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Architecture for an All IP network
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Contents

Foreword

This Technical Report has been produced by the 3GPP.

The contents of the present document are subject to continuing work within the TSG and may change following formal TSG approval. Should the TSG modify the contents of this TS, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

Version x.y.z

where:

    x   the first digit:

        1   presented to TSG for information;

        2   presented to TSG for approval;

        3   Indicates TSG approved document under change control.

    y   the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.

    z   the third digit is incremented when editorial only changes have been incorporated in the specification;

1   Scope

This technical report will propose an architecture that provisions an all-IP architecture option for release 00. The purpose of the technical report is to

    -         identify key issues and affected ongoing 3GPP work that need to be resolved and

    -         propose a high level work plan for completion of an all IP release 00 UMTS standard

in order to provide this architectural option within Release 2000.

2   References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.

- For a specific reference, subsequent revisions do not apply.

- For a non-specific reference, the latest version applies.

- A non-specific reference to an ETS shall also be taken to refer to later versions published as an EN with the same number.

[1]   TS 22.101 version 3.6.0: Service Principles

[2]   TS 23.121: Architectural Requirements for Release 99.

[3]   TS 22.121: The Virtual Home Environment

[4]   TS 23.002: Network architecture

[5]   "Compressing TCP/IP Headers for Low-Speed Serial Links", IETF RFC 1144, V. Jacobson

3   Definitions and abbreviations

3.1   Definitions

For the purposes of the present document, the [following] terms and definitions [given in ... and the following] apply.

existing service: services supported in Release 99 and earlier releases for both GSM and UMTS.

All IP core network: core network of release 2000 that uses IP for transport of all user data and signalling

ERAN is defined as an evolved GSM BSS supporting EDGE modulation schemes on a 200kHz basis and real time packet services

3.2   Abbreviations

For the purposes of the present document, the following abbreviations apply:

<ACRONYM>	<Explanation>
2G	second generation
3G	third generation
AMR	Adaptive Multi Rate
AS	Application Server
BSC	Base Station Controller
BTS	Base Station
CAMEL	Customised Applications for Mobile Network Enhanced Logic
CAP	CAMEL Application Part
CC	Call Control
CCF	Call Control Function
CN	Core Network
CS	Circuit Switched
CSCF	Call State Control Function
CSE	CAMEL Service Environment
DN	Directory Number
DNS	Directory Name Server
EDGE	Enhanced Data for GSM
EGPRS	Enhanced GPRS
FFS	For Further Study
GGSN	Gateway GPRS Support Node
GMSC	Gateway MSC
GPRS	General Packet Radio Service
GSN	GPRS Support Node
GTP	GPRS Tunnelling Protocol
H-CSCF	Home CSCF
HN	Home Network
HSS	Home Subscriber Server
ICGW	Incoming Call Gateway
IN	Intelligent Network
INAP	IN Application Part
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISP	Internet Service Provider
ISUP	ISDN User Part

LAN	Local Area Network
LN	Logical Name
MAHO	Mobile Assisted Handover
MAP	Mobile Application Part
MCU	Media Control Unit
MExE	Mobile Execution Environment
MGCF	Media Gateway Control Function
MGW	Media Gateway
MM	Mobility Management
MO	Mobile Originated
MRF	Media Resource Function
MSC	Mobile Switching Centre
MT	Mobile Terminated/Terminal
NPA	Numbering Plan Area
O&M	Operations and Maintenance
ODB	Operator Determined Barring
OSA	Open Service Architecture
PCU	Packet Control Unit
PDP	Packet Data Protocol
PDU	Packet Data Unit
PS	Packet Switched
PSTN	Public Switched Telephony Network
QoS	Quality of Service
R00	Release 2000
R99	Release 1999
RA	Routing Area
RAN	Radio Access Network
RLC/MAC	Radio Link Control/Media Access Control
RNC	Radio Network Controller
R-SGW	Roaming Signalling Gateway
RTP	Real Time Protocol
SAT	SIM Application Toolkit
SCF	Service Control Function (IN) and Service Capability Features (VHE/OSA)
SCP	Service Control Point
S-CSCF	Serving CSCF
SGSN	Serving GPRS Support Node
SIP	Session Initiated Protocol
SLA	Service Level Agreement
SN	Serving Network
SRNC	Serving Radio Network Controller
SSF	Service Switching Function
TCP	Transmission Control Protocol
TE	Terminal Equipment
T-SGW	Transport Signalling Gateway
UDP	User Datagram Protocol
UE	User Equipment
UMS	User Mobility Server
UTRAN	UMTS Terrestrial Radio Access Network
VHE	Virtual Home Environment
VLR	Visitor Location Register
VoIP	Voice over IP
VPN	Virtual Private Network
WAP	Wireless Application Protocol
WIN	Wireless IN (ANSI-41)

4   Requirements

In order for TSG-SA2 to conduct a study of the architecture issues relating to the introduction of an All IP architecture within UMTS, assumptions were made as to the requirements for this architecture. TSG-SA1 are invited to validate and to extend these requirements as part of the work package on requirements for Release 2000.

4.1   General

The aim of the all IP architecture is to allow operators to deploy IP technology to deliver 3rd Generation services, that is an architecture based on packet technologies and IP telephony for simultaneous real time and non real time services. This architecture should be based on an evolution from Release 99 specifications and should be compatible with IMT-2000, providing global terminal mobility (roaming) [1].

The IP network should provide wireless mobility access based on ERAN and UTRAN with a common core network, based an evolution of GPRS, for both. In this context, an E-GPRS radio access network is a 200kHz GSM based network supporting EDGE and evolved to support real time packet services. Although EDGE is not within the scope of 3GPP, there are requirements for the core network of the all IP architecture, to be common to both access technologies.

The characteristics of this network are

- Based on an evolution of GPRS

- Common network elements for multiple access types including UTRAN and ERAN

- Packet transport using IP protocols

- IP Client enabled terminals

- Support for voice, data, real time multimedia, and services with the same network elements.

The report also covers the support of CS services on IP technologies.

The benefits of this approach include

- Ability to offer seamless services, through the use of IP, regardless of means of access (e.g. common features used by subscribers whether accessing via conventional land telephony, cable, wireless, HIPERLAN 2 etc.)

- Synergy with generic IP developments and reduced cost of service

- Efficient solution for simultaneous multi-media services including voice, data, and advanced real time services.

- Higher level of control of services

- Integrated, and cost reduced OA&M through IP

- Take advantage of Internet applications by supporting terminals which are IP clients.

- Cost reduction through packet transport

4.1.1   General Requirements

1. The overall aim of the all IP network is to support similar services to GSM release '99 and new innovative services. Where appropriate these services should inter-work with exisiting GSM services.

2. In addition it should also possible to support existing (R99 and before) services/capabilities (speech, data, multimedia, SMS, supplementary services, VHE,...)in a manner that is transparent to the users of these services [1]. That is, the network needs to provide the service capabilities required in such a way as to support interworking of these services between the R00 all IP network option and the other family networks two domain architecture option (GSM pre Release 99, UMTS release 99).

3. The standard shall enable the all IP core network to support release 99 CS terminals. This shall be standardised in such a way as to allow operators to decide whether or not they wish to support Release 99 CS only terminals.

4. The support of existing services shall not preclude the extension of service capabilities possible through the use of an all IP architecture.

5. When the all IP networks are deployed, there will be services and databases provided for existing networks which are non-IP based e.g. local number portability, free phone numbers, specialised corporate services. The all IP architecture will need to be able to access these services.

6. R'00 all IP core network shall allow implementations having a CS and a PS domain, that are separated like both these domains in the R'99 architecture. This implementation allows the two domains to evolve independently, e.g. to combine an all IP R'00 PS domain with a STM based R'99 CS domain. Furthermore it shall be possible to implement a CS domain that uses all IP based architecture and in distinct service areas of the same network a CS domain based on ATM/STM. This allows a smooth migration to an all IP based core network.

The R'00 all IP architecture shall support that all services share bearer level transport and bearer control.

R'00 architecture shall allow an operator to migrate a R'99 network into a R'00 network, without need for change of transport network technology, node numbering scheme etc. R'00 networks shall also allow connection of R'99 UTRAN over Iucs, to provide the operator with flexibility in the network implementation.

Note: In the general R'00 architecture other transport technology than IP shall be possible.

4.2 Service Capabilities

4.2.1 General

The following general service capabilities are identified:

1. Legal interception has to be possible in the R00 All-IP network.

2. Emergency calls shall be supported in the R00 All-IP network.

4.2.2 Basic requirements for the Service and Application Platforms

List of requirements on the "Application and Service" block:

1. Service capabilities are to be made available to the Applications through Service capability features;
2. Service capability features are provided by one or more service capabilities (possibly directly provided by network functions, e.g. HSS, CSCF etc.), as illustrated in Figure 4-1;
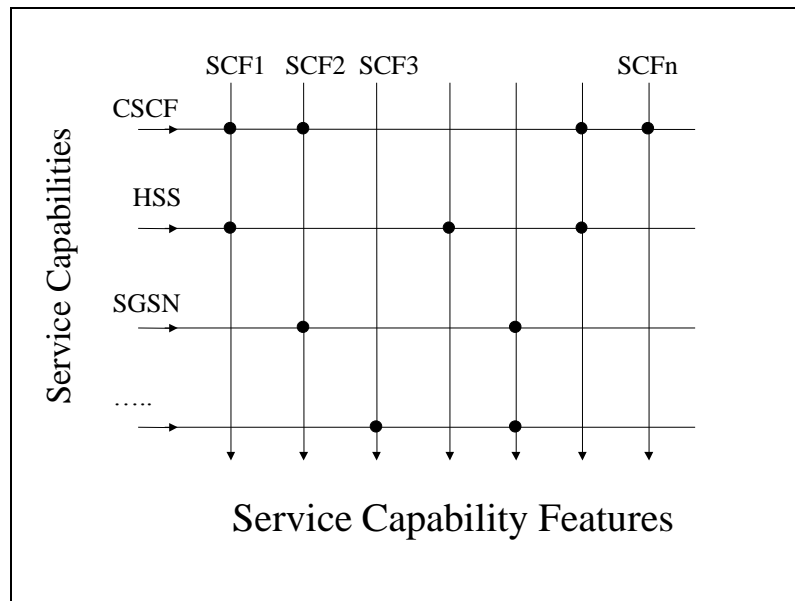
Figure 4-1: Relationship between Service Capability Features (SCFs) and Service Capabilities.

3.   Incremental introduction of service capability features by the network operator has to be possible, for operator specific service capabilities;

4.   A standard interface, called the application interface, has to be introduced for access to Service capability features by Applications. This interface has to provide a controlled, secure and accountable relationship between Applications and Service capability features;

5.   The Application interface must provide access to the Service capability features based on user subscription profile;

6.   Applications can be located either on servers and/or on (mobile) terminals;

7.   Applications can only access the service capabilities through the service capability features;

8.   Applications can utilize both capabilities provided by the Mobile Network functions, and functions as provided by IT systems, through the service capability features.

4.3   Numbering Schemes

The standards shall allow mobile terminated communications to be routed to the user's terminal on the basis of a single identifier e.g. MSISDN. This does not preclude multiple addresses being used for different services and capabilities (e.g. data, Fax, SMS). The network will route the call to the terminal over the available resources, dependent upon, for example, terminal capability, traffic loading and coverage. Networks migrating to an all IP architecture will require the ability to route based on a single identifier to maintain service transparency.

4.4   R99 Terminals

See section 4.1.1 above: The following requirements for the support of R99 terminals is an operator specific option.

1.   The standards shall enable the All-IP core network to support UMTS R99 terminals.

2.   Speech services including emergency calls shall be provided in All-IP networks to any UMTS R99 terminal supporting these services.

3.   To ensure roaming of non VoIP capable UMTS R99 terminals, speech services including emergency calls shall be possible based on CS capabilities of these terminals.

4.5   Radio aspects

1.   The radio resource usage should be optimised within the architecture for both service and signalling support.

2.   Separation of the radio related and radio un-related functionalties between the core network and the radio access network

3.   Separation of the user plane and control plane protocol stacks in the radio access networks

4.   Fast uplink access and fast resource assignment procedures in both uplink and downlink for multiplexing different types of traffic on the same air link.

5.   Optimization of end-to-end IP transport for certain class of real time applications (e.g. header compression, header stripping)

6.   Network controlled handover procedure with short interruption to support real time applications (see handover requirement, section 4.9.)

7.   Protocol stacks in the access network to support a range of services with different QoS requirements

8.   Inter-working/interoperability of the QoS mechanism developed for the radio access network and the QoS mechanism used in the packet core network.

9.   Bearer differentiation capability at the access network for multiplexing different types of traffic on the air to achieve maximum spectrum utilization

10.   Optimal coding and interleaving for some applications such as voice.

11.   Support of multiple data flows with different QoS per IP address (as defined in QoS framework in release 99)

12.   Spectrum efficiency shall be maximized (e.g. statistical multiplexing).

13.   The ERAN shall support GPRS and EGPRS services for pre-Release 2000 terminals.

4.6   Interworking

1.   The All-IP core network shall support interworking to external IP and non-IP networks (e.g. circuit-switched networks (PSTN, ISDN, GSM PLMN, UMTS PLMN,...).

4.7   Mobility management

1.   The All-IP core network shall provide streamlining and CN operated hand-over procedures for UMTS.

4.8   Roaming

1.   The standard shall enable the All-IP core network to support roaming with 2G GSM/GPRS networks and R99 UMTS networks.

4.9   Handover

The support of handover between release 98, release 99 and release 2000 network technologies is essential in maintaining adequate network coverage. Table 4-1 illustrates the necessary handover scenarios and the status of development of mechanisms.

Table 4-1: Handover requirements for UMTS All IP network

Between	2G-GSM cs	2G-GPRS	UMTS cs (R99)	UMTS ps (R99)	UMTS IP (PS services)	UMTS IP (CS services)
UMTS IP (PS services)	Req R00*	Req R00	Req R00*	Req R00	OK	Not Required
UMTS IP (CS services)	Req R00	Not Req	Req R00	Not Req	Not Req	OK

Key:

OK   Same technology

Req R00   Required for release 2000

* The implications of the requirement for PS to/from CS handover in R00 are the subject of much debate. Alternatives for either the support of handover or to provide service coverage need to be investigated.

4.9.1   Handover Categories

1   Intra network handover

       Handover inside one all IP network

         1a Intra RAN handover

         1b Inter RAN handover

2   Inter network handover

       Handover between two all IP networks

3   Inter-system handover

       Handover between an all IP network and other systems

4.9.2   General Definition

Reselection and handover are two methods of supporting mobility during an active session. Reselection is the process whereby the mobile station autonomously determines which cell the mobile will receive services on. Handover is the process whereby the network determines which cell the mobile will receive services on.

4.9.3   General Requirements

For real-time services handover procedure shall be used. The network shall control the handover procedure.

Handover shall be the selected method of mobility if one or more active sessions have requested handover in a multi session call with different QoS requirements

Performance requirement on speech interruption (i.e. "mute" period) shall be equivalent to or better than than GSM or ANSI-136 handover for telephony services. TBD for other services offered by an all IP Network.

Maintain maximum packet loss limit (i.e. less than TBD) and maximum delay limit (i.e., less than TBD) during handover.

Non-real-time services shall use either handover or cell reselection depending on the QoS parameters in combination with network parameters.

The Subscribed QoS level should be maintained across the handover boundary. However QoS negotiation (if neceassary) should be possible before, during and after the handover (The application may reject the offered QoS)

Handover procedure shall utilize radio resources efficiently.

Handover shall not compromise the security of: the network providing the new radio resources; the (possibly different) network providing the original radio resources; and the terminal UE.

There shall be efficient handling of multiple bearers, e.g. if voice and email transfer is going on simultaneously.

Essential IP/UDP/RTP header information (for inter and intra all IP network handover), as seen by an IP end point, shall be preserved across handover boundary. The required essential header information depends on the bearer.

4.9.4　　MS Requirements

The mobile station shall be capable of supporting both reselection and handover.

The mobile station shall aid the RAN in the handover decision by supplying RF environmental information (e.g. received signal strength from serving cell and neighbour cells).

4.9.5　　RAN Requirements

Handover decisions shall be based in the RAN.

Maintain the RAN QoS parameters, associated with the mobile station, across a handover boundary. Note, RAN QoS parameters for a mobile station are based upon the negotiated set of QoS parameters.

Facilitate admission control to optimize radio resources.

Select a handover target based on criteria such as RF environmental information, radio resources of the neighboring cells, QoS requirements of the session, etc.

4.10　　Call Control and Roaming

The following requirements need consideration for call control and roaming support in an all IP based network.

1 Routing of signalling and transport needs to be optimised, for the purposes of call control and roaming between networks.

2 Whenever possible, tromboning of the user's voice or data communication session back to their home environment should not be used to provide the user with services when roaming outside their home network.

3 The Release 2000 all IP network must comply with the mandated requirements for Emergency Services.

4 The Release 2000 all IP network must comply with the mandated requirements for Number Portability.

5 The Release 2000 all IP network must support multiparty voice and data communications sessions including the capability for the user or service logic to dynamically add or delete users from an active communications session.

6 The Release 2000 all IP network must be able to accept and re-route incoming voice or data communication requests that are addressed to the user's directory number during periods of realignment of the national numbering plans (e.g., NPA splits in North America).

7 Transcoding of the traffic (voice, data, video) should be minimised. For example,, if the terminal equipment of the called and calling party have the same vocoder, no transcoding of the voice traffic, within the network, would occur.

8 The Release 2000 all IP network must provide connection to the services of the legacy 2G and release 99 networks.

9 The Release 2000 all IP network shall support VHE for roamers.

10 A minimum set of services for roamers shall be provided within the serving network. This minimal set of user services is still being defined. However, the following is anticipated to be in this minimal set of user services:

　a Speech call and data session origination.

    b   Speech call and data session termination.

    c   Call Waiting for voice calls in the case of monomedia session

    d   Call Forwarding services for voice calls.

    e   Calling party identification information

    f   SMS

11   In the event that the Release 2000 all IP operator does not have a legacy network in the market that a Release 2000 all IP network is being deployed into and the Release 2000 all IP operator does not have any business relationships with the operators of the legacy networks. Consequently, the design of the Release 2000 all IP network can not assume that the requirements for mandated services or operator-specific services can be satisfied by forwarding the Release 2000 all IP call to the legacy network. The following are examples of Operator Services that may need to be handled directly by the Release 2000 all IP networks:

    a   Directory Assistance

    b   Third party billing

    c   Collect calls

    d   Calling card calls

12   When a Release 2000 all IP user roams from a Release 2000 all IP network to another Release 2000 all IP network and gets access to both transport services (e.g. GPRS) and application level services (e.g. multimedia calls), services may be provided by a CSCF in the serving network or by a CSCF in the home network. The serving network shall contain the information to contact the user's home network for the user's profile information. The CSCF of the serving network shall have access to the necessary information for the invocation and control of the user's advanced/ supplementary services at the user's home network.

13   The user shall be able to gain access to their ISPs or corporate LAN application level services..

14   Both dynamic and dedicated IP addresses shall be supported.

15   Release 2000 all IP networks will be capable of providing VPN functionality. VPN refers to both the GSM VPN and Intranets. The VPN features supported require further study and analysis.

16   When the UE uses the service of a CSCF/MSC server, the CSCF/ MSC Server needs to authenticate the UE. The way this is handled when the UE uses the services of a CSCF in the visited network requires further study and analysis.

4.11   Security

The general principle for security for the all IP network implementation is to reuse the same mechanisms developed for 3GPP Release 99 wherever possible.

5   Architecture for an all IP PLMN

5.1   Reference Architecture

The reference architecture provides two options:

Option 1: has been developed with the goal of allowing operators to deploy an all IP based architecture to deliver 3rd Generation wireless/mobile services. This architecture is based on packet technologies and IP telephony for simultaneous real time and non real time services.

