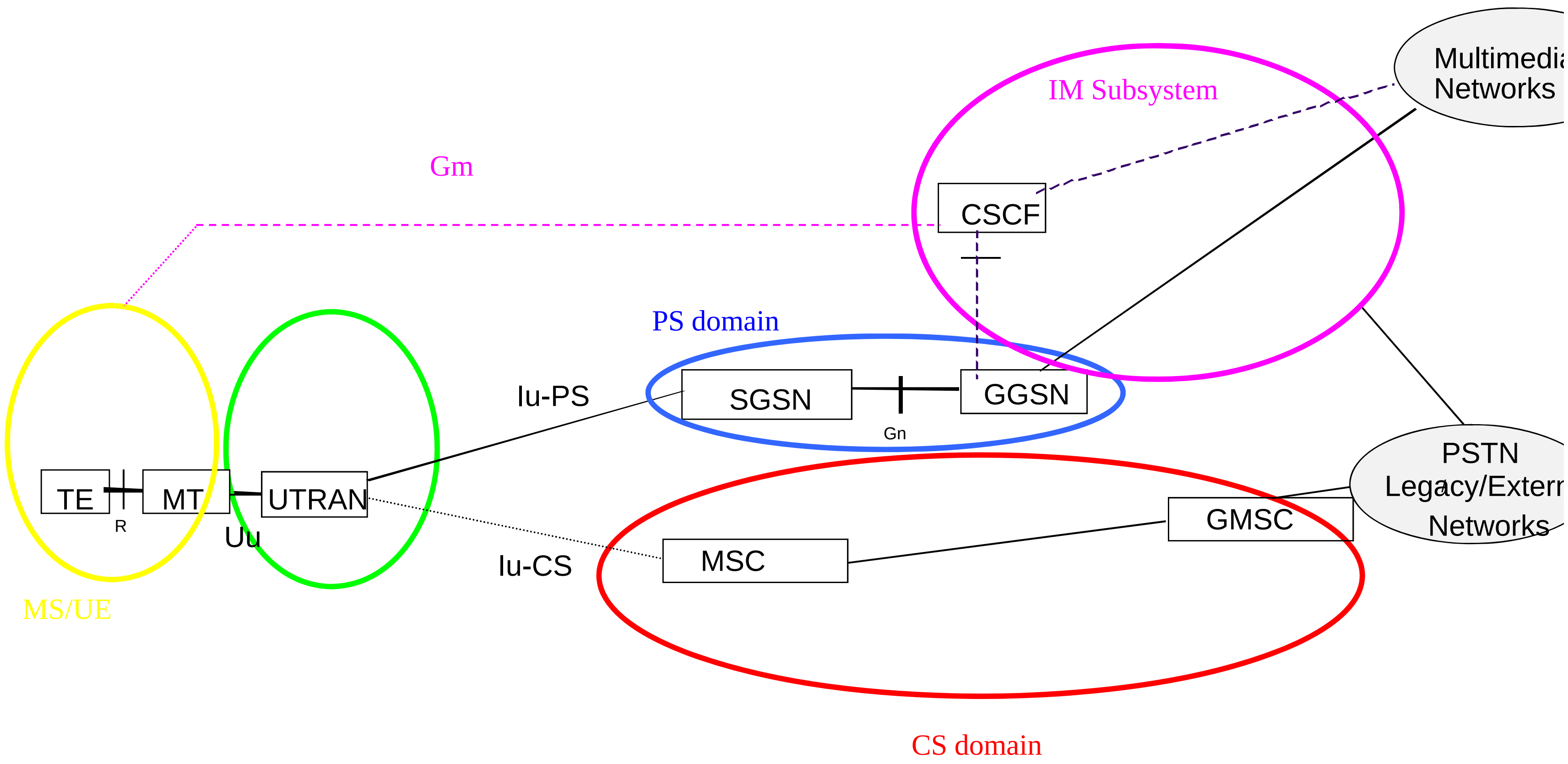
# Multimedia networks and services

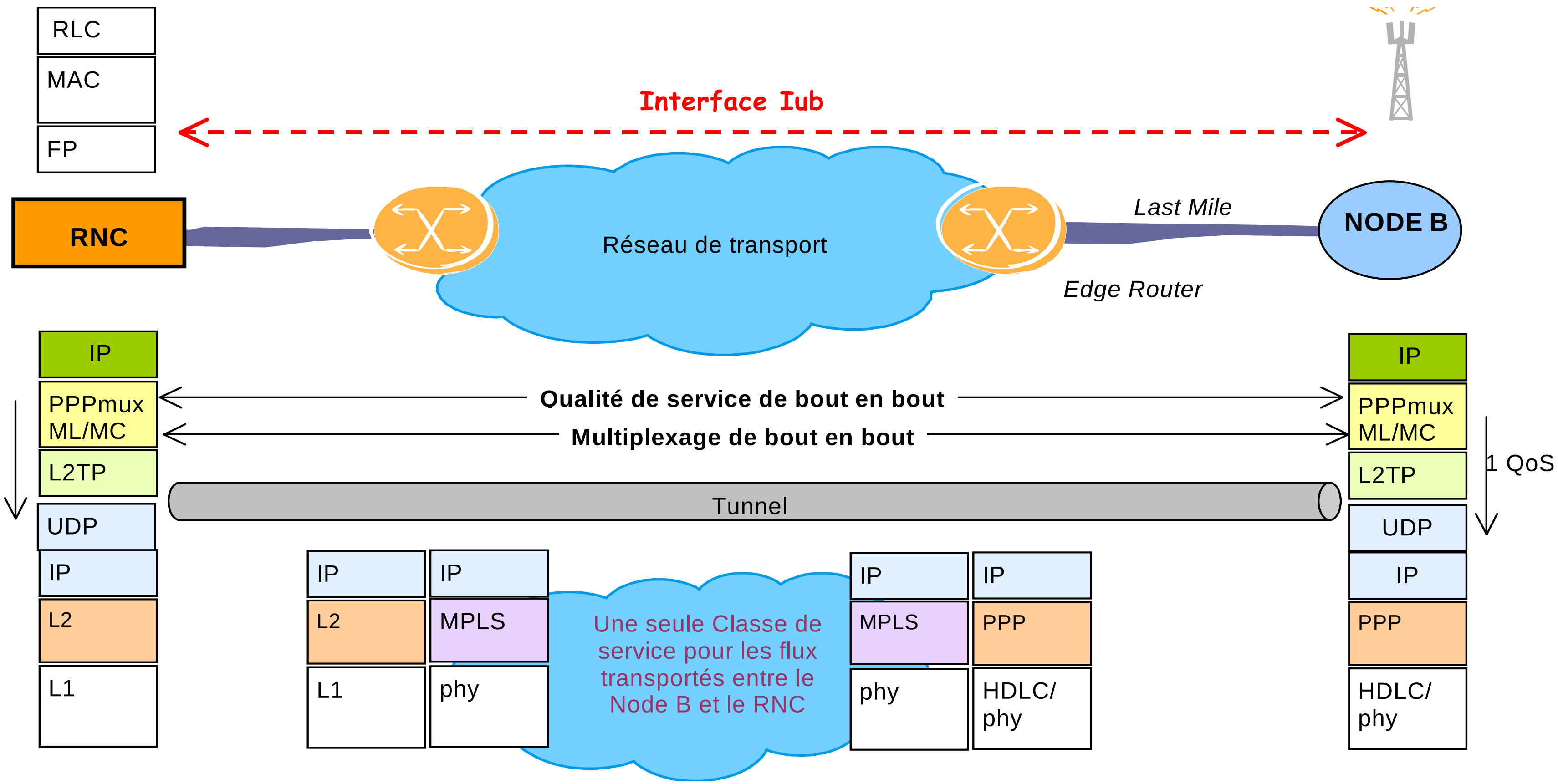
Samir Tohmé
PRISM Laboratory
Université de Versailles

# MM Networking Applications

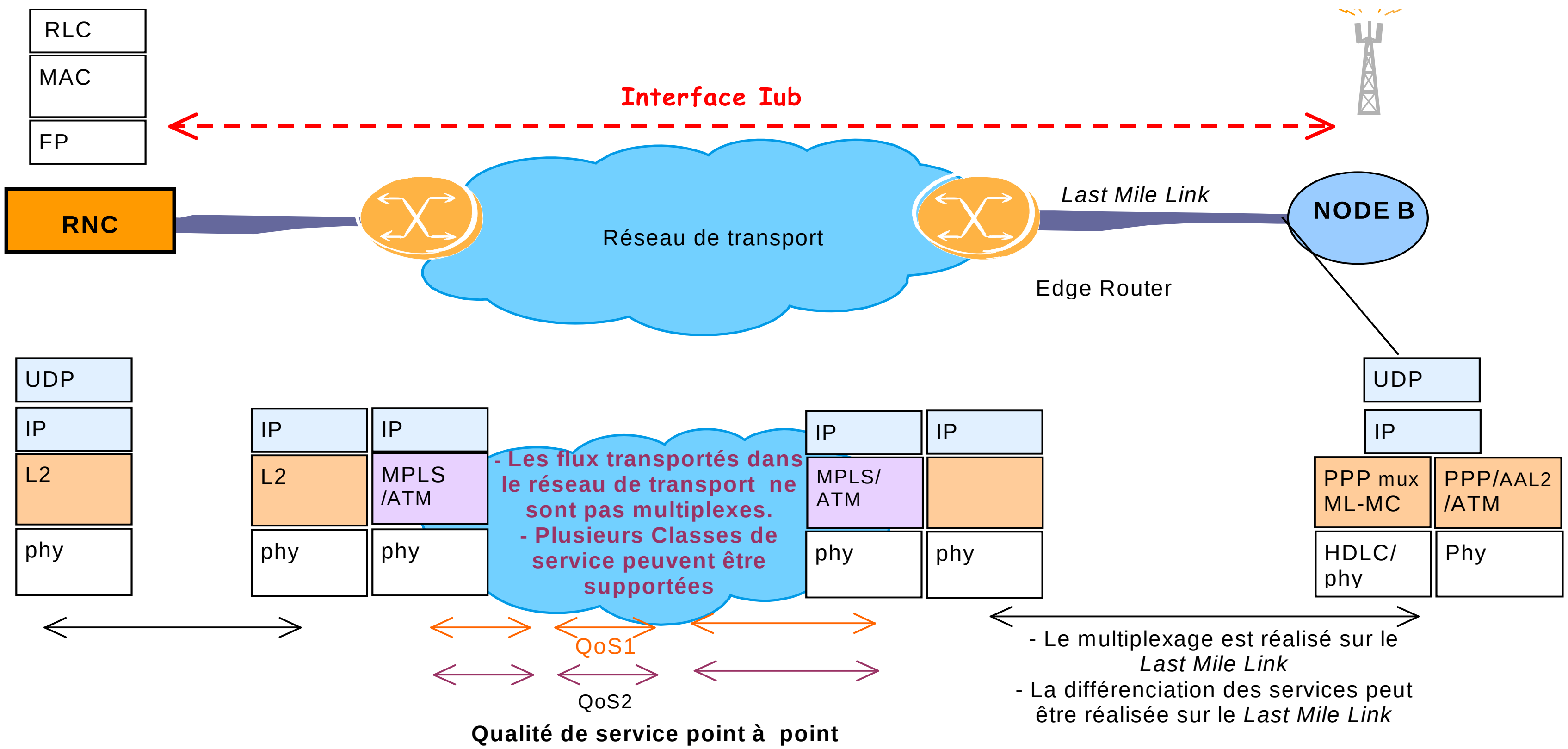
- Classes of MM applications
  - Streaming stored audio and video
  - Streaming live audio and video
  - Real-time interactive audio and video
- Fundamental characteristics:
  - Typically delay sensitive
    - end-to-end delay
    - delay jitter
  - But loss tolerant

# UMTS Rel 5 Architecture





RLC
MAC
FP
RNC
Interface Iub
Réseau de transport
Last Mile
NODE B
Edge Router
IP
PPPmux ML/MC
L2TP
UDP
IP
L2
L1
Qualité de service de bout en bout
Multiplexage de bout en bout
Tunnel
1 QoS
IP
L2
L1
IP
MPLS
phy
Une seule Classe de service pour les flux transportés entre le Node B et le RNC
IP
MPLS
phy
IP
PPP
HDLC/ phy
IP
PPPmux ML/MC
L2TP
UDP
IP
PPP
HDLC/ phy


RLC
MAC
FP
Interface Iub
RNC
Réseau de transport
Last Mile Link
NODE B
Edge Router
UDP
IP
L2
phy
IP
L2
phy
IP
MPLS
/ATM
phy
- Les flux transportés dans le réseau de transport ne sont pas multiplexes.
- Plusieurs Classes de service peuvent être supportées
IP
MPLS/
ATM
phy
IP
phy
UDP
IP
PPP mux
ML-MC
PPP/AAL2
/ATM
HDLC/
phy
Phy
QoS1
QoS2
Qualité de service point à point
- Le multiplexage est réalisé sur le Last Mile Link
- La différenciation des services peut être réalisée sur le Last Mile Link

# UMTS core network in Rel 5

- UMTS Rel 5 defines a new Core Network architecture divided into three domains :

