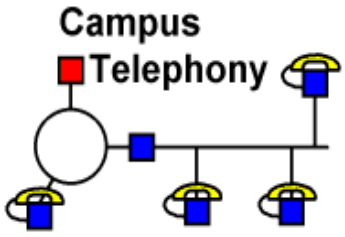
Packet Based Telephony



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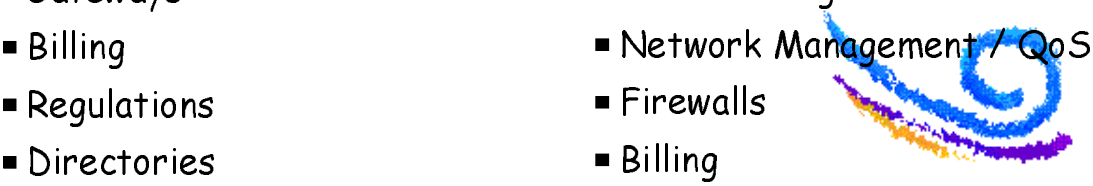
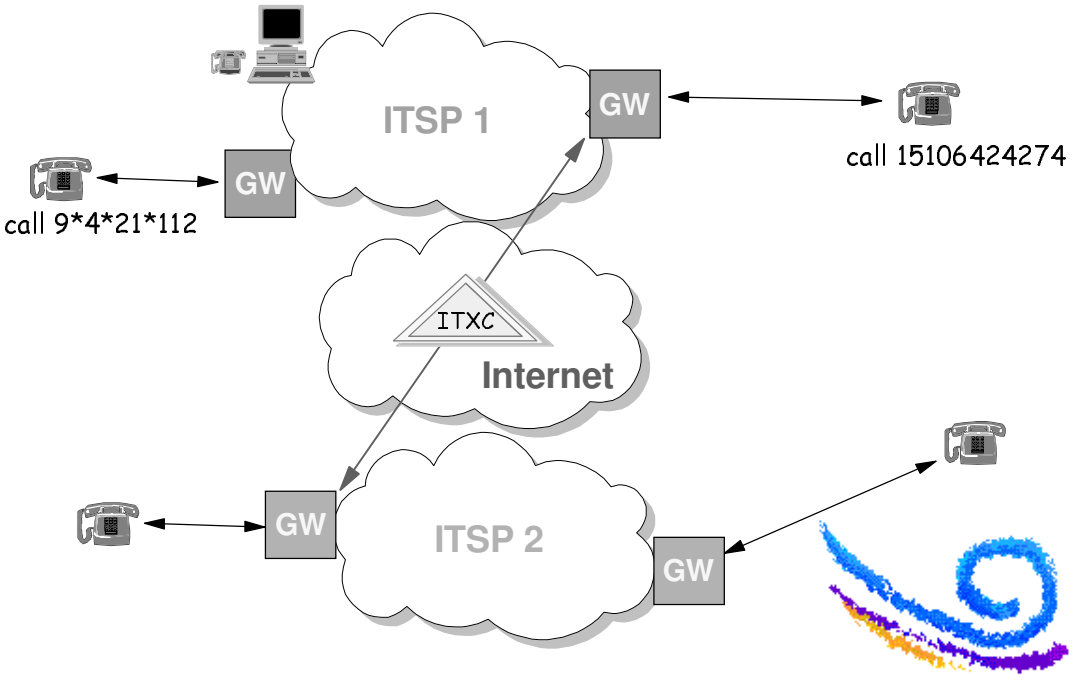
- Business Case
- Products / Companies
- Standards
- Problems and Solutions

Why Use Packet Networks for Voice?