Option 2:. One purpose of option 2 is also to allow support of release 99 CS domain terminals. In addition option 2 also supports the IP based services of option 1.

5.1.1   Reference Architecture – Option 1

As described earlier in the Requirements section 4.1, the architecture shown in Figure 5-1 has been developed with the goal of allowing operators to deploy an all IP based architecture to deliver 3rd Generation wireless/mobile services. This architecture is based on packet technologies and IP telephony for simultaneous real time and non real time services.

The architecture shown and the components of which are described in subsequent sections allow for flexible and scaleable mechanisms to support global roaming and interoperability with external networks such as PLMN, 2G Legacy networks, PDNs and other multimedia VOIP networks.

The end-to-end architecture consists of the following key segments:

a)   Radio Network

b)   The GPRS network

c)   The Call Control

d)   Gateways to external networks

e)   The Service architecture

The Radio network part consists of the equipment associated with the mobile user, the Radio Airlink and the Radio Access Network. The RAN supports both the UTRAN and the EDGE technologies.

Section 4 indicates that the intent of the core network part of the all IP architecture is that it should be designed to allow operators to use other access networks, for example ERAN and HIPERLAN 2. Within Figure 5-1, the ERAN is shown explicitly, where as the other access networks are represented by the bubble labelled "Alternative Access Network". For the purposes of this report,

> the **ERAN** is defined as an evolved GSM BSS supporting EDGE modulation schemes on a 200kHz basis and real time packet services

The support of alternative access networks, and the impact of the All IP architecture on the ERAN are outside the scope of this activity. However, to avoid the loss of information, the report does indicated where requirements are known to apply to these access networks.

Within this report, the reference point between the ERAN and the core network is designated as the Iu_ps'. That is, the reference point is Iu and the implementation is expected to be similar to that of the Iu_ps.

Note: the use of the reference label Iu_ps' is confusing. During the standardisation activity a more suitable label should be chosen.

The GPRS network part has the GSNs which provide the mobility management and the PDP context activation services to the mobile terminal as they do in the R99 GPRS PS domain network. The HLR functionality for the GPRS network is provided by the Home Subscriber Server, (HSS).

The Call Control part of the architecture is the most critical functionality. The CSCF, MGCF, R-SGW, MGW, T-SGW and the MRF comprise the Call Control and signalling functionality to deliver the real-time mobile/wireless services. The CSCF is similar to the H.323 GateKeeper or a SIP Server. The architecture has been intentionally kept generic and is not based on a specific call control mechanisms such as H.323 or SIP. Such a choice is for further study.

The user profiles are maintained in the HSS. The signalling to the multimedia IP network is interface solely via the CSCF while the bearer is interfaced directly with the GGSN. The MRF interfaces with all bearer components for bearer media and with the CSCF for signaling. The MRF provides for media mixing, multiplexing, other processing and generation functions.

The interconnectivity to external networks such as PLMN, other PDNs, other multimedia VOIP networks and 2G Legacy networks (GSM or TDMA) is supported by the GGSN, MGCF, MGW, R-SGW and T-SGW functional elements. Other PLMNs are interfaced for both bearer and signalling via their respective GPRS components. The CSCF is a new component which also participates in this signalling. The signalling to legacy mobile networks is interfaced via the R-SGW, CSCF, MGCF, T-SGW and HSS, while the bearer is interfaced to and from the legacy PSTN network via an MGW.

Legacy landline circuit switch signalling is interfaced via the CSCF, MGCF and T-SGW while the bearer is interfaced to and from the legacy PSTN network via an MGW.

The Service Architecture part of the network is currently depicted as an external entity and is described in detail in the section 10. Non-standard services are provided via interfaces to an application services layer. The HSS, the SGSN and the CSCF interface with the application and services bubble.

The details for each of the functional elements of the architecture are provided in subsequent subsections of this document.
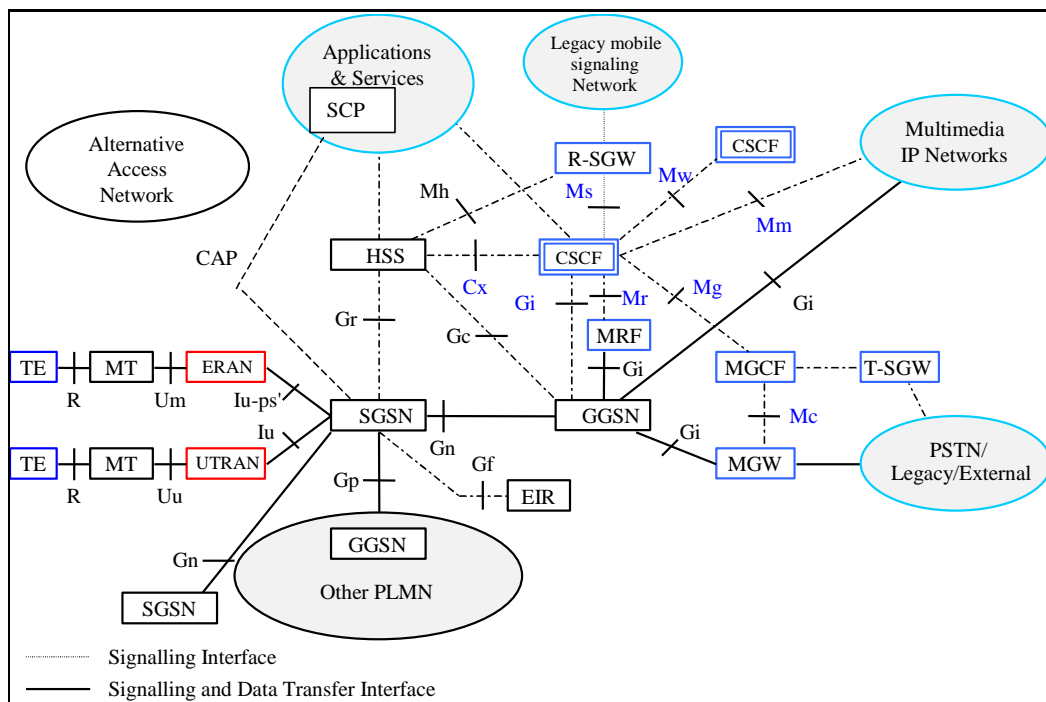

Figure 5-1: Reference Architecture for Option 1

The Gm interface between the UE and the CSCF consists of the user to network multimedia signalling. It is carried on radio, Iu, Gn and Gi interfaces.

The SGSN and GGSN are the same functional elements as defined in 23.002 [4] for R99 of UMTS.

Mobility management procedures for Release 2000 all IP network MSs in Release 2000 all IP networks are based on the Routing Area identifier (RAId). Mobility management procedures for CS capable MSs in Release 2000 all IP networks are based on the Routing Area identifier (RAId) and on Location Area Identifiers. In case of roaming between Release 2000 all IP networks to 2G and vice versa, a mechanism to convert the format of identities is needed (i.e. MS roaming from Release 2000 all IP network to 2G network has knowledge only of old RA but Location Area Identifier also needs to be given to the 2G-MSC). When roaming from 3G-R00 to 2G (without possibility of combined update), how the 2G-MSC can retrieve the IMSI from the all IP core network is an open issue.

MS identity (IMSI) is protected over the radio interface through the adoption of a single temporary identifier (P-TMSI) allocated by the SGSN during the MS registration procedure. P-TMSI can be reallocated at every following registration or routing area update. [relevance to roaming scenarios between Release 2000 all IP networks and legacy cellular networks]

The Access Network Nodes (GSN(s), RNC) are not aware the multimedia signaling protocol between the UE and the CSCF. They are even not aware that a given UE sends or does not send signalling to the CSCF.
Note1: this does not preclude that to optimize the radio, the RNC might support specific RAB for the individual flows of the multimedia user plane. These RAB are requested by the UE at PDP context activation.

Note2: the SGSN may have a role in the choice of the CSCF by the UE. This is FFS. But even if the SGSN participate in the CSCF address determination, the SGSN does not carry out the multimedia registration on behalf of the UE.

Different PDP contexts carry multimedia signalling and user flows due to different requirements on QoS for these PDP contexts. The Access Network Nodes (GSN(s), RNC) are nevertheless not aware whether a given PDP context carries multimedia signalling or not.

Working Architectural approach

1. The All IP Core network is engineered primarily to use a common technology (IP) to support all services including multimedia and voice services controlled by H.323/SIP or ISUP.

2. Network architecture is based upon IP packet technologies for simultaneous real-time and non-real-time services.

3. Network architecture is based upon an evolution of GPRS.

4. For support of R99 CS domain services the R99 CS domain CC mechanism may be reused. (NOTE: This does not prevent alternative mechanism such as H.323, SIP or evolved forms of R99 CS domain CC mechanisms being used by operators to deliver R99 CS domain services)

5. For the support of release 00 terminals are IP based, and the integration of services is obtained through IP.

6. Network architecture should support personal mobility and interoperability between mobile and fixed networks for both voice and data services.

7. Maintain or improve quality of service levels when compared to today's networks.

8. Maintain or improve network reliability when compared to today's networks.

9. All IP interfaces and associated network interfaces should be enhanced to support real-time multimedia services.

10. Network architecture will provide a separation of service control from call/connection control.

11. Network architecture will replace SS7 transport with IP.

12. Network architecture will be independent of network transport layers of Layer 1 (L1) and Layer 2 (L2).

13. Regardless of service type, ISUP based or IP based, IP transport shall be possible for all signalling and data transport.

5.1.2 Reference Architecture – Option 2

As described earlier in the Requirements section 4.1.1 item 3 and 6, the architecture shown in Figure 5-2 allows operators to migrate from a R'99 UMTS network into a R'00 All IP network. One purpose of option 2 is to allow support of release 99 CS terminals. Option 2 allows the two domains of R'99 to evolve independently.

As for option 1 the architecture of option 2 allows that all services supported by option 2 share bearer level transport and bearer control. Various underlaying transport mechanisms shall be allowed (e.g. RTP/UDP/IP, AAL2/ATM or STM).

Reference architecture option 2 includes the SGSN/GGSN/CSCF based services of reference architecture option 1. The definitions and working architectural approach described in chapter 5.1.1 therefore also applies to the SGSN/GGSN/CSCF part of option 2.

[Note: The requirement in section 4.3 for the support of routing on a single MSISDN or to allow operators to move subscribers from the CS services to the All IP service without changing MSISDN will be solved as part of R99.]

Two control elements, related with R'99 CS domain architecture, are added in option 2; the MSC server and GMSC server.

Option 2 benefits from the Iu architecture of R'99, having transport of user data separated from control, to allow UTRAN to access the core network via a MGW separated from the MSC server. Between UTRAN and MSC server the control part of Iu, RANAP, is used.

[Note: 1) Exact definition of ERAN *in relation to GSM/teleservice speech and ERAN's relations to MSC server/MGW needs to be defined. 2) The possibility of interfacing GSM/BSS to MGW and MSC server shall be further studied.]* .

By allowing servers to terminate MAP and the user-network signalling (04.08+ CC+MM), requirements related to service and network migration of R'99 UMTS CS domain services and network migration can be fulfilled. The requirements that requires a network architecture according to option 2 are:

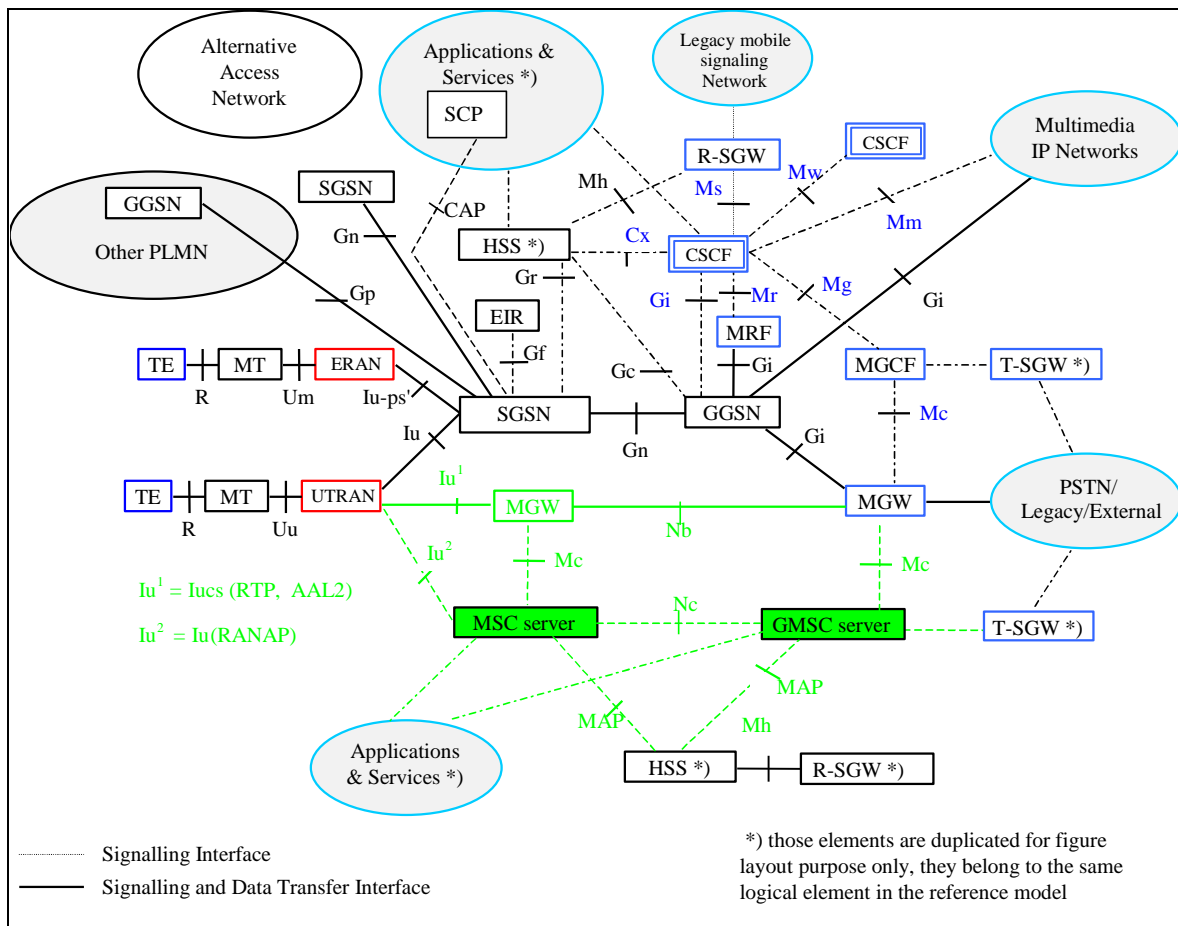| Section	Requirement Number
4.1.1	2 (to meet the timescales of R00)
4.1.1	3
4.1.1	6
4.4	all
4.8	1 (the need for option 2 to meet this requirement will depend upon the roaming solution chosen)

Figure 5-2: Reference Architecture for Option 2

Iu is the reference point between UTRAN and all IP core network. Between UTRAN and SGSN Iu is IP based. Between UTRAN and MGW - Iucs (RTP, AAL2) - Iu may be based on different transport technologies.

MAP is operated between HSS and MSC server and GMSC server respectively.

5.1.3 What is the Border of the Network

Open Issue: What is the Border of the Network

In the case that the GGSN is seen as the border of the network towards the IP network or the GGSN+MGW is seen as the network border towards the PSTN/Legacy network, then the following issue need to be clarified: how to determine the MGW and how to ensure the most optimal routing towards this MGW. On call setup, the CSCF needs to determine the appropriate MGW for the call. For example, it needs to determine if the call is to the PSTN (and in this case towards which PSTN network) or to a Voice Over IP network. This determination can only be ensured when call set-up signalling has been analysed by the CSCF and possibly by the SCF. This analysis may change the called party number (for instance modify the called party address from the address corresponding to an IP terminal to an address corresponding to a foreign PSTN terminal). It is only at that point that the best MGW can be determined. This determination cannot be done before by the SGSN. The MGW address is sent back (through H.225 signalling) to the UE. The UE can then activate a PDP context (for the support of user plane traffic) towards the appropriate network (i.e. the network that best allows to reach the MGW to be used by the call).

Note [FFS]. The issue of optimal routing when an MRF is added cannot be determined until there is agreement on what is the border of the network.

5.2 New Functional Elements

5.2.1 Call State Control Function (CSCF)

In the following section, CSCF has been divided into several logical components.

Currently, these logical components are internal to the CSCF. The need for external components to be able to address directly one of the logical components of the CSCF is for FFS.

Every CSCF acting as a Serving CSCF (see section 9) has a CCF function.

ICGW (Incoming call gateway)

♦ Acts as a first entry point and performs routing of incoming calls,

♦ Incoming call service triggering(e.g. call screening/call forwarding unconditional) may need to reside for optimisation purposes,

♦ Query Address Handling (implies administrative dependency with other entities)

♦ Communicates with HSS

CCF (Call Control Function)

♦ Call set-up/termination and state/event management

♦ Interact with MRF in order to support multi-party and other services

♦ Reports call events for billing, auditing, intercept or other purpose

♦ Receives and process application level registration

♦ Query Address Handling (implies administrative dependency)

♦ May provide service trigger mechanisms (service capabilities features) towards Application & services network (VHE/OSA)

♦ May invoke location based services relevant to the serving network

♦ May check whether the requested outgoing communication is allowed given the current subscription.

SPD (Serving Profile Database)

♦ Interacts with HSS in the home domain to receive profile information for the R00 all-IP network user and may store them depending on the SLA with the home domain

♦ Notifies the home domain of initial user's access (includes e.g. CSCF signalling transport address, user ID etc. needs further study)

♦ May cache access related information (e.g. terminal IP address(es) where the user may be reached etc.)

AH (Address Handling)

♦ analysis, translation, modification if required, address portability, mapping of alias addresses

♦ May do temporary address handling for inter-network routing.

Other functions such as admission control, multiple session knowledge within one CSCF, multiple CSCFs serving one terminal for multiple services, role of multiple CSCFs serving a network etc. need further investigation.

Interfaces need to be further studied and defined.

5.2.2 Home Subscriber Server (HSS)

The **Home Subscriber Server** (HSS) is the master database for a given user. It is responsible for keeping a master list of features and services (either directly or via servers) associated with a user, and for tracking of location of and means of access for its users. It provides user profile information, either directly or via servers. It is a superset of the Home Location Register (HLR) functionality., for example as defined in GSM MAP, but differs in that it needs to also communicate via new IP based interfaces. The HSS shall support a subscription profile which identifies for a given user for example:

♦ user identities

♦ subscribed services and profiles

♦ service specific information

♦ mobility management information

♦ authorization information

Like the HLR, the HSS contains or has access to the authentication centers/servers (e.g. AUC, AAA).
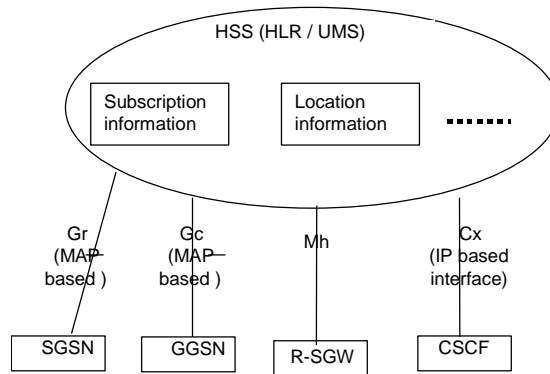
Figure 5-3: Example of a Generic HSS structure and basic interfaces
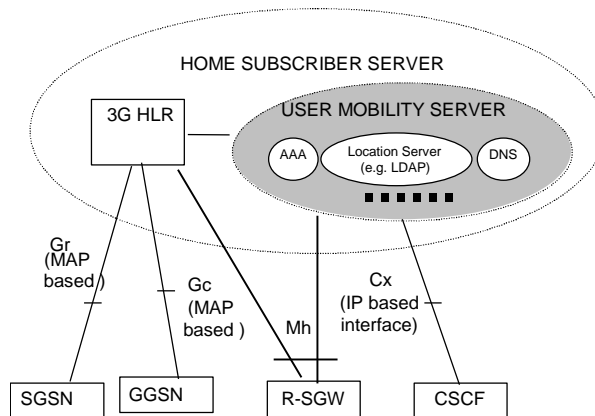
Figure 5-4: Example of HSS structure with UMS Specific Functionality

The HSS may consist of the following elements as shown in the Figure 3:

1) User Mobility Server (UMS): it stores the Release 2000 all IP network Service Profile(see section 9.1) and stores Service Mobility or Serving CSCF related information for the users. UMS might also generate, store and/or manage security data and policies (e.g. IETF features). UMS should provide logical name to transport address translation in order to provide answer to DNS queries. UMS role and functional decomposition are for further study.