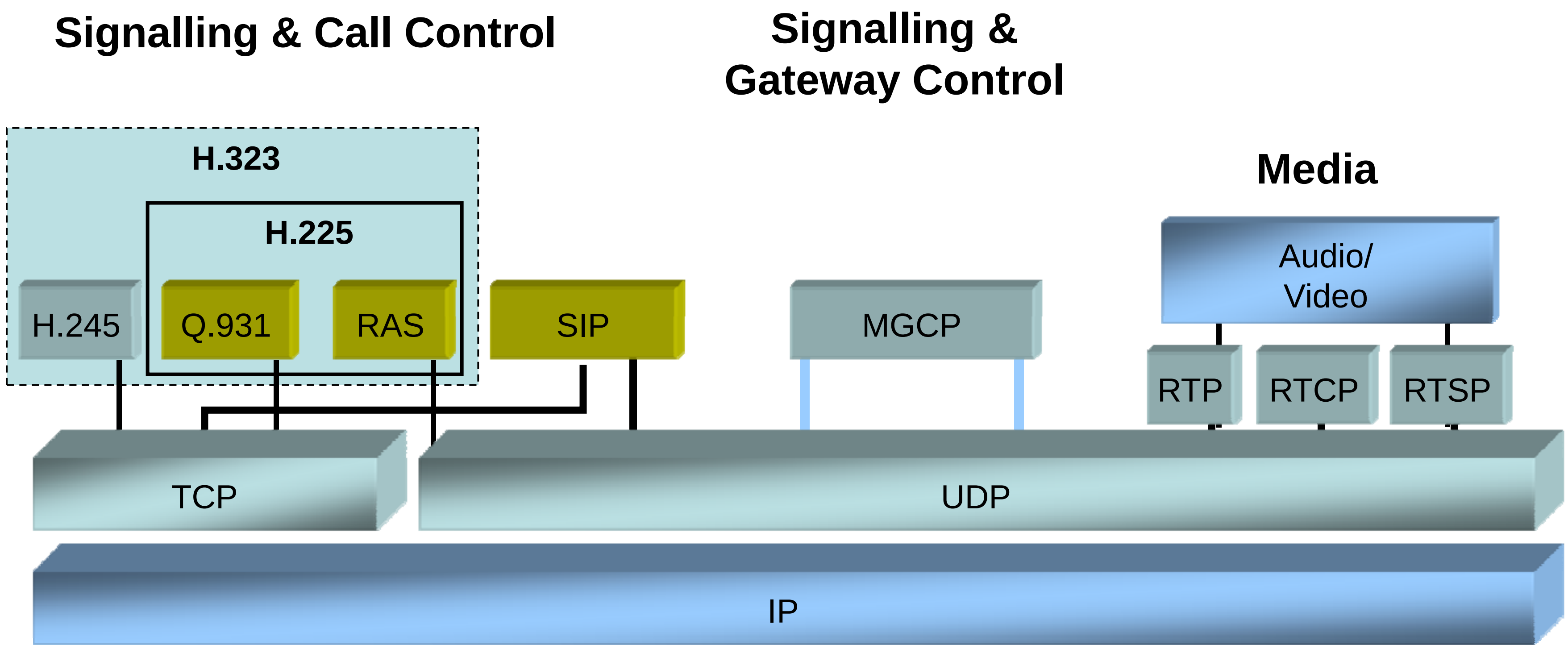
  PS domain : is the evolution of GPRS packet domain, it offers bearers for IP based applications.

  CS domain : is the evolution of GSM NSS, it offers bearers for circuit services.

  - From Rel 4, 3 GPP group has introduced to the CS domain the NGN (*Next Generation Network*) concept which allows to separate control part from the transport part

  IP Multimedia Subsystem : this domain is the innovative part of UMTS Rel5. It is expected to support IP multimedia applications between two end users. The control protocol used is SIP

# Control plane / User plane Protocols



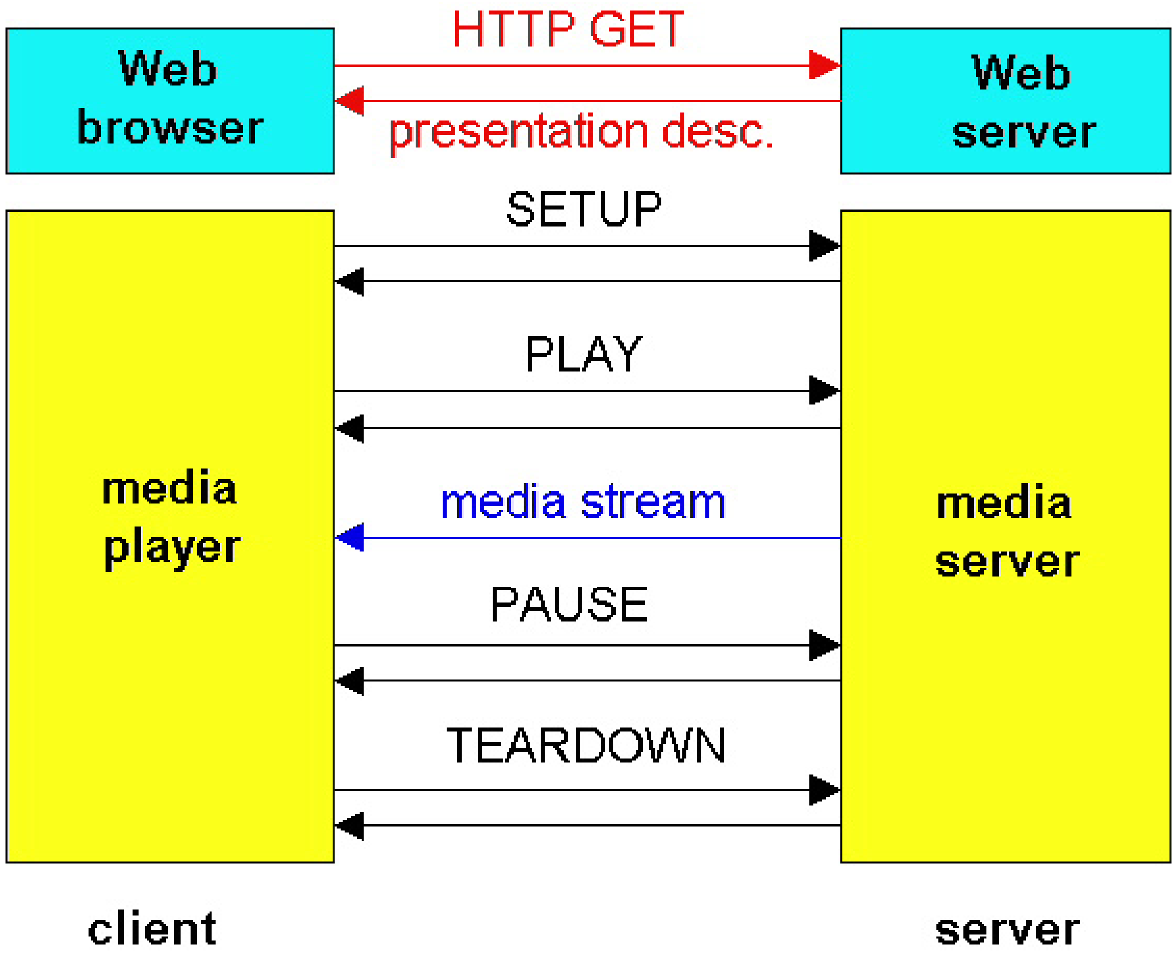
# User Control of Streaming Media: RTSP

- HTTP
  - Does not target multimedia content
  - No commands for fast forward, etc.
- RTSP: RFC 2326
  - Client-server application layer protocol.
  - For user to control display: rewind, fast forward, pause, resume, repositioning, etc…
- What it doesn't do:
  - does not define how audio/video is encapsulated for streaming over network
  - does not restrict how streamed media is transported; it can be transported over UDP or TCP
  - does not specify how the media player buffers audio/video

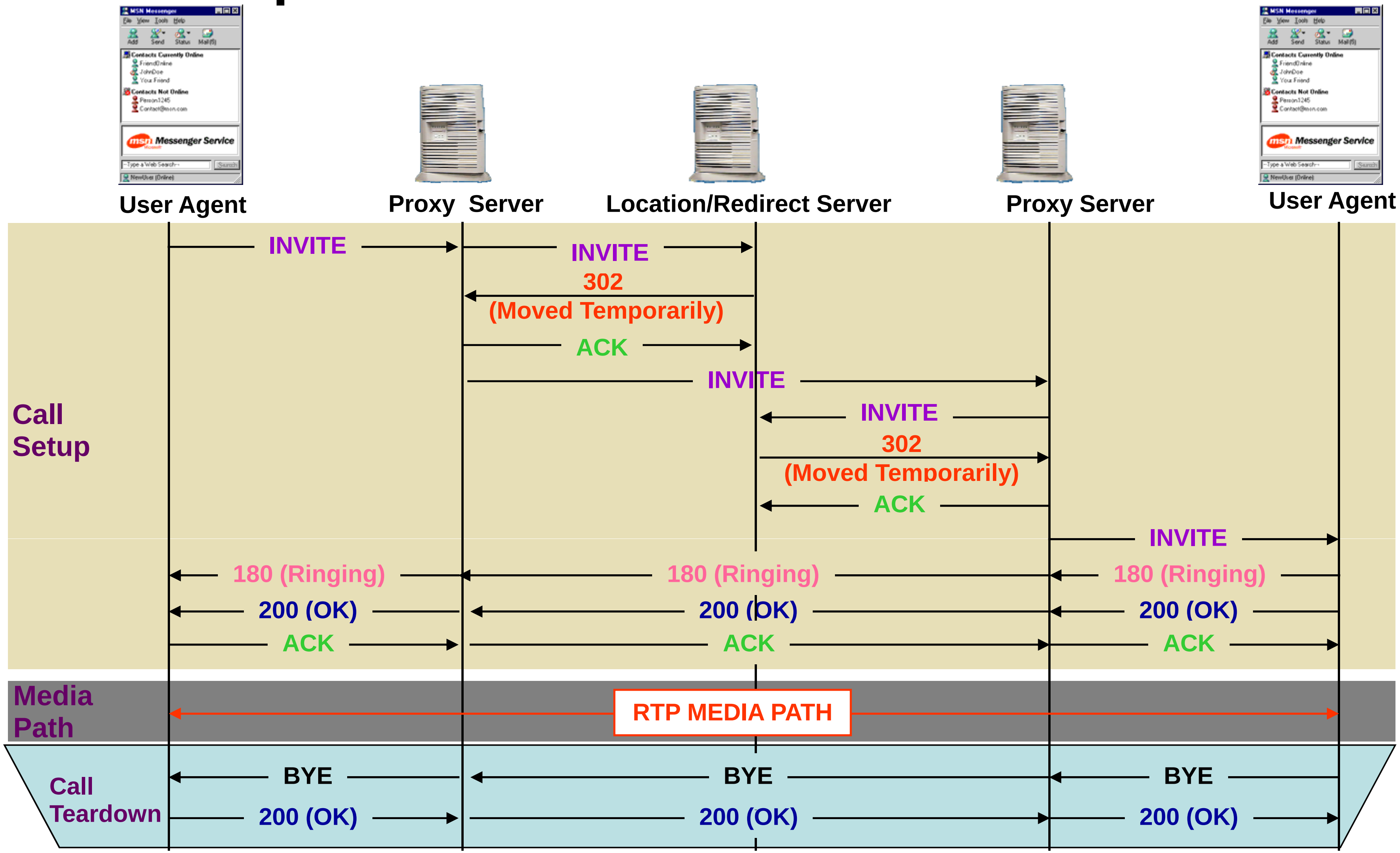
# RTSP: out of band control

- FTP uses an “out-of-band” control channel:
  - A file is transferred over one TCP connection.
  - Control information (directory changes, file deletion, file renaming, etc.) is sent over a separate TCP connection.
  - The “out-of-band” and “in-band” channels use different port numbers.
- RTSP messages are also sent out-of-band:
  - RTSP control messages use different port numbers than the media stream: out-of-band.
    - Port 554
  - The media stream is considered “in-band”.

