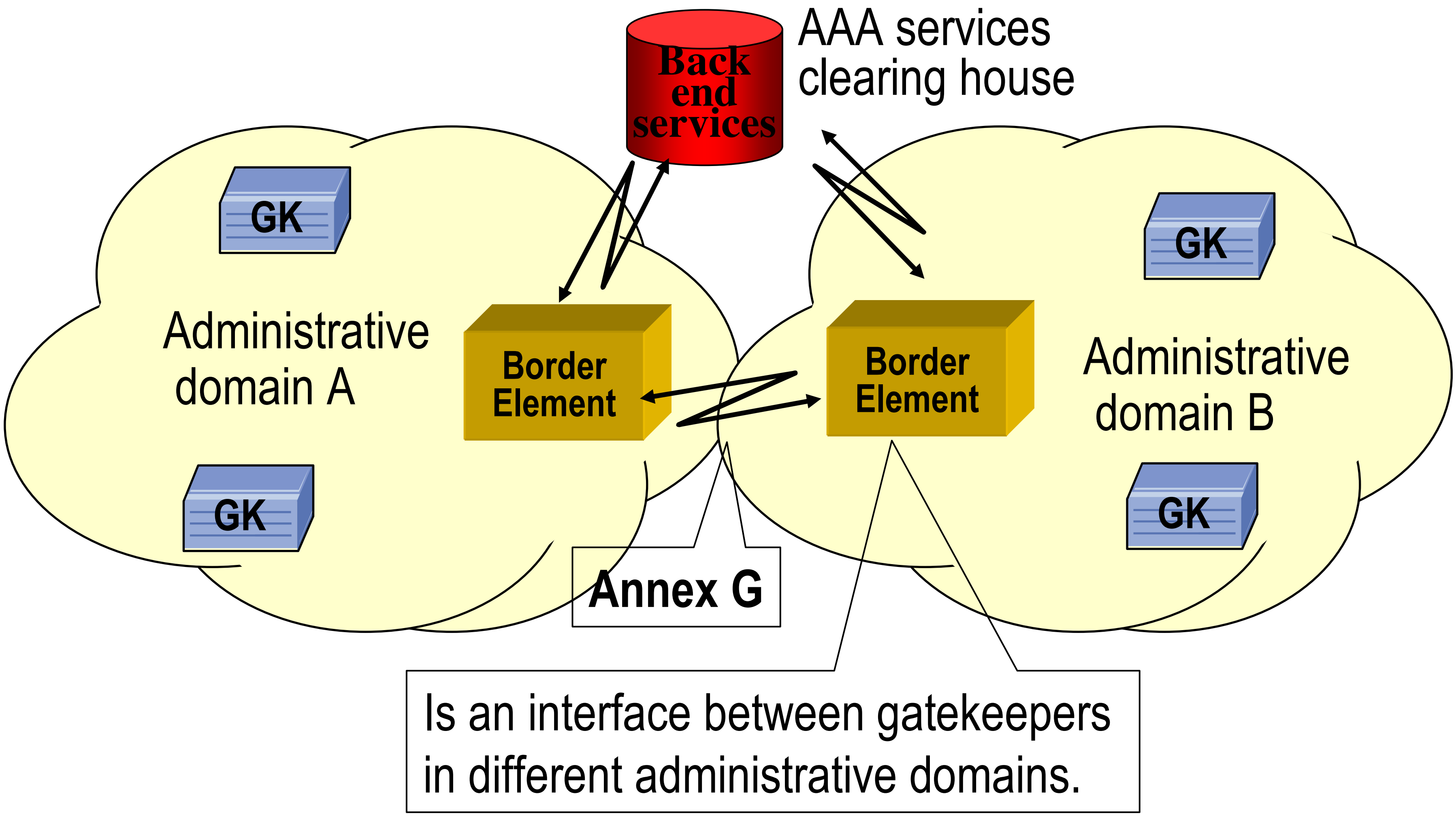


H.225: Annex G

- Introduces methods to implement fundamental services when a call is set-up across multiple administrative domains
  - global address resolution
  - access authorization
  - usage reporting

- Introduces a new netwokr element: the Border Element

H. 225 : Annex G


Back end services
AAA services clearing house
GK
GK
Administrative domain A
Border Element
Border Element
Administrative domain B
GK
GK
Annex G
Is an interface between gatekeepers in different administrative domains.

H. 225 : Annex G

- Border Elements (and Clearing Houses) exchange information on:
    - reachability
    - cost

- "I'll route calls to 1303*, and I'll charge 8 cents a minute peak, 5 cents a minute off peak"
- "I can resolve everything for 33*"
- "I can resolve everything for *@cisco.com

H.323 H.225.0 architecture under Annex G