2) 3G HLR: A GPRS HLR enhanced to support Release 2000 all IP networks GPRS specific information.

Gr and Gc use MAP which may be implemented using MAP transported over IP, however the issue of roaming to a network that supports MAP over SS7 needs to be considered. The Cx interface requires further study: it may be implemented via IETF protocols such as DNS or via MAP procedures.

Following functionality may need to be supported in the HSS and are for further study:

♦ it stores the R00 all- IP network Service Profile and stores location information for the users.

♦ it may also generate, store and/or manage security data and policies (e.g. IETF features).

♦ may need to provide logical name to transport address translation.

♦ The HSS interacts with the R-SGW to communicate with VLRs and other Mobility managers which do not use IP. HSS interfaces with CSCF via Cx which is for further study.

♦ Other R00 all-IP based network functions such as AAA, DNS etc. and their interactions with HSS is for further study.

♦ Interface(s) between UMS and 3G HLR is for further study.

note: If the user profile is split across different databases then there should either be no duplication of information elements or the consistency of the data should be maintained.

5.2.3 Transport Signalling Gateway Function (T-SWG)

This is component in the R00 all-IP network is PSTN/PLMN termination point for a defined network. The functionality defined within T-SGW should be consistent with existing/ongoing industry protocols/interfaces that will satisfy the requirements.

- Maps call related signalling from/to PSTN/PLMN on an IP bearer and sends it to/from the MGCF.
- Needs to provide PSTN/PLMN <-> IP transport level address mapping.

Interfaces need to be further studied and defined.

5.2.4 Roaming Signalling Gateway Function (R-SGW)

The role of the R-SGW described in the following bullets is related only to roaming to/from 2G/R99 CS and GPRS domain to/from R00 CS and GPRS domain and is not involving the multimedia/VoIP domain.

- In order to ensure proper roaming, the R-SGW performs the signaling conversion at transport level (conversion: Sigtran SCTP/IP versus SS7 MTP) between the legacy SS7 based transport of signaling and the IP based transport of signaling. The R-SGW does not interpret the MAP / CAP messages but may have to interpret the underlying SCCP layer to ensure proper routing of the signaling.

- (For the support of 2G / R99 CS terminals): The services of the R_SGW are used to ensure transport interworking between the SS7 and the IP transport of MAP_E and MAP_G signalling interfaces with a 2G / R99 MSC/VLR

For the Multimedia/VoIP domain, MAP interworking at the R-SGW is for Further Study.

5.2.5 Composite Gateway

Composite gateway: A logical entity composed of a single MGC and one or more MGs that may be reside on different machines. Together, they preserve the behaviour of a gateway as defined in H.323 and H.246 (this may include SIP servers and MSC servers in release 2000).

5.2.6 Media Gateway Control Function (MGCF)

This component in the R00 all-IP network is PSTN/PLMN termination point for a defined network. The

functionality defined within MGCF should be consistent with existing/ongoing industry protocols/interfaces that will satisfy the requirements.

♦ Controls the parts of the call state that pertain to connection control for media channels in a MGW.

♦ Communicates with CSCF.

♦ MGCF selects the CSCF depeneding on the routing number for incoming calls from legacy networks.

♦ Performs protocol conversion between the Legacy (e.g. ISUP, R1/R2 etc.) and the R00 all-IP network call control protocols (this is still under further study within the industry).

♦ Out of band information assumed to be received in MGCF and may be forwarded to CSCF/MGW.

Interfaces need to be further studied and defined.

5.2.7 Media Gateway Function (MGW)

This component in the R00 all-IP network is PSTN/PLMN transport termination point for a defined network. For the architecture option 2, the component is also used for interfacing UTRAN with the All IP core network over Iu.

The functionality defined within MGW should be consistent with existing/ongoing industry protocols/interfaces that will satisfy the requirements.

A MGW may terminate bearer channels from a switched circuit network (i.e., DSOs) and media streams from a packet network (e.g., RTP streams in an IP network). Over Iu MGW may support media conversion, bearer control and payload processing (e.g. codec, echo canceller, conference bridge) for support of different Iu options for CS services: AAL2/ATM based as well as RTP/UDP/IP based. *[Note: in the general R'00 architecture different core network transport technologies shall be possible for example: ATM, STM or IP.]*

♦ Interacts with MGCF, MSC server and GMSC server for resource control.

♦ Owns and handles resources such as echo cancellers etc.

♦ May need to have codecs.

In band signalling impacts to MGW and R00 all-IP network is for further study.

Functionality on the delivery of ring tone towards PSTN/ PLMN are for further study.

The MGW will be provisioned with the necessary resources for supporting UMTS/GSM transport media. Further tailoring (i.e packages) of the H.248 may be required to support additional codecs and framing protocols, etc.

For architecture option 2, the MGW bearer control and payload processing capabilities will also need to support mobile specific functions such as SRNS relocation/handover and anchoring (note that these functions are provided by the SGSN/GGSN in architecture option1 and are not required in the MGW). It is expected that current H.248 standard mechanisms can be applied to enable this. Solutions of how to use the H.248 generic bearer control mechanisms for mobile specific functions needs further studies.

Interfaces need to be further studied and defined.

5.2.8 Multimedia Resource Function (MRF)

♦ This component performs multiparty call and multi media conferencing functions. MRF would have the same functions of an MCU in an H.323 network.

♦ Responsible for bearer control (with 3G-GGSN and MGW) in case of multi party/multi media conference

♦ May communicate with CSCF for service validation for multiparty/multimedia sessions.

Handling of resources such as two stage dialling, announcements etc. are for further study.

Interfaces need to be further studied and defined.

5.2.9 MSC Server

MSC server mainly comprises the call control and mobility control parts of a GSM/UMTS R99 MSC.

The MSC Server is responsible for the control of mobile originated and mobile terminated 04.08CC CS Domain calls. It terminates the user-network signalling (04.08+ CC+MM) and translates it into the relevant network – network signalling. The MSC Server also contains a VLR to hold the mobile subscriber's service data and CAMEL related data.

MSC server controls the parts of the call state that pertain to connection control for media channels in a MGW.

5.2.10 Gateway MSC Server

The GMSC server mainly comprises the call control and mobility control parts of a GSM/UMTS R99 GMSC.

5.3 Description of Reference Points

5.3.1 Cx Reference Point (HSS – CSCF)

This reference point supports the transfer of data between the HSS and the CSCF.

When a UE has registered with a CSCF, the CSCF can update its location towards HSS. This will allow the HSS to determine which CSCF to direct incoming calls to. On this update towards the HSS, the HSS sends the subscriber data (application related) to CSCF.

For a MT call, CSCF asks the HSS for call routing information.

5.3.2 Gm Reference Point (CSCF – UE)

This interface is to allow UE to communicate with the CSCF e.g.

- register with a CSCF,

- Call origination and termination

- Supplementary services control.

5.3.43 Mc reference point (MGCF – MGW)

The Mc reference point describes the interfaces between the MGCF and MGW, between the MSC Server and MGW, and between the GMSC Server and MGW. It has the following properties:

- full compliance with the H.248 standard, baseline work of which is currently carried out in ITU-T Study Group 16, in conjunction with IETF MEGACO WG.

- flexible connection handling which allows support of different call models and different media processing purposes not restricted to H.323 usage.

- open architecture where extensions/Packages definition work on the interface may be carried out.

- dynamic sharing of MGW physical node resources. A physical MGW can be partitioned into logically separate virtual MGWs/domains consisting of a set of statically allocated Terminations.

- dynamic sharing of transmission resources between the domains as the MGW controls bearers and manage resources according to the H.248 protocols.

For architecture option 2, the functionality across the Mc reference point will need to support mobile specific functions such as SRNS relocation/handover and anchoring. It is expected that current H.248/IETF Megaco standard mechanisms can be applied to enable this. Solutions of how to use the H.248 generic bearer control mechanisms for mobile specific functions needs further studies.

5.3.4   Mh Reference Point (HSS – R-SGW)

This interface supports the exchange of mobility management and subscription data information between HSS and R99/legacy mobile networks. This is required to support All IP users who are roaming in a 2G network.

5.3.5   Mm reference Point (CSCF – Multimedia IP networks)

This is an IP interface between CSCF and IP networks. This interface is used, for example, to receive a call request from another VoIP call control server or terminal.

5.3.6   Mr Reference Point (CSCF - MRF)

Allows the CSCF to control the resources within the MRF

5.3.7   Ms reference Point (CSCF – R-SGW)

This interface allows CSCF to contact legacy network elements, e.g. 2G HLR, for location management (location update and subscriber data download), and call control (eg 2G HLR enquires for routing number (RN) for a roaming 2G user).

5.3.8   Mw Reference Point (CSCF – CSCF)

The interface allows one CSCF (e.g. home CSCF) to relay the call request to another CSCF (eg serving CSCF).

5.3.9   Nc Reference Points (MSC Server – GMSC Server)

Over the Nc reference point the Network-Network based call control is performed. Examples of this are ISUP or an evolvement of ISUP for bearer independent call control (BICC). In the all IP core network Nc reference point uses an IP based signalling transport. *[Note: in the general R'00 architecture different options for signalling transport on Nc shall be possible.]*

5.3.10   Nb Reference points (MGW-MGW)

Over the Nb reference point the bearer control and transport are performed. The transport may be RTP/UDP/IP or AAL2 for transport of user data. The bearer control over Nb is FFS, it may be based on RTP, H.245 or corresponding. *[Note: in the general R'00 architecture different options for user data transport and bearer control shall be possible on Nb, for example: AAL2/Q.AAL2, STM/none, RTP/H.245.]*

5.3.11   SGSN to Applications and Services

The interface from the SGSN to the SCP in the Applications and services domain is the interface defined for GPRS to support Charging Application Interworking.

5.4   Usage of MAP/CAP - Protocol stack below MAP / CAP – General considerations

Below MAP and CAP, the protocol stack within the All IP CN is as shown in Figure 5-5:

-    SCCP and TCAP are used below CAP and MAP. Indeed CAP and MAP both rely on services provided by these underlying protocols (e.g., transaction capabilities, global title translation). Alternatives to providing the services of SCCP is for further study.

-    The lower transmission layers are compliant with the IETF Sigtran protocol suite used to carry telecommunication signalling on top of an IP backbone.
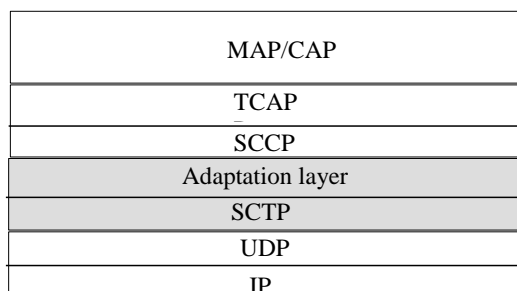
MAP/CAP
TCAP
SCCP
Adaptation layer
SCTP
UDP
IP

Figure 5-5: All IP R00 protocol stack for MAP/CAP

This protocol stack is used to carry MAP/CAP flows:

•   inside the All IP CN, between nodes terminating the MAP/CAP: e.g. between HSS or SCP and the All IP CN functions (SGSN, MSC servers …) handling Call / Session and needing to dialog with the HSS or SCP for user mobility management / subscriber data retrieval. This is for example the case of the Gr, Gc interfaces.
It may also be envisaged (although this requires further study) to use MAP on Cx for user mobility management / subscriber data retrieval.

•   in case of interworking with nodes not supporting this MAP/CAP over IP stack (e.g. 2G or UMTS R99 networks nodes) but needing a MAP / CAP dialog with a node supporting this MAP/CAP over IP stack. This protocol stack is used between the nodes terminating MAP/CAP and the R-SGW. The Mh interface shall use MAP over IP, for the roaming scenarios involving the R00 CS domain and GPRS (excluding the VoIP/Multimedia domain), hence this does not exclude other protocols being used. The use of MAP on the Ms is FFS.

6   QoS

The work currently being done within the S2 QoS Ad Hoc is reflected within TR 23.907 and the QoS section of TR 22.105. The R99 version of these specifications will support real time applications on a packet switched network which includes the ability of UMTS to transparently support multi-media applications that utilize the H.323 protocol. The all-IP architecture as described in this document defines the implementation of a call control function which can be based on either SIP or H.323 within the PLMN. Therefore, the R00 QoS work will include any changes required to support QoS

capabilities necessary for support of multi-media based on H.323 or SIP within the PLMN. Doing this is not expected to introduce any new QoS requirements at the UMTS bearer level.

In additon, please note that the desire also exists to have the all-IP architecture support multi-media applications that utilize the SIP protocol. However, QoS work related to the SIP protocol would only be undertaken within 3GPP when 3GPP itself undertakes the work to support the SIP protocol. However, we do anticipate that the QoS work centered on H.323 is directly applicable to SIP.

In addition, since the work on the all IP network includes EDGE support as identified by the ERAN architecture, the QoS work within the ERAN needs to be in alignment with QoS support within UMTS.

A preliminary review of the current version of TR 23.907 leads us to believe that it is largely sufficient to meet the objectives an all IP network. The review includes the following observations :

- TR 23.907 includes the specification of a QoS conversational class which includes voice. TR 23.907 identifies the fundamental characteristics of this class as:

- Preserve time relation (variation) between information entities of the stream.

- Conversational pattern (stringent and low delay)

- These characteristics apply whether voice is carried within a circuit or as packets. So there should be no need to modify the QoS classes currently defined.

- Implementing the H.323 call model within the PLMN is not expected to affect the R99 TR 23.907 identified QoS technical requirements, the overall architecture, nor the functions identified therein. However, a brief study will be necessary to verify this.

- The Radio Access Bearer Service attributes currently defined will need to be reviewed in light of an all IP network but minimal additions in this area would be expected for UMTS. However, for EDGE, work should be anticipated. Obviously the mapping from bearer to radio bearer is also affected.

- The issue of interworking the packet voice capable GPRS with other networks needs to be studied at least as it pertains to an acceptable voice delay budget.

7 Handover

Within this section, the topics to be studied and standardised to support handover for real time services in the PS domain have been identified. This section has investigated various handover scenarios, however the fact that the scenario has been studied here does NOT imply a requirement for the support of that scenario. The requirements for handover relating to the All IP architecture of R00 in UMTS will be determined by S1 as part of the R00 Service Requirement specification work.

7.1 SRNC Relocation within a UMTS R00 IP network

Within UMTS, work has already been undertaken to provide handover for real time PS domain services. The UTRAN does not distinguish between circuit and packet services, it simply provides for real and non-real time services, hence Intra RAN handover for real time services is available.

7.1.1 Support Required within ERAN

The goal of the All IP architecture is to provide a common core network for both UTRAN and ERAN. The specification of this work is outside the scope of this study, however, it is worth noting that the ERAN will need to support the following procedures:

- SRNC Relocation

- Mobile Assisted Network Controlled handover for the real time packet services.

7.1.2   All IP UTRAN to All IP ERAN Handover

The need to support this handover scenario is for FFS.

In this scenario, a CSCF will support terminals in both the ERAN and the UTRAN. The terminal will have access to the same Media Gateway from both RANs, hence the same media codec will be used in the network.

7.2   SRNC Relocation/Handover Between All IP and CS Domain/GSM

7.2.1   Requirement

The need to support these handover scenarios is for FFS.

The expected scenarios:

- Inter system handover, where target system does not support the necessary RT requirements for its packet domain (e.g. Inter system hand-over towards R97)

To fulfill this potential requirement, 2 solutions have been currently proposed (other may be studied). Any solution would face the following issues:

1. The MS has one or more PDP context for the signalling and the traffic. As the MS after the HO is handled by the CS domain, these PDP contexts need to be switched out? How to signal to the MS that its traffic is now handled by the CS domain? There is no H323 (H225 / H245) that could be used for such purpose?

2. Multimedia CC messages sizes may be larger than supported in the CS domain. The feasibility of transfering the Multi-media protocol messages on top of GSM CS signalling radio and A interface connections as well as on the MAP E interface needs to be investigated.

3. How does the SGSN determine that sessions involve the CSCF?

7.2.2   Solution with CSCF supporting MAP E

The following text considers the scenario when a UE has at least one session active which involves the CSCF.

On receipt of an SRNC relocation required message, the SGSN determines that the SRNC relocation results in a change of SGSN to one, which does not support the All IP services. One option is to force the serving RNS to hold the sessions until those involving the CSCF have been torn down, alternatively the SGSN needs to transfer the sessions not using the CSCF to the SGSN and the CSCF involved sessions to a 3G-MSC.

The signaling described below is based on the procedures for SRNC Relocation in 23.121. It shows that the use of the anchor MSC concept could be applied in order top maintain the Multimedia CC signaling to be tunneled back to the CSCF which acts at the "Anchor MSC". However, issues still arise as to how to remove the PDP context to terminate it within the network.

1. On receipt of "SRNC relocation Required Message" SGSN checks for

   - Do any sessions involve the CSCF? If so,

   - is the target SGSN an E-GSN?

2. SGSN signals to the CSCF that a handover to an MSC is required. I.e. sends "Forward SRNC Relocation" message

3. CSCF signals "Prepare SRNC Relocation" to Non-anchor MSC including the information received from the Source RNC.

4. Non-anchor MSC starts the Relocation process, treating the CSCF as the Anchor MSC. This allows the Multi-media client CC messages to be tunneled through the Non-anchor MSC to the CSCF.

5. The CSCF instructs the GW to prepare to transfer the traffic between the PSTN and the Non-Anchor MSC. I.e. to take the GGSN out of the path.

6. Successful switch of the bearers at the RNC, takes the SGSN and GGSN out of the path. The GTP tunnels are then released.

Open Issues:

1. How does the SGSN determine which sessions involve the CSCF?

2. How does the SGSN inform the CSCF to request a connection from the MSC via a MAP-E interface? In the case of the CSCF being in an external network, it may not be possible for the SGSN to know the CSCF address.

3. For a Mobile to PSTN call, the CSCF will need to signal to the GW to route PSTN traffic to the 3G-MSC

7.2.3   Inter-System handover using the ISHF

The mechanism described in this section, identifies a new functional element, the ISHF. This isolates MAP/E from the CSCF. Further work is required to identify if this approach, or the approach of supporting MAP/E on the CSCF (see section 7.2.2) should be adopted.

7.2.3.1   General

Based on the handover requirements given in Table 4-1, the following intersystem handover scenarios should be accommodated by the All IP architecture.