# RTSP Operation

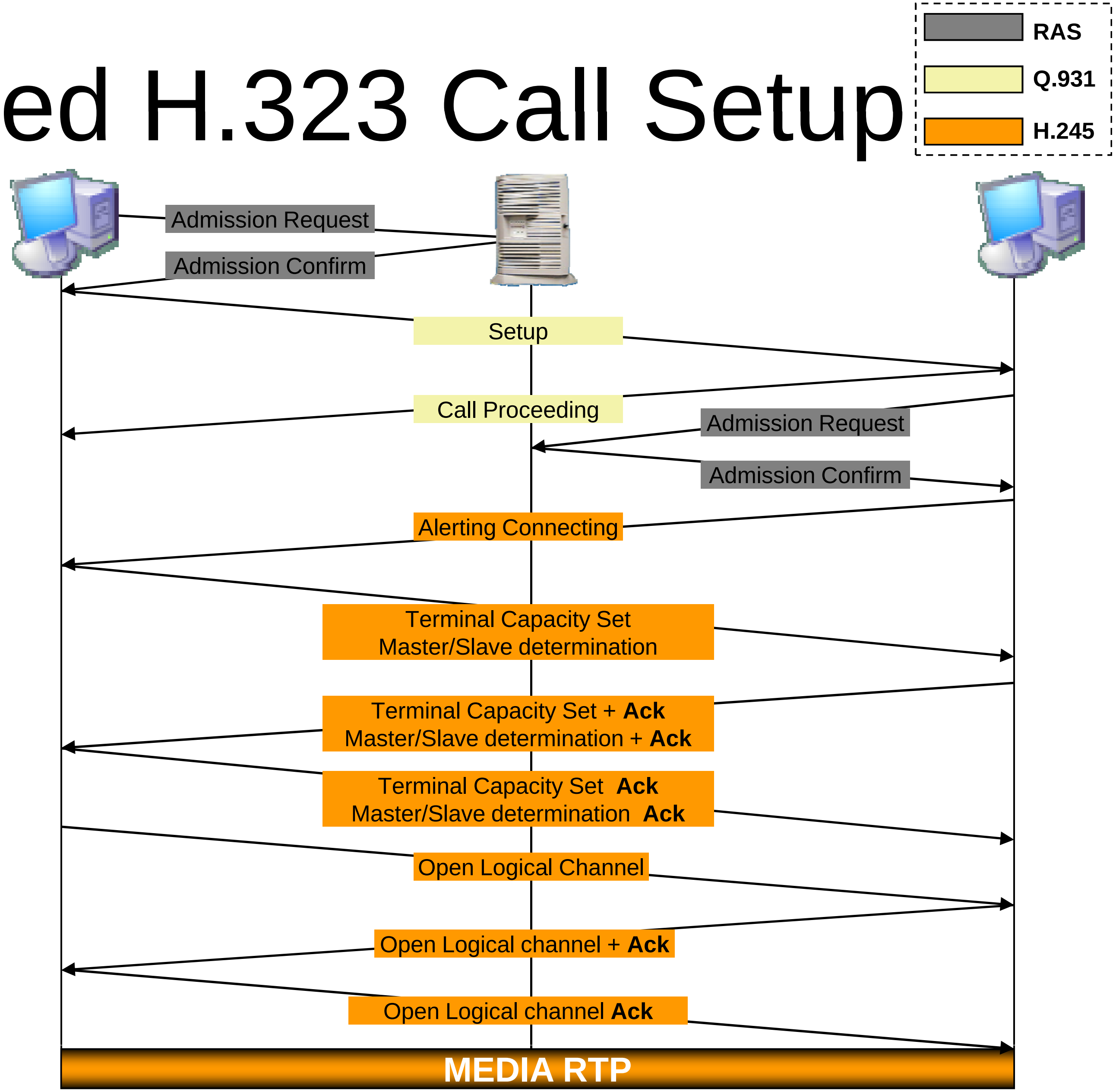


# Simplified Call Model with SIP



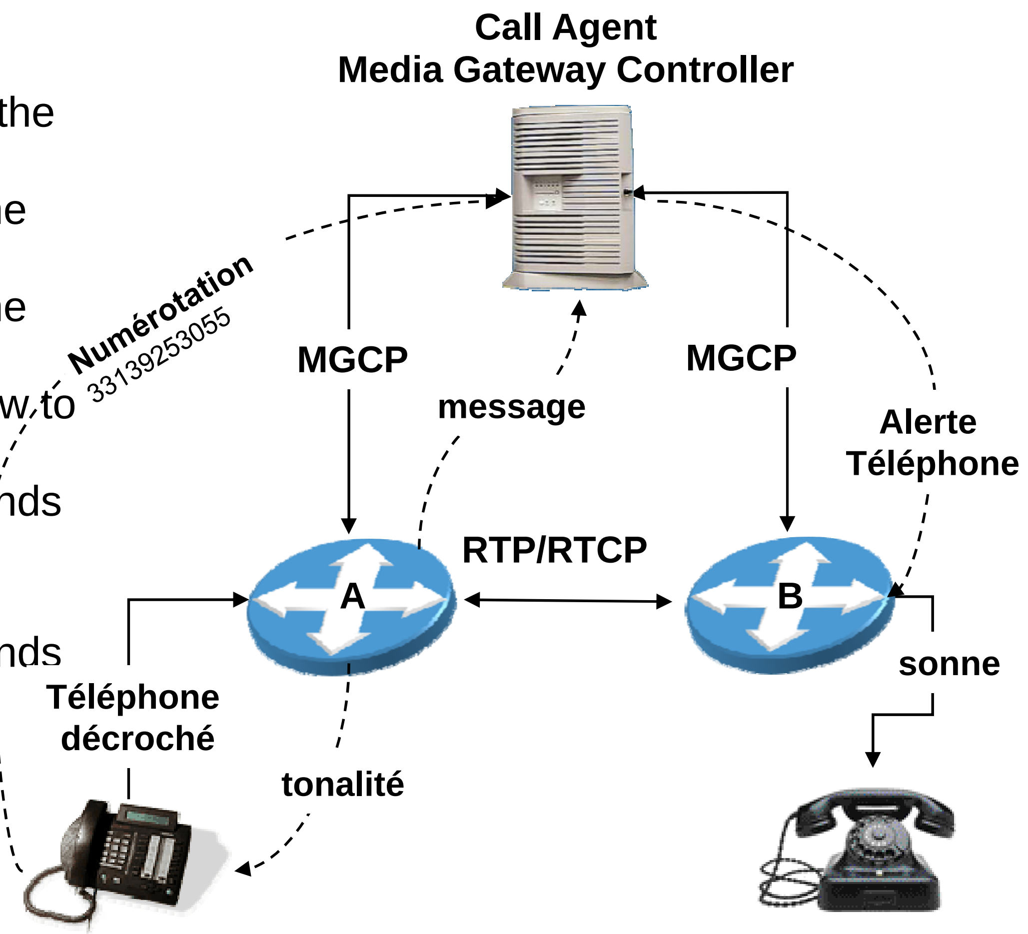
# Simplified H.323 Call Setup

- Both endpoints have previously registered with the gatekeeper.
- Terminal A initiate the call to the gatekeeper. (RAS messages are exchanged).
- The gatekeeper provides information for Terminal A to contact Terminal B.
- Terminal A sends a SETUP message to Terminal B.
- Terminal B responds with a Call Proceeding message and also contacts the gatekeeper for permission.
- Terminal B sends a Alerting and Connect message.
- Terminal B and A exchange H.245 messages to determine master slave, terminal capabilities, and open logical channels.
- The two terminals establish RTP media paths.



# Simplified Call Flow

- When Phone A goes offhook Gateway A sends a signal to the call agent.
- Gateway A generates dial tone and collects the dialed digits.
- The digits are forwarded to the call agent.
- The call agent determines how to route the call.
- The call agent sends commands to Gateway B.
- Gateway B rings phone B.
- The call agent sends commands to both gateways to establish RTP/RTCP sessions.



# MGCP, SIP and H.323

- divides call setup/control and media establishment functions.
- does not replace SIP or H.323.
- interoperates with H.323 and SIP.

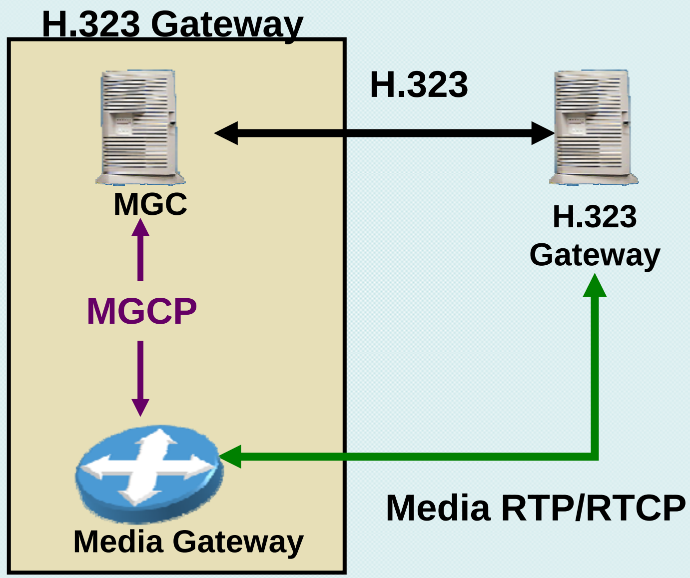
  - *A call agent accepts SIP or H.323 call setup requests.*
  - *The call agent uses MGCP to control the media gateway.*
  - *The media gateway establishes media sessions with other H.323 or SIP endpoints.*