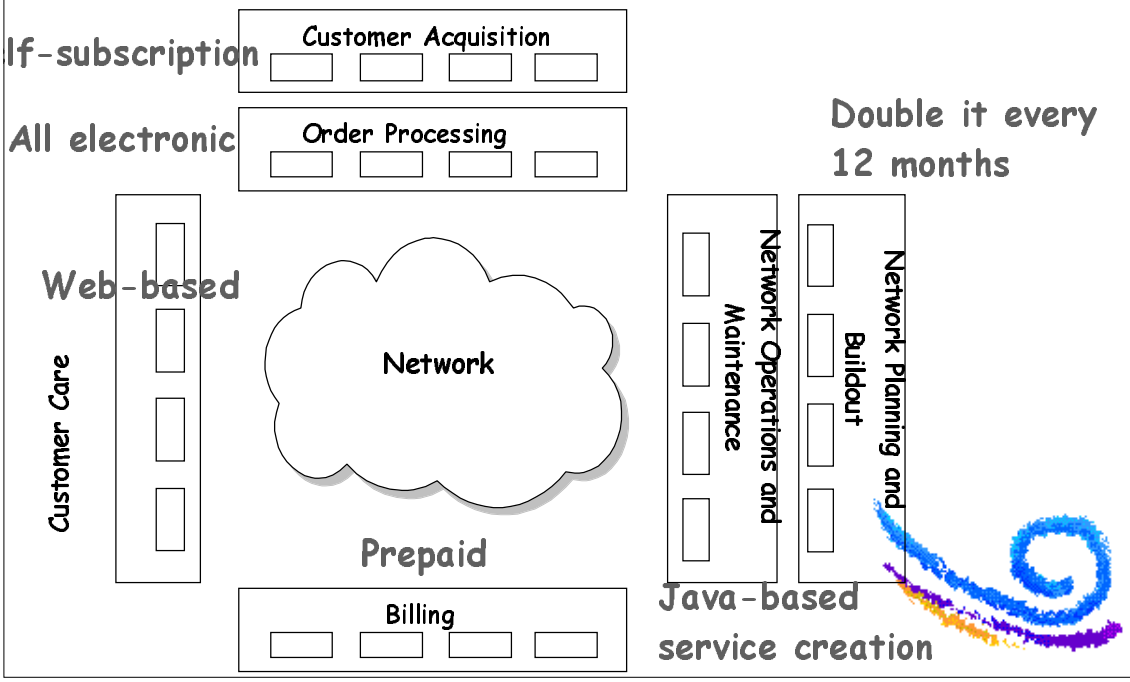
- **Internet**
  - cheap long distant calls/fax
- **Problems**
  - Network Delay / QoS
  - OS Delay
  - Service Agreements between Providers
  - Gateways
  - Billing
  - Regulations
  - Directories
- **Intranet**
  - only one network to install and maintain
  - cheap extensibility
  - new features:
    - mobility / multiple lines
    - multimedia conferencing
- **Problems**
  - Interworking with PBX
  - Network Management / QoS
  - Firewalls
  - Billing