- UMTS R 00 IP network to/from 2G GSM network handover

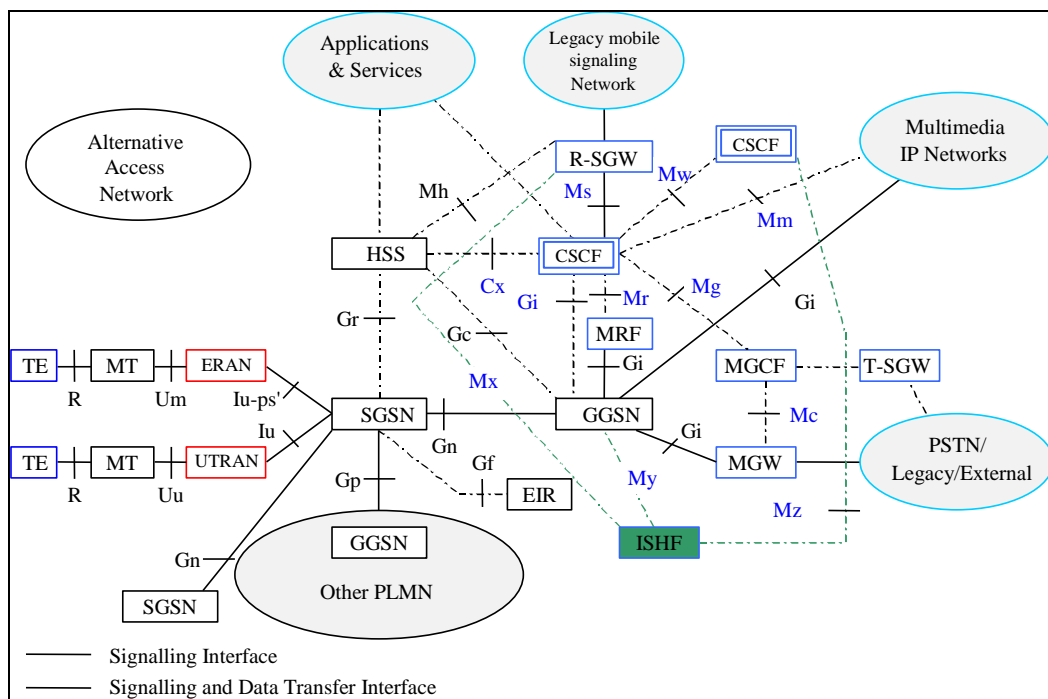These procedures listed shall not require change to the terminal.

Figure 7-1: Support of InterSystem Handover

To support intersystem handover it is proposed that a new function is added to the architecture, called the InterSystem Handover Function (ISHF), see Figure 7-1.

The changes/additions to the baseline architecture given in Figure 7-1 include:

1. **ISHF** is the Inter-System Handover Function for handover between the UMTS R 00 IP networks and UMTS (PS, CS) networks and between UMTS R 00 IP networks and legacy networks. The ISHF is responsible for the handoff signaling procedures to another core network in addition to the establishment of a bearer connection between the source and target networks.

2. **Mx** is the interface between ISHF and the R-SGW. This interface is MAP/E and is used to signal handoff messaging between networks.

3. **My** is the interface between the ISHF and the GGSN. This interface relays handover related Iu signaling between the UTRAN and the ISHF. Note this is interface is tunneled through the SGSN.

4. **Mz** is the interface between the ISHF and the CSCF. This interface is used setup bearer resources between the source and target networks for inter-system handover.

It is assumed that the R-SGW function will interwork the UMTS R 00 IP handover procedures to the handover procedures of the source or destination network, that is the ISHF resides within the R-SGW. It may be desirable to create a separate function that performs interworking between core network protocols, this is for FFS.

7.2.3.2 UMTS R 00 IP network to/from 2G network handover

This example shows how handover (Hard Handover) is performed from UMTS R 00 IP network to a legacy GSM network. This demonstrates the signaling required between the networks and assumes a trunk circuit bearer between the networks. Other bearer connection schemes are possible, but not addressed in this example. (Note that all the air interface messages are not shown for clarity in the diagram. In addition, the MGCF and T-SGW are shown as a combined node and the messaging between these functions are omitted for clarity.)
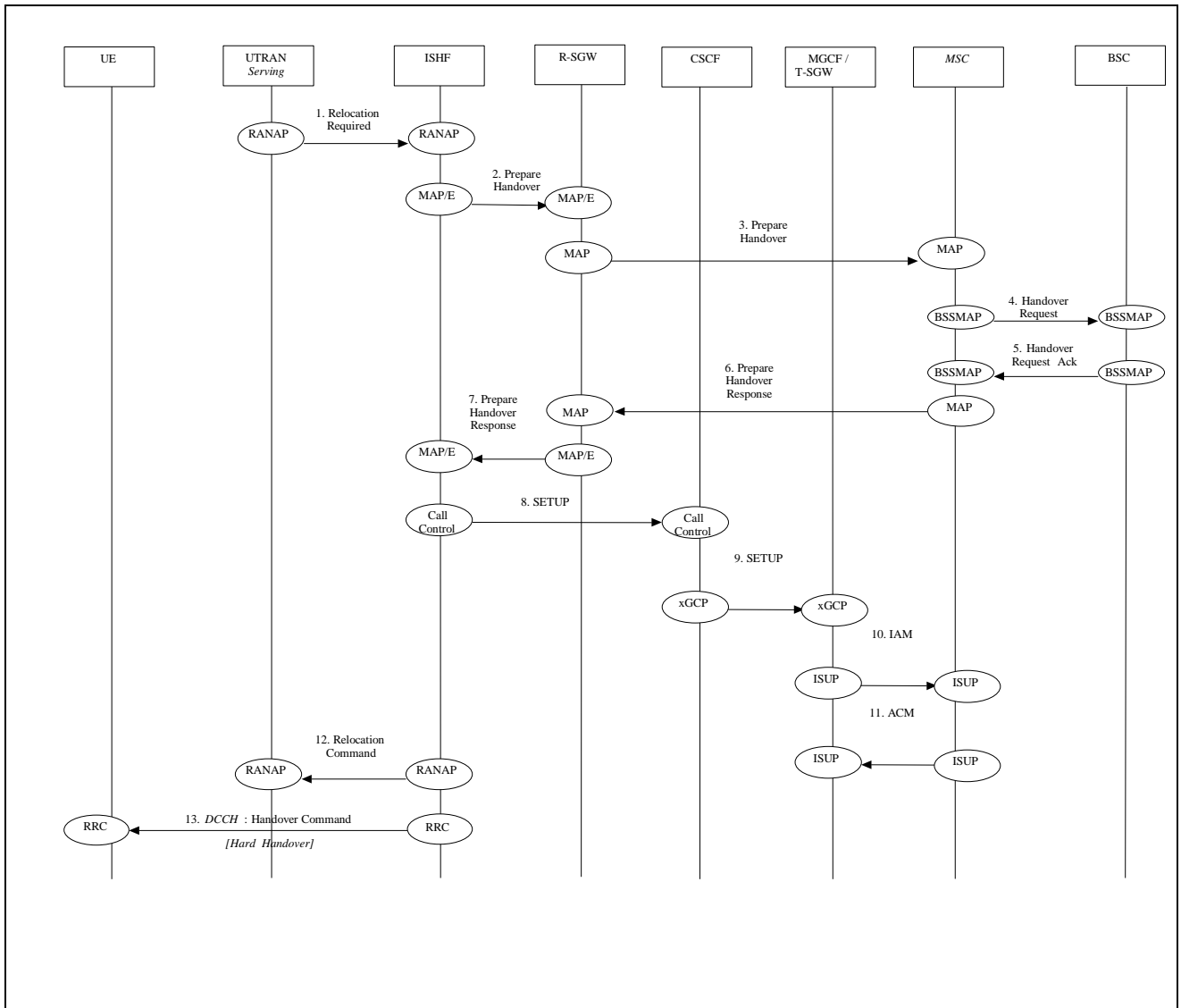
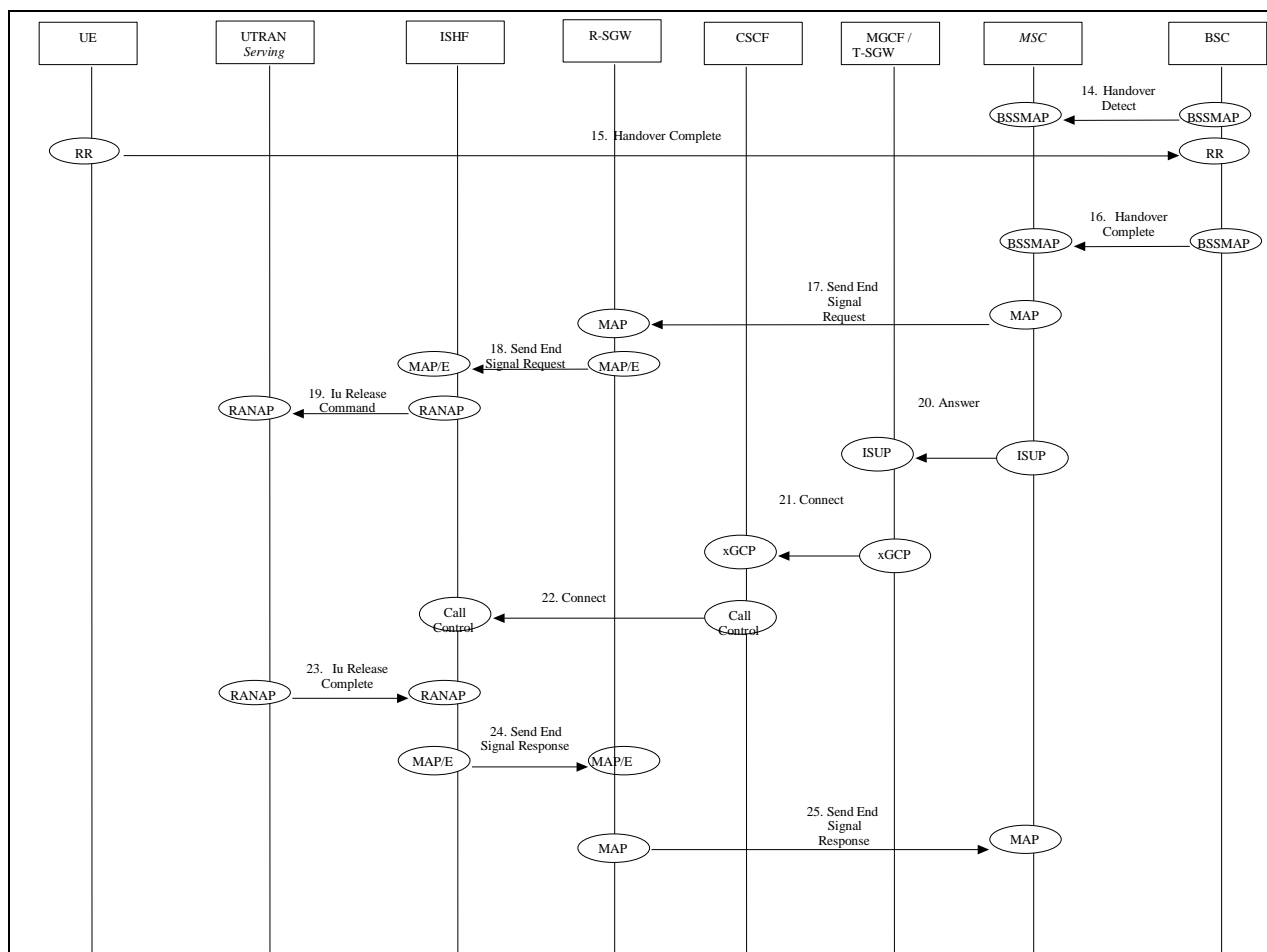**Figure 7-2: UMTS R 00 IP to GSM handover**

Figure 7-3: UMTS R 00 IP to GSM handover (continued)

UMTS R 00 IP ⇒ GSM handover

1. Upon detection of a trigger SRNC sends RANAP message **Relocation Required** to the ISHF.

2. The ISHF will send the MAP/E **Prepare Handover** to the R-SGW.

3. The R-SGW Interworks (if required) the **Prepare Handover** to the appropriate network protocol (in this case GSM MAP) and sends the message to the other network (MSC)

Note: Steps 4&5 follow the normal GSM procedures and are shown only for clarity.

6. Once initial procedures are complete in GSM MSC/BSS the MSC returns MAP message **Prepare Handover Response** to the R-SGW.

7. The R-SGW converts (if required) to the MAP/E protocol and sends the resulting **Prepare Handover Response** message to the ISHF.

8. The ISHF initiates procedures to establish bearer resources between the networks. In this case a trunk circuit is established. The ISH sends a **CONNECT** message to the CSCF to initiate a call to set up the bearer.

9. The CSCF sends a **CONNECT** to the circuit gateway (MGCF, T-SGW shown combined for simplicity) to establish an outgoing call to the MSC.

10. The circuit gateway sends the ISUP **IAM** message to the MSC.

11. The MSC responds with the ACM message.

12. ISHF responds to the initial request from SRNC by sending RANAP message **Relocation Command** to the SRNC.

13. Via existing RRC connection, SRNC sends RRC message **Handover Command** (Hard Handover) to the UE.

Parameters: Handover type.

Note: Procedures related to synchronization etc. to GSM BSS are not shown.

Note: Step 14-16 follow normal GSM procedures and are shown only for clarity.

17. Detection of the UE within the GSM coverage results in the MSC sending MAP message **Send End Signal Request** to the UMTS R 00 IP network (R-SGW)

18. The R-SGW forwards the **Send End Signal Request** to the ISHF.

19. ISHF initiates release of resources allocated by the former SRNC (**Iu Release Command**).

20. **ISUP Answer** is sent from the MSC to the Circuit Gateway.

21. **Connect** is returned to the CSCF function

22. **Connect** is relayed back to the ISHF.

23. Previously allocated bearer resources are released within UMTS (e.g. using RANAP and ALCAP protocols [ALCAP not shown]) (**Iu Release Complete**).

24. Procedure is concluded from UMTS point of view by ISHF sending MAP/E message **Send End Signal Response** (this message is not sent until the end of the call).

25. The R-SGW will send the MAP **Send End Signal Response** to the MSC.

7.3 Areas for Further Study

The following areas may require further study.

- Bearer set-up/control between networks during handover

- Anchoring bearer in the UMTS R 00 IP network

- MAHO support

- Inter-RNC Soft handover

- Inter RAN to RAN of same type streamlining

- Inter RAN to RAN of different type streamlining

8 Radio Aspects

Note: This section requires support from the RAN group.

8.1 General

1) CN – RAN interface definition

   (a) Functional split between CN and RAN- new radio access network called EGPRS Radio Access Network (ERAN) is considered. The interface between the radio access network such as ERAN or UTRAN and the CN needs to be defined/extended and should allow different air

interface technologies to access to the CN. The detailed functional split between CN and RAN needs further investigation.

    (b) Impact on existing GPRS/EGPRS/UTRAN implementations and deployments

    (c) Migration scenarios-2G to release 99 (BSS is not IP-based) to release 2000 (all IP-based network)

    (d) Protocol stack evaluation (including evaluation of control and user planes from CN to both RAN and MS)

2) ERAN architecture (Refer to SMG2)

    (a) ERAN reference model-network entities, protocol stacks and logical or functional elements

    (b) Functional split between elements

    (c) Definition of interaction between elements

    (d) Impact on existing GPRS/EGPRS deployments (BSC, PCU, and BTS) and mitigation strategies

3) UTRAN architecture extensions

    (a) Identification of required extensions

4) Realtime Handover for Packet Domain

    (a) ERAN issues (Refer to SMG2)

    (b) UTRAN issues

5) QoS support

    (a) Evaluation of S2 QoS Ad Hoc progress for real-time data support

    (b) Signalling mechanism

    (c) CN issues

    (d) Alignment of GPRS with UMTS QoS

    (e) Realization of QoS on radio link

6) (E)GPRS Radio issues (Refer to SMG2)

(a) Real-time support including handover and QoS

(b) Spectrum efficiency/performance (e.g. statistical multiplexing and source/channel coding)

(c) RLC/MAC enhancements

(d) The effect of various deployment scenarios (e.g. spectrum availability) and traffic mix, such as voice and data, on spectrum efficiency should be considered.

7) UTRA Radio issues

    (a) Real-time packet data support including handover and QoS – validation and possible enhancement

    (b) Radio efficiency/performance (e.g. statistical multiplexing and source/channel coding)

    (c) RLC/MAC enhancements if needed

    (d) The effect of various deployment scenarios (e.g. spectrum availability) and traffic mix, such as voice and data, on spectrum efficiency should be considered.

Note: The RLC/MAC line items about (sections 6(c) and 7(c)) may include

- Enhanced for radio resource allocation.
- Radio access bearer definitions (i.e., define for the various traffic classes the path through the protocol stack and the bearer to be used).
- Flow classification (e.g. mapping of user traffic onto appropriate radio access bearer)
- Enhancement of EGPRS protocol to support real time service and QoS management (e.g. Fast channel allocation schemes).

8.2   Airlink Optimisation for Real-Time IP

8.2.1   Introduction

In the all-IP architecture, a fundamental objective is to support IP-based real-time and non real-time traffic for a mobile terminal while achieving spectral efficiency and error robustness. In the case of real-time voice, spectral efficiency and error robustness have a performance baseline coming from the current cellular systems. There is also a baseline in voice quality. It is natural to expect that the all-IP architecture has to meet this existing baseline for voice services. The question is then how to meet the objectives of spectral efficiency and error robustness and the existing baseline for real-time voice when the all-IP paradigm is applied to cellular systems.

For IP-based real-time multimedia, RTP protocol is predominantly used on top of UDP/ IP. The size of the combined IP/UDP/RTP headers is at least 40 bytes for IPv4 and at least 60 bytes for IPv6, while the voice payload is short, typically shorter than the IP/UDP/RTP header. Clearly, if the headers were sent "as is" over the air interface to conform to the pure IP paradigm, it is not possible to meet or even get close to the baseline spectral efficiency of existing circuit voice. Some header adaptation technique is required, whereby a transformation is applied to the IP/UDP/RTP headers to reduce their size before transmission on the air interface, and the reverse transformation applied after crossing the air interface, to restore the original header size and values. Reduction of the header size is done by removing redundancy in the originally coded header information and/or removing header field information and thereby losing functionality. Impact on transparency and robustness to errors have to be fully understood in order to design the appropriate adaptation techniques (Transparency for a given header field is defined as the property whereby the value after transformation/reverse transformation is the same as in the original header).

This section explores the range of possible adaptation techniques and proposes two adaptation techniques, header stripping and header compression. The two techniques must be further studied as regards error robustness, voice quality and IP transparency.

8.2.2   user plane adaptation

In the following we refer to the functionality that does transformation/reverse transformation as User Plane Adaptation (UPA), and explore the range of possible adaptation techniques, along with their pros and cons.

8.2.2.1   Full opacity (no adaptation)

The UPA has no knowledge of the internal structure of the headers or payload, and no transformation is done on the IP/UDP/RTP headers which are sent in full over the air interface. Error protection is applied evenly to all the bits in the header, and evenly to all the bits in the payload. The header part will likely require stronger error protection than the payload, since a header loss will require to discard the corresponding packet, and no error concealment or mitigation can be applied to the header. This technique achieves full transparency, which allows to support protocols such as IPSEC on an end-to-end basis. An obvious con is the high overhead caused by the headers, which results in very poor spectrum efficiency.

8.2.2.2   Payload opacity (header adaptation only)

In this case, the UPA only needs to know the internal structure of the IP/UDP/RTP header but not of the payload. Only the headers are adapted, either by header compression or header stripping.

8.2.2.2.1   Header compression/decompression

IP/UDP/RTP headers are compressed before transmission over air interface and decompressed at the receiving end. Like before, headers require stronger error protection than payload. The most wellknown header compression algorithm is the Van Jacobson algorithm (RFC 1144, Compressing TCP/IP Headers for low speed serial links). In general compressed headers are more vulnerable to errors than uncompressed headers. The current standardised algorithms has therefore proven to be less efficient over lossy links such as a radio interface.

The benefit of compressing the IP/UDP/RTP header is nevertheless obvious as it significantly reduces the required overhead per packet.For an efficient header compression scheme, the IP/UDP/RTP headers can be compressed down to 2 bytes.

New header compression schemes, adapted to cellular radio link reliability characteristics, will be developed in the future. Such schemes adapted to radio environment may be able to compress the IP/UDP/RTP headers down to 2 bytes. One example among others would be a scheme currently being proposed for development in the IETF in which the compressed header carry a checksum computed over the header before compression. This provides a reliable way to detect and repair errors and increases error robustness.