**In this example, an H.323 gateway is "decomposed" into:**

**–A call agent that provides signaling.**

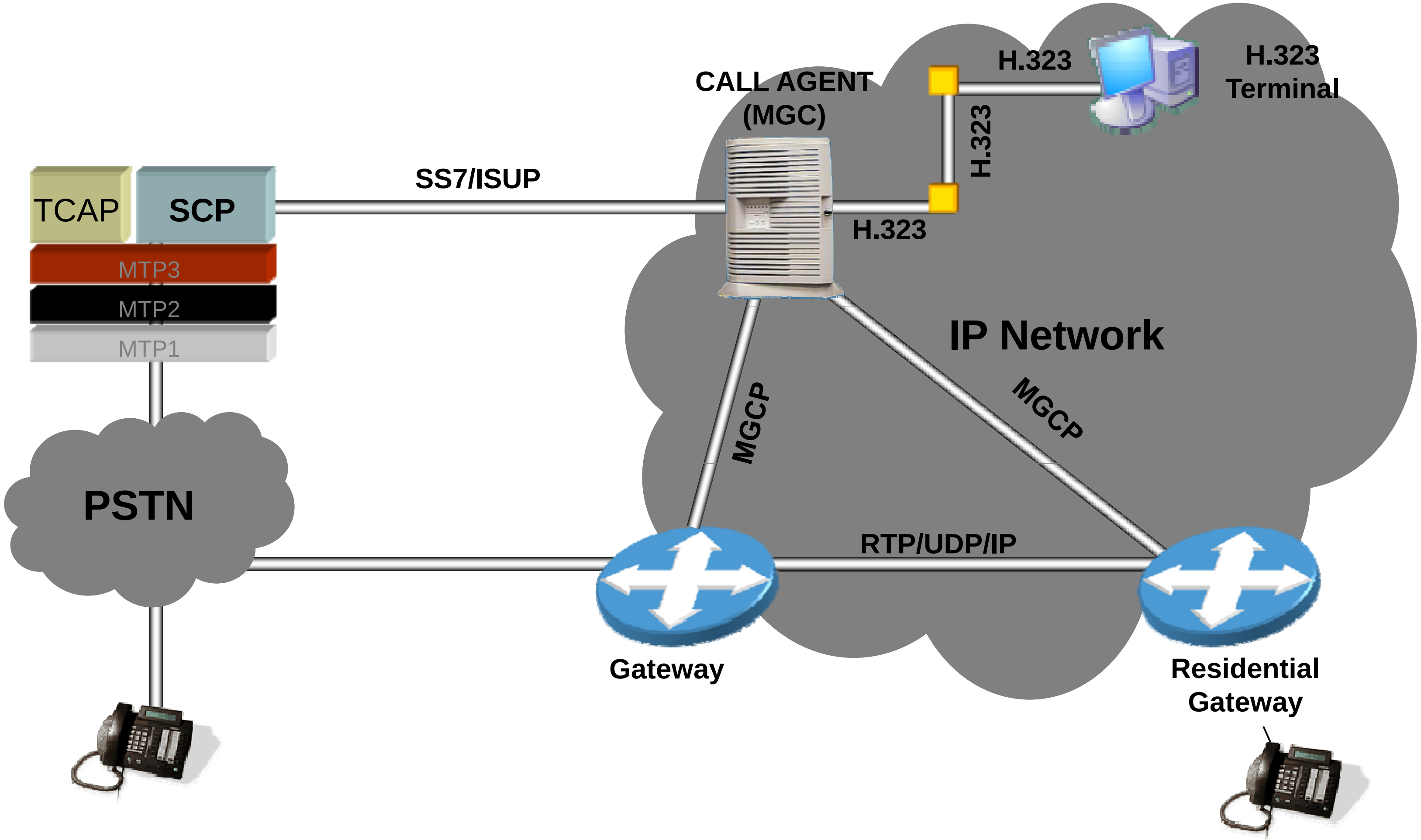
**–A gateway that handles media.**

**MGCP protocol is used to control the gateway.**

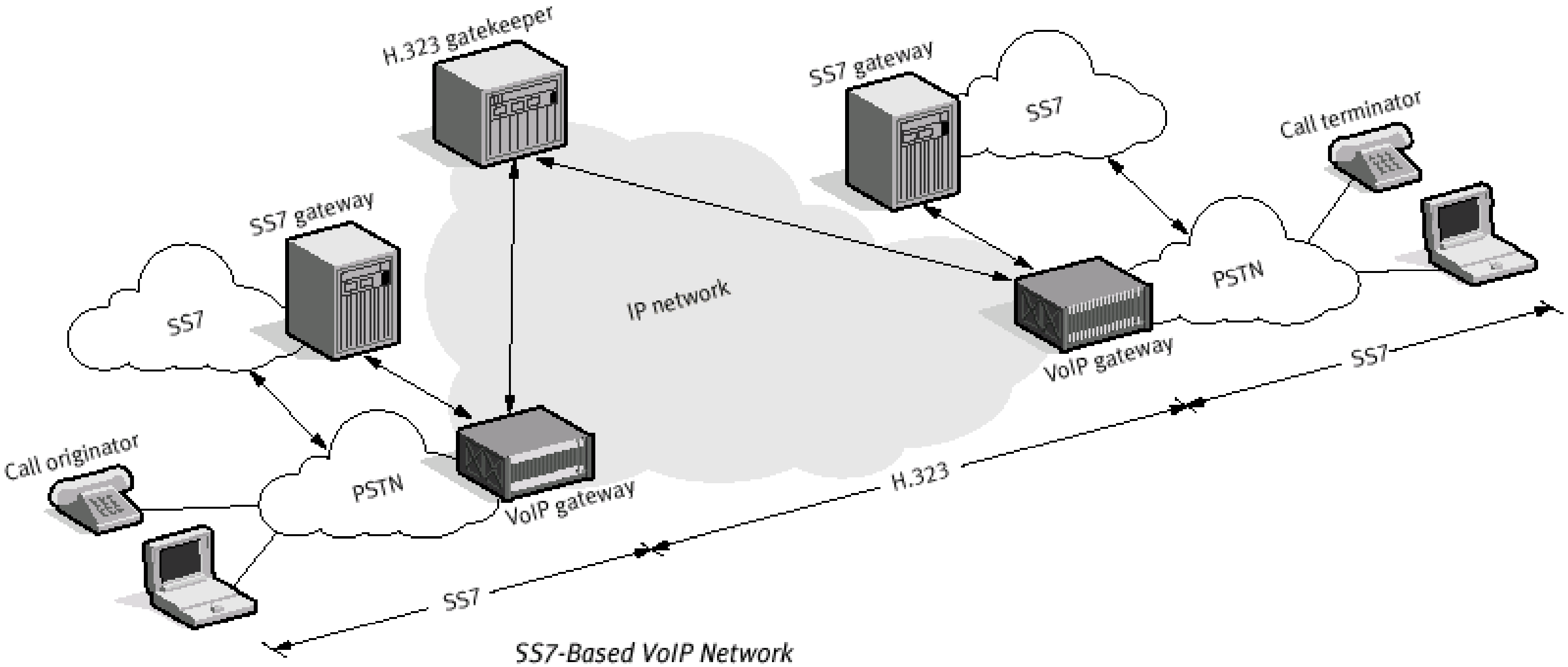


# MEGACO/H.248

## MGCP – System Architecture



# VoIP Network




*SS7-Based VoIP Network*

# VoIP: more on Protocols and Packets


Carry the speech frame inside an RTP packet

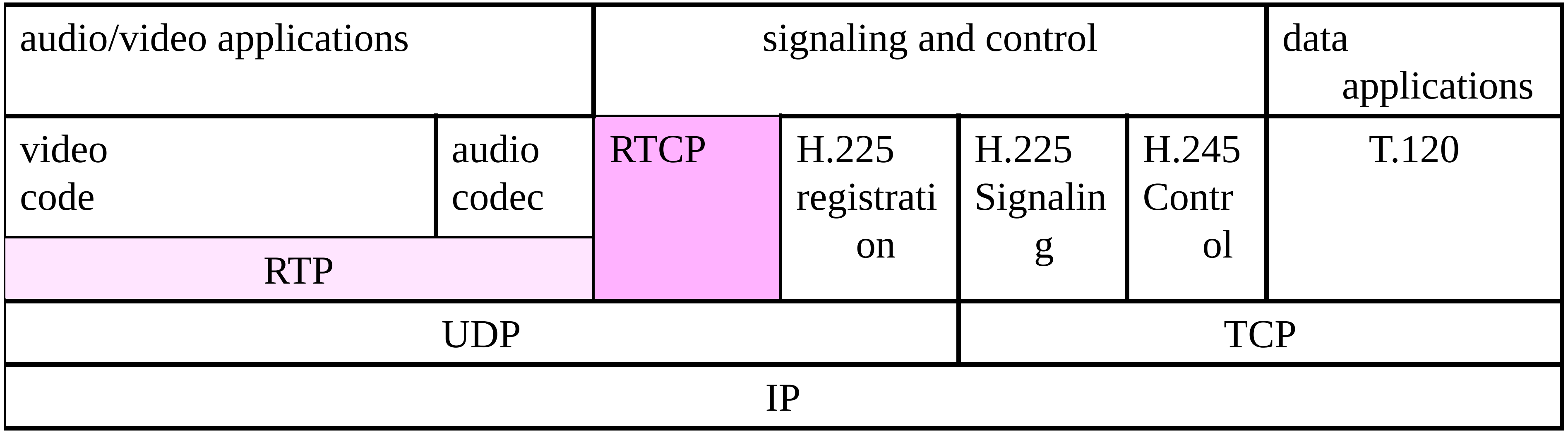
Typical packetization time of 10-20ms per audio frame.

This should be compared to the durations relevant to speech phenomena:

- "10 ms: smallest difference detectable by auditory system (localization),
- 3 ms: shortest phoneme (plosive burst),
- 10 ms: glottal pulse period,
- 100 ms: average phoneme duration,
- 4 s: exhale period during speech." (from slide titled 'What is a "short" window of time?'

# VoIP: Details

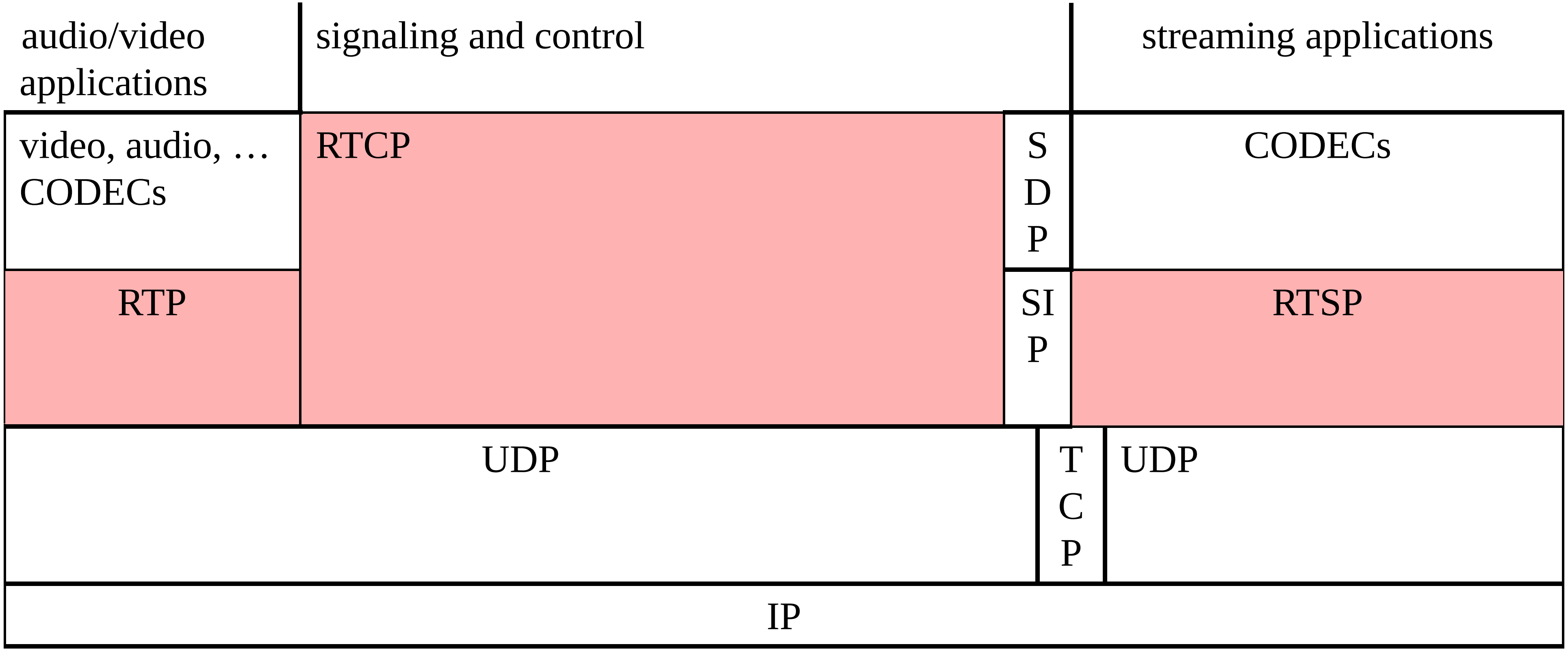
## RTP and H.323 for IP Telephony




| | |
|---|---|
| H.323 | framework of a group protocols for IP telephony (from ITU) |
| H.225 | Signaling used to establish a call |
| H.245 | Control and feedback during the call |
| T.120 | Exchange of data associated with a call |
| RTP | Real-time data transfer |
| RTCP | Real-time Control Protocol |

# VoIP cont.

## RTP, RTCP, and RTSP



# VoIP cont.

- Real-Time Delivery
  - In a real-time application : data must be delivered with the same time relationship as it was created (but with some delay)
  - Two aspects of real-time delivery (for protocols):
    - Order : data should be played in the same order as it was created
    - Time : the receiver must know when to play the packets, in order to reproduce the same signal as was input
  - We keep these separate by using a sequence number for order and a time stamp for timing.
  - Consider an application which transmits audio by sending datagrams every 20ms, but does silence detection and avoids sending packets of only silence. Thus the receiver may see that the time stamp advances by more than the usual 20ms, but the sequence number will be the expected next sequence number. Therefore we can tell the difference between missing packets and silence.

# RTP: Real-Time Transport Protocol

- First defined by RFC 1889, now defined by RFC 3550
  - Designed to carry a variety of real-time data: audio and video.
  - Provides two key facilities:
    - Sequence number for order of delivery (initial value chosen randomly)
    - Timestamp (of first sample) - used for control of playback
  - Provides no mechanisms to ensure timely delivery

# Timestamps

- The initial timestamp is to be chosen randomly (just as the initial sequence number is selected randomly):
  - to avoid replays
  - to increase security (this assumes that the intruder does not have access to all the packets flowing to the destination)
- The timestamp granularity (i.e., the units) are determined by the payload type {often based on the sampling rate}

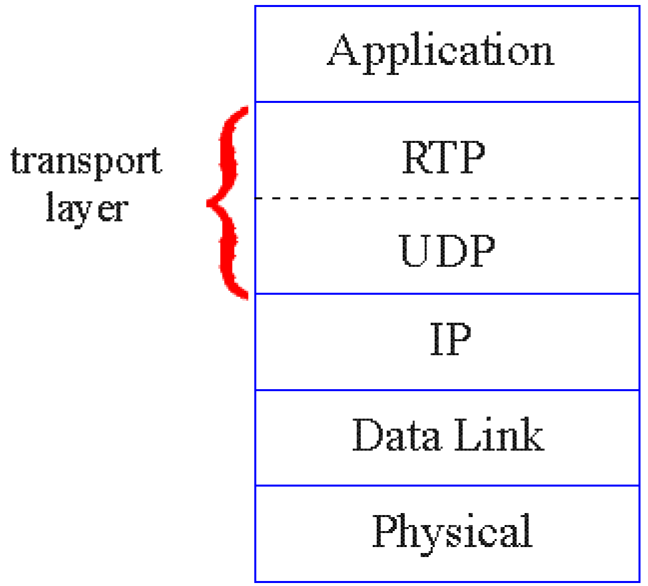
# RTP (cont.)

- RTP specifies a packet structure for packets carrying audio and video data
- RTP packet provides : payload type identification, packet sequence numbering, timestamping
- RTP runs in the end systems.
- RTP packets are encapsulated in UDP segments
- Interoperability: If two Internet phone applications run RTP, then they may be able to work together

# RTP runs on top of UDP

RTP libraries provide a transport-layer interface that extend UDP:

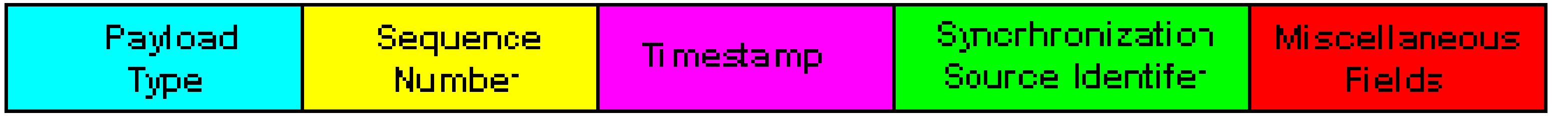
- port numbers, IP addresses
- payload type identification
- packet sequence numbering
- time-stamping



# RTP and QoS

- RTP does not provide any mechanism to ensure timely delivery of data or provide other quality of service guarantees.
- RTP encapsulation is only seen at the end systems: it is not seen by intermediate routers.
- Routers providing best-effort service do not make any special effort to ensure that RTP packets arrive at the destination in a timely matter.

# RTP Header




**RTP Header**

**Payload Type (7 bits):** Indicates type of encoding currently being used. If sender changes encoding in middle of conference, sender informs the receiver through this payload type field.

Payload type 0: PCM mu-law, 64 kbps
Payload type 3, GSM, 13 kbps
Payload type 7, LPC, 2.4 kbps
Payload type 26, Motion JPEG
Payload type 31. H.261
Payload type 33, MPEG2 video

**Sequence Number (16 bits):** Increments by one for each RTP packet sent, and may be used to detect packet loss and to restore packet sequence.

# RTP Header

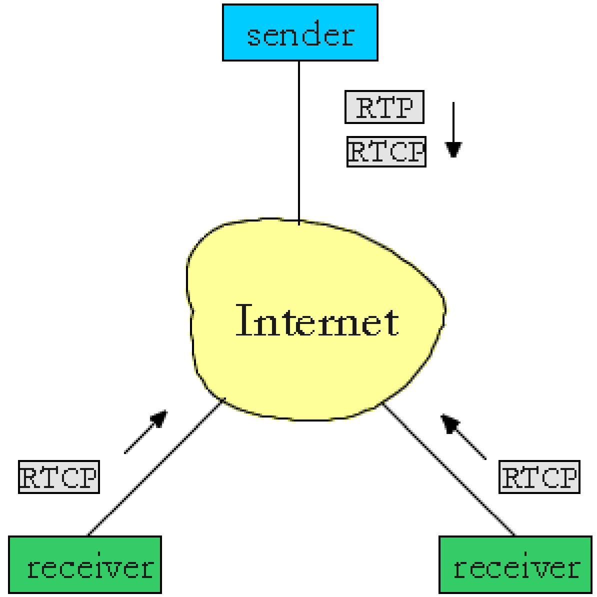
- **Timestamp field (32 bytes long).** Reflects the sampling instant of the first byte in the RTP data packet.
  - For audio, timestamp clock typically increments by one for each sampling period (for example, each 125 usecs for a 8 KHz sampling clock)
  - if application generates chunks of 160 encoded samples, then timestamp increases by 160 for each RTP packet when source is active. Timestamp clock continues to increase at constant rate when source is inactive.

- **SSRC field (32 bits long).** Identifies the source of the RTP stream. Each stream in a RTP session should have a distinct SSRC.