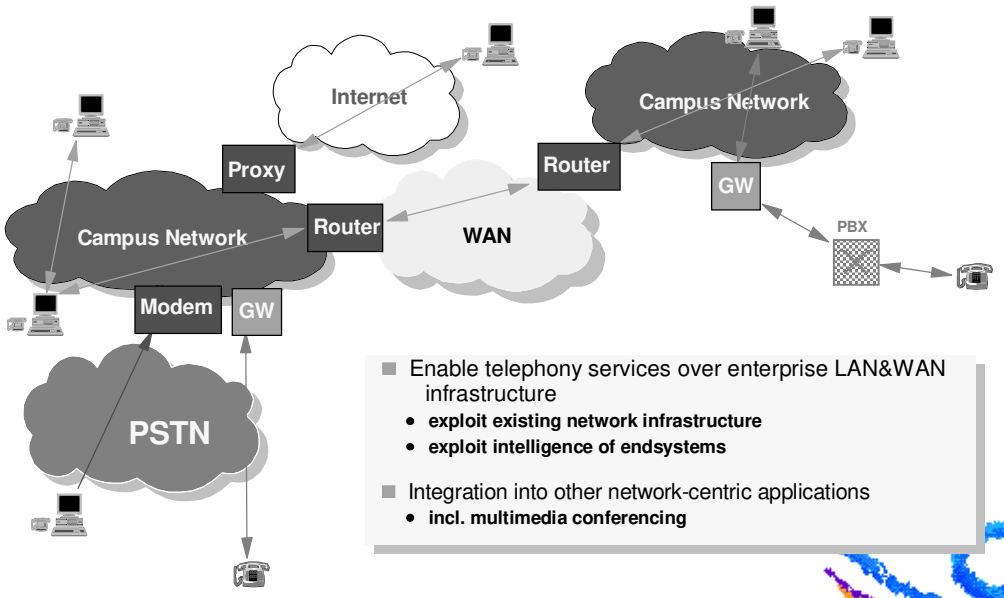
Internet Telephony



How do ITSP work



Intranet Telephony

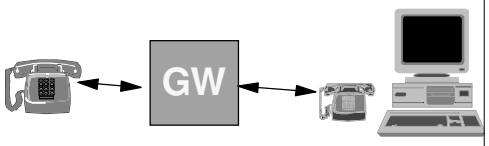


- Enable telephony services over enterprise LAN&WAN infrastructure
  - **exploit existing network infrastructure**
  - **exploit intelligence of endsystems**
- Integration into other network-centric applications
  - **incl. multimedia conferencing**

Products: IP Phones

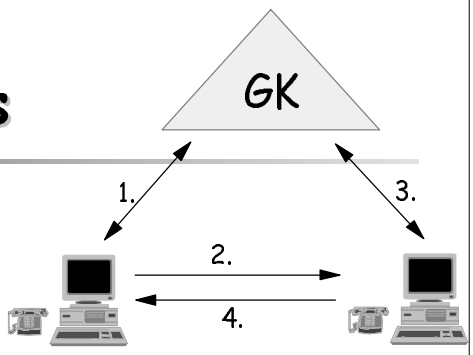
- **Companies**
  - Microsoft (Netmeeting, free, crippled conference support)
  - IBM / Vocaltec (~$40) / Intel
- **Features**
  - some with and some without conferencing
  - no directory support or non standard LDAP or ILS
  - some use non standard signaling or crippled signaling
  - Some support application sharing, whiteboard, video
  - no OS support (especially in Windows systems)
  - no QoS

Products: Gateway



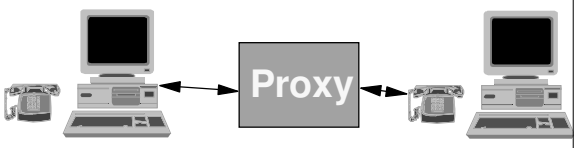
- **Companies**
  - IBM
  - Vocaltec
- **Features / Problems**
  - mostly support only a few (10-20) lines, needed are gateways for hundreds of lines
  - no standard gateway/gateway protocol to discover gateways
  - no billing between gateways

Products: Gatekeepers



- **Companies**
  - Databeam (IBM)
  - Radvision
  - PictureTel
- **Features**
  - All standard compliant H.323 IP Telephony Products need to ask their Gatekeeper for permission for each call.
  - Supports billing in H.323 networks.
  - crude bandwidth control (no real QoS yet)
  - Supports Directories in H.323 networks
  - no Gatekeeper/Gatekeeper protocol

Products: Proxy



- **Companies**
  - IBM
  - Intel
  - Cisco
- **Features**
  - Supports call through Firewalls
  - an H.323 Proxy needs to listen into the call signaling and forward TCP and UDP streams (breaks call security!)
  - needs directory integration

Products: PBX Gateways

- **Companies**
  - none
- **Features**
  - seamless integration between IP users and PBX users
    - forwarding
    - conferencing
    - group calling
  - PBX's have propriatory interfaces
  - PBX vendors have no interest to provide those systems

Products: MCU

- **Companies**
  - Databeam (IBM)
  - PictureTel
- **Features**
  - mixes audio and sends out one stream to each participant
  - mixes video (different modes possible)
  - can invite other participants to the conference

Products: assorted other ones

- H.323 / H.320 gateways
- Routers with phone jacks (Cisco)
- special cards to connect a POT to a computer via cards or serial I/O connection
- cards to connect analog lines, ISDN, T1, and other connections to computers.
- Telephones with Ethernet connection