A general drawback with most header compression schemes are incompatibility with end-to-end security (IPSEC) and bandwidth management, since compressed headers have variable size.

Requirements and evaluation criteria for header compression schemes to be used over the radio interface is summarised in the table 8-1.

Figure 8-1 shows a conceptual diagram of header compression used in conjunction with the lower layers in cellular. Voice is used as an example. The lower layers may perform interleaving and channel coding. For simplicity, the effect of interleaving and channel coding on the bit stream transmitted over the air interface is not shown. The effect of possible link level multiplexing with other traffic streams is not shown either. There is an MS-based UPA point and a network-based UPA point. The MS-based UPA acts as header compressor and header decompressor for the uplink and downlink respectively, while the network-based UPA acts as header decompressor and header compressor for the uplink and downlink respectively.

The requirements for header compression are described in the table below. In each case, some justification for the requirement is also provided.

#	Focus Area	Requirement	Justification
1	**Performance / Spectral Efficiency**	Must provide low relative overhead (as defined in [1]) under expected operating conditions	In general, a primary goal is high spectral efficiency. Reduction of overhead has direct impact.
2	**IPv6 or IPv4**	Must include support for IPv6 and IPv4	Ipv4 and Ipv6 terminals are expected to coexist for some time
3	**Ubiquity**	Must NOT require modifications to existing IP (v4 or v6), UDP, or RTP protocol stack implementations	Enables use of current devices/services which employ generic IP/UDP/RTP stacks.
4	**Cellular HO**	Must support the cellular handoff operation, in an efficient manner; All fields must be transparent to the HO process, i.e, are exactly regenerated subsequent to handoff.	Target application is for adaptation of the user plane on cellular air interfaces; therefore this operation **must** be supported. Efficiency requirement is due to potential impacts on spectral efficiency and voice quality if HO is not properly handled.
5	**Integrity**	*The header compression process must be lossless*	Would like to maintain the end-to-end integrity of IP
6	**Error Propagation**	Error propoagation due to header compresion should be kept to a absolute minimum or avoided if at all possible.; *error propagation* is defined as the loss of packets subsequent to the one where the error actually occurred, even when those subsequent packets contain no errors	Error propagation results in lower spectral efficiency and lower voice quality; this is a serious problem for existing schemes such as [5].
7	**Delay**	Must operate under all expected delay conditions; header compression process must not contribute significantly to system delay budget	The user may be in different types of environments with different characteristics; additional delays will have adverse effects on conversational voice

8	**Packet Loss**	Must operate under all expected packet loss conditions; prefer that header compression efficiency is as independent of packet loss rate as possible	The user may be in different types of environments with different characteristics
9	**Media Supported**	Must function regardless of media type in RTP payload (in general, there is NO *required* knowledge of payload)	The algorithm should be applicable to any type of RTP/UDP/IP data flow; note that this does not preclude optional optimizations for certain media types
10	**Independence with respect to call type**	Must function for mobile-mobile and mobile-landline calls; performance in each case should be comparable to existing cellular (in terms of both quality and spectral efficiency)	Both types of calls will occur in All-IP cellular systems; each is equally important

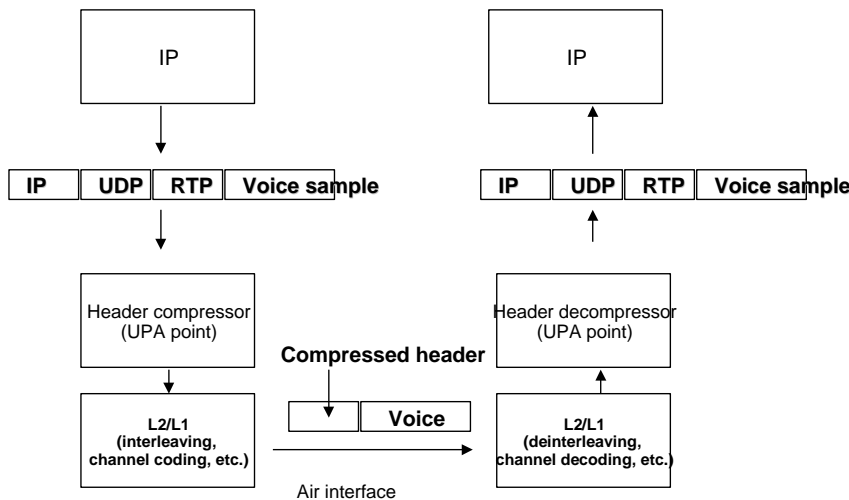**Table 8-1: Requirements for Header Compression**

Figure 8-1: Header compression

8.2.2.2.2 Header stripping/regeneration

IP/UDP/RTP headers are stripped before transmission over air interface and regenerated at the receiving end. Essentially only the payload is transmitted, but some additional header-related information needs to be transmitted to enable the header regeneration. The degree of header transparency achieved is variable, depending on the amount of header-related information that one wants to transmit. No header error protection is needed when the header information is completely removed. However, the necessary information for header regeneration requires a header, at least for some packets. When the payload has constant size, bandwidth management issue is virtually eliminated since the payloads can be carried on a constant bit rate channel. The constant bit rate channel also eliminates QoS (delay and jitter) problems. As before, end-to-end security cannot be accomodated.

Figure 8-2 shows a conceptual diagram of header stripping used in conjunction with the lower layers in cellular. Voice is used as an example. The lower layers may perform interleaving and channel coding. For simplicity, the effect of interleaving and channel coding on the bit stream transmitted over the air interface is not shown. The effect of possible link level multiplexing with other traffic streams is not shown either. There is an MS-based UPA point and a network-based UPA point. The MS-based UPA acts as header stripper and header regenerator for the uplink and downlink

respectively, while the network-based UPA acts as header regenerator and header stripper for the uplink and downlink respectively.
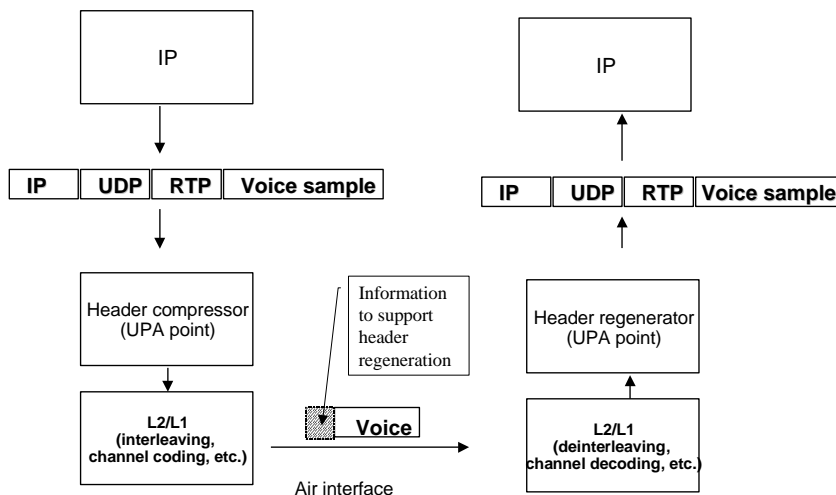

Figure 8-2: Header stripping

8.2.2.3   No opacity (full adaptation)

The UPA knows the structure of the headers and the payload. Headers can be compressed or stripped. In addition, payload transmission is optimised by techniques such as unequal bit protection, channel and error coding optimised for the payload structure, etc.

8.2.3   Application to all-IP network

The all-IP network is expected to provide real-time bearer services intended to carry

- Basic conversational voice (service equivalent to voice in current cellular)

- Real-time Multimedia (includes voice which is seen as a component of multimedia)

8.2.3.1   Basic voice

For basic voice, the emphasis is on meeting and if possible exceeding the baseline of traditional cellular in terms of spectrum efficiency, error robustness and voice quality. Traditional cellular systems achieve that baseline by using well known techniques such as unequal bit protection, channel and error coding optimised for the payload, etc. In addition, speech frames do not incur any header (in the IP sense) overhead. In the all-IP system, we propose to define a "basic voice" bearer tailored for conversational voice and possible other media with the same characteristics.

The basic voice will use payload optimisation and unequal bit protection of the payload similar to traditional cellular. Packing more speech frames into one packet will improve the relative overhead, but at the expense of added delay, which negatively impacts voice quality. Transmission of header-related information and/or compressed header will require strong error protection.

Two options eexist to achieve required characteristics for Basic Voice:

Header Stripping: At a minimum, header stripping for basic voice will have to achieve transparency for the static IP/UDP/RTP fields (those that do not change during the call) and the RTP time stamp and RTP sequence number. This bearer corresponds to the full adaptation case above with header stripping.

Header compression adapted to radio characteristics: By using a robust header compression scheme the overhead per packet is reduced to 2 bytes. Also this case shall correspond to the full adaptation case above with header compression.

Additional optimisation techniques may be contemplated to further improve the spectrum efficiency.

The two options and specific algorithms shall be evaluated according to the criteria of table 8-1 of chapter 8.1.2.1.

8.2.3.2 Real-Time multimedia

Real-time multimedia is a new service that does not exist in traditional 2G cellular systems. A new bearer is proposed. For that bearer, transparency for all the IP/UDP/RTP fields is crucial. Under the transparency constraint, we want to optimise spectrum efficiency and error robustness, but unlike voice, there is no baseline to be used as target. The transparency objective naturally leads to choosing header compression as the user plane adaptation. Payload will have some error protection and compressed header will have even stronger protection. The ability to provide unequal bit protection of the payload also for this service needs to be studied. This bearer corresponds to the header adaptation only case above with header compression. Specific algorithms applied to Real-Time multimedia shall be evaluated according to the criteria of table 8-1 of chapter 8.1.2.1.

8.2.3.3 Pure IP

The Pure IP service can be provided to accommodate end-to-end protocols such as IPSEC. In order to achieve this accomodation, the bearer does not do any adaptation and corresponds to the "No adaptation" case above. Header adaptation may also apply for Pure IP. Specific algorithms for header adaptation shall be evaluated according to the criteria of table 8-1 of chapter 8.1.2.1.

8.2.4 Conclusions

IP/UDP/RTP packets require adaptation to the radio link to meet the spectrum efficiency and error robustness requirements of cellular systems. It shall be investigated if a single scheme can simultaneously and fully meet the above requirements and IP transparency. An alternative to a single scheme is a gradation of schemes tailored to the particular

type of application. Applications may then use different kinds of bearers optimized for their particular current needs **.**

real-time bearers can be categorised in Seen from the header compression point of view basic voice (BV) and real-time multimedia (RTMM). BV bearer is intended to carry voice, as a service equivalent to the one in traditional cellular systems. RTMM bearer is intended to carry generic multimedia traffic, which can include voice. In addition, a pure IP service may be contemplated for support of applications, which require full transparency.

- BV will use header stripping or header compression with unequal bit protection in the payload.

- RTMM will use header compression with equal bit protection in the payload. The possibility to support unequal bit protection in the payload shall be investigated.

- Pure IP service will support data transport without transforming the header. Header compression shall also be possible for Pure IP. Pure IP uses equal bit protection in the payload.

In all cases, header (if present) requires strong error protection.

9 Call Control

9.1 Terminology for Call Control

The terminology in this section is that terminology used that is new or has been changed from that defined for R99. The terminology defined in this section has not been the object of a real debate and hence cannot be considered as agreed. This section needs to be aligned with the terminology used in R99. R99 terminology should be used unless a new or changed concept is introduced. Changes to terminology defined by S1 require S1 agreement.

This section defines a common set of terms on which the present document is based on. The following list of terms is the first attempt to define some terminology.

The terminology defined here have not been matched with the existing 3GPP terminology and this matching will need to be done. Moreover, 3GPP has not defined all the terms that are needed for an all IP based network yet.

1   **Access Profile:** contains subscription profile information relevant to a specific bearer network. As an example, E-GPRS profile plus the radio bearer features (e.g. QoS) to which the user can access is an Access Profile. Access specific roaming information for access to and from legacy networks is a part of the access profile . As an example for CS terminals, Access Profile contains information on the allowed LA, on security data for authentication

2   **Release 2000 all IP networks Service Profile:** contains service subscription data relevant to the Release 2000 all IP network services the user has subscribed to. As an example, Service Profile contains user's identifier, user's aliases, user's temporary location information (e.g. pointer to the current Serving Domain), indication of the multimedia services and capabilities the user has subscribed to, service triggers, status of supplementary services, etc.

3   **User Profile:** is a combination of one or more Access Profiles and zero, one or more Release 2000 all IP networks and Roaming Service Profiles. The User Profile is maintained in the HSS. (FFS)

4   **PDP Flow**: it is a PDP context without the restriction that a different IP address has to be assigned to different PDP contexts. Differentiation between PDP flows is based on protocol type (e.g. TCP, UDP etc.) and port number. (FFS)

5   **Bearer Network**: it is a set of network elements that provides a user with means to connect to a serving/home domain to use services or facilities of the network the user is roaming in and gain access to the home domain or other service networks. Examples of bearer networks are:

- E-GPRS plus one or more different RANs;

- Cable Access Network, etc.

6.   **Domain:** A domain is a logical association of network elements. A domain may contain any number of HSS, CSCFs and MRF. A domain can be Home Domain for some users (those whose subscription profile is stored in the HSS in that specific domain) and Serving Domain for other users (those whose subscription profile is stored in the HSS in a distinct domain). A domain can connect to a multitude of bearer networks. (FFS) The purpose of introducing the domain concept in release 00 is to enforce access Independence in the core, support of NAI addresses, scalability, additional services and servers expected (ex. Email/CSCF interworking), allow the use of DNS and directories for translations. Domain's already provide the glue for many useful services and functionalities. A domain is a logical realm that indicates a system or zone. A domain is used to associate services and servers with a common identifier for translation purposes. A domain is used as a key into databases in order to obtain the network addresses of nodes for respective services associated with the domain. The actual location of nodes supporting these services is not restricted in any way by a domain. The term domain used here refers to the DNS definition of domain.

Open Issue: The use of the term, domain, home domain and serving domain requires further clarification and analysis.

7   **Home Domain**: it is a domain that contains an HSS. In particular, the Home Domain of a user is the domain containing the HSS that stores the user profile. Home Domain may or may not contain the Home CSCF, the MRF or service logic. Home Domain:

- provides and maintains the user and user's subscription data in a HSS for the user belonging to the domain;

- supports also access independent users and users profile such as browser bookmarks and phone lists;

- user provides and updates the currently visited serving domain with user's profile;

- store routing and mobility information that enables service delivery to users and users roaming outside and inside the home network;

- maintains roaming agreements and service level agreements with other networks;

- Home Domain is seen as the initial termination point from the originating network when it contains the Home CSCF;

- may collect and consolidate charging data.

Other functions are FFS.

8   **Serving Domain**:

- stores roaming profile as received from the home domain

- provides services or access to services in the home domain (as per terminal capabilities and service level agreement if different operator domain) with the same "look, touch and feel" as much as possible;

- stores routing and mobility information that enables service delivery to users roaming in the service domain;

- optionally collects data for billing and statistics;

- can provide local services such as location based services and information (e.g. advertisements, operator announcements to local events, etc.);

- provides optional resources such as conference devices, multimedia call control etc. (Resource could be provided in home network)

- Home Domain can act as Serving Domain when the user is registered in the home domain. (FFS)

9   **Release 2000 all IP networks**: A Release 2000 all IP networks comprises of the following logical components:

- One or more domain(s);

- Any number of bearer network(s);

- Connectivity to one or more MGCFs and MGWs.

- Zero, one or more MSC/GMSC Servers

10 **Home Network (HN):** considering a specific user, the Home Network is made of zero, one or more serving domains, any number of bearer networks, zero, one or more MGCF/MGW, and the home domain for that specific user. (FFS)

11 **Serving Network (SN):** considering a specific user, the Serving Network is made of zero, one or more serving domains and one or more bearer networks, zero, one or more MGCF/MGW, and does not contain the home domain of that specific user. (FFS)

12 **Service Logic Domain**: includes the following functions:

- contains existing telecom service capabilities (i.e. SCP for IN);

- contains WAP type capabilities for Web-based services;

- allows easy access to services by the users (notifications of new services, activation/deactivation of services in the network, capability for payment and upgrade for new services, etc.);

- updates profile data in the home domain in the event of modifications by the user or by the provider of the Service Logic Domain operator;

- provides access to or capabilities to reach other Service Logic Domain;

- some location based services can also reside in the Service Logic Domain.

The Service Logic Domain could have a tight coupling with the Home Network in order to manage/charge/provide application service capability set uniformly to the users. In particular, if CSCF handles service triggers the Service Logic Domain and Home Domain may be providing both the user's user profile and subscription profile to CSCF in a co-ordinated/transparent fashion. The Service Logic Domain could also be stand-alone, i.e. independent from the network location. (FFS) It is for further study whether or not the CSCF can be logically associated with the service logic domain.

13 **Home User**: is a user of the home network having a subscription in the home domain. A user is considered a home user when the user is located in a serving domain in its home network. The user may or may not be located in their home domain.

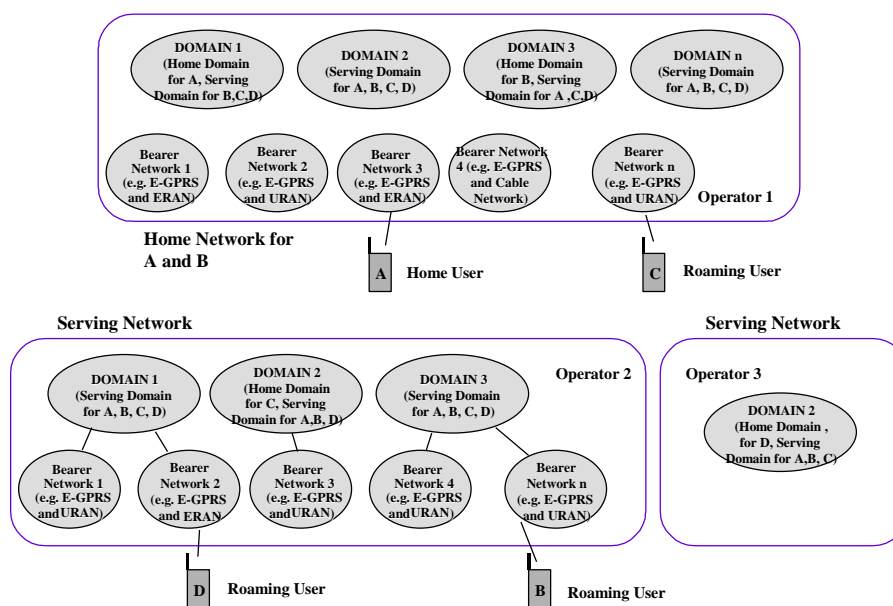14 **Roaming User**: is a user roaming outside its Home Network and being served in a Serving Network. (FFS)

Figure 9-1: Modelling of the network in domains

15 **Legacy Network**: a legacy network can be:

- SS7 based networks (e.g. PLMN, PSTN and ISDN) as well as CAS based networks;

- GPRS networks. (FFS)

16 **Multimedia IP Network**: it is an external IP network with support of real-time multimedia services (using H.323 and/or SIP protocols), and includes SIP/H.323 network elements and terminals, and possibly gateways to interface with Release 2000 all IP networks. (FFS)

17 **Serving CSCF (S-CSCF)**: it is a CSCF in the Serving Domain with which the user is registered and that is providing the services depending on the Service Profile(s) obtained from the Home Domain of the user. (FFS)

18 **Home CSCF (H-CSCF):** the Home CSCF is a CSCF in the Home Domain or Private Domain. The Home CSCF is associated to a user at the subscription time and, if the user is identified by aliases that might require translation to an IP address (e.g. logical names translated by DNS), the transport address of the Home CSCF will be provided as translation of the aliases. (FFS)

9.2   Assumptions

The following assumptions have been considered in the development of the roaming models described in the present version of the document.