# Real-Time Control Protocol (RTCP)

- Works in conjunction with RTP.
- Each participant in RTP session periodically transmits RTCP control packets to all other participants.
- Each RTCP packet contains sender and/or receiver reports
  - report statistics useful to application
- Statistics include number of packets sent, number of packets lost, interarrival jitter, etc.
- Feedback can be used to control performance
- Sender may modify its transmissions based on feedback

# RTCP - Continued




- For an RTP session there is typically a single multicast address; all RTP and RTCP packets belonging to the session use the multicast address.

- RTP and RTCP packets are distinguished from each other through the use of distinct port numbers.

- **To limit traffic, each participant reduces his RTCP traffic as the number of conference participants increases.**

# RTCP Packets

**Receiver report packets:**

- fraction of packets lost, last sequence number, average interarrival jitter.

**Sender report packets:**

- SSRC of the RTP stream, the current time, the number of packets sent, and the number of bytes sent.

**Source description packets:**

- e-mail address of sender, sender's name, SSRC of associated RTP stream.
- Provide mapping between the SSRC and the user/host name.

# Synchronization of Streams

- RTCP can synchronize different media streams within a RTP session.
- Consider videoconferencing app for which each sender generates one RTP stream for video and one for audio.
- Timestamps in RTP packets tied to the video and audio sampling clocks
  - not tied to the wall-clock time
- Each RTCP sender-report packet contains (for the most recently generated packet in the associated RTP stream):
  - timestamp of the RTP packet
  - wall-clock time for when packet was created.
- Receivers can use this association to synchronize the playout of audio and video.

# VoIP Issues

- **Voice quality**
  - ✓ Can delay be acceptable reduced?
  - ✓ Can VoIP be as good or better than PSTN?
  - ✓ How good is good enough?
- **Reliability**
  - ✓ How long until VoIP can provide lifeline service?
  - ✓ Will companies trust VoIP as their only long distance?
- **Inter-working with the public Switched Telephone Network**
  - ✓ SS7/IN capabilities
  - ✓ Number portability
- **Standards**
  - ✓ Call control
  - ✓ Service access

# VoIP Issues (cont.)

- **Interoperability**
  - ✓ Vendor gateway to vendor gateway
  - ✓ Vendor gateway to PSTN for advanced features

- **Incumbent carrier migration**
  - ✓ Circuit switched net migration to IP
  - ✓ Does VoIP make sense for POTS subscribers?

- **Numbering plans**
  - ✓ E.164 vs. IPv6 vs. ?

- **Regulations**
  - ✓ What happens to industry growth if arbitrage opportunity disappears?

# Some Conclusions concerning the VoIP

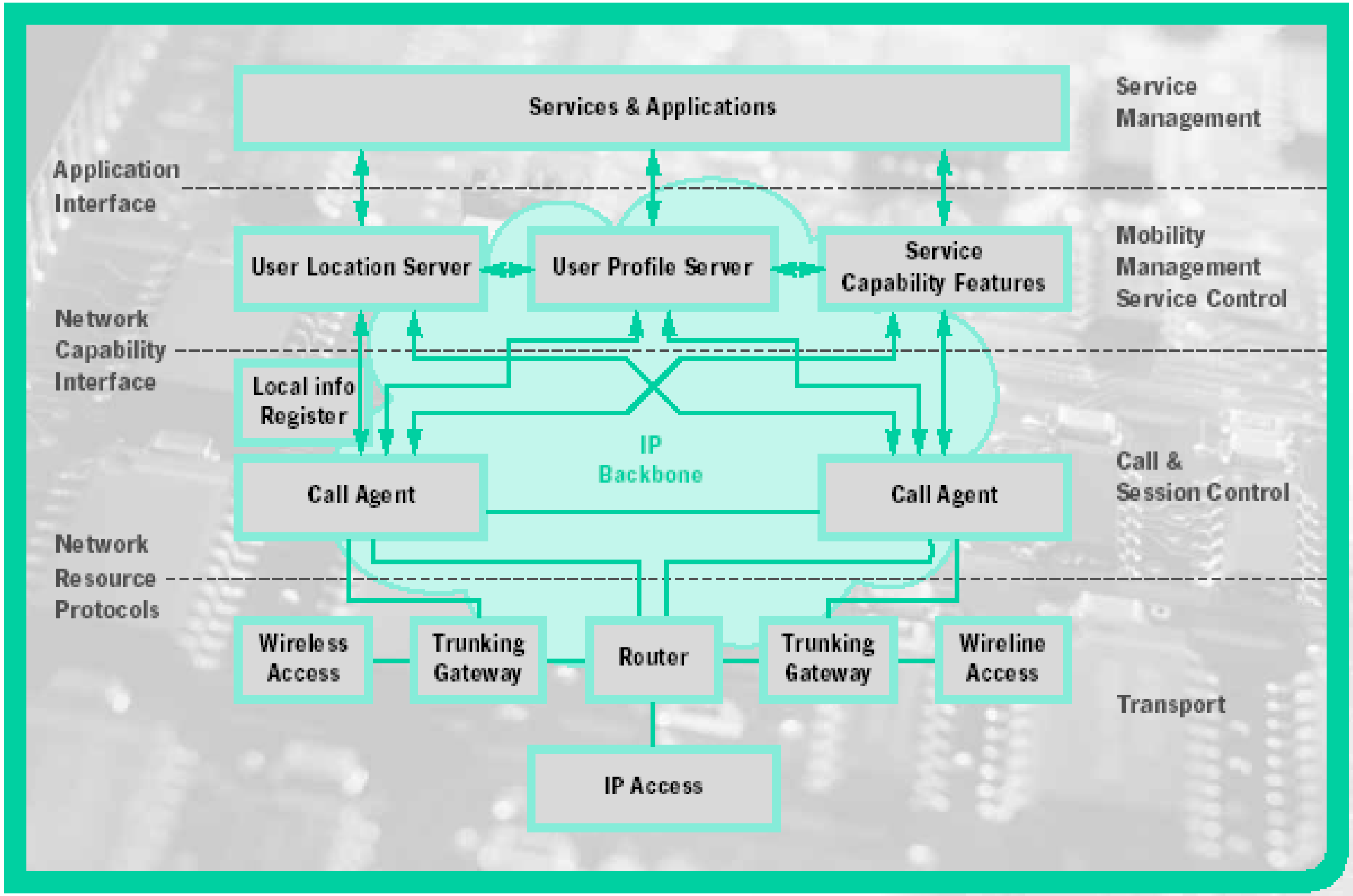
Today, the focus of Voice over IP is cheap long distance

Longer term, the focus is on a more efficient infrastructure and the ability to create new applications

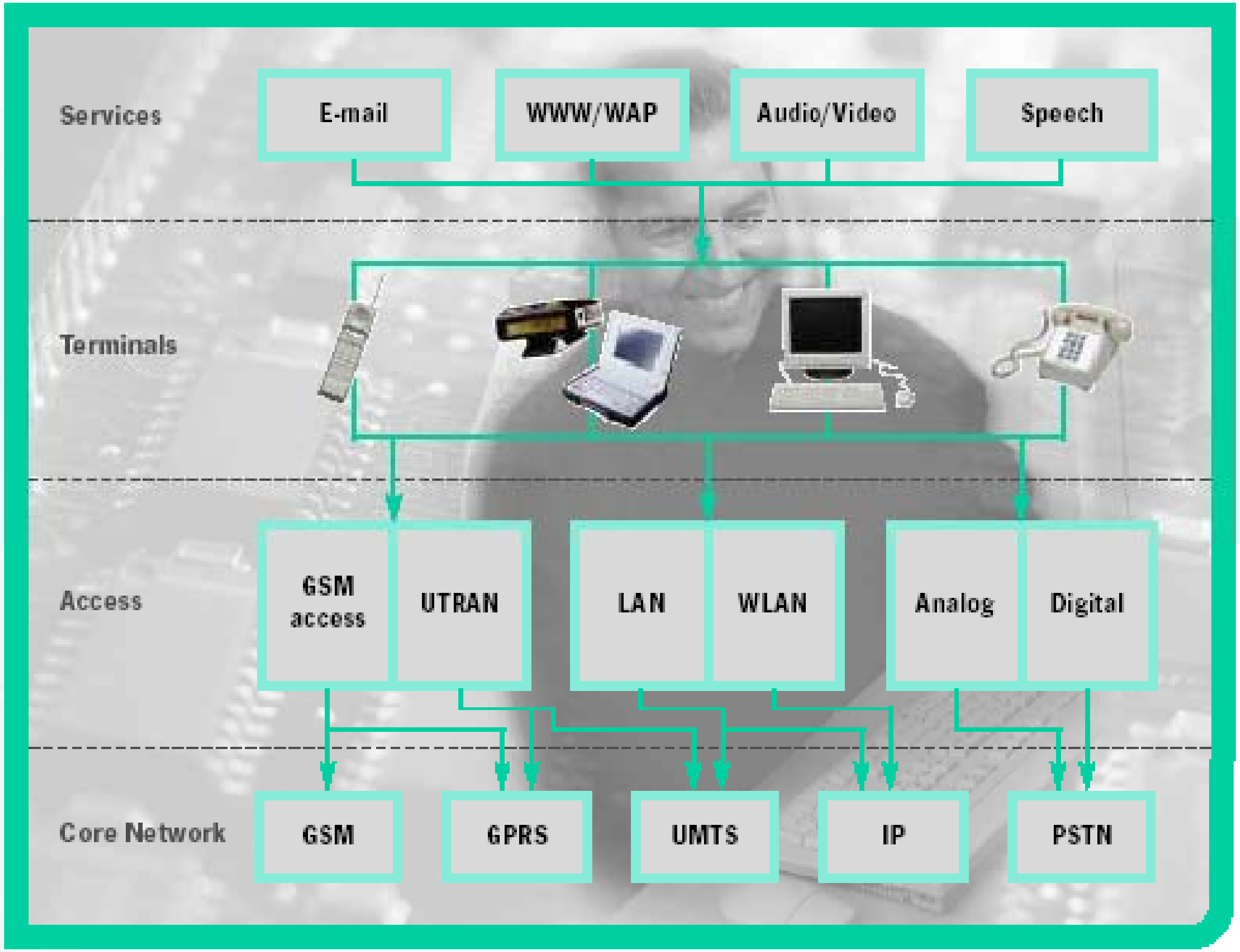
There is much work to be done on quality, reliability, standards, service creation, and integration with the Public Switched Telephone Network

# Trends

- Trend 1 ( All IP UMTS network)
  - Packet-switched – circuit switched
  - Multimedia support in core network
- Trend 2 (Open Service Architecture,OSA)
  - Provide third party service provider access to their UMTS service architecture
  - The concept of service portability was called VHE in 3GPP standardization
  - VHE philosophy:
    - Make it possible for third party service providers to develop UMTS applications