Standards for IP Telephony

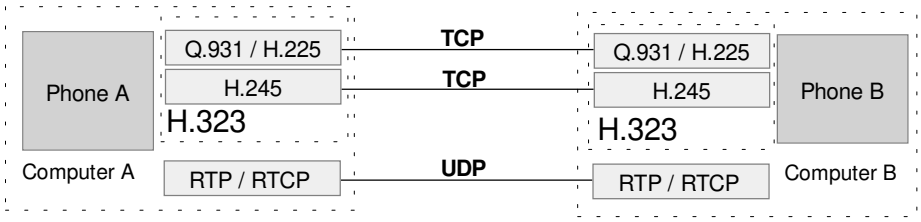
- H.323/H.245/H.225/H.235 (signaling)
- SIP/SDP (signaling)
- RTP/RTCP (media streaming)
- T.120 (data conference)

- All those protocols have no need for a centralized entity, not even for conferencing!

H.323

- defined by the ITU
- telco background (builds on ISDN)
- uses ASN.1 PER encoding
- very flexible capability exchange
- too many options make interoperability except for basic features problematic
- security extension (H.235) is too open
- uses crippled RTP for media streaming

The H.323 Protocol



Q.931 / H.225:

- initial connection
- ringing
- name / authentication
- hold / retrieve
- forward / transfer
- hang-up
- H.245 connection parame

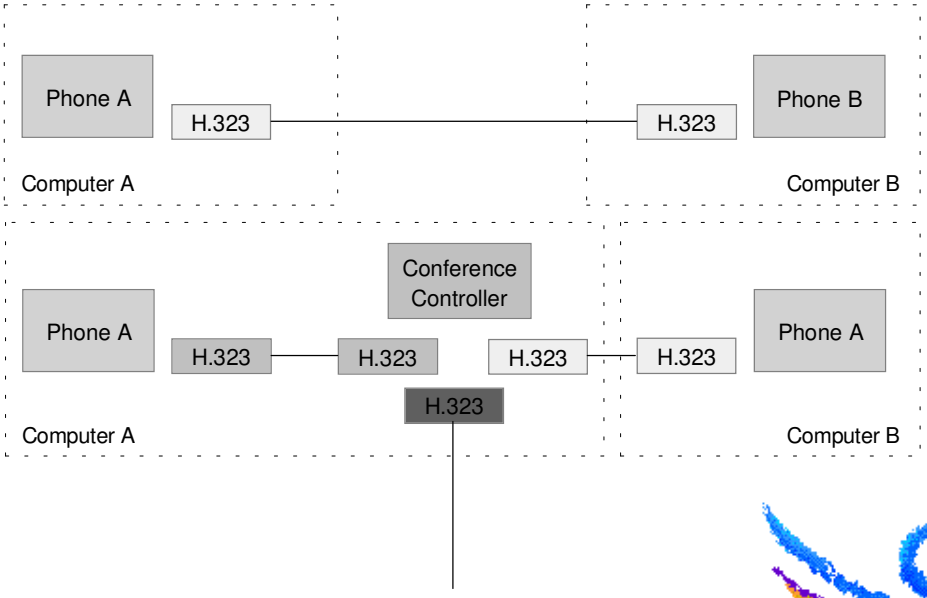
H.245:

- Capability exchange / selection (codec)
- hang-up
- Conferencing
- name / authentication (C)
- hold / retrieve (C)
- RTP connection parameters