1   The addressing requirements and mechanisms will be based on the requirements and mechanisms identified by 3GPP in 3G TR 22.975 and 3G TS 33.003.

2   Call admission/denying and call re-routing will be considered. Details of call admission (e.g. authentication and QoS) will not be discussed here.

3   Re-routing of incoming voice or data communication requests that are addressed to the user's directory number during periods of realignment of the national numbering plans will be considered. Probably, a non final solution (e.g. re-direction of calls based on databases) will be provided.

4   A specific PDP flow (called Signalling PDP flow), distinct from PDP flows carrying media PDUs, is adopted to carry signalling between UE and CSCF (e.g. call set-up, in-call signalling such as flash requests). The signalling PDP flow does not need to have an IP address different from the one of the PDP flows carrying the media.

5   The user profiles are stored in a permanent way into a database/server in the Home Domain.

6   In case of roaming the user profile in the home network must be interrogated by the serving network at least at registration and part of user profile could be temporarily stored into the serving network.

7   The roaming architecture will be optimised with the assumption that IP addresses will be allocated dynamically.

8   No new requirements would be placed on the legacy (e.g. PSTN, 2G PLMN) and Multimedia IP Networks to interconnect with the Release 2000 all IP networks. Release 2000 all IP networks would have to ensure inter-operability with the existing legacy and Multimedia IP Networks. In case a 2G HLR (e.g. GPRS HLR) is re-used to hold data for Release 2000 all IP network users, the 2G HLR will be upgraded to support Release 2000 all IP network user and its interfaces might need to be upgraded.

10. An MS registration procedure consists logically of a bearer level registration (e.g. GPRS attach) and, if so specified/allowed by the MS subscription profile, an application level registration. MS registration is performed, as an example, at MS power up.

11. The bearer level registration procedure entails registration with the GPRS nodes, according to GPRS-derived procedures.

12. The application level registration procedure entails registration with a serving CSCF/MSC server in order to inform the CSCF/MSC server of the MS presence and to allow the CSCF/MSC server to retrieve the user information from the HSS.

13. Bearer level registration and application level registration are considered to be two separate procedures. Bearer level registration completeion may trigger in the UE a signalling PDP flow activation as a possible mean of supporting the application level registration procedure.

14. The MS location is tracked at the GPRS level using GPRS mobility management, and user mobility is tracked at the application level through a specific procedure aimed at updating the user information maintained by the CSCF / MSC Server and, possibly, location information held by the HSS."

15. HSS keeps track of the user mobility in terms of the current CSCF/MSC server or Serving Domain.

16. Support of multiparty voice and data communication sessions (including the capability for the user or service logic to dynamically add or delete users from an active communication session) is not considered in the present version of this document.

17. Roaming agreements (static or dynamic) between the co-operating operators must exist.

18. Service Level Agreements (SLA) between network operators are defined (statically or dynamically) to ensure consistent level of services (e.g. end-to-end QoS, security etc.).

19. All network components that require address analysis and address translation for routing of terminating calls (e.g. CSCF, MSC server, MGCF) are capable of doing so or has access to consistent translation databases or to the HSS in order to resolve routing/call termination issues.

20. O&M functions exist to connect various components of a Release 2000 all IP network to each other, making it possible for each component to know how to address/access any other component within the network domain.

21. O&M functions exist to make network provisioning profiles (e.g. 800 triggers, tone information) available in each domain whenever.

22. In order to obtain user-plane optimisation, the serving GGSN will be preferably located in the serving network.

23. Service profile (subscription and activation status) and service triggers are either maintained by the HSS and/or updated by the HSS towards the CSCF/MSC Server.

24. CSCF-routed call signalling is assumed for real-time services.

25. Every CSCF is associated to one or more MRF, and every MRF can be controlled by more than one CSCF. For sake of clarity, if H.323 terminology is used to describe the MRF, MCU is part of the MRF. MRF could be located in the home network (when the Home CSCF is controlling the current call) or in the serving network (when the serving CSCF is controlling the current call).

26. Some of the functions of the A Home CSCF is are needed in all the roaming scenarios in order to support:

- incoming calls addressed to a DN from other Release 2000 all IP networks or Multimedia IP Networks with optimised routing (i.e. calls not routed through PSTN);

- incoming calls from other Release 2000 all IP networks or Multimedia IP Networks originated with a LN (Logical Name);

- implementation of supplementary services and Incoming Call Screening-like functions for terminated calls (e.g. Call Forwarding Unconditional).

9.3 Roaming Within All IP networks

In the follow, a set of roaming scenarios is described.

Editor's Note: please note that the network interfaces and the names shown in the diagrams from 6.4 to 6.7 may not always be correct.

9.3.1 Call Model

The call model described by the following statements has been adopted in the present document:

- Calls from/through PSTN are routed to an MGCF with connectivity to the Home Network corresponding to the dialled DN.

- Calls from a Release 2000 all IP network to a different Release 2000 all IP networks originated with a DN can be optimised (i.e. not routed to PSTN) only if the originating Release 2000 all IP networks has knowledge of the numbering plan of the destination Release 2000 all IP networks and can route the call directly to the destination Release 2000 all IP networks without leaving the IP domain. Further optimisations could be possible if the network where the call is originated has also access to the HSS of the network to which the call is destined and corresponding to the dialled DN. The same applies to calls from a Multimedia IP Network to a Release 2000 all IP network originated with a DN. (FFS)

- Assuming scenario 1, for incoming calls (i.e. mobile terminated calls), the call setup request always arrives at the ICGW which interrogates the HSS, implements Incoming Call Screening and relays (without performing any call control function) the request to Serving CSCF / MSC Server. If the originating Release 2000 all IP networks had the capability to interrogate HSS of the destination network, it would be possible to address the call setup request to the Serving CSCF directly.

- Regarding originating calls, the call request is handled by the Serving CSCF/MSc server (when present), or the Home CSCF when a Serving CSCF is not present.

This section covers only PS services.

9.3.2   Scenario 1, Traditional Model

The following pictures show respectively the roaming scenario 1 applied to roaming inside a single network and applied to roaming between networks.
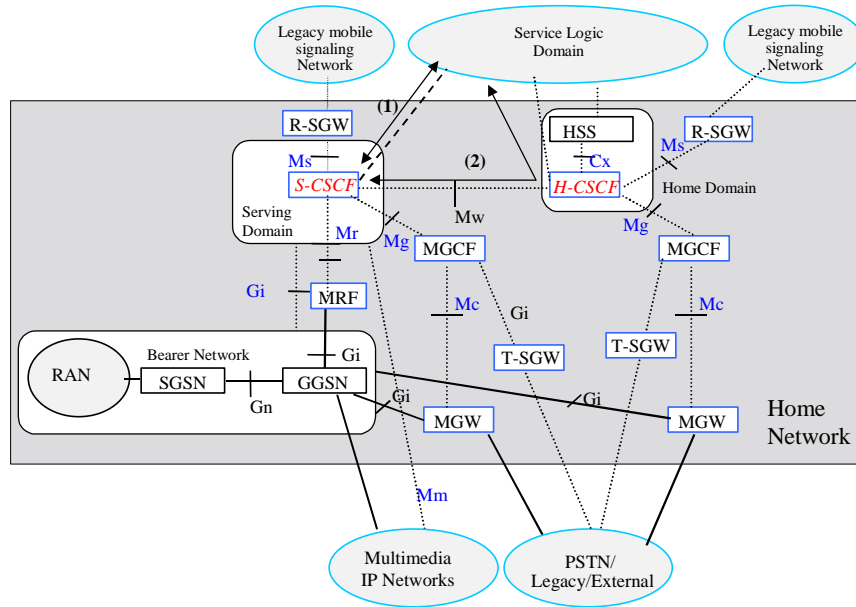

Figure 9-2: Scenario 1 applied to roaming inside a single network
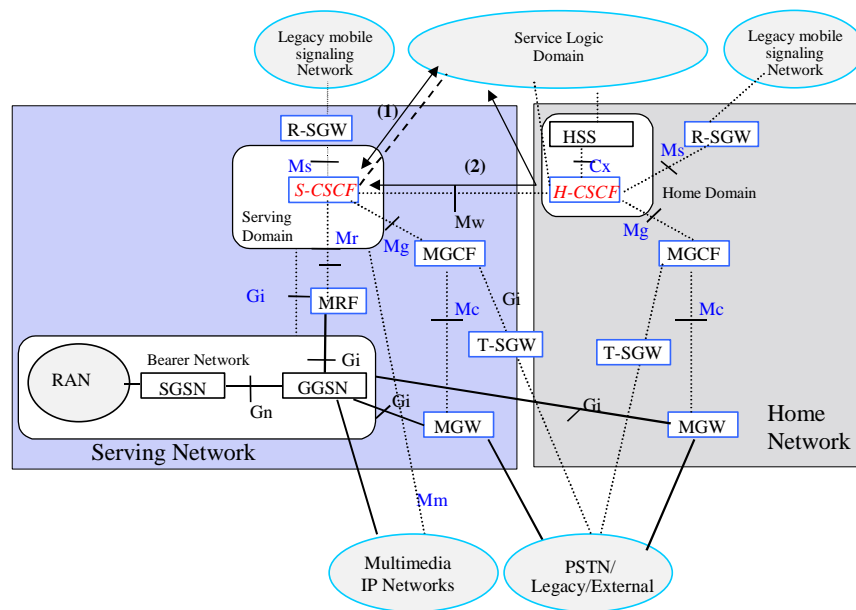

Figure 9-3: Scenario 1 applied to roaming between networks

The following points characterise scenario 1:

- both a Home CSCF and Serving CSCF are present and active;

- MT calls are routed to the Serving CSCF through the Home CSCF;

- non-basic services (e.g. supplementary services, etc.) invoked by MO calls and requiring interaction with service logic specific to the home network operator can be provided in two ways:

- Serving CSCF has a direct interface to the service logic (e.g. direct interface to a SCP in case of IN), case (1) in the above pictures;

- only Home CSCF has access to the service logic and the two CSCFs co-operate in order to provide the service, case (2) in the above pictures;

- MT Calls which reach the Serving CSCF are handled by the Servering CSCF for basic voice services;

- MT calls invoking non-basic services are handled as described for MO calls;

- user plane for MT calls is routed from the originating network to the serving network through the home network (i.e. from PSTN to MGW to GGSN in the serving network);

- user plane for MO calls to non Release 2000 all IP networks (and for non-optimised Release 2000 all IP networks to Release 2000 all IP networks calls) is routed from the serving network to PSTN through a MGW in the serving network, thus optimising the routing of user plane. In case of roaming within the same network, MGW can be chosen "close" to the bearer network again in order to optimise the routing of user plane;

- Home CSCF implements Incoming Call Screening triggers (i.e. triggers for supplementary services and IN services for incoming calls) and relays the call control signalling to the Serving CSCF address retrieved during the HSS interrogation.

9.3.3   Scenario 2

The following pictures show respectively the roaming scenario 2 applied to roaming inside a single network and applied to roaming between networks.
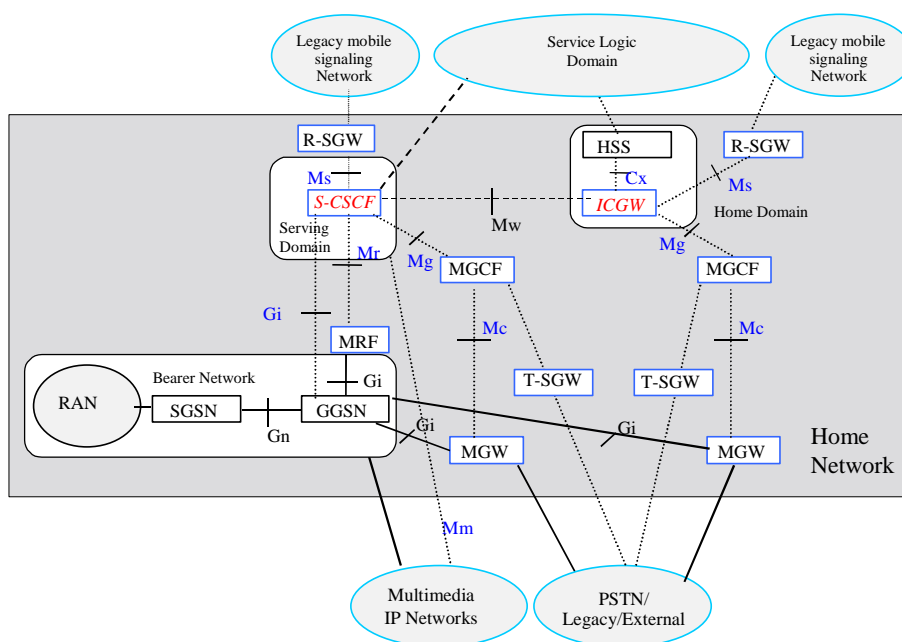

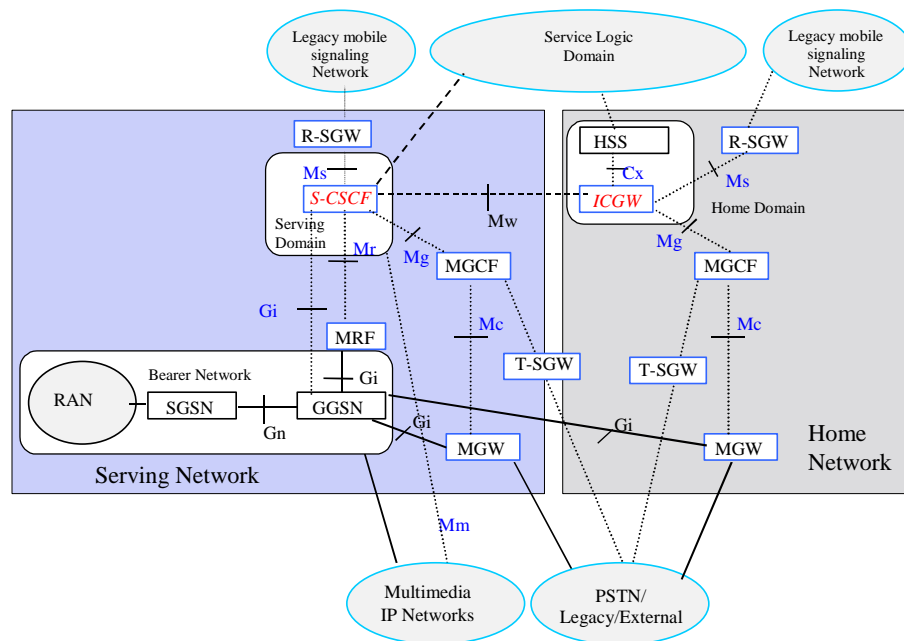**Figure 9-4: Scenario 2 applied to roaming inside a single network**

Figure 9-5: Scenario 2 applied to roaming between networks

The following points characterise scenario 2:

- only a Serving CSCF is present;

- MT calls are routed to the Serving CSCF through the ICGW in the Home Domain/Network;

- all the services (basic and non-basic) invoked during MT and MO calls are provided by Serving CSCF;

- If "non basic" service means non standardised, these services will involve serving CSCF and elements within the Home domain and service logic domain.]

- user plane for MT calls is routed from the originating network to the serving network through the home network (i.e. from PSTN to MGW to GGSN in the serving network);

- user plane for MO calls to non Release 2000 all IP networks (and for non-optimised Release 2000 all IP network to Release 2000 all IP network calls) is routed from the serving network to PSTN through a MGW in the serving network, thus optimising the routing of user plane. In case of roaming within the same network, MGW can be chosen "close" to the bearer network again in order to optimise the routing of user plane.

9.3.4　Scenario 1: Information Flows for Validation

In order to validate scenario 1 proposed above, information flows for registration, location management and call delivery/origination are provided in this section.

The information flows presented in the Call Control and Roaming proposal do not provide many details. Generic names have been chosen for signalling messages. Any resemblance to known existing protocols is due to an attempt simply to proposing information flows immediately comprehensible. Only a restricted subset of the possible information flows is included in the document.

9.3.4.1　Registration and Location Management

- In this version of the Technical Report, only a basic registration procedure is considered.

- The basic registration procedure is composed of three steps:

- GPRS attach: is a plain GPRS attach procedure;

- PDP context activation: a PDP context is set up to support application level signalling;

- application level registration with CSCF.

  The latter is considered in the registration flows reported in the follow.

  Two basic cases of registration are shown here, in order to provide an initial picture of the application level registration.
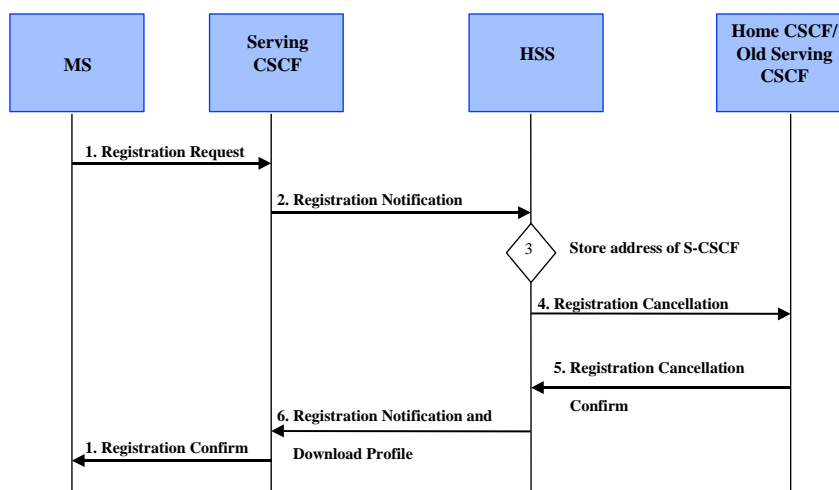

Figure 9-6: Release 2000 all IP network user registering in Release 2000 all IP network Serving Domain

Steps 4 and 5 are optional and take place only in two cases:

- if the user was previously registered in a different Serving CSCF;

- if the user was not previously registered but, during a MT call, HSS determined that a service profile was needed in the Home CSCF to handle the supplementary services triggered by the incoming call (e.g. a forwarded-to multimedia call triggered by the MT call).

Other registration and location management scenarios are FFS.

9.3.4.2   MT/MO Calls

Two call information flows are presented in this version of the Technical Report. The call flows are based on the following call delivery model:

- a call from PSTN towards a DN corresponding to the user is received by one of the MGCF of the Home Network, ISP or corporate LAN domain in the IP multimedia network;

- MGCF reaches the Home CSCF translating the DN;

- Home CSCF interrogates the HSS to retrieve information regarding the user corresponding to DN;

- Home CSCF receives information on how the call has to be routed: in the flows shown here, HSS returns the signalling address of a Serving CSCF, but in general could be the transport address of the MS if there is no

Serving CSCF or a forwarde-to number). Also, HSS might return service profile information in case the user is not registred.

- Home CSCF forwards call signalling to the retrieved address

Regarding user-plane, packets are routed from the MGW in the corporate LAN domain or home network to the GGSN in the bearer network where the user is presently located.

No optimisation has been considered for user-plane of MO calls. In case MO routing optimisation is desired, the MGCF used to route the call towards PSTN can be chosen using different possible criteria.