Generic Model for All-IP core network architecture



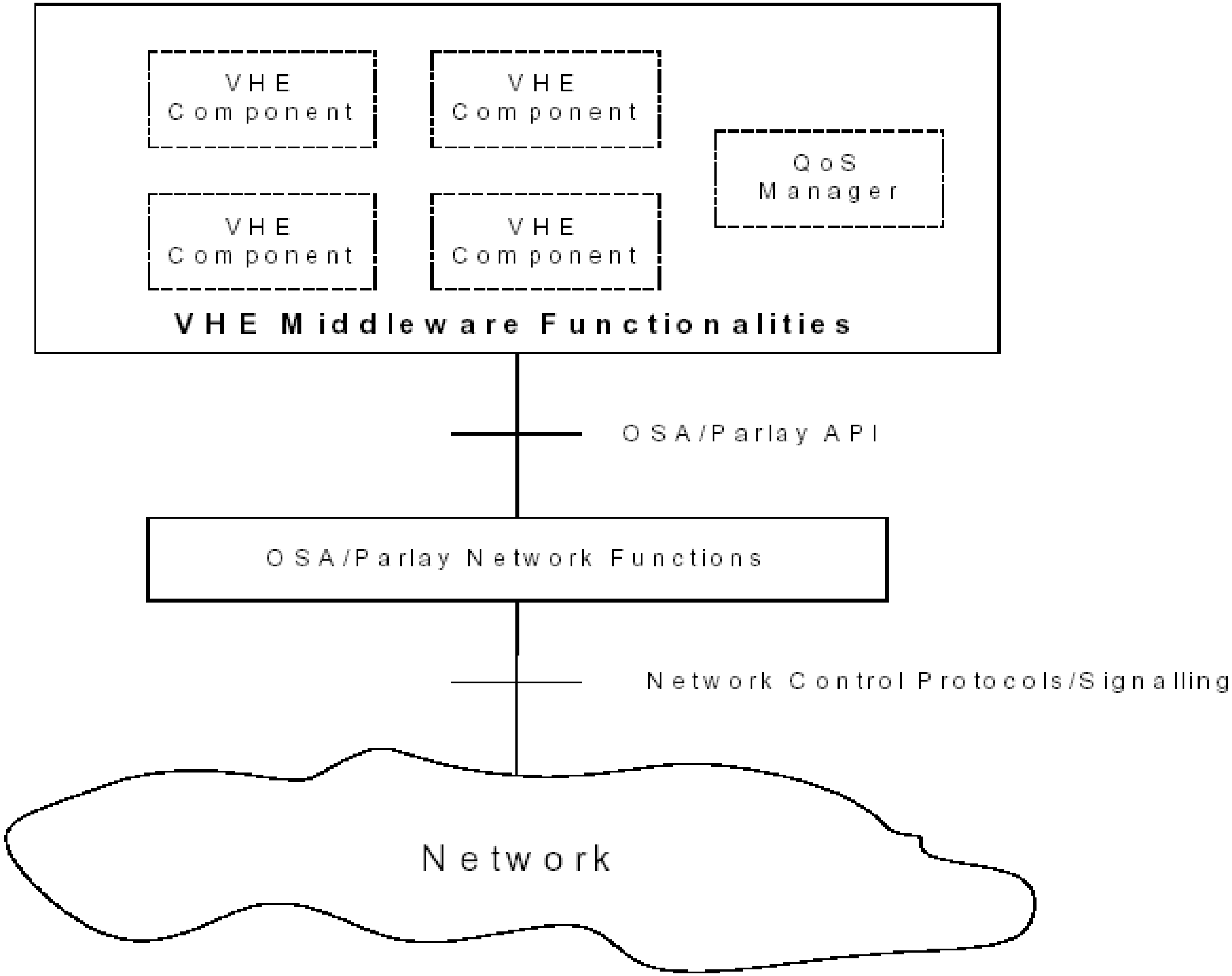

Uniform service presentation over different networks and terminals

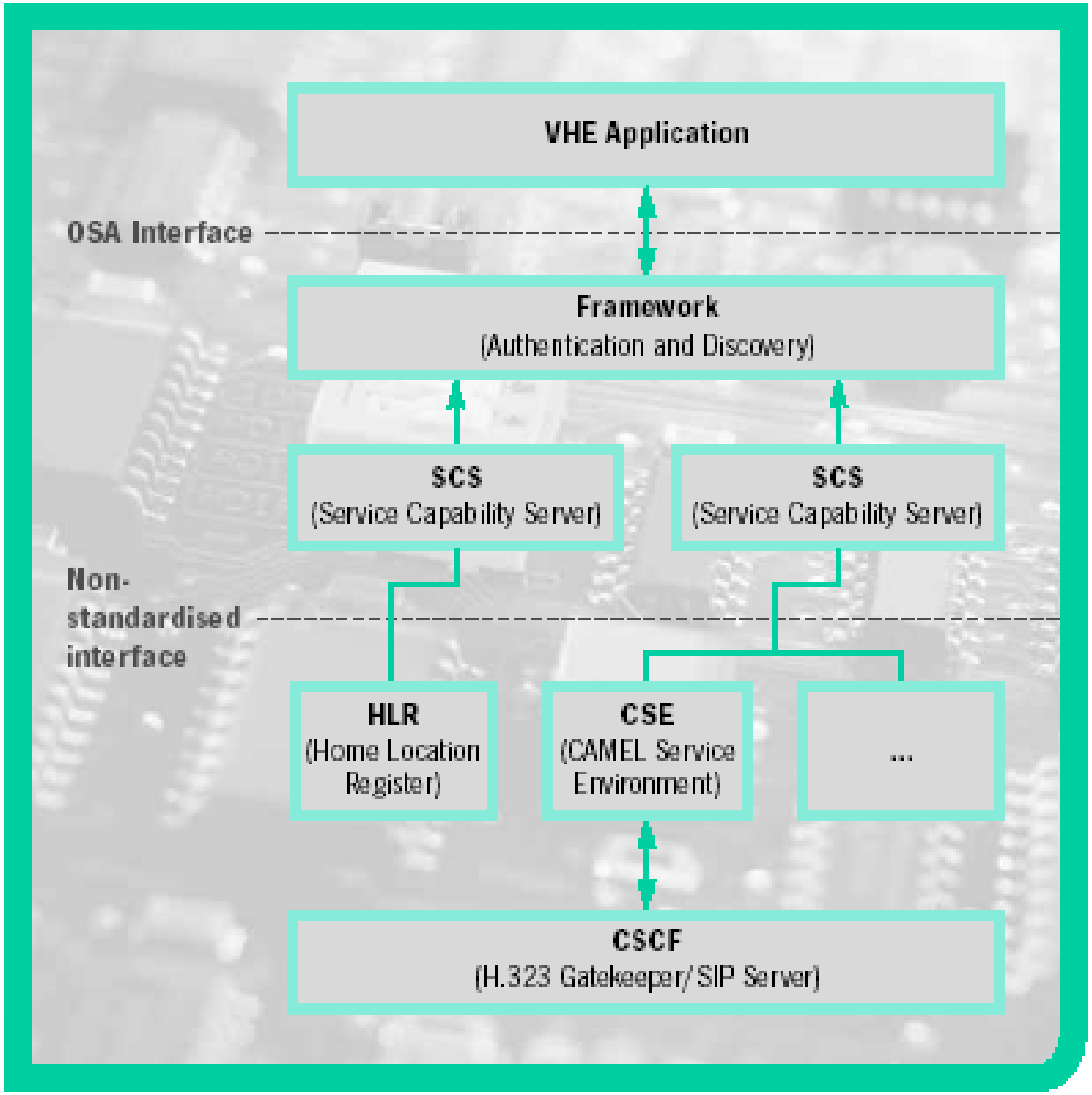
UTRAN: UMTS Radio Access Network
WLAN: Wireless LAN
WAP: Wireless Application Protocol
PSTN: Public Switched Telephone Network
GPRS: General Packet Radio Service

| | Definitions |
|---|---|
| **3GPP** | "The VHE ensures that users are consistently presented with the same personalised features, User Interface customisation and services in whatever network and whatever terminal (within the capabilities of the terminal and network), where ever the user may be located" [5] |
| **GSM MoU** | "Virtual Home Environment (VHE) is a system concept for service portability in the Third Generation across network borders" [6] |
| **ITU / IMT2000** | "VHE is a capability whereby a User is offered the same service experience in a visited network as in his Home system." [7] |
| **UMTS Forum** | "VHE means that the user will have the same interface and service environment regardless of location (personalised user interface independent from the current serving network)." [8] |
| **Eurescom P920-GI** | "The Virtual Home Environment is an environment which presents the user with a common look and feel interface and service experience regardless of location, network and terminal type. The VHE is based on standardised service capabilities and personalised features that are consistently presented so that the user always "feels" that he is on his home network even when roaming across network boundaries" [9] |
| **IST VESPER** | "VHE main feature is that the customised environment will be following the user while he/she is roaming within different networks and using different terminals" [10] |

# VHE(service capabilities)

- Three fundamental architecture improvements from GSM
  - Wideband access: high bit rate
  - Mobile-fixed-Internet convergence
    - Cross domain service
    - e.g. Tracking a user’s location
    - Automatically adapting the content of his incoming messages to SMS, voice message, fax or e-mail.
    - VHE is the enabler of this service portability across networks and terminals in the different domains.
  - Flexible service architecture


VHE
Component
VHE
Component
VHE
Component
VHE
Component
QoS
Manager
VHE Middleware Functionalities
OSA/Parlay API
OSA/Parlay Network Functions
Network Control Protocols/Signalling
Network




VHE application invocation based on OSA

# VHE(service capabilities)

- VHE
  - A system concept for personalized service portability across network boundaries and between terminals. “virtual at home”
  - Allows a user to personalize the set of services across different types of networks- mobile, PSTN, Internet and terminals-mobile, laptop, fixed phone, PDA,PC
  - e.g. “ from 9h00 to 17h00 I want to receive incoming messages from my office”
  - Layered architecture, see figure.1
    - Network layer
    - Service layer
    - Allow faster, easier and more flexible creation, deployment, and operation of new personalized applications/services

# VHE(service capabilities)

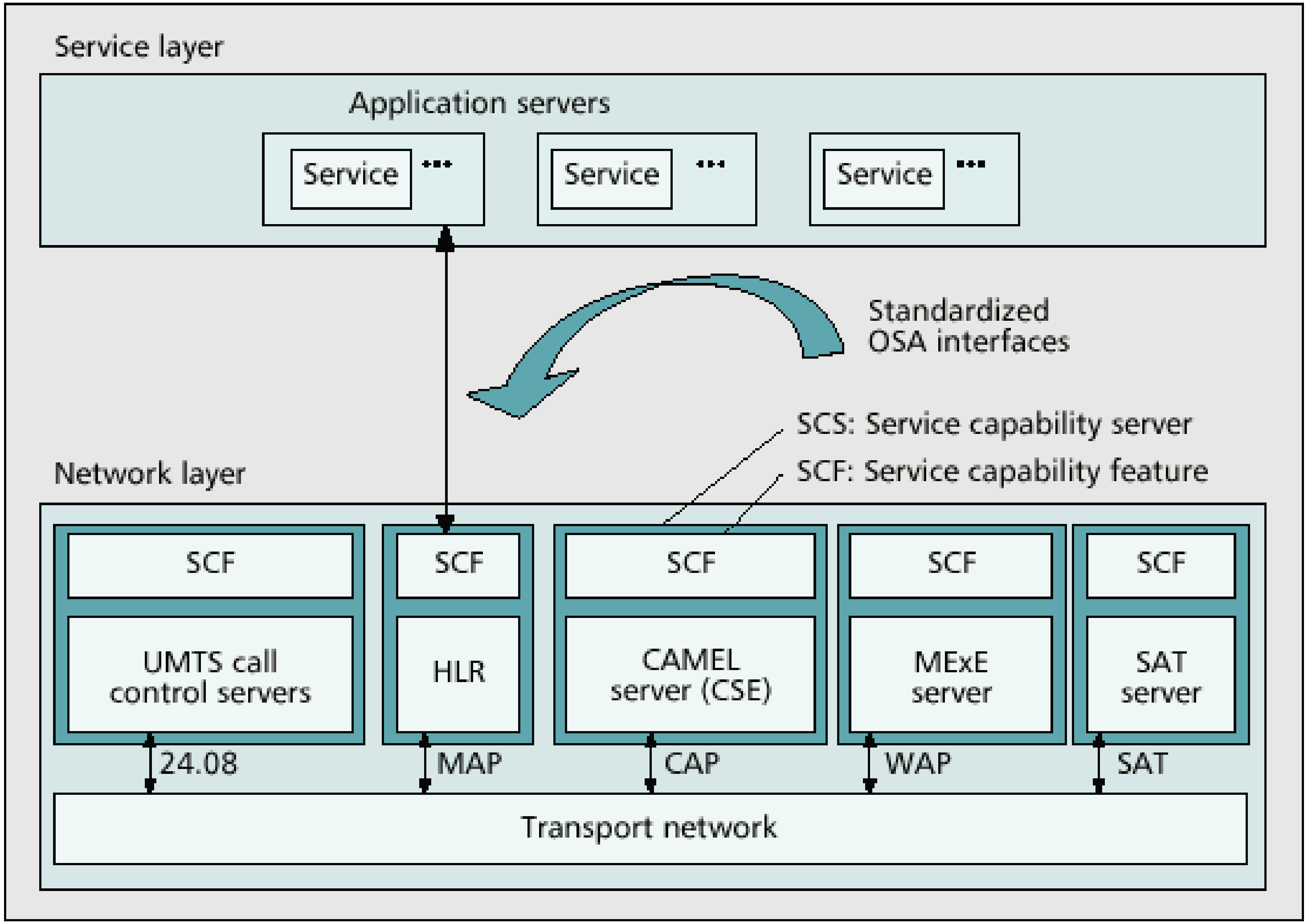



Figure 1. *The concept of the virtual home environment (R99).*

# VHE(service capabilities)

- VHE (continued)
  - Open interface OSA
    - Object oriented Application Programming Interface (API)
  - SCSs (Service Capability Servers)
    - defined as all those servers in the network that provide functionality used to construct services.
    - Group into software interface classes.
  - SCFs (Service Capability Features)
    - The classes of the OSA interface
    - e.g. call control, location.positioning and notification.
  - Secure
    - Service layer access to the SCFs of all the SCSs in the network layer
    - Additional authentication, authorization, accounting and management

# More definitions

- OSA Interface: standardized Interface used by application to access service capability features
- Personal Service Environment (PSE): contains personalized information defining how subscribed services are, provided and presented towards the user, the PSE is defined in terms of one or more User Profiles.
- Service Capabilities (SC): bearers defined by parameters, and/or mechanisms needed to realize services, these are within networks and under network control.
- Service Capability Feature (SCF): functionality offered by service capabilities that are accessible via the standardized OSA Interfaces
- Service Capability Server: Functional Entity providing OSA interfaces towards an application
- Value Added Service Provider: provides services other than basic telecommunications service for which additional charges may be incurred
- Virtual Home Environment: concept for personal service environment portability across network boundaries and between terminals

# VHE(Open Standardized Interface with Service Capability Servers SCSs)

- The functionality represented by the SCFs is offered via an open standardized interface, OSA interface.
- GSM/UMTS protocol
  - MAP
  - CAP
  - WAP

# VHE(Open Standardized Interface/SCSs)

- UMTS call control servers
  - Only MSC for Circuit Switch Call Control, 24.08 CC is UMTS Call Control protocol
- HLR( Home Location Register)
  - Location and subscriber information, MAP
- MExE server (Mobile Execution Environment )
  - JVM, WAP browser, WAP and WTP (Wireless Telephony Application)
- SIM Application Toolkit (SAT) server
  - SIM card contains certain subscriber and security related information.
  - Some small application (phone book, calendar..)
  - Pro-active command from SIM
- CAMEL( customized Application for Mobile Network Enhanced Logic)
  - IN (prepaid service)
  - Invoke via trigger