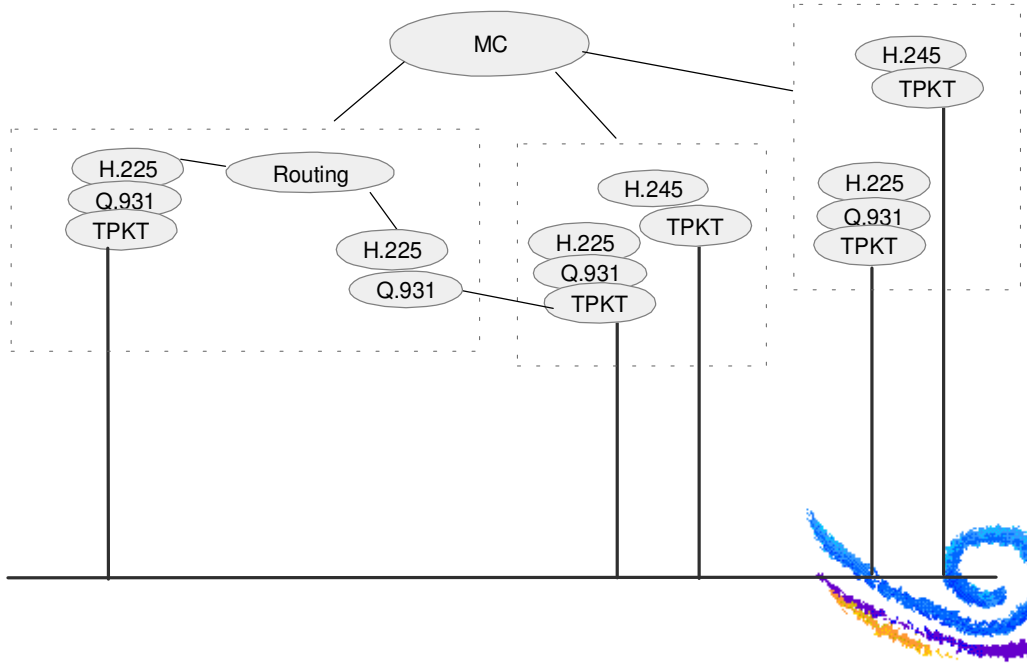
RTP / RTCP:

- Audio / Video transport
- network monitoring
- voice activity

H.323 ad hoc based Conferencing



Inviting an additional Participant



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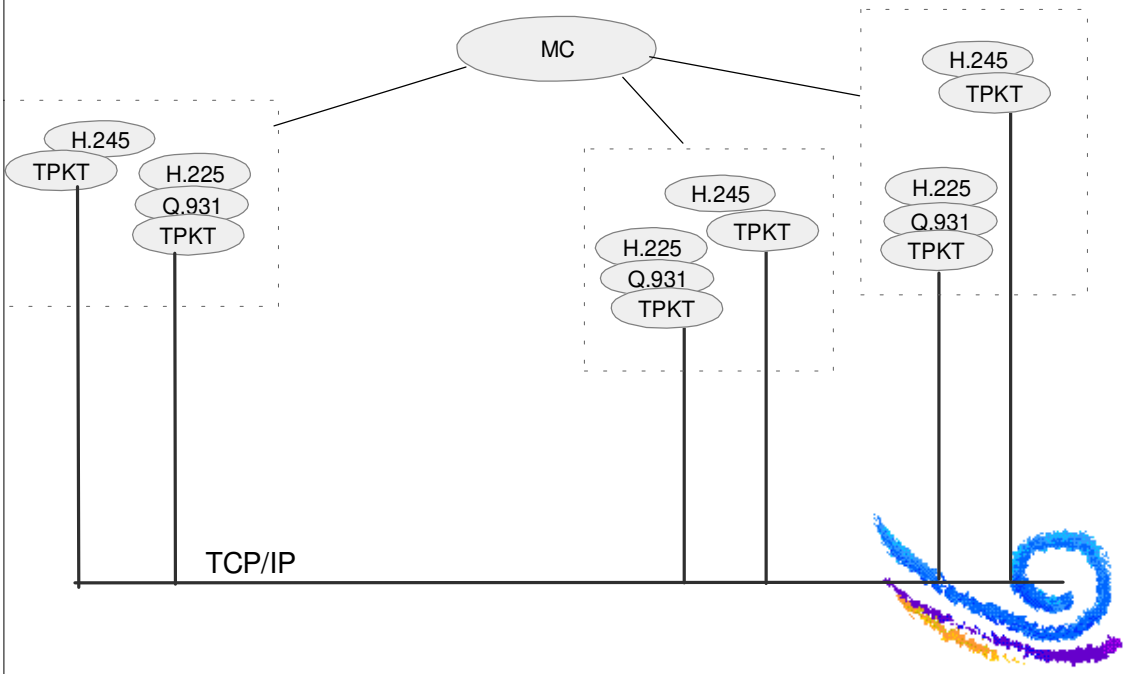

SIP

- Defined by the IETF
- Internet background
- uses ASCII protocol
- uses RTP for media streaming
- flexible security scheme
- no business support except for MCI
- does not keep the signaling connection open during the call

RTP/RTCP

- Media Streaming protocol defined by the IETF
- used in the MBone architecture
- too flexible for telephony and multimedia conferencing

RTP / RTCP

Sender:

- there can be many senders and receivers
- a computer can send several media streams
- one session has either video or audio

Receiver:

RTP - Packet:
- sequence number
- timestamp
- unique identifier (SSRC)
- codec used

RTP Session
either Multicast or
Multi/Unicast

RTP - Packet:
- sequence number
- timestamp
- unique identifier (SSRC)
- codec used

RTCP Sender Report:
- packets sent / bytes sent
- unique identifier (SSRC)
- name/email address/...