9.3.4.2.1 Incoming call from PSTN to a Release 2000 all IP network

The following flow describes the call delivery for a MT call from PSTN to a Release 2000 all IP network user addressed through a DN.
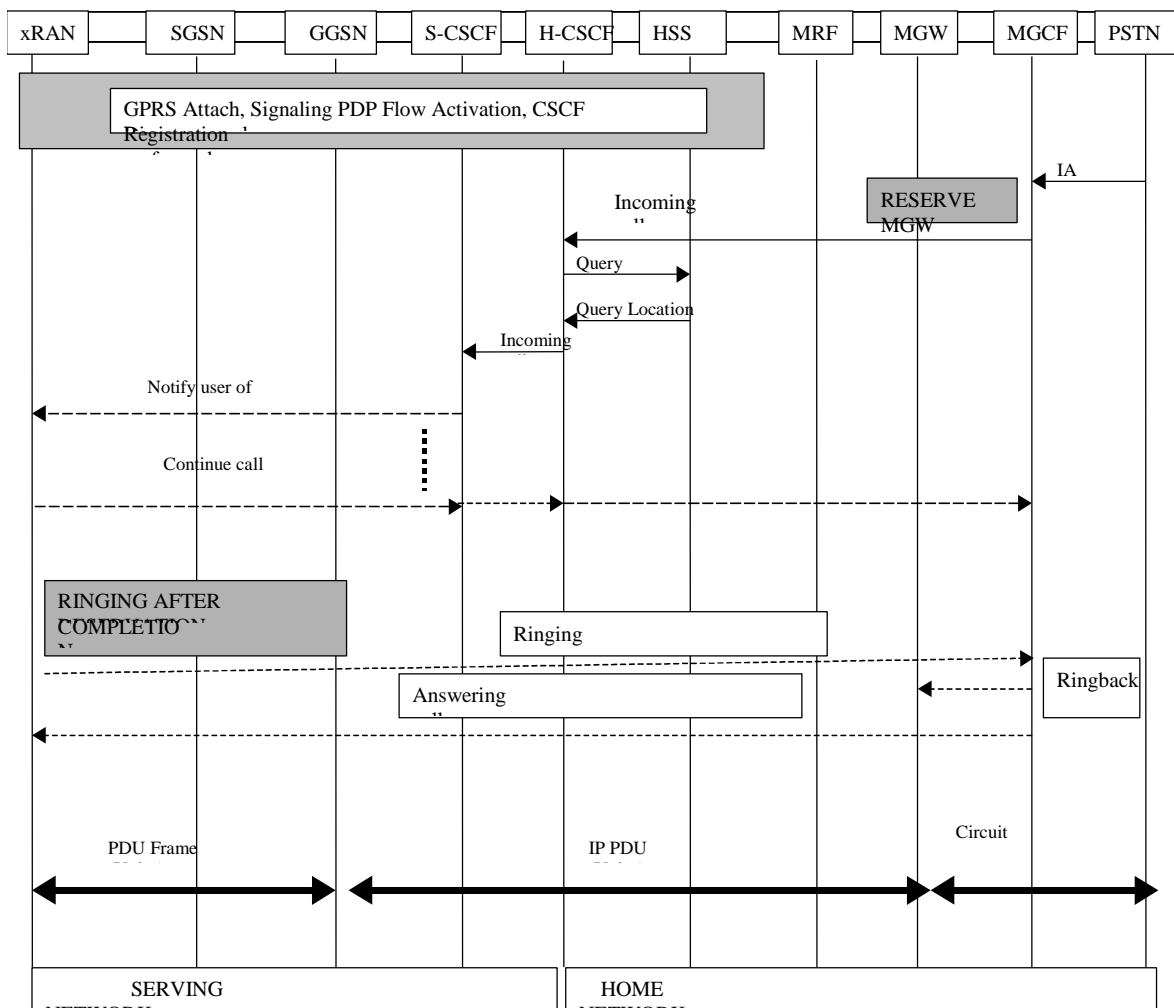
Figure 9-7: Incoming call from PSTN to a Release 2000 all IP network

Issues such as QoS negotiation, policy management, etc. are FFS.

9.3.4.2.2 Call from an 3GPP IP based network/Multimedia IP Network to 3GPP IP network

The following flow assumes that a Release 2000 all IP network user has roamed into a visited network. The flow describes a call from a different Release 2000 all IP network terminated into the home domain of the called user. It is assumed that

the Release 2000 all IP network where the call is coming from, is aware of the addressing plan information (IP based addressing) that allows the call to directly terminate into the H-CSCF (otherwise, it may have been routed through the PSTN and terminated into MGCF).

Details will need to be worked on.

Figure 9-8: Call from Release 2000 all IP network to Release 2000 all IP network
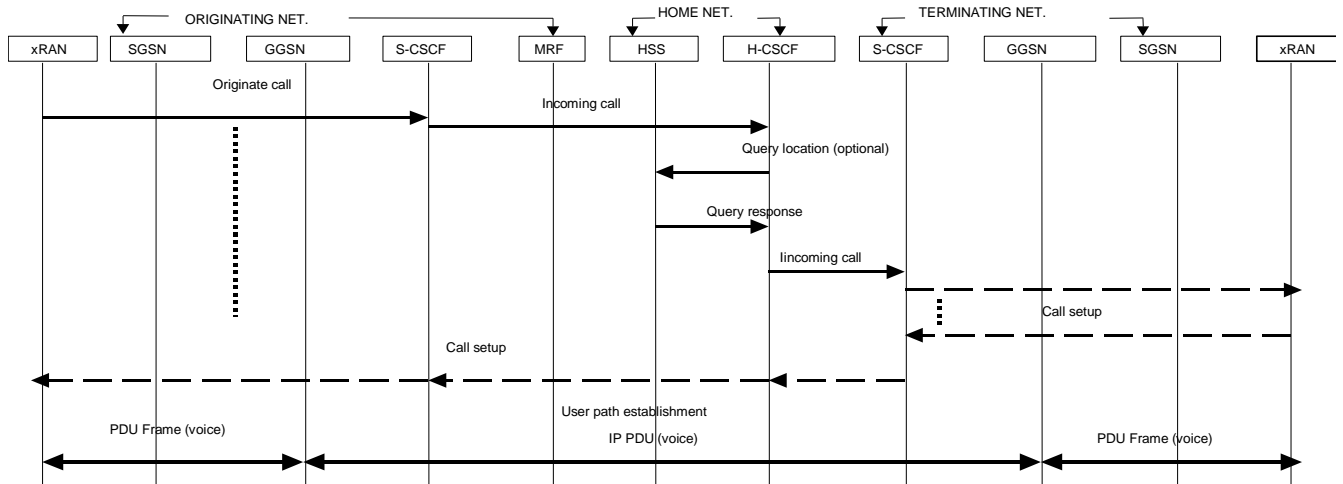
9.3.5 Scenario 2: Information Flows for Validation

No flow will be shown for this version of the document.

9.4 Roaming to Other Networks

In order to ensure compatibility and easy roaming between 2G GSM/GPRS, UMTS R99 and UMTS R00 CS and GPRS domain (excluding the VoIP/multimedia domain), the same mobility procedures are used within and between the 3 kind of networks (storage of the current location in the HSS, use of MAP to update the HSS with the current subscriber location of an user and to download / update the subscriber data in the visited node).

Some enhancements with regard to current mobility procedures are obviously needed:

- in case of a R00 terminal roaming in a MSC/SGSN of a 2G / R99 network:
  - R-SGW relays all the MAP / CAP messages between the HSS / SCP and the functions (MSC, SGSN, …) handling Call / Session in the 2G / R99 CN.
  - the R00 HSS sends to the MSC/SGSN a MAP_R99 translation of the subscriber data compatible with the data a R99 MSC/SGSN can handle. This should be classical for MAP application context handling.
  - in case of a R00 terminal roaming in R99 network and requesting Multimedia service: as in R99 there is no **standard** way to have a visited GK,
    - Either, in order to get a customized service, the R00 terminal requests service from its Home CSCF in its R00 network. The R-SGW is not impacted in this case.
    - Or, the service of a GK in the visited PLMN is used and the service cannot be customized. The R-SGW is not impacted in this case.

- in case of a R99 terminal roaming in a MSC/SGSN of a R00 network:
  - R-SGW relays all the MAP / CAP messages between the 2G / R99 HLR / SCP and the functions (SGSN, …) handling Call / Session in the All IP R00 CN. The R00 VLR (in SGSN and/or CS domain) or Call / Session handling function is able to interpret MAP_R99 / CAP_R99 received from R99 HLR / SCP

- in case of a R99 terminal roaming in R00 network and requesting Multimedia service: as in R99 there is no **standard** way to have a visited GK,
  - Either, in order to get a customized service, the R99 terminal requests service from its Home CSCF in its R99 network. The R-SGW is not impacted in this case.
  - Or, the service of a GK in the visited PLMN is used and the service cannot be customized. The R-SGW is not impacted in this case.

There are currently two proposed solutions for roaming. These solutions are not completely in contradiction with each other but actually include a good set of commonalities. However, further work is needed to identify the commonalities and to consolidate the proposals. The two proposals are summarised in this section with references to more detailed descriptions. The presented roaming cases and key issues addressed in the contributions should be further considered and taken as a baseline for further work on roaming model definition for R00 networks. Alternative solutions for roaming are for further study.

9.4.1   Roaming Procedures for R00 networks

One possible solution for the support of roaming in R00 networks is described in Tdoc S2k99117. The contribution covers both roaming between R00 networks and roaming to mobile legacy networks. The contribution only covers the PS-only architecture in UMTS R00. Roaming from UMTS R99 considered in this document is in terms of USIM roaming. The contribution makes certain assumptions on mobility management and network identities that have to be considered as part of the future work. A basic registration procedure is introduced in order to allow the discussion on roaming.

The roaming scenarios described are:

- Roaming within R00 PS domain networks;

- Roaming from R00 PS domain networks to 2G/UMTS R99 networks;

- Roaming from 2G/UMTS R99 networks to R00 PS domain networks;

9.4.2   Overlaid solution to roaming

One roaming solution is to introduce an overlaid personal number service, which keeps track of users registrations (attached to 2G/3G CS and/or PS MultiMedia) and call reception preferences. This enables inter-service as well as inter network roaming for Telephony as classical TeleService Speech in 2G/3G networks and Telephony as the voice component of a MultiMedia service.

The proposed roaming solution is further detailed in Tdoc S2K-99070. The Tdoc elaborates on the driving forces for overlaid roaming and includes examples of user registration and call reception cases.

9.5   Open Issues

The following issues need to be discussed and solved through interaction with the other working groups in Release 2000 all IP network and might require discussion in the plenary.

- Support of multiparty voice and data communications sessions (including the capability for the user or service logic to dynamically add or delete users from an active communications session). The impact on the control plane (e.g. CSCF) and on the user plane (e.g. use of MRF vs. IP multicast) has to be studied.

- UMS structure and functionality has to be defined. As an example, UMS might have additonal interface (e.g. GRIC CSP) to the clean houses as defined in H.225. GRIC Communications, Inc. develops a global intelligent transaction platform (GRIC CSP) that allows diverse networks to interoperate. This platform allows ISPs, Telcos, and emerging carriers to offer multiple IP-based services such as IP Telephony, E-commerce, Internet Roaming, and Internet Faxing. Leveraging its GRIC Alliance Network, a worldwide membership of over 450 major ISPs and telcos in more than 140 countries, GRIC has aggregated an addressable user base of 30 million dial-up users and an estimated 40 million corporate users. For more information on GRIC see http://www.gric.org/ Routing

optimisation for calls originated in a Release 2000 all IP network towards a MS of another Release 2000 all IP network and addressed using a DN (Directory Number) in order to bypass the PSTN has to be discussed.

- Also, routing of MT calls towards LN has to be specified.

- CSCF discovery has to be discussed and solutions have to be presented at the next meetings.

- Impact of requirements regarding backward compatibility with 2G networks (e.g. RAN, terminals, call control and roaming, services) needs to be addressed.

- Location confidentiality issues have to be considered versus optimisation of signalling and transport paths.

- The Release 2000 all IP network must provide the capability for service logic to deny or re-route voice and/or data communication requests. This capability has to apply to both incoming and to user initiated communication requests.

- Also, signalling flooding problems and malicious attacks to the network have to be considered.

- The issue could affect the architecture in terms of where more firewall functions will have to be implemented.

- Addressing of multiple PDP flow by means of the same IP address has to be solved as an issue if not yet standardised for GPRS.

- Details regarding the set of vocoders supported in Release 2000 all IP network and the vocoder negotiation mechanism need to be investigated.

- Signalling PDP flow can be activated when the MS attaches/registers with the network and kept alive for all the duration of the attach/registration session (dorment signalling PDP flow), or it can be activated on demand. The choice between the two options has to be discussed considering paging issues, load due to MM for the dorment PDP context even when the MS is idle, and the impact on the MS.

- Is T.120 considered as real-time application, i.e. do T.120 components of multimedia calls have to be controlled by a CSCF?

- Database queries in the HSS and other network databases has to be discussed and defined.

- Addressing of network entities and address translation need to be defined (e.g. It is assumed that MGCF has the ability to retrieve the CSCF/MSC server corresponding to a DN; how MGCF does it is FFS).

- Storage of 2G profiles in Release 2000 all IP network in order to support roaming to 2G legacy networks has to be discussed. In particular, the presence of a 2G HLR functionality in HSS need to be discussed and possible alternatives evaluated.

- Long term issues and assumptions, i.e. not related only to R00, should be considered in order to have a future proof solution. As an example, long terms requirements on addressing mechanisms (e.g. dynamic vs. static, IPv4 vs. IPv6) should be considered.

- a "VLR" type functionality capable of caching the service profile for the user in a serving domain needs to be defined. The functionality should not be related to location management but focus on service profile.

- Differences in the definition of the term "Location" as understood within the cellular/wireless community and in the IETF community have not been addressed yet in this document. This needs to be harmonized.

- Provisioning of location-based services (e.g. service based on geo-location information) and the impact on the architecture need to be considered.

- QoS and security issues have not been addressed yet.

10 Service Platform Impacts

10.1 3GPP Release 2000 Service Architecture

This section describes how the 3GPP release 99 service architecture [3] can be applied to the 3GPP release 2000 network by extending the VHE/OSA concept to the Multi-Media core network. This can be done by providing an application interface (as described in VHE specification [3]) from the CSCF, see Figure 10-1.. As VHE expect the service to be located in the home domain of the end–user (the Home Environment), other network elements besides the CSCF may be needed to provide an roaming architecture that allow the serving domain to pass control to the home domain where the service logic resides. [1]

Note: The architectures in Figure 10-1 and Figure 10-2 show the CAMEL Service Environment (CSE) which is not shown in the Reference Architecture in Section 5.
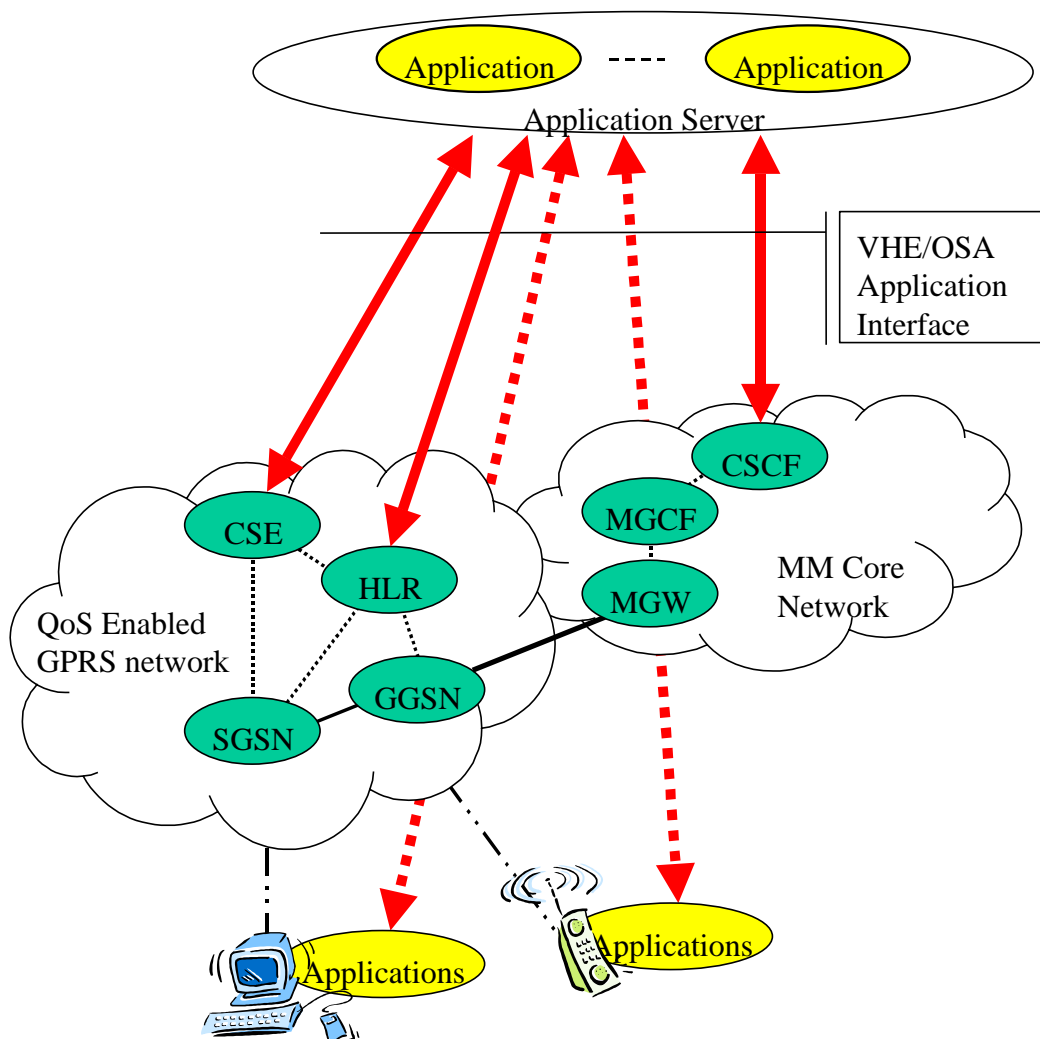

Figure 10-1: 3GPP Release 2000 service architecture

The Open Service Architecture consists of three parts, as illustrated in Figure 10-1(note that the figure is not meant to be exhaustive of all interrelationships):

Within the current 3GPP specifications this is achieved via CAMEL providing the CAP interface between serving and home network.

-   **Applications**, e.g. VPN, conferencing, location based applications. These applications are implemented in one or more Application Servers;

-   **Framework**, providing the applications with basic services that enable applications to make use of the service capabilities in the network. Examples of framework services are Authentication, Registration and Discovery;

-   **Service Capabilities**, providing the applications with services that are abstractions from underlying network functionality. Examples of services offered by the Service Capabilities are Call Control, Message Transfer and Location. Services are possibly provided by more than one Service Capability Server. For example, the Call Control service might be provided by CAMEL and MexE. The Service Capability Servers taken into account for UMTS Release 99 are CAMEL, MExE, SAT and HLR.
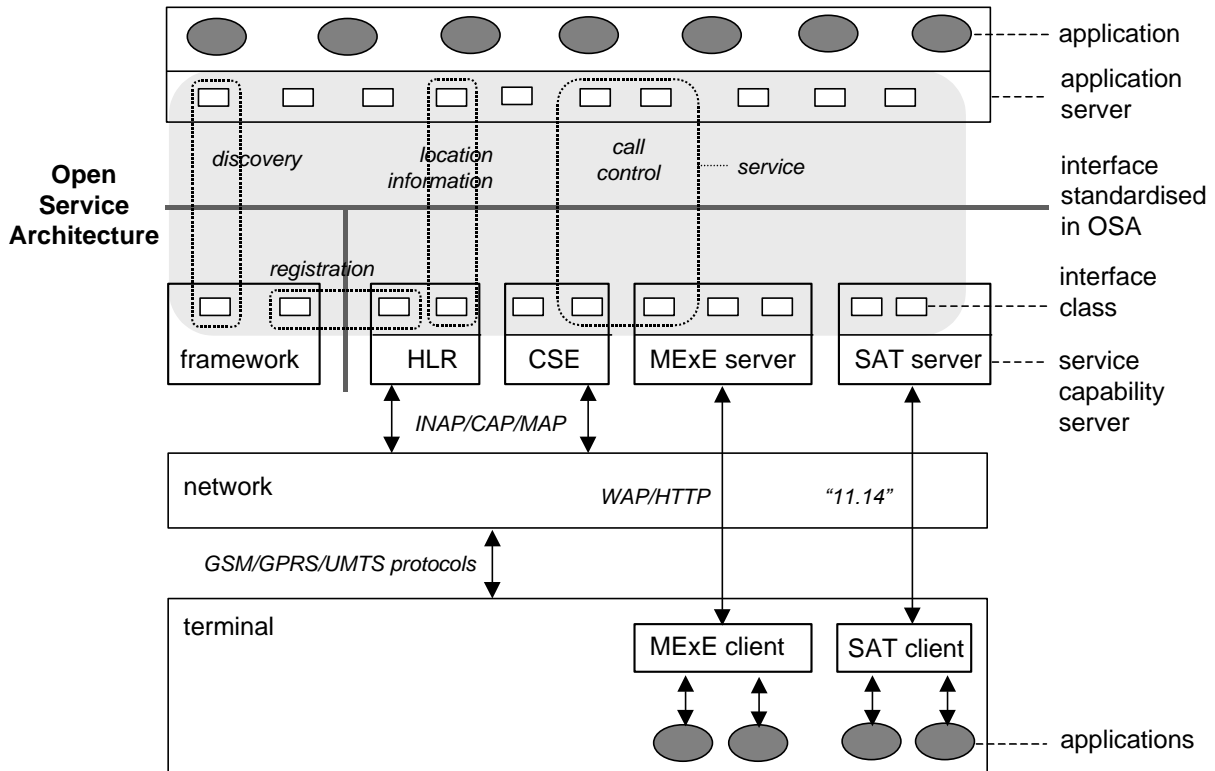
Figure 4

Figure 10-2: Overview of Open Service Architecture

Note: This may not be in line with the latest version of the VHE/OSA Stage 2 document

10.2   IN based Services

The IN based service is one example of legacy services and the IN based service logic is one example of how legacy services may be introduced to the 3GPP Release 2000 networks. This IN based service logic may need to be enhanced in 3GPP Release 2000 networks, based on the proposed architecture, when full support for multi-media is required.