# VHE(Open Standardized Interface/SCSs)

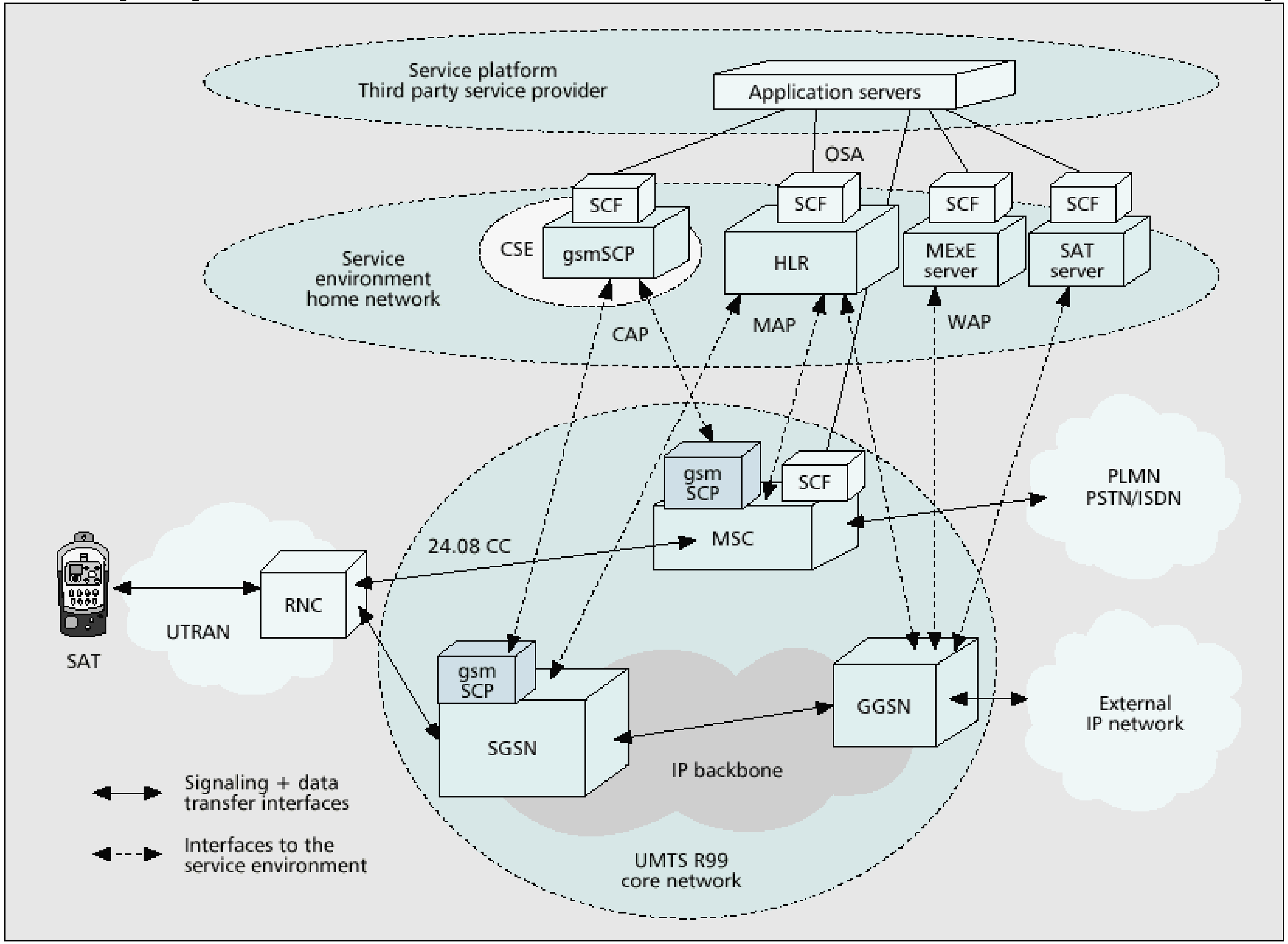



Figure 2. *Mapping of SCFs to the Release 99 network architecture.*

# VoIP issues in Rel 5 of UMTS

- New features introduced in All IP Release
  - Provisioning of IP-based multimedia service
  - Packet based network transport. Replace circuit switched transport
  - IP transport within the UTRAN
  - Network architecture is independent of the transport layer, based on IP or ATM
- All IP Release support two types of real-time service
  - Circuit switched voice service
  - IP-based multimedia service
- All IP Release's MSC is split into
  - A control part : MSC server
  - A transport part : Media Gateway Controller MGC

# VoIP(All-IP UMTS solution)

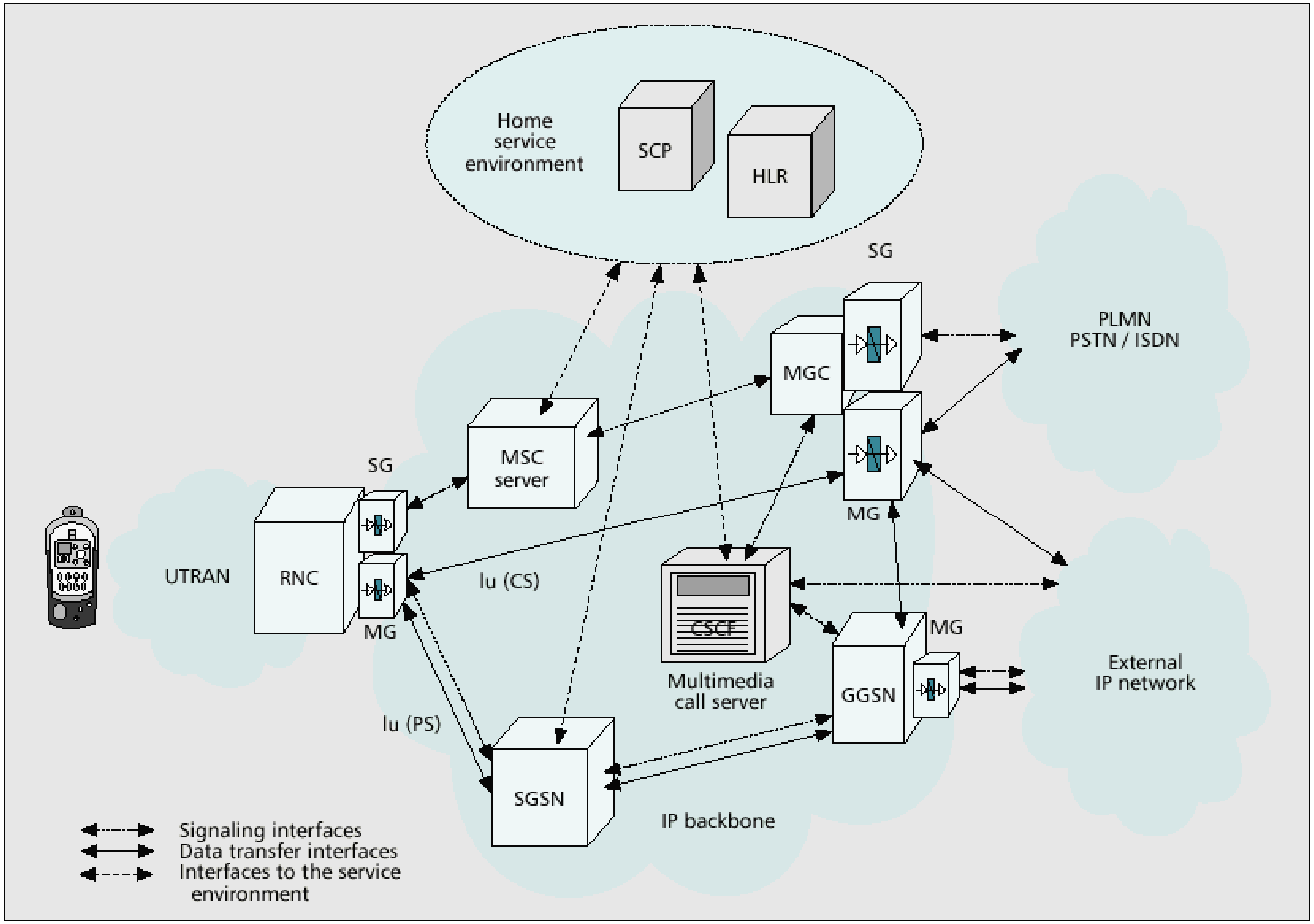
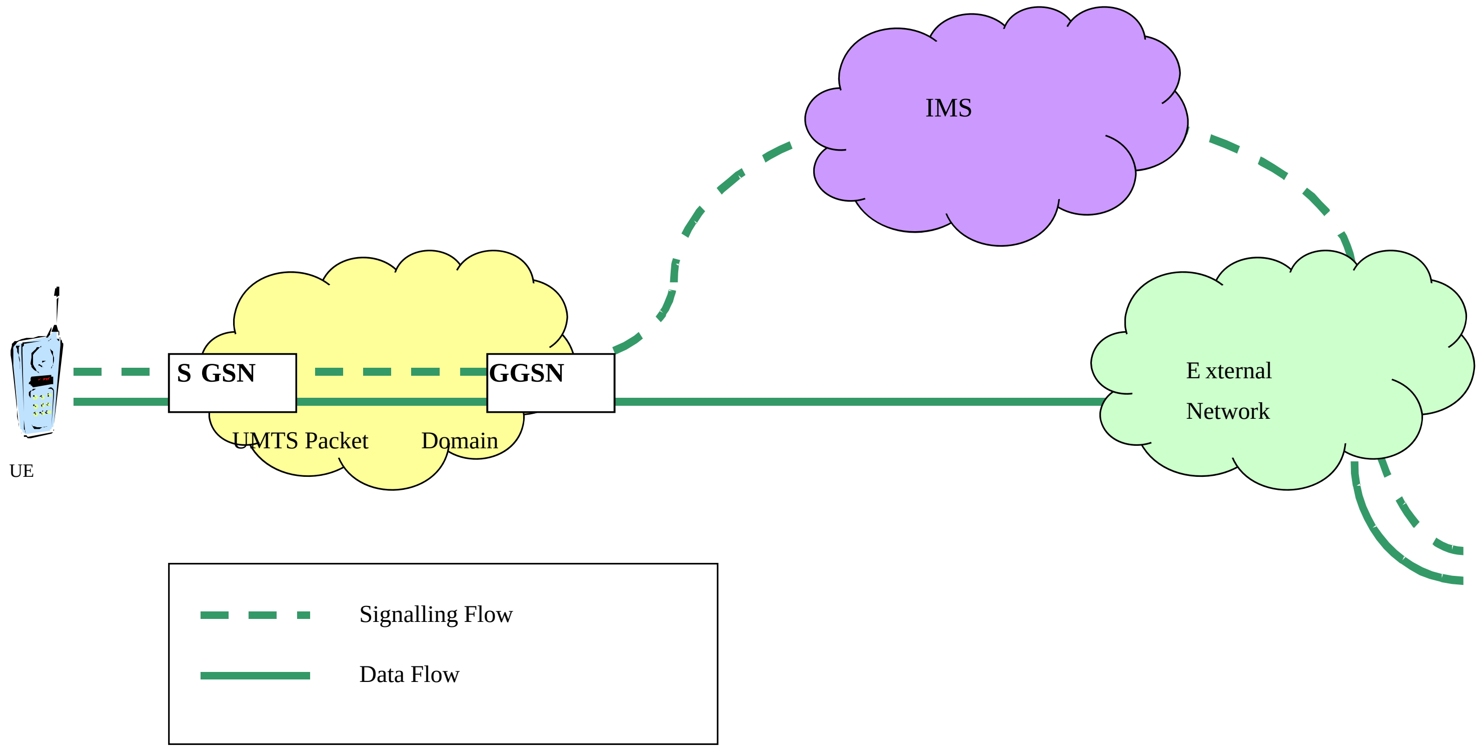


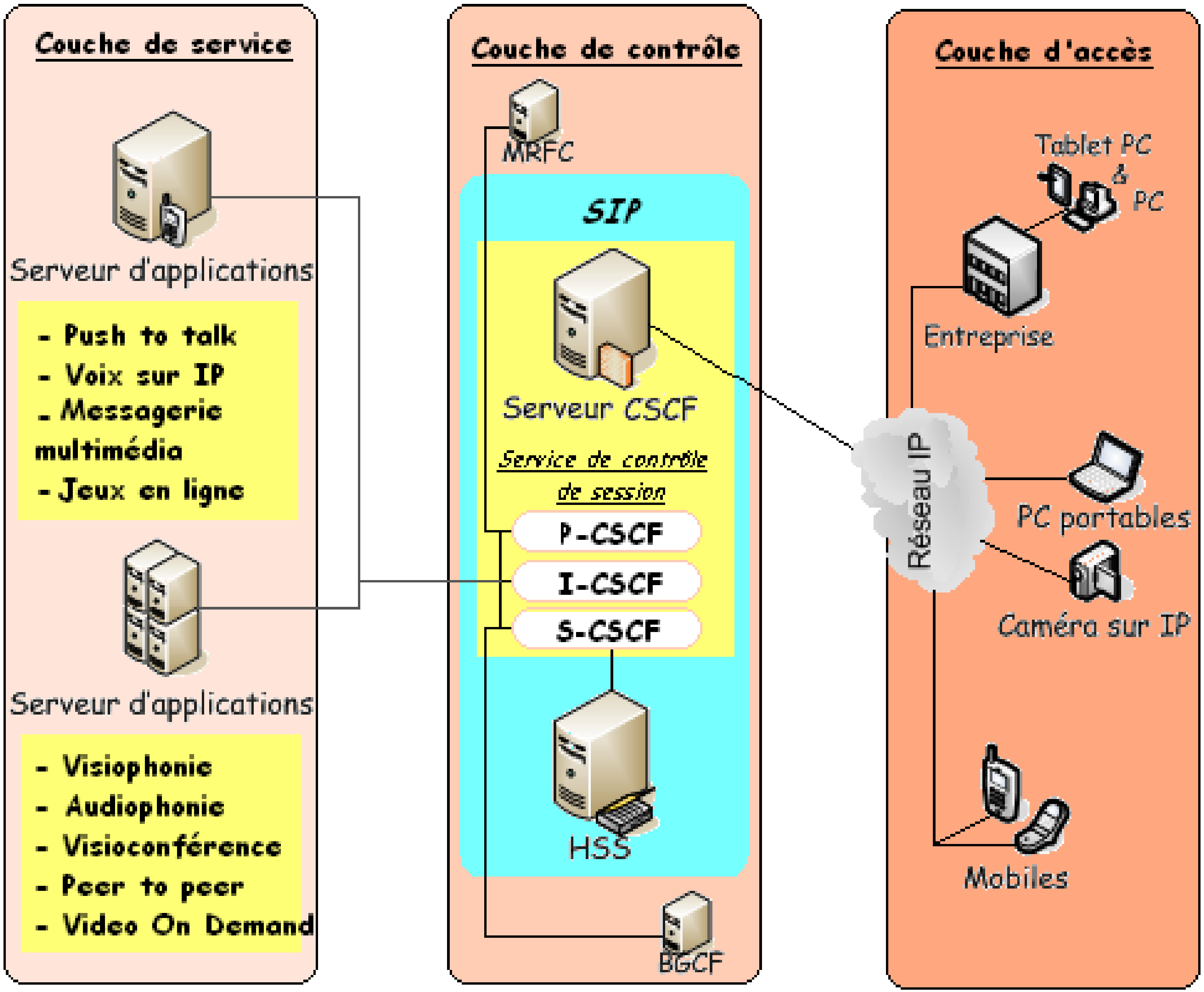

Figure 3. *A simplified Release 2000 all-IP architecture.*

# About IMS

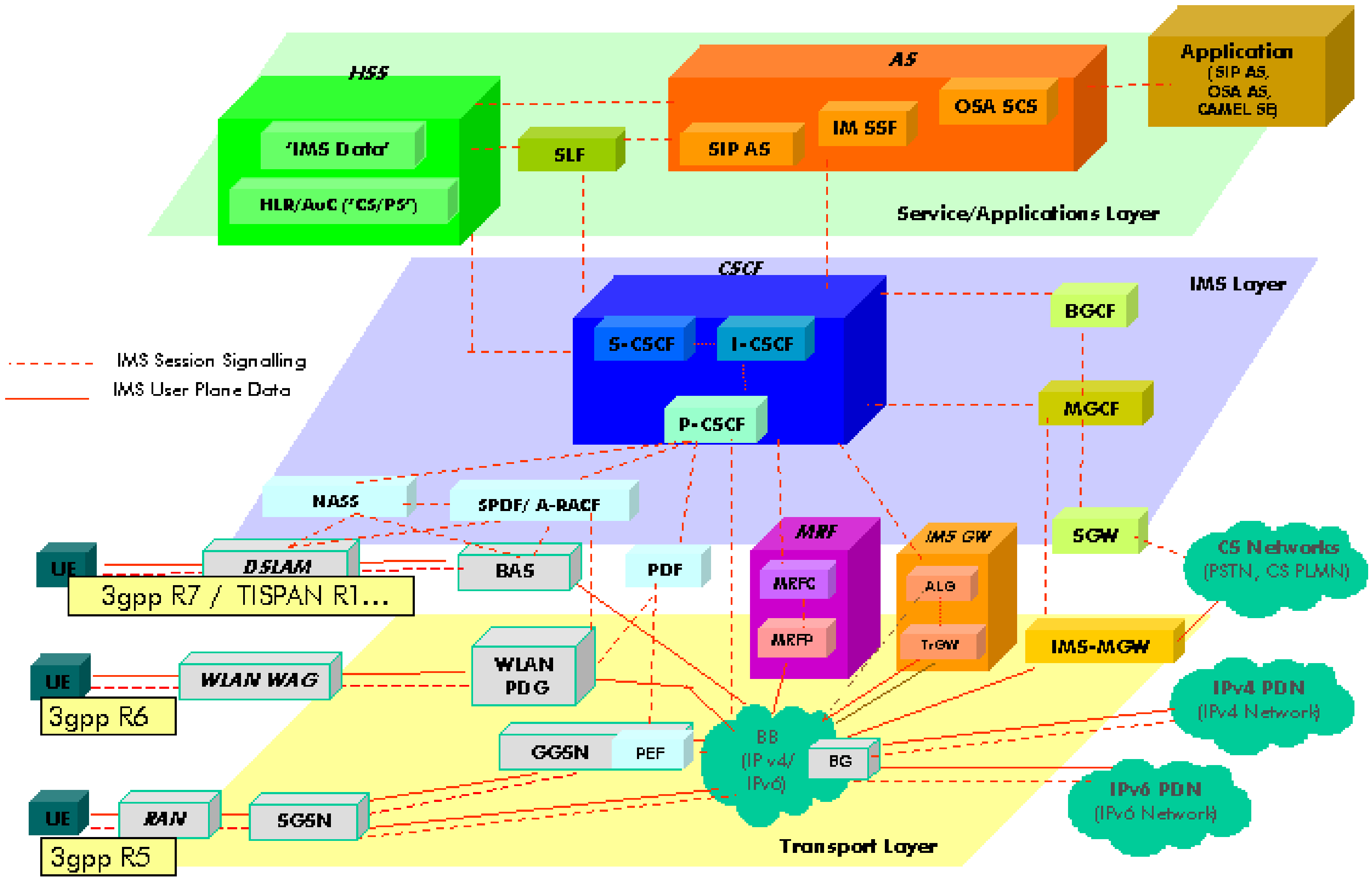
- IMS :
  - Provides UMTS packet domain with session control plane
    - Protocol chosen to perform call control is SIP (Session Initiation Protocol)
    - QOS is negotiated end-to-end at the application level via SIP protocol, then this QOS is translated to UMTS QOS to allocate CN bearer.
  - Enables UMTS customers to be reachable anytime anywhere thanks to the SIP identifier
    - Note that in R99 and Rel 4 mobiles can not receive packet based calls from end users.
  - Enables enhanced services in conjunction with service platform and architecture (OSA, CAMEL).
  - New services via packet domain : VoIP, videoconference, games…

# Path followed by IMS Signaling and Multimedia Data

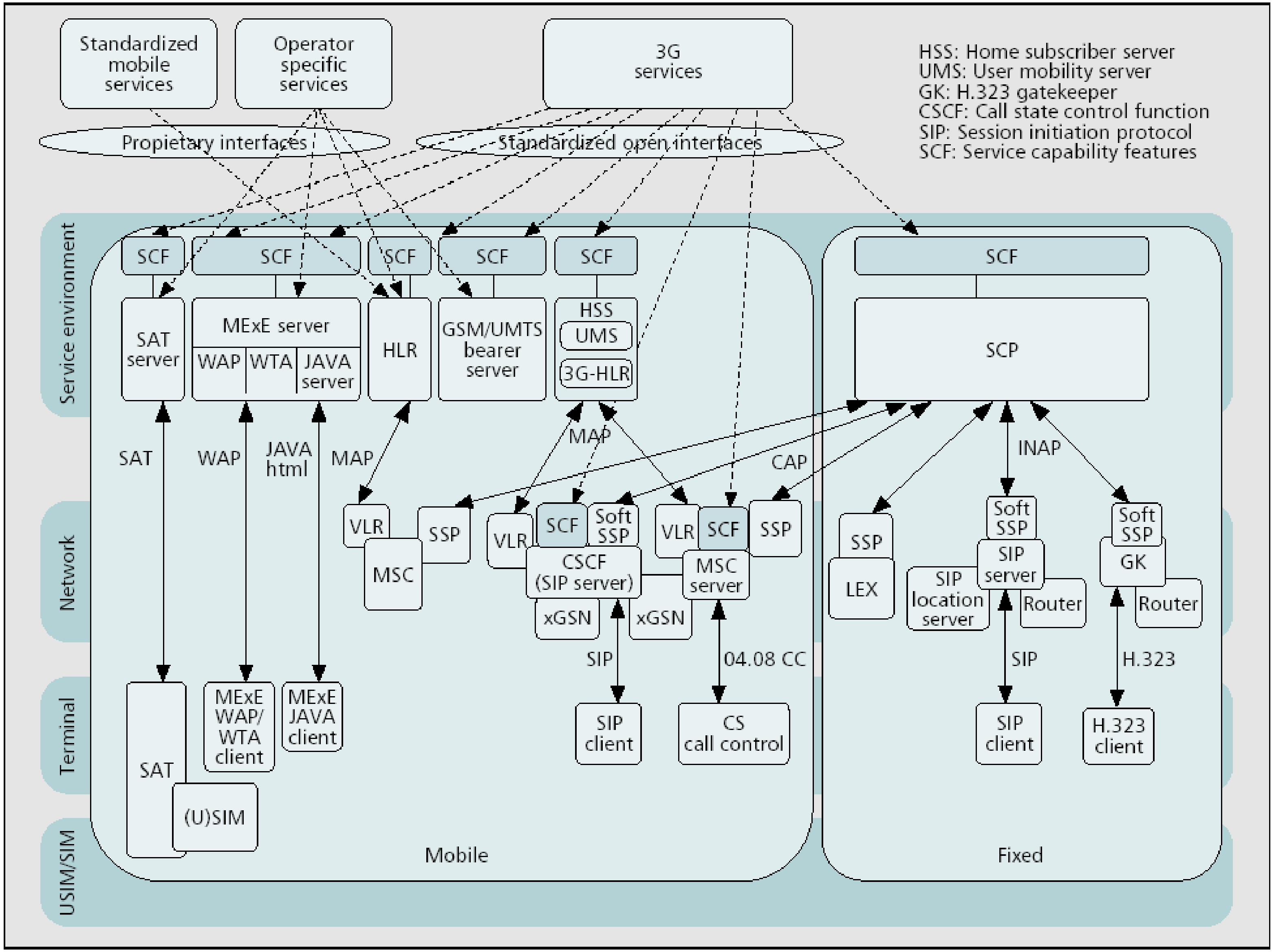



Couche de service
Serveur d'applications
- Push to talk
- Voix sur IP
- Messagerie multimédia
- Jeux en ligne
Serveur d'applications
- Visiophonie
- Audiophonie
- Visioconférence
- Peer to peer
- Video On Demand
Couche de contrôle
MRFC
SIP
Serveur CSCF
Service de contrôle de session
P-CSCF
I-CSCF
S-CSCF
HSS
BGCF
Couche d'accès
Tablet PC & PC
Entreprise
Réseau IP
PC portables
Caméra sur IP
Mobiles

# IMS layers

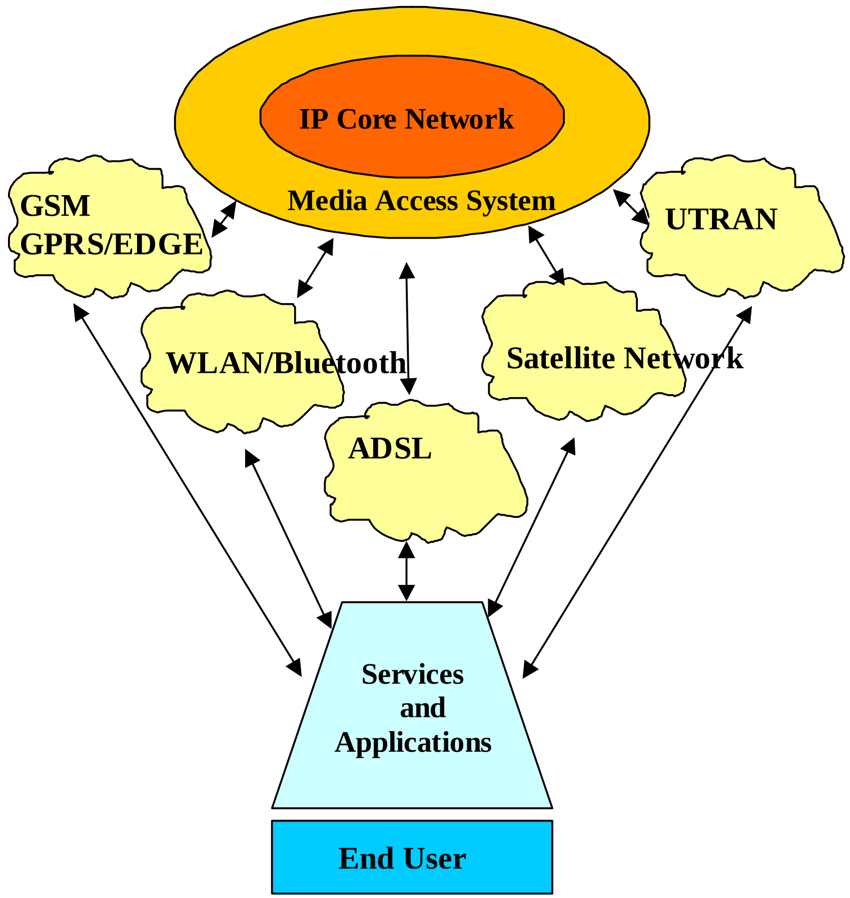


HSS
'IMS Data'
HLR/AuC ('CS/PS')
SLF
AS
SIP AS
IM SSF
OSA SCS
Application
(SIP AS,
OSA AS,
CAMEL SE)
Service/Applications Layer
CSCF
S-CSCF
I-CSCF
P-CSCF
IMS Layer
BGCF
MGCF
IMS Session Signalling
IMS User Plane Data
NASS
SPDF/ A-RACF
UE
DSLAM
BAS
PDF
MRF
MRFC
MRFP
IMS GW
ALG
TrGW
SGW
CS Networks
(PSTN, CS PLMN)
3gpp R7 / TISPAN R1...
UE
WLAN WAG
WLAN
PDG
IMS-MGW
IPv4 PDN
(IPv4 Network)
3gpp R6
GGSN
PEF
BB
(IP v4/
IPv6)
BG
IPv6 PDN
(IPv6 Network)
UE
RAN
SGSN
3gpp R5
Transport Layer




UMTS Service Architecture

# Fixed/Wireless Convergence



# Network Design Issues

- Access network and core network design
- Traffic modelling
- Traffic engineering
- Handover management
- Roaming management
- Hierarchical cellular organisation