RTCP Receiver Report:
- packets lost
- delay jitter
- name / email address / ...

RTCP Sender Report:
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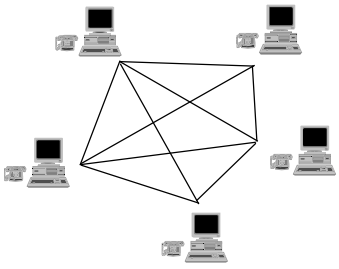
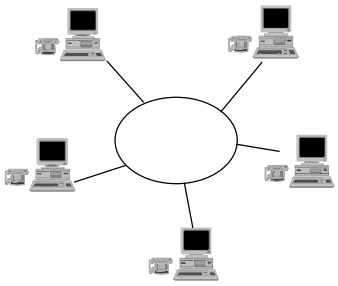
Media Streams in Conferencing

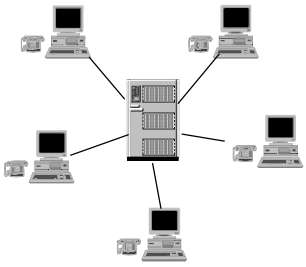
Multi/Unicast (full mesh)



Multicast



Centralized Mixing



)ifferent medias can
ıse different models

T.120

- Defined by ITU
- Protocol for shared whiteboard, chat, file transfer, application sharing
- Includes conference control
- will be controlled by H.323 in H.323 phones
- uses ASN.1 PER encoding

Problems

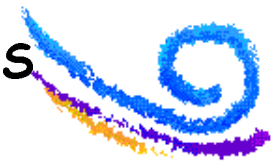
- Billing
- QoS
- Interoperability
- Directories
- Security
- Proxies

Billing

- A centralized entity is needed for billing, but there is none!
- Gatekeeper are supposed to solve the problem, but they can be bypassed
- Proxies / Gateways can be used
- Interoperability with PSTN network?
- Security of billing?
- what will be billed (bytes? time? QoS services?
- How do we pay (CC?, Prepaid?, Invoice?)

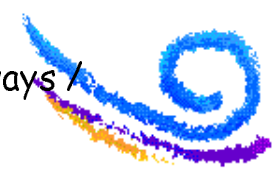
Quality of Service

- RSVP is not scaleable and not everywhere accessible
- ATM is not delivered to the desk
- VPN make QoS an even bigger problem
- Solutions include Differentiated Services and splitting QoS service from normal service
- Security issues: who can use QoS features for which applications?



Interoperability

- works fine for basic service
- does not work well for features like conferencing, forwarding, mobility, ...
- we need Interworking between
  - Gateways and PBX
  - Gateways / Gateways
  - Gatekeepers / Gatekeepers
  - Gatekeepers / SS7 (IN)
  - Directories / Directories / Proxies / Gateways / Gatekeepers



Directories

- LDAP can be used, but there is no standard mask
- ILS server don't interoperate and leak to much information to the caller
- we need a distributed directory service

Security

- security and call tracing/billing don't work together too well
- Proxies need to get at least some information, therefore TLS/SSL cannot be used end-to-end
- export and use regulations
- should work together with existing security architectures (i.e. kerberos)

Proxies

- support usually only one signaling type
- need to look at call info, so a real application is needed
- they need directory information
- none of the public directory servers knows how to work with proxies

The Future

- Telcos will move from voice network to packet networks.
- Permanent IP connection is coming (ISDN, xDSL, Cable, Wireless and Electricity).
- all new PCs support videoconferencing, you just need a camera and a mike.
- QoS networks for business users will be necessary.