When only voice/audio has to be supported several options exist:

1. Re-route call to legacy system. This is applicable to very specific services such as 800- and 900- services.

2. Provide 'INAP' like interfaces between the 3GPP Release 2000 network functional blocks (e.g. CSCF) and a legacy SCP. These interfaces will be used for inter- and intra-network connections, and as such should be based on a suitable INAP protocol (e.g. CAP).

3. Provide new interfaces between legacy IN and 3GPP Release 2000 network functions, allowing the AS to access the application in the legacy SCP.

It shall be noted that the Operator Specific Services defined for the QoS enabled GPRS network are still available and apply for the access bearers towards the MM core network. This will for instance enable the pre-paid charging of the GPRS bearer.

As indicated in the sections below, option 2 (section 10.2.1) will allow the possibility that existing services can be upgraded to provide 3G users with a seamless transition of familiar 2G services, especially whilst roaming. Option 3 (section 10.2.2) does not have this advantage but benefits from future proofing. It would thus seem appropriate to allow the two options to co-exist, where support of legacy CAMEL services would be carried via option 2 and enhanced/future services can be provided via OSA principles as outlined via option 3 or a combination of both.

10.2.1 'INAP' based interface between legacy SCP and R00 network entities

In this option, the classical IN model is extended to include the CSCF as a node capable of supporting a service switching function (based on the 23.078 specification of the gsmSSF) and a transaction based protocol (TCAP) with CAP. Conceptually, a new functional entity is introduced between the CSCF and the CSE. This functional entity, called a *softSSF*, can potentially be based on a modified release 99 GMSC, VMSC and VLR functionality, where call control, billing and database functions are retained or enhanced. This softSSF interacts with the CSE via CAP and interacts with the CSCF either via an internal interface (if it is co-located with the CSCF) or via an open interface based on OSA concepts (if the softSSF is deployed as a Service Capability Server). Some changes to CAP can be expected to take into account the impact of the underlying IP call control. The CSE is able to offer its services via defined open interfaces based on OSA principles. These services can be implemented by the CSE via the CAP interfaces to the SGSN and the CSCF. (Note the CSCF may also offer its services via defined open interface based on OSA principles).

This option benefits for the extensive re-use of standardised process (gsmSSF), protocol (CAP) and already deployed services.
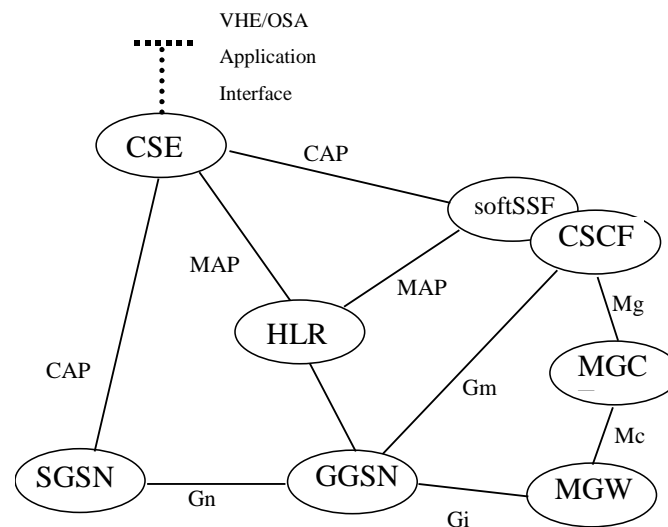
Figure 10-2 Functional Architecture to support option 2

10.2.1.1 Advantages

♦ Maximises the re-use of existing functional entities, protocols and services. Such reuse decreases the development and ownership costs allowing existing familiar 2G services to be provided to 3G users from an early stage.

♦ Minimum changes to the CSE for the support of legacy services. There are several IN/CAMEL services already deployed PrePaid, VPN, Mobile Number Portability, etc which can be used in a voice over IP network.

♦ Potentially, CAP/TCAP can be carrier over IP (this requires further study/contributions)

♦ This approach is in line with the work currently underway in ETSI SPAN 3 (Services and Protocols), in particular a work item addressing IN support for voice over IP on the H.323 architecture and associated protocols in association with the TIPHON project. The study will investigate how an H.323 gatekeeper can act as a virtual Service Switching Point (SSP). It is worth noting that ETSI plan to harmonise the fixed line IN protocol (ETSI Core INAP CS3.1) and mobile equivalent (CAP Phase 3) into a common protocol targeted for ETSI Core INAP CS4.

10.2.1.2 Disadvantages

♦ Introduces new functional entity 'softSSF', which provides the necessary mapping between the CSCF and the CSE. However, this functional entity is based on the functions already provided by a VMSC/GMSC, where already standardised process such as the gsmSSF can be reused. The interface between the CSCF and the softSSF requires further study.

10.2.2 New open interface between legacy SCP and R00 network entities

This option adopts the OSA principles where the service capability servers such as the CSCF and the legacy SCF have defined APIs that allow applications in separate application servers to use the features offered by these SCS. This approach allows the service features that are provided by the SCS to be made available to applications designers with out having detailed knowledge of the specific protocols. Currently, the SCSs reflect the service capabilities in UMTS phase1, and CAMEL is one example. From TS 22.121 - "A service capability server consists of one or several server components. Taking CAMEL Services as an example, the server components could be Call Control, Location/Positioning, PLMN Information & Notifications. Each of these server components offers its services via defined open interfaces, and implements these by using GSM/UMTS protocols".

The problem is that within an all IP network, the above mentioned server components are not all available via the CSE as there is no underlying protocol between the CSE and the network for call control (apart form the CSE/SGSN interface). To provide the same functionality provided by existing legacy services, new applications will have to be created that make some limited use of the features that may be offered by the CSE (for example database lookup), plus service features offered in the CSCF.

APIs supporting client-server models will exist. These APIs will enable the users to access service logic via the UE and specialist servers (e.g. MexE). The interface these specialist servers and the CSCF is for further study.
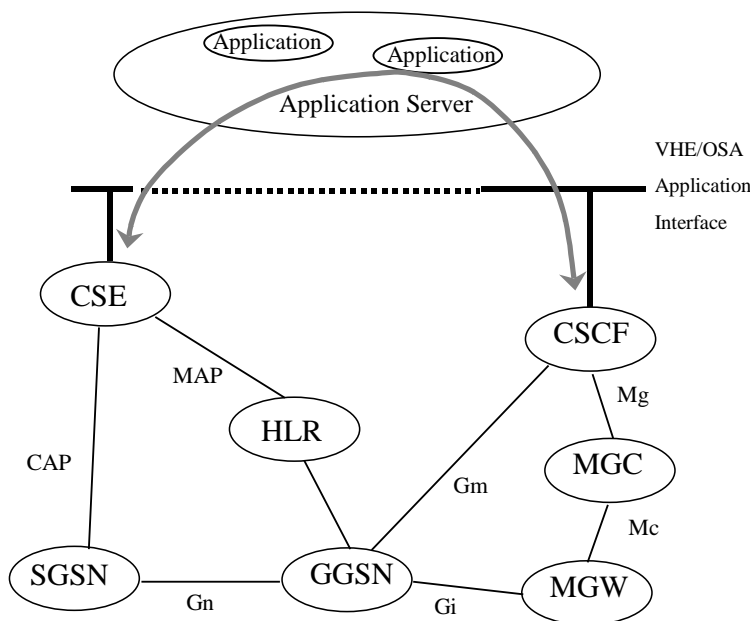

Figure 10-3 Functional Architecture for option 3

10.2.2.1   Advantages

♦ An open interface based on OSA concepts. APIs that may interface with the CSCF and the CSE are expected to become available allowing specific protocols to be hidden from the service/application designers

♦ Easier deployment of new/enhanced services for multimedia applications.

10.2.2.2   Disadvantages

♦ When considering existing CSE based services, little re-use is made of already standardised protocols, services and processes. More significantly, new processes and protocols must be re-defined and re-implemented for services that already exists, increasing development and ownership costs.

♦ Requires new applications on an application server to be created in order to support legacy services.

♦ Impact on the legacy CSE and services considered greater than option 2.

10.3   Issues requiring further contributions

The following issues require further contributions:

• Applications may reside not only in Application Servers (AS) but also in terminals.

• Options for sharing applications or parts of them between AS and terminals

• Which elements, beside the CSCF, will provide API for application design (aligned with VHE/OSA)

• Terminal shall also provide API for application design (aligned with MExE)

• Which new Service Capabilities/Service Capability Features are needed for 3GPP Release 2000 (e.g. WIN)

• Specific implementation cases of the proposed architecture should be provided.

• If and how 3GPP Release 2000 service features could be made accessible to 2G terminals via 2G networks

• If and how 3GPP Release 2000 service features could be made accessible to dual mode 2G/3G terminals via 2G networks

-

11   Security

There will be a common authentication scheme for the terminals operating in the all-IP mode, which will be SIM/USIM based. It is required that all-IP terminals will be able to register and provide basic service when used with a 3GPP SIM/USIM.

12   Work Plan

12.1   Milestones for Release 00

3GPP has the objective of producing the second release of specification for UMTS by the end of 2000. The project management for this work will need to include the elements of work package definition, the interdependency of these work packages and their scheduling. As the work is undertaken in the various TSGs and WGs in 3GPP there will need to be agreement across these groups on the overall plan. Subsequent communication between these groups on such issues as changes to schedules and requirements will be essential. An additional task relating to communication is the

reviewing of requirements documents to identify technical problems in the implementation at an early stage. The completion of release 99 has the potential to impact the availability of resources and hence the time scales for release 00.

Note: "The text that follows is of more general nature than the rest of the document. This has been included to show the framework for release 2000, of which an All IP Architecture may be one Work Item. The dates shown are assumed dates and need to be verified by TSG-S2."

12.1.1   Release 00 milestones

3GPP has not yet agreed overall milestones for release 00. For the purposes development of a high-level work plan the following key milestones are proposed.

July 99	3GPP All-IP network feasibility study started
Sept 99	TSG-S2 R00 Ad Hoc will submit the results of the TSG-S2 for approval
Oct 99	After TSG-S2 approval, TSG-S2 R00 Ad Hoc Group results will be submitted to TSG-S for approval
Oct 99	After TSG-S approval Project planning work for R00 (including all IP option) will be started at TSG-S2 project planning ad hoc groups (These groups have adequate participation from all relevant TSGs/WGs).
Dec 99	After TSG-S2 approval the detailed workplan for R00 (including all-IP network option) will be submitted to TSG-S for approval.
Dec 99	Release 00 service requirements available
Jan 00	TSG-SA2 completes first draft architecture for all-IP network.
Jan 00-Dec 00	Work within 3GPP TSGs and WGs proceed according the TSG-S approved R00 Project Plan
Dec 00	Release 00 specifications completed including all-IP option

A high level PERT chart is given overleaf to form the basis of the planning for releases beyond R99. The reviewed first drafts of the service and architecture specifications should be available before the end of 1999.

12.1.2   Detailed activity plan

Date	Meeting group	Proposed activity
August 23 – 27	S2	Progress architectural study
September 13 - 17	S2	Joint S1 – S2 activity on R00 - Finalize the requirements for the architectural study and identify the key issues. Finalize the proposed R00 architecture at the TSG-S2 R00 ad hoc group.
Late September/early October	S2 R00 Ad Hoc Group meeting (provisional)	Possible refinements to the outcome of the ad hoc group
September 29 - 1	S1	Review input on service requirements for R00
October 11 - 13	SA Plenary	Approve the results of TSG-S2 R00 Ad Hoc Group
October 25 – 29	S2	Start the specification of the architecture, and detailed work plan. Initiate the S2 project coordination adhoc groups work for R00 based on the approved results from TSG-SA Plenary as well as results from other

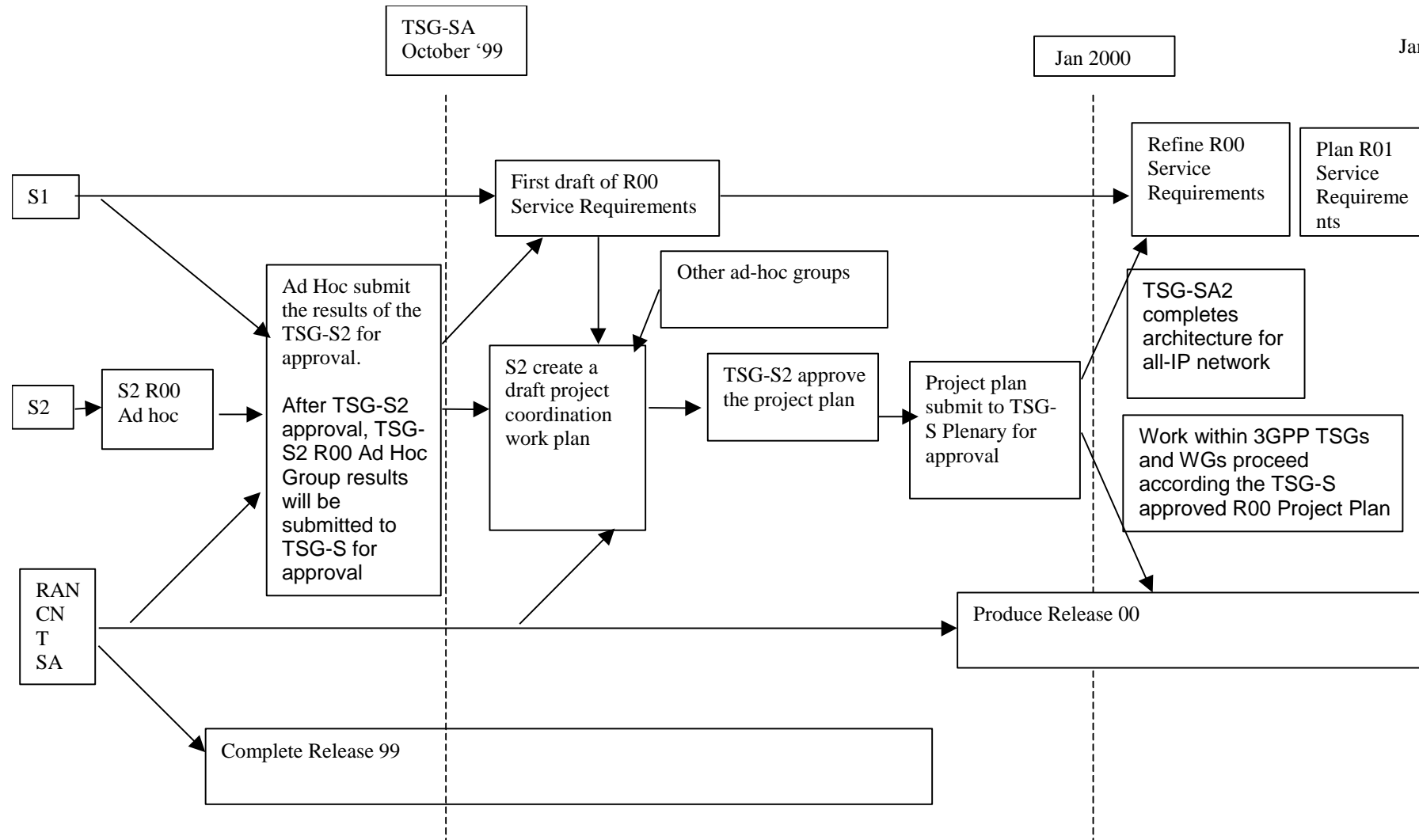|		groups, e.g. the Mobile IP ad-hoc group.
November 29 - 3	S1	Review service requirement and architecture
November 29 - 3	S2	Develop architecture specifications
December 15 - 17	SA Plenary	Approve detailed work plan from TSG-S2 project coordination adhoc groups

TSG-SA
October '99

Jan 2000

Jan 2001

S1

First draft of R00
Service Requirements

Refine R00
Service
Requirements

Plan R01
Service
Requireme
nts

Ad Hoc submit
the results of the
TSG-S2 for
approval.

After TSG-S2
approval, TSG-
S2 R00 Ad Hoc
Group results
will be
submitted to
TSG-S for
approval

Other ad-hoc groups

S2

S2 R00
Ad hoc

S2 create a
draft project
coordination
work plan

TSG-S2 approve
the project plan

Project plan
submit to TSG-
S Plenary for
approval

TSG-SA2
completes
architecture for
all-IP network

Work within 3GPP TSGs
and WGs proceed
according the TSG-S
approved R00 Project Plan

RAN
CN
T
SA

Produce Release 00

Complete Release 99

Figure 14-1: 3GPP Standardisation Activities Beyond Release 99

History

Document history		
V0.0.0	July 1999	Creation of document.
V0.0.1	11 Aug 1999	Updated after R2000 Ad Hoc Swindon, 10-11th August. Scope is clarified and new text added to requirements, Service Platform sections and a new architecture sub-section on support for R99 terminals. Architecture changes: new CSCF <-> CSCF interface. MGW split into MGW and Transport signalling gateway. SGW to legacy network labelled as roaming gateway.
V0.0.2	1 Sept 1999	Updated after R00 Ad Hoc, based on tdocs s2k99030, s2k99031, s2k99032, s2k99035, s2k99045, s2k99049
V0.0.3	6th Sept 1999	Updated with new sections reviewed by e-mail. Contains marked changes from v0.0.2
V0.1.0	22 Sept 1999	Updated as per meeting week beg. 13th Sept (Bonn). New text on handover and IP header compression /stripping. Definitions on Mc reference point and on the Media Gateway function al blocks.
V0.1.1	28th Sept 1999	Draft for Ad Hoc Helsinki, changes include addition of HSS in section 5, agreeed solution for support of CS terminals.
V0.1.2	29th Sept 1999	2nd draft for Ad Hoc Helsinki, changes include addressing editor's notes in section 9, addirtion of text to sections on QoS and Security and new subscetions to service Platforms
V0.1.3	30th Sept 1999	Interim draft at R00 Ad Hoc, Helsinki
V0.1.4	1st Oct 1999	Final draft for approval
V.1.0.0	7th Oct 1999	Prepared for presentation at SA#5. Technical content identical to v.0.